Khaled Elleithy
Editor

Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering

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Edited by

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To my father and mother

Contents

Preface

This book includes Volume II of the proceedings of the 2007 International Conference on Systems, Computing Sciences and Software Engineering (SCSS). SCSS is part of the International Joint Conferences on Computer, Information, and Systems Sciences, and Engineering (CISSE 07). The proceedings are a set of rigorously reviewed world-class manuscripts presenting the state of international practice in Advances and Innovations in Systems, Computing Sciences and Software Engineering.

SCSS 07 was a high-caliber research conference that was conducted online. CISSE 07 received 750 paper submissions and the final program included 406 accepted papers from more than 80 countries, representing the six continents. Each paper received at least two reviews, and authors were required to address review comments prior to presentation and publication.

Conducting SCSS 07 online presented a number of unique advantages, as follows:

- All communications between the authors, reviewers, and conference organizing committee were done on line, which permitted a short six week period from the paper submission deadline to the beginning of the conference.
- PowerPoint presentations, final paper manuscripts were available to registrants for three weeks prior to the start of the conference.
- The conference platform allowed live presentations by several presenters from different locations, with the audio and PowerPoint transmitted to attendees throughout the internet, even on dial up connections. Attendees were able to ask both audio and written questions in a chat room format, and presenters could mark up their slides as they deem fit.
- The live audio presentations were also recorded and distributed to participants along with the power points presentations and paper manuscripts within the conference DVD.

The conference organizers and I are confident that you will find the papers included in this volume interesting and useful. We believe that technology will continue to infuse education thus enriching the educational experience of both students and teachers.

Khaled Elleithy, Ph.D.
Bridgeport, Connecticut
June 2008

Acknowledgements

The 2007 International Conference on Systems, Computing Sciences and Software Engineering (SCSS) and the resulting proceedings could not have been organized without the assistance of a large number of individuals. SCSS is part of the International Joint Conferences on Computer, Information, and Systems Sciences, and Engineering (CISSE). CISSE was founded by Professor Tarek Sobh and me in 2005, and we set up mechanisms that put it into action. Andrew Rosca wrote the software that allowed conference management, and interaction between the authors and reviewers online. Mr. Tudor Rosca managed the online conference presentation system and was instrumental in ensuring that the event met the highest professional standards. I also want to acknowledge the roles played by Sarosh Patel and Ms. Susan Kristie, our technical and administrative support team.

The technical co-sponsorship provided by the Institute of Electrical and Electronics Engineers (IEEE) and the University of Bridgeport is gratefully appreciated. I would like to express my thanks to Prof. Toshio Fukuda, Chair of the International Advisory Committee and the members of the SCSS Technical Program Committee including: Abdelaziz AlMulhem, Alex A. Aravind, Ana M. Madureira, Mostafa Aref, Mohamed Dekhil, Julius Dichter, Hamid Mcheick, Hani Hagras, Marian P. Kazmierkowski, Low K.S., Michael Lemmon, Rafa Al-Qutaish, Rodney G. Roberts, Sanjiv Rai, Samir Shah, Shivakumar Sastry, Natalia Romalis, Mohammed Younis, Tommaso Mazza, and Srini Ramaswamy.

The excellent contributions of the authors made this world-class document possible. Each paper received two to four reviews. The reviewers worked tirelessly under a tight schedule and their important work is gratefully appreciated. In particular, I want to acknowledge the contributions of the following individuals: Alejandro Regalado, Alexander Kalashnikov, Ali Moallemi, Arturo Mora-Soto, Chris Panagiotakopoulos, Daniela López De Luise, Daniele Magazzeni, Elias Pimenidis, Fazal Rahman, Fuensanta Medina-Dominguez, Hai Lin, Han-Ching Wu, Igor Miskovski, Jaak Tepandi, Jeongkyu Lee, Jing Li, Jin-tun Zhang, Jorge Ochoa Somuano, Kristina Zgodavova, Leandro Oliveira, Linfeng Zhang, Marian P. Kazmierkowski, Matti Koivisto, Michel Owayjan, Mirca Popa, Mohammad hadi zahedi, Mohammed Younis, Natalia Romalis, Nicoleta Liviana Tudor, Oleg Starostenko, Sandeep Sadanandan, Serguei Mokhov, Shivakumar Sastry, Steve Roach, Tamas Szabo, Veljo Sinivee, Virve Siirak, Xiang Hua Miao, Xiaojun Lu, Xiaopeng Zhang

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Bluetooth: A Case Study for Industrial Applications

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***Abstract* — Connectivity and physical mobility requirements for industrial applications have brought an exponential growth in wireless systems. In a production factory parameters like temperature, humidity level, vibration level and light etc., are to be monitored and controlled. In this study we provide a complete prototype wireless solution for such industrial applications through Bluetooth. A prototype system with a user interface is build and interfaced with a process model to monitor and control its parameters. The required driver software is also written for the communication between user interface and the plant. MSP430 microcontroller from Texas Instruments and Merlin BlueCOM embedded module are used in our implementation design. The communication is established between micro-controller MSP430 which continuously transmit/receive data from different sensors like temperature, air pressure, light and humidity of the process model and communicates via BlueCOM embedded module to the operator station (PC) using Bluetooth dongle. The idea is to use a BlueCOM module in an industrial plant to implement a cheaper and reliable system for monitoring and controlling the parameters efficiently. Our system provides excellent quality of service by utilizing the resources efficiently.**

***Key terms* — Bluetooth, Industrial Plant, Reliable, Prototype, Control, Monitor, Simulation.**

I. Introduction

In recent years, the desire for connectivity and physical mobility have provided a new momentum in wireless systems for industrial automation [1][2]. The information being communicated in industrial environments is typically state information. In normal operations it takes the form of recurring streams of small packets. At the same time, these packets are associated with harsh environments and may have strict timing requirements for critical tasks. In addition it may include extreme temperatures, high humidity levels, intense vibrations, explosive atmosphere, corrosive chemicals and excessive electromagnetic noise. Thus, in general, the required data throughput of the network is relatively low, however its reliability needs are high. In industrial environments, apart from lower installation and maintenance costs, the wireless connectivity offers ease of equipment upgrading [3]. Also practical deployment of mobile robotic systems and micro-electromechanical systems alike the process model which is used in our study is possible. In a production factory, where human activity is very high, good visibility and ventilation is of utmost importance, so is the room temperature and humidity level. Keeping in mind such parameters for monitoring and controlling we selected this process model to integrate with Bluetooth wireless system.

The selected parameters of the process model are *Light, Airflow, Temperature and Moisture*. Obviously a good control scheme is required for such a task. To achieve this goal we considered this small Process Model portraying an industrial plant to be an appropriate choice to build a prototype system. The system basically collects and analyzes the real time prevailing conditions of this plant. Our wireless solution design for industrial applications through Bluetooth is made up of software plus hardware and both have to be reliable as well as efficient. The hardware gathers and feeds data into the system, while software does the actual processing on the information, and data is displayed on the screen to the administrator (operator/supervisor) using Bluetooth wireless link.

The wireless solution for such type of industrial applications must have a human-machine interface to control and monitor. To have an effective monitoring of the processes data acquisition and historical functions

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must be incorporated in the setup, explained later in the text. The aim of this study is to build a real time wireless Bluetooth prototype system by incorporating a small process plant with real monitoring conditions.

II. PROCESS MODEL

The production plant (process model) to be monitored and controlled is shown in figure 1. The environment to be controlled is the glass chamber (production room). The whole set-up comprises of the process plant, communication link between the plant, MSP430 microcontroller and the operator station.

A. Parametrs to be Monitored and Controlled

The parameters are: Bulb (illumination), Humidity (moisture), Air flow and Temperature. The process model can be controlled in two modes local or remote:

Local mode: Controlled by four potentiometers at the control panel. They are heat, ventilator, electric bulb and valve to regulate both humidity and temperature levels in the plant.

Remote mode: Controlled by external electric control signals (0 – 20 mA / 0 – 5 volt) from any computerized system like MSP430 micro-controller etc.

To keep good and healthy conditions in production plants ventilation is key parameter to be maintained for good respiration of the workers. In this plant air circulation is controlled by a centrifugal ventilator. Good visibility is also of utmost important in industrial plants therefore a bulb is attached to a light dimmer controlled by (MV Light) and an illumination meter (XT Lux) to measure luminous intensity.

III. BLUETOOTH COMMUNICATION

For decades, industrial automation has used direct wire connections between the sensors, actuators and the controlling computers [4]. Wireless connectivity is a growing alternative for industrial automation connections. This connectivity has several benefits over wired connections for example cost, flexibility and portability [3]. Bluetooth has a tremendous potential in moving and synchronizing information in a localized setting. Potential for Bluetooth applications is huge e.g., by installing a Bluetooth network in industrial setting one can get away with the complex and tedious task of networking between the computing devices, yet having the power of connected devices. Additionally one is no longer bound to fixed locations in a network. Each Bluetooth device could be connected to 200 other devices making the connection with every other device possible. Since it supports both point to point and point to multipoint it virtually make connectivity of unlimited devices. One potential disadvantage using Bluetooth is that it operates in the free 2.4 GHz - 2.48 GHz band other devices like wireless LAN and microwaves also use this band therefore the band can get crowded.

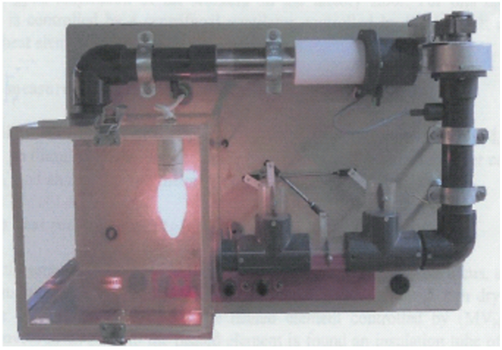

Figure 1: Process Plant

IV. SYSTEM DESCRIPTION

We incorporated the BlueCOM module (detailed description later in the text) with a microcontroller MSP430 to send and receive various parameters discussed earlier from the process plant to our controlling station (PC). The conceptual diagram of the system is shown in figure 2.

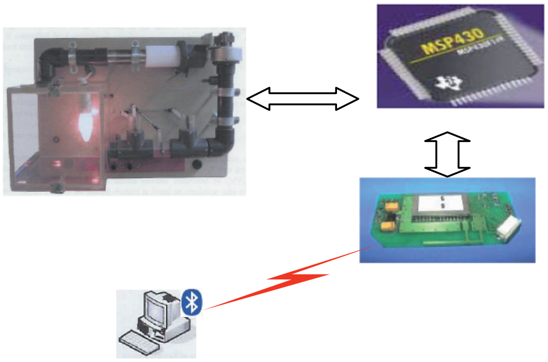

Figure 2: System Description

The BlueCOM is a compact device which makes it excellent device for large scale usage. Additionally, Merlin BlueCOM control unit is very low cost and have low power consumption. The system design has a microcontroller MSP430 to run the L2CAP layer and for lower layer commercial Bluetooth module is used. The process plant driver for communication with HCI interface is written. Which is involved with the Bluetooth Module in sending commands, receiving data and communication with the upper L2CAP protocol residing on the microcontroller. The detailed description of the block follows:

1. Serial Interface (RS232 port): This part consists of standard RS232 port. That enables the microcontroller to communicate with the data source therefore a complete driver software module is implemented for this communication.

2. Microcontroller (TI-MSP430F149): This is the control module of the implementation. It provides the functionality to the L2CAP layer of the Bluetooth stack [5][6]. L2CAP is an adaptation layer for hiding the base-band protocols from higher layer transport protocol and applications. Its functions are:-

Protocol Multiplexing: This allows the integration of various higher layer protocols e.g. TCP/IP, SDP, etc. That enables various protocols to be transmitted over the same channel. We do not need this functionality in our study and hence we have not implemented it.

Segmentation and Reassembly: From the data logger point of view, we need to take in chunks of data and send it across to central monitoring unit. Hence segmentation is essential and hence being implemented. The L2CAP layer ensures that the monitoring device's maximum limits does not exceed while transmitting the data.

Group Abstractions: It permits L2CAP to be efficiently mapped onto piconets for group protocols. It is not required in our study as we have only established a point-to-point connection.

Quality of Service: It is necessary to monitor the resource utilization and ensure that the quality of service between applications is honoured at all times. We have made sure through our software implementation that this requirement is reasonably fulfilled.

A. The Bluetooth Module

This is a commercially available module which has the lower layers of the stack (up to HCI) implemented on it. We have used Merlin Mid 56, compatible with the Bluetooth 1.1 specification as shown in figure 3. Antenna operates in the ISM Band and is part of the module. The board is complete with radio and voltage regulators and is directly connected to the MSP430. The MSP430x14x has two hardware universal synchronous/asynchronous transmit/receive (USART0 and USART1) peripheral modules for serial data communication. The USART supports synchronous SPI and asynchronous UART communication protocols, using double-buffered transmit and receive channels. The USART used on the MSP430 is USART0.

V. SYSTEM OPERATION & IMPLEMENTATION

Operator Station (PC) is continuously sending and receiving data from process model. An interface is required to display the data in proper format. Figure 4 shows the graphical user interface of this prototype system. Four parameters of process model are observed (monitored) and control is performed through two parameters. GUI consists of two main panels one (top) for monitoring and the other (bottom) for controlling.

The monitoring panel consists of four sub-panels one for each sensor of process model. The panel for temperature consists of a text field, a progress bar and three toggle buttons red, green and yellow. Text fields displays the value for example in the case of temperature sensor it shows value in degree centigrade. The progress bar shows the percentage value of temperature sensor. Three toggle buttons shows different levels of temperature sensor value in graphical format like red when temperature rise more than a defined level and it will start blinking, green when temperature level is normal (within defined range) and yellow start blinking when temperature level is lower compared with a predefined level. The same method and graphical components are used for other three sensors Air Flow, Moisture and Light. The controlling panel consists of two sub-panels one for air flow control and other for light control. Air flow panel consists of a text field and a slider bar. Text field is used to display the value of air flow, controlled by the slide bar. Slider bar is used to change air flow values (valve). An 8 bit D/A converter is used it means slider bar can have values in the range 0 – 255. The same method and graphical components are used for light panel.

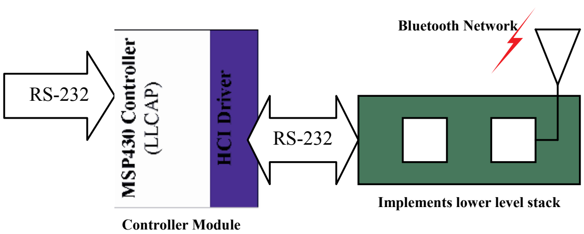

Figure 3: Bluetooth Module

A. Control Strategy for Microcontroller and PC

Microcontroller is continuously receiving data from different process model's sensors (temperature, air pressure, light and humidity) and transmitting them to the PC (operator Station) using Bluetooth wireless link. It is difficult to differentiate the data coming from different sensors as a data stream. It must be differentiated in both ways (PC to microcontroller and vice versa). Microcontroller receives values from four different sensors. To differentiate the sensors values an assignment code is used, as shown below:

tempCode = 0x10; //code for temperature value

pressureCode = 0x20; //code for Air Pressure value

moistureCode = 0x30; //code for Moisture value

lightCode = 0x40; //code for Light value

The assignment code is attached to the sensor value and send to the PC through wireless link. As 12 bit ADC converter is used however, memory control register is 16 bit therefore, actual value is only 12 LSB. The 4 MSB are set to zero at initialization time. First 8 bits are taken into a variable and passed to the putchar() function and then 8 bits of ADC value are shifted and the value is pushed into a new variable (thi). The XOR of thi value with tempCode is obtained and put into putchar(thi) function. Same method is used for other three sensor values but with different codes.

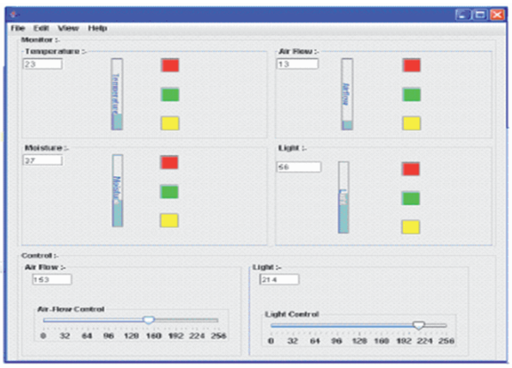

Figure 4: User Interface for Monitor-Control

B. *Operator Station's (PC) Control Strategy*

Operator Station is continuously receiving data from microcontroller on serial port. Code mask scheme is used as signature identification to differentiate the data from different sensors. Multiplexing can also be used, but during multiplexing data should be buffered to save the previous states, however process model does not have any built-in buffer. When data is available on serial port and an event is generated, the data is taken from the read buffer and put into the queue. When queue length is greater than 1, printNumber() function is called to display the value. Two variables, codeMask and numberMask have been declared in printNumber() function. The observations are removed from the queue and simple AND operation between codeMask/numberMask provide us the exact number.

C. *Airflow Measurement & Control*

The airflow transmitter FT which is connected to channel 6 pin 31 of the A/D board is for monitoring airflow in the chamber and is located in the air passage of the chamber. The model of the signal control and GUI on PC is shown below in the figure 5. The airflow velocity in the Process Model is transmitted through the FT and converted to voltage. This is further converted to bits and communicated to PC through wireless link. The voltage range in Process Model is 0 – 5 volts which is scaled to 0 – 2.5 to match with MSP30F149. Total numbers of bits are 4096, therefore 1 bit corresponds to 0.0006 v, air velocity range is 0 – 5 m/s. For example considering the air speed in the chamber to be 2 m/s.

Then 2m/s = (bits/4096)*5 which equals to 1640 bits

Voltage = 1640 bits = 1V

In the reverse case a motor is mounted in the plant which runs a blower to generate airflow in the system and this blower is connected to channel 6 pin 14 of the D/A board. It means to control the airflow in the system we should control the blower. The JSlider function is implemented in the user interface to change the value of blower. The manipulated value of the Airflow controller having a range of 0 – 1 (0 – 100 %) is fed into the MSP430F149. The output from the MSP430F149 is digital value which is passed to the D/A through channel 6 pin 32. The process can be analyzed by the following calculations;

1640 bits from the MSP430F149 controller

(1640/4096)*5m/s = 2 m/s

Voltage = (1640/4096)*2.5v which equals to 1 volt

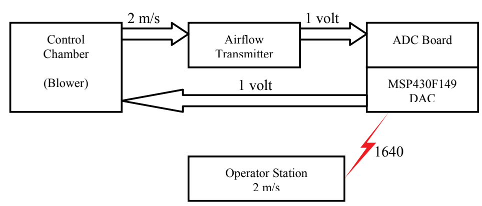

Figure 5: Airflow

The final voltage signal is sent to the blower as a preset value to control hence it means we are controlling the airflow of the Process Model. Figure 6 shows the delay in seconds to reach the required level for these parameters (required level is shown as average normalised values). Fifty samples are observed for these parameters; airflow takes 3 – 7 seconds, temperature takes 10 – 30 seconds and moisture takes 7 – 25 seconds to settle down to the required values. It is understandable that moisture takes similar amount of time as it is dependent both on airflow and temperature. However, light takes 30 – 35 milliseconds which is the same amount of delay a typical Bluetooth wireless connection encounters.

VI. CONCLUSION

Bluetooth has a tremendous potential as mentioned earlier in moving and synchronizing information in a localized setting. The Bluetooth technology can connect many peripherals wirelessly. The typical requirement for all industrial applications is the real time performance and in most data communications the objective is to send data over the established link without any errors. We have found that our simple process model for short distance works fine provided an acceptable delay range of about 0 – 30 seconds. The performance can be better perhaps with better sensors and optimized software implementation.

It has also been described in many references about the Bluetooth potential use in wireless transmission link even in quite hash industrial environment. Our proto-type implementation can be a functional unit in any industrial setting e.g., car manufacturing, airport luggage system etc. Although in our study we have not dealt with security and error correction schemes [7]. We would like to explore more in our future work and we are certain that more adaptive error correction protocols will increase the applicability and secure information will increase its potential in more areas of wireless connectivity. In the end we propose that our system can be expanded by cascading, such an arrangement could be very useful for example in a small industry setting where small scattered units are monitored/controlled through a single operator station.

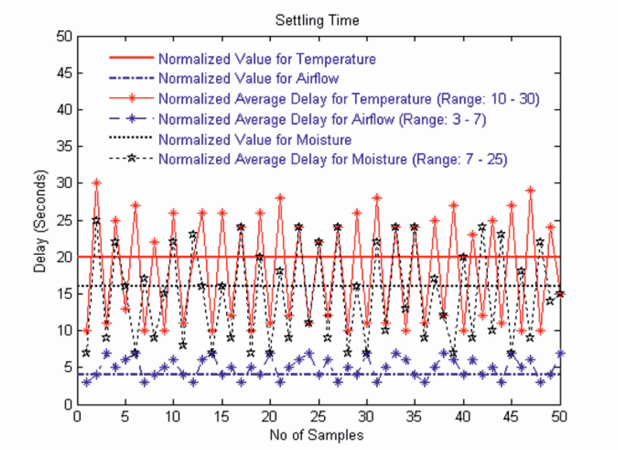

Figure 6: Time Delay

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An Enhanced Retinal Vessel Detection Algorithm

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***Abstract*- Retinal Vessel detection is one of the most studied fields of vessel detection. Study of retina is important as it can help to the diagnosis of many diseases. In this article a combinational algorithm which is based on Gabor wavelet and classification is modified. The modified algorithm gains more than 2% accuracy and outperforms the first algorithm.**

I. Introduction

Study of vascular and non-vascular tissues has been one of the main tools of diagnosis for many years. Study of retinal vasculature is important as it can help to the diagnosis of hypertension, diabetes, arteriosclerosis, cardiovascular disease and stroke. Computer aided disease diagnosis has spread as it eases the diagnosis, increase the accuracy and speeds up the whole process. Most of these techniques depend on algorithms realizing vessels from non-vessel parts of a tissue.

Different methods have been developed in case of vessel detection. These methods are grouped into three, Kernel based, trackers and classifier based methods[1]. In kernel based methods response of one or more kernels is used to detect the vessel. Most of kernel based algorithms require sort of post processing to enhance the kernel response and gain better accuracy. An algorithm that utilizes a tracker is the one which starts from one or more points and keeps tracks of features extracted by means of a model. The tracked features form the vessel. These methods mostly use a threshold to find the starting points that looks to be a vessel.

The classifier based methods extract a group of features that can be suitable for classifying the pixels of an image into vessel and non-vessel. The simplest form of these algorithms uses threshold based algorithms to detect the vessel.

Recently, enhanced algorithms are based on combination of these methods; a good example of these algorithms could be found in [2-5]. Combinational methods are center of attention because of their accuracy and optimum response. Combining kernel based methods and a supervised classification technique is the most well-known method of combinational group. In this case response of kernels is used as features of a classifier. There is always a triad off between accuracy and training time of the classifier. In this article we concentrate on a combinational method proposed by Soares et al[2] and it is shown that the result can be improved by modifying feature extraction part of the algorithm.

Next section will be an introduction to the Soares' method. In section III the improvement scheme is explained. Section IV contains the simulation results of the two algorithms and comparison of methods. The last part is dedicated to conclusion.

II. Soares' Method

Gabor wavelet response as feature vector and using a supervised classifier to classify the pixels of retinal images into vessel and non-vessel is the main idea of Soares et al[2]. In this section the major parts of the algorithm, feature extraction by use of Gabor wavelet and the classification technique is explained.

A. Feature Extraction

Soares et al uses continuous form of wavelet defined by(1), where c_ψ, ψ, b, θ and a denote normalization constant, analyzing wavelet, displacement vector, rotation angle and scale.

$$T_\psi(b,\theta,a) = c_\psi^{-1/2}\left\langle \psi_{b,\theta,a} \middle| f \right\rangle \tag{1}$$

Continuous wavelet can be implemented using Fourier transform described by(2).

$$T_\psi(b,\theta,a) = c_\psi^{-1/2}\int \exp(jkb)\hat{\psi}^*(ar_{-\theta}k)\hat{f}(k)d^2k \tag{2}$$

where $j=\sqrt{-1}, \hat{\psi}^*$ and $\hat{f}$ denote the Fourier transform. Soares et al used Gabor kernel to calculate the wavelet. Gabor is a directional kernel; this property makes it an excellent choice for vessel detection. Gabor wavelet is defined by(3).

$$\psi_G(x) = \exp(jk_0)\exp\left(-\frac{1}{2}|Ax|^2\right) \tag{3}$$

where $A = \mathrm{diag}\left[\varepsilon^{-1/2},1\right], \varepsilon \geq 1$ is a 2×2 diagonal matrix which contains the anisotropy of the filter and k_0 is a vector that defines the frequency of the complex exponential. Soares et al sets the ε parameter to 4 and $k=[0,3]$ to perform the calculations. They also used the maximum response of the wavelet over all possible orientations, which is calculated by(4).

$$M_{\psi(b,a)} = \max_\theta \left|T_\psi(b,\theta,a)\right| \tag{4}$$

Thus Gabor wavelet transform is computed for □ spanning from 0 up to 170° at steps of 10° and the maximum is taken.

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The maximum modulus of the wavelet transform over all angles is calculated for multiple scales are then taken as pixel features. Each image's extracted feature is normalized by its own mean and standard deviation by use of(5).

$$\hat{v}_i = \frac{v_i - \mu_i}{\sigma_i} \quad (5)$$

where $\hat{v}_i$ is the i_{th} normalized feature value, v_i is the i_{th} feature value, μ_i and σ_i are respectively mean and standard deviation of the i_{th} feature.

B. Classification

Classification can be performed by use of different tools. Histograms[6], threshold[7-10] and statistical classifiers[2, 3, 5, 11] are all methods of classification that has been used in case of retinal vessel detection.

Use of classification is based on the basis of classifying pixels into vessel and non-vessel groups. The classifier used by Soares et al is Gaussian Mixture Model. *G.M.M* is kind of Bayesian classifier which uses a linear combination of Gaussian functions as class-conditional probability density function[12]. Decision making is done using Bayesian rules defined by(6).

$$\begin{aligned} &\textit{decide } \text{vessel } \textit{if } P(v \mid \text{vessel}).P(\text{vessel}) > \\ &\qquad P(v \mid \text{nonvessel}).P(\text{nonvessel}) \qquad (6) \\ &\qquad \textit{else decide } \text{nonvessel} \end{aligned}$$

Number of Gaussians is an important factor in this method which results in different accuracy. Besides, it affects the time of training, more Gaussians takes more time to train.

III. Enhancement

Kernel-based methods have been used for vessel detection for a long time. Different kernels have been used in literature. One of the most famous kernels that have been used is Gabor kernel. This kernel sometimes is used indirectly, for example Vermeer et al[1] used a modified Laplace kernel which results in a Gabor Kernel. Definition of Gabor kernel given by (2) can be extended and written as(7).

$$\begin{aligned} &\psi_G = \exp\left\{-\left(\frac{x'^2}{\sigma_x^2} + \frac{y'^2}{\sigma_y^2}\right)\right\}\cos(2\pi x'/\lambda) \\ &\text{where } \begin{bmatrix} x' \\ y' \end{bmatrix} = \begin{bmatrix} \cos\theta & \sin\theta \\ -\sin\theta & \cos\theta \end{bmatrix}\begin{bmatrix} x \\ y \end{bmatrix} \end{aligned} \quad (7)$$

Zhang et al[7] used the above definition and studied the best possible parameters of the kernel, their studies shows for extracting a vessel with the width of *d*, while using window size of *h,* it is best to select $\lambda = 1.5h$, $\sigma_x = \sigma_y = 0.5\lambda$ and $0.5\lambda \geq d$. So by selecting the above definition and the suggested values of parameters, it is expected to get a better vessel realization. Beside the tuned Gabor kernel Zhang et al[7] suggests variance of responses over all possible orientations as defined by (8).

$$V_{\psi(b,a)} = \underset{\theta}{\text{var}} \left| T_\psi (b,\theta,a) \right| \quad (8)$$

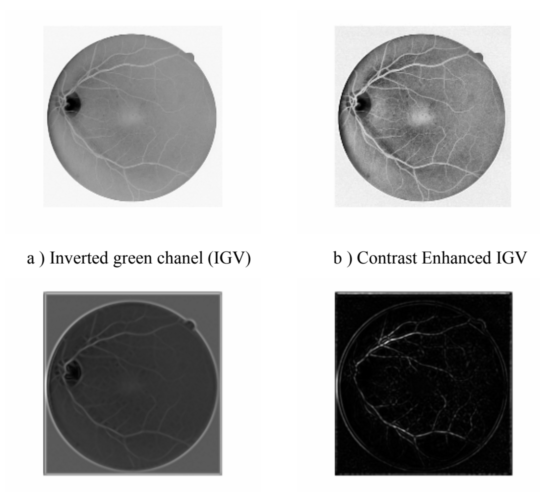
As Fig. 1 shows variance of responses results in a noise insensitive response in comparison to the maximum response. But it needs a preprocessing, such as adaptive contrast enhancement[13] to generate an acceptable response in which vessels can be fully realized.

So it is suggested to let the parameters be adaptive with the width of vessels instead of using a fixed set of values. Also using the variance of responses instead of maximum response is preferred. Using these suggestions the original algorithm was modified and tested on a dataset.

IV. Results

The two algorithms were implemented and tested on the DERIVE dataset. Four different scales of Gabor response plus the inverted green channel of the image were used as feature vector. One million random samples used to train the classification. The implementation was done by using Matlab 7, on a system with 2.4Ghz CPU and 1Gb of RAM. Fig. 2 shows the result of segmentation of the enhanced algorithm for a sample retinal image.

Table I, summarizes result of simulation for the two algorithms, as it is shown the enhanced algorithm that utilizes adaptive Gabor kernel and variance of response outperforms the other algorithm.



a) Inverted green chanel (IGV) b) Contrast Enhanced IGV

c) maximum Gabor response d) variance of response

Fig. 1. Response of Gabor over all orientations, comparison of maximum and variance of response

V. CONCLUSION

In this article it was shown by using the adaptive Gabor kernel and variance of response, instead of maximum response and fixed Gabor kernel in a combinational algorithm, the accuracy increases. In this case a gain of 2.36% is reached.

Fig. 2. Sample of vessel detection using the enhanced algorithm

TABLE I
ACCURACY EVALUATION OF ALGORITHMS
USING DERIVE DATA SET

Algorithm	Accuracy
Soares (K^*=2)	91.04%
Proposed algorithm (K^*=2)	93.40%

$^*K$ is the number of Gaussians used in *G.M.M.*

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A New Simulation Approach to Investigate Avalanche Behavior

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***Abstract*— Experiments revealed a dependence of sustained avalanche current on epitaxial layer structure, ambient temperature and chip size. With higher on-resistance, the avalanche current normally increases as well. A new mixed-mode simulation model proposed before explained the dependencies of avalanche current on device structure due to inherent instabilities. In this work, the simulation model was extended to explain found dependencies of avalanche current on chip size and therefore the chip layout and chosen external gate resistance.**

I. Introduction

There is an ongoing demand for low-voltage power MOSFET with low on-state resistance and good switching behavior. These devices are used for example in DC-DC power supplies, AC-DC adapters, Class-D amplifiers and motor drives. In all these applications, atypical switching conditions can occur, particularly during high voltage peaks, driving devices into the avalanche mode due to the presence of a small parasitic inductance.

Previous work was done to predict, by means of numerical simulations, the maximum avalanche current I_{as} that the transistor is able to sustain. It was possible to realistically simulate the strong dependence of avalanche current on process variations, especially for small values of the parasitic inductance [1].

Under such conditions the devices were usually destroyed at current values lower than the expected limit for the case of thermal destruction. Here, the device finally fails because the dissipated energy due to the large but homogenous current flow leads to a temperature at which the carrier concentration becomes too large and the device behaves intrinsic. In difference, avalanche events triggered by a small inductance often result in earlier destruction of the device due to the formation of current filaments, which is known as non-thermal destruction [2].

The developed model could be used to explain the avalanche behavior in dependence on process variations and therefore of the device structure. Of course, there are other effects taking influence on the avalanche ruggedness of a device. One is the often found dependence of the normalized avalanche current ($I_{as}/I_{nominal}$) a device is able to sustain on the chip size and external gate resistance. Thus, a larger chip shows a reduced avalanche current at lower gate resistance value compared to a smaller chip although the device structure is identical. If the external gate resistance value is higher, both devices are able to sustain a multiple of their respective nominal current.

Smaller values of the resistor correspond with smaller avalanche currents. In this work the model is extended to account for such effects. It can be used to predict the influence of switching behavior on avalanche ruggedness due to effects of inhomogeneous switching on the chip which commonly occur immediately after or even during the switching process with the transistor not yet fully switched off.

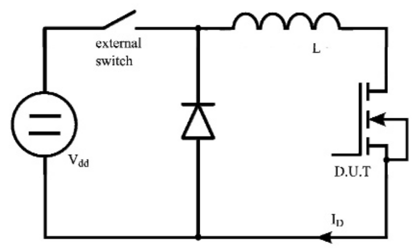

Fig. 1: Circuit to determine the UIS behavior of a MOSFET

II. Experimental Findings

To determine the maximum avalanche current the device is able to sustain under defined conditions, the UIS (Unclamped Inductive Switching) test is used. Fig. 1 shows the basic circuit as used for these measurements.

Typical current and voltage waveforms of an avalanche measurement are shown in Fig. 2. The current ramps up in proportion to the inductance in the circuit and the applied voltage. After turning off the device, the energy stored in the inductance must be dissipated in the transistor.

Since the current continues to flow through the inductance and cannot change instantaneously, the transistor is forced to maintain the current. Thus the voltage over the device increases until the stationary breakdown voltage is exceeded and the device enters the avalanche mode. In case of thermal destruction, the device fails before the current flow has ceased. In difference to that, non-thermal destruction occurs earlier during an avalanche event as shown in Fig. 3. In this example, the device is destroyed almost immediately and the voltage waveform just shows a peak. Non-thermal destruction is often

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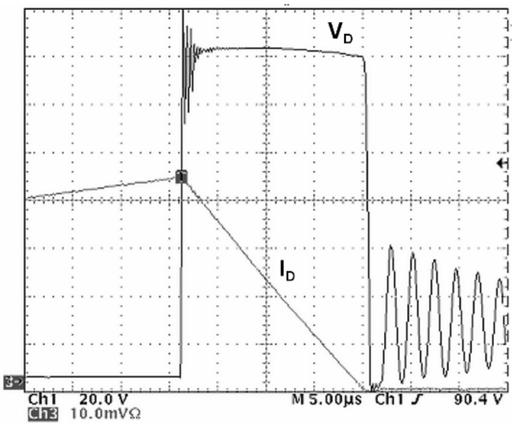


Fig. 2: Current and voltage waveform for a typical avalanche event

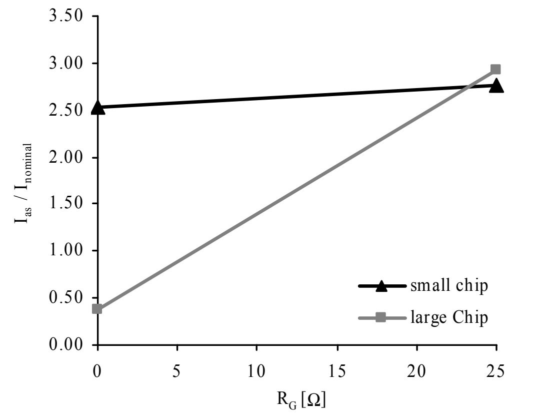


Fig. 4: Avalanche current in dependence of chip size and value of external gate resistor for small value of load inductance.

found in case of small values of the inductance L, which corresponds to large rates in current change di/dt as well as in relatively large current values itself.

Fig. 4 illustrates the dependence of avalanche current on chip area and value of the external gate resistor. While the small chip shows only a slight dependence on the value of gate resistor, the large chip degrades significantly.

To better understand the nature of these effects the chip layout must be taken into consideration. Fig. 5 shows the most important items of a typical chip layout.

We assume the cells are arranged in stripes (dark grey in the figure). Each stripe represents a trench in which a gate contact is included. In our example the gate stripes are filled with highly doped silicon and are contacted to a metal runner at the trench ends. These metal runners are connected to the gate pad which represents the interface to the external circuitry. Considering this layout one can assume that not all parts of the chip experience the same internal gate resistance. In our example, the region (2) sees a gate resistance R_{G2} consisting of the poly silicon R_{Poly2}, the metal runner R_{Metal2} and the metal runner R_{Metal1}. Region (1), on the other hand, has a reduced gate resistance consisting only of R_{Metal1}. In case of a switching event, Region (1) may respond faster compared to Region (2).

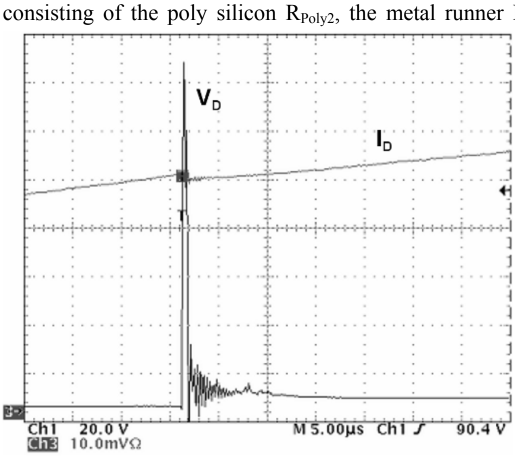


Fig. 3: Example for non-thermal destruction during avalanche mode

III. SIMULATION MODEL

A. Generalized Approach

Three general approaches in 2D simulations can be distinguished. For some effects, single cell simulations proved to be sufficient [1],[3]- [6]. It is also possible to model effects of filaments by means of multi-cell approaches, that is incorporating cell entities into one structure [7]. Additionally, simulation models using the mixed-mode simulation approach have been reported, e.g. [8]-[10]. They typically use two or more cells varying in size with one cell exhibiting a weakness in structure or other design parameters, e.g. higher body resistance or different geometry.

Based on the model presented in [1] and [11], we propose a new generalized model shown in Fig. 6. It consists of N basic

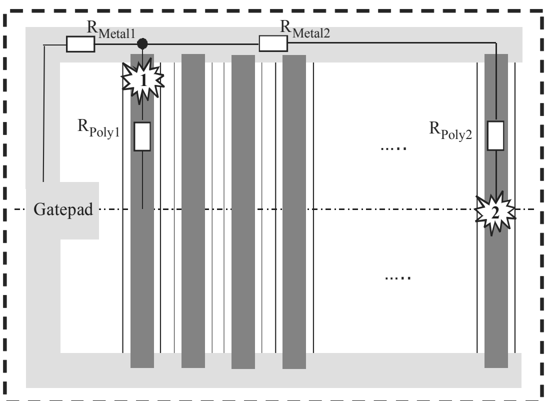


Fig. 5: Simplified model of chip layout

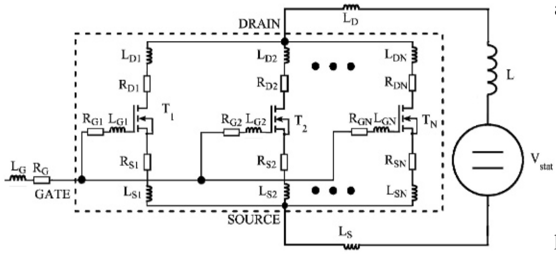


Fig. 6: Generalized simulation approach along with external circuitry

cells that may or may not be identical and various external and internal elements. A dashed rectangle symbolically separates the external circuitry from internal elements. External elements are related to the external circuitry including parasitic elements of the testing environment. Internal elements are introduced to model different effects related to the behavior of different parts on the chip as described before.

The internal resistive elements must be scaled according to the size of the cells to get similar results. We define the area ratio vector $\vec{Y} = (y_1 \quad y_2 \quad \ldots \quad y_N)$:

$$\vec{A} = \vec{Y} \cdot A_{chip}, \tag{1}$$

with

$$\vec{Y} \bullet \vec{1}^T = \sum_{i=1}^{N} y_i = 1, \tag{2}$$

meaning the sum of all cells is identical to the chip size. Then we can define the internal elements of cell i:

$$\vec{R}_i = \frac{1}{A_i} \cdot (R_{Di} \quad R_{Si} \quad R_{Gi}) \tag{3}$$

$$\vec{L}_i = \frac{1}{A_i} \cdot (L_{Di} \quad L_{Si} \quad L_{Gi}) \tag{4}$$

Another feature is the current inhomogeneity vector $\vec{X} = (x_1 \quad x_2 \quad \ldots \quad x_N)$:

$$\vec{J} = \vec{X} \cdot J_{total}, \tag{5}$$

with

$$\frac{1}{N} \cdot \vec{X} \bullet \vec{1}^T = \frac{1}{N} \sum_{i=1}^{N} x_i = 1 \tag{6}$$

It can be treated as a disturb signal imposed on the system. If instabilities exist they should become apparent in the cells with a (slightly) higher current density.

If all internal resistances and inductances are electrically equal:

$$R_{Xi} = R_{Xj} \quad \forall X, i, j \tag{7}$$

$$L_{Xi} = L_{Xj} \quad \forall X, i, j \tag{8}$$

then all cells show identical switching behavior. If this is not the case then asymmetries in the switching behavior are apparent.

Differences in the time response of the different cells can also lead to current inhomogeneities in this model. We assume a generalized dependency for the current density in cell i:

$$J_i = \frac{1}{A_i} K_i \cdot f_i\left(V_{DS_i}, V_{GS_i}\right), \tag{9}$$

with

$$K_i = \mu_n C_{ox} \frac{w_{ch,i}}{l_{ch,i}} \tag{10}$$

Here, μ_n is the electron mobility, C_{ox} the oxide capacity per device area, $w_{ch,i}$ the channel width and $l_{ch,i}$ the channel length of device i. (Note that $l_{ch,i}$ is also a function of V_{DSi}.)

Then the normalized current density of the m[th] cell calculates to:

$$\frac{J_m}{J_{total}} = \frac{1}{y_m} \cdot \frac{K_m \cdot f_m}{\sum_{i=1}^{N} K_i \cdot f_i}. \tag{11}$$

In every cell the maximum temperature T_{max} is extracted separately. Destruction is defined, if any T_{max} exceeds a defined critical temperature T_{crit} as discussed e.g. in [12]. In our analysis we considered a system of $N = 2$ cells according to Fig. 7 simplifying the generalized simulation model in Fig. 6 to focus solely on the influence of internal gate resistances.

The model elements defined in Eq. (1) to Eq. (6) are set to:

$$\vec{L}_i = \vec{0} \tag{12}$$

$$\vec{Y} = (1 - y_2 \quad y_2) \tag{13}$$

$$\vec{X} = (2 - x_2 \quad x_2) \tag{14}$$

$$\vec{R}_1 = \frac{1}{A_1}\left(\frac{\Delta V_{DS}}{(1 - y_2)(2 - x_2) J_{total} A_{chip}} \quad 0 \quad 0\right) \tag{15}$$

$$\vec{R}_2 = \frac{1}{A_2}(0 \quad 0 \quad 0) \tag{16}$$

This model proved to be sufficient to clarify various effects. Device simulations were performed using the device simulation software package MEDICI [13].

B. *Model Verification*

Firstly, the influence of area ratio parameter $\vec{Y}$ and the current inhomogeneity factor $\vec{X}$ apart from internal resistance value variations will be shown.

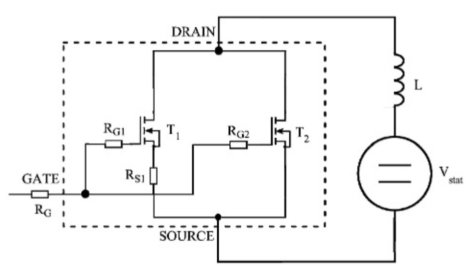
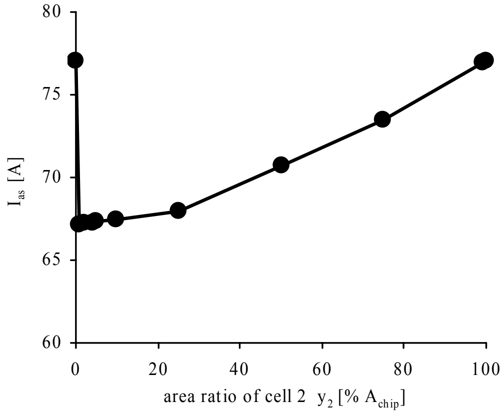
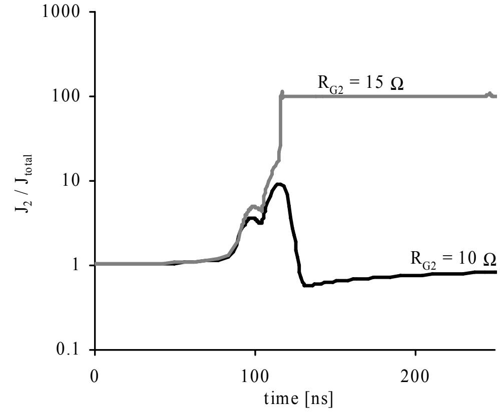


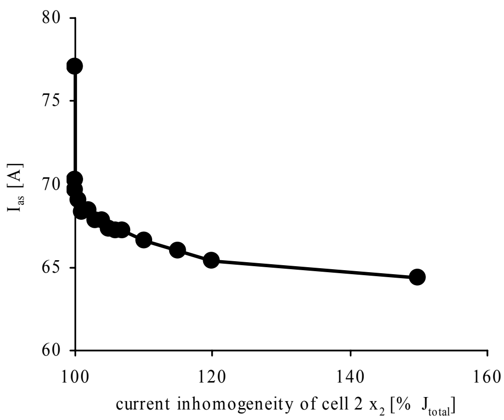
Fig. 7: Simplified simulation model



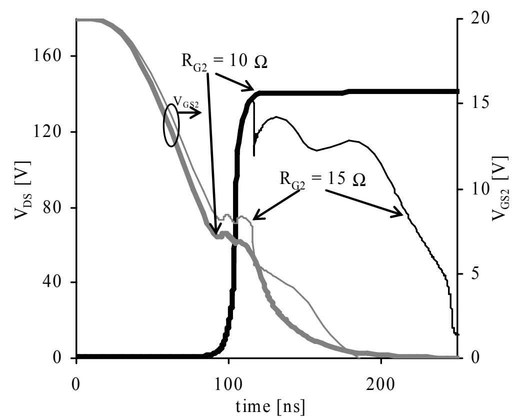
a)



a)



b)



b)

Fig. 8: Influence on simulated I_{as}
a) of area ratio of cell 2 with x_2 = 105%
b) of current inhomogeneity factor for cell 2 with y_2 = 1%

Fig. 9: Simulated current density ratio J_2/J_{total} in cell 2 a) and drain source voltage V_{DS} (b) for two different values of R_{G2}.

with ΔV_{DS} representing the drain source voltage difference between cell 1 and 2 due to either current inhomogeneity or different on-state resistances.

The dependence on the area ratio vector $\vec{Y}$ is shown in Fig. 8(a). If y_2 approaches 1 (or zero) the obtained avalanche current I_{as} is identical with the result obtained by means of single cell simulation. As expected, the current values obtained for $y_2 = 0\%$ and $y_2 = 100\%$ are identical, representing the result of a single cell simulation. The influence is obviously strongest if the "weaker" cell has the lowest possible area since current concentration is highest. For that reason, we chose an area ratio vector $\vec{Y} = (0.99 \quad 0.01)$, supposing the "weaker" cell 2 to be small.

Secondly, a reasonable current inhomogeneity vector $\vec{X}$ needed to be identified. For our purposes, an inhomogeneity factor of $\vec{X} = (0.95 \quad 1.05)$ was acceptable. This value was chosen somewhat arbitrarily to see an influence according to Fig. 8b). A deviation of current in a small part of device of 5% seemed to be realistic and no qualitatively different behavior could be achieved with other values of current inhomogeneity factor $\vec{X}$.

IV. Simulation Results

We are now able to relate the internal gate resistances to the actual resistance values on the chip. Regarding Fig. 5 we can set R_{G1} and R_{G2} as follows:

$$R_{G1} = R_{Metal_1} \tag{17}$$

$$R_{G2} = R_{Metal_1} + R_{Metal_2} + R_{Poly_2} \tag{18}$$

All simulations are done with the parameters $\vec{X}$ and $\vec{Y}$ discussed in the previous section:

$$\vec{X} = (0.95 \quad 1.05) \tag{19}$$

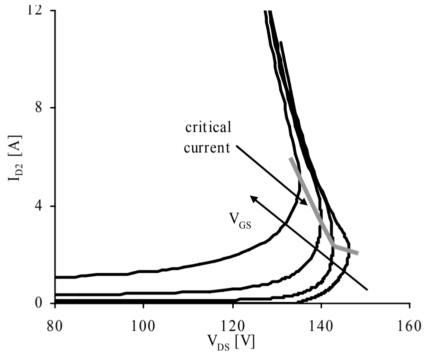


Fig. 10: Isothermal simulation (300 K) of drain current over drain voltage at varying gate voltages to extract the critical current. Grey line indicates the extracted critical current by connecting the snap back currents.

$$\vec{Y} = \begin{pmatrix} 0.99 & 0.01 \end{pmatrix} \tag{20}$$

Therefore, cell 2 is the smaller and "weaker" cell and should experience any existing instabilities.

To begin with, the effect of asymmetric switching will be presented. As already stated, different gate source voltages V_{GS} lead to different current densities on the chip. Fig. 9 depicts simulation results obtained for different internal gate resistances R_{G2} for a small constant value of R_{G1} = 1 mΩ.

Apparently, the switching asymmetry between the two cells increases as the internal gate resistance increases. If the current density exceeds a certain value as shown in Fig. 9(a) the device enters a critical region indicated by a sudden drop in drain source voltage shown in Fig. 9(b). A high value of current density ratio leads to destruction at lower total device currents since a small part of the device must sustain a very high current density and thus a very high temperature.

It is well known that devices with Negative Differential Conductivity (NDC) in their IV characteristics are prone to instabilities, e.g. oscillations or filaments [14]. This leads to a

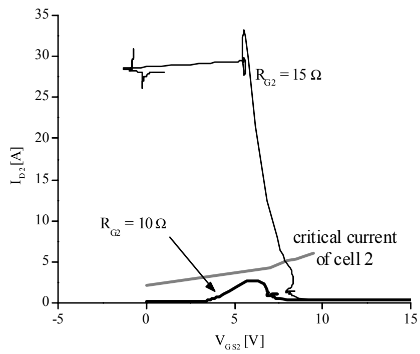


Fig. 11: Critical and actual current for different R_{G2} over gate source voltage. If actual current exceeds critical current a stable current filament develops.

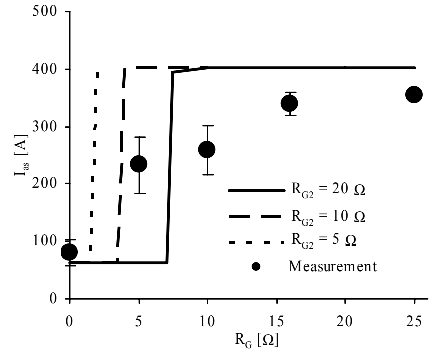


Fig. 12: Comparison of measurement and simulation of avalanche currents for a large chip. A larger (external) gate resistance leads to an improvement of avalanche behavior (L = 10 µH). R_{G1} was set to 1 mΩ

possible explanation of this effect: An applied gate voltage alters the blocking characteristics and hence the critical snap back current. For gate voltages far above the threshold voltage the channel is in strong inversion. This leads to additional impact ionization multiplication of saturation current in the space charge region [15]. The isothermal breakdown behavior can be determined if the blocking characteristics for different gate voltages are considered. In Fig. 10 the drain current I_{D2} of the small cell over drain source voltage V_{DS} is shown for various gate source voltages V_{GS2}. It can be seen that a certain drain current exists at which the drain source voltage starts to decrease for higher currents.

This current is defined as critical current since the drain source voltage reaches a maximum for this current and decreases for higher currents. This can be regarded as negative differential resistance leading to inherent instabilities. For a given device, it depends on gate source voltage and temperature. The grey line in Fig. 10 is the connection of the critical currents for different V_{GS} for a constant temperature of 300 K. This extracted critical current can be applied to determine the stability regime during transients. Considering again the switching transient in Fig. 9 we can depict the transient drain current over applied gate voltage thus eliminating the time dependency. This method is shown in Fig. 11. In the same graph, the critical current extracted as described above is shown. Thus, Fig. 11 provides a linkage between transient and quasi stationary simulations.

If the gate resistance of the small cell 2 is set to R_{G2} = 10 Ω, the current in cell 2 lies below the critical current, thus no destructive state can develop and the device safely turns off. If, on the other hand, R_{G2} is raised to a value of R_{G2} = 15 Ω, then the current I_2 in cell 2 continues to rise until all current flows through this cell thus leading to possible destructive temperatures.

This analysis was done for several chip currents and model settings. The results for a large chip, the potentially critical case, are shown in Fig. 12. Here, the external gate resistance

R_G was varied both in simulation and measurement. Measurements showed a strong degradation of avalanche current I_{as}. Simulations revealed the dependency of this degradation on internal gate resistance values.

Furthermore, a higher external gate resistance could remedy this effect. This in turn leads to slower switching behavior. Naturally, the effect was much stronger in simulation owing to the simplicity of the model. In experiment, no sudden drop in avalanche current I_{as} occurred for lower gate resistance values. The model nevertheless revealed the nature of this degradation.

V. Conclusion

Experimental results revealed a dependency of avalanche current on chip size and on external gate resistance.

A new model for the simulation of avalanche behavior was developed taking into account the influence of asymmetric switching on avalanche current. It was shown that this effect could lead to current concentration in small regions of the chip thus leading to destructively high temperatures even at current levels far below the thermal destruction limit.

This effect of asymmetric switching is especially apparent at high currents occurring at low load inductance values. It represents another destruction mechanism in the non-thermal regime compared to the well-known triggering of the parasitic bipolar transistor. A potential means to prevent the onset of destructive current concentration could be the enlargement of external gate resistance. In continuation we will study the influence of temperature and self-heating on this behavior. This also includes the consideration of other parasitic elements.

Acknowledgment

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Qualitative Choice and Logit Models Applied to Price Optimization

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Abstract - **Multinomial Logit Model can be implemented in a Qualitative Choice Situation in order to generate an optimal pricing policy. The analysis is based on a simple case where the seller has to define the best combination of prices to offer for two products, considering a single-type of costumers. Three alternative approaches are compared for the computation: Multinomial Logit, Binary Logit, and Two Stage or Logit-Logit. The analysis of the problem involves three phases: Simulation of sales data, Estimation of parameters and Price optimization.**

I. INTRODUCTION

The Revenue Management doctrine and dynamic pricing techniques were originally implemented in the 1980's for the Airline industry in order to allocate capacity and seat control, today these models and techniques have spread out to several other industries such as: car rental, hotels, retailers, etc. [1]. The generic problem can be stated as "selling the right product to the right customer at the right time, and for the right price [2]," therefore, pricing policies are a critical element to be considered in order to solve this issue. Price is one of the most effective variables that managers can manipulate not only to encourage or discourage demand, but also to regulate inventory and production pressures. Nowadays, with the rapid evolution of technologies and e-commerce, dynamic pricing policies and models have become increasingly more popular within the management community [3]; they are key drivers of the companies' performance. These new technologies along with strategies of distribution channels more and more sophisticated, have helped to create new opportunities and challenges for the dynamic pricing area, not only because changes of prices are made faster, more frequently, and with negligible costs, but also because prices have become more visible to consumers, since they can easily compare shopping alternatives with just the "click of a mouse." Customers choose among products based upon their preferences and the prices of all the alternatives offered.

A. Qualitative Choice Model (QCM)

A Qualitative choice situation [4] is one in which a decision-maker faces different options among which one has to be selected. Choice of an airline for example is a typical application of discrete choice models. The nature of the choice to be made depends upon the problem faced by the decision-maker. The restrictions placed upon the choices are:

a. Finite number of alternatives,
b. Alternatives are mutually exclusive, and
c. The set of alternatives is exhaustive

Consumers are utility maximizers. They select the product they believe has the highest overall utility or "happiness" from a competing set of products. The option chosen depends on the characteristics of the individual choices, which are influenced by habit, inertia, experience, advertising, peer pressure, environmental constraints, opinion, etc. Associated with each choice is the choice probability, and it can be specified as a parametric function of the following form [5]:

$$P_{in} = f(x_{in}, x_{jn} \;\; for\ all\ j\ in\ J_n\ and\ j \neq i, \;\; S_n, \beta) \qquad (1)$$

Where, x_{in} is the vector of characteristics of alternative i observed by decision-maker n, J_n is the set of all alternatives, S_n is the observed characteristics of decision-maker n such as income, age, etc., and β is a vector of parameters.

Specific qualitative choice models such as ***logit*** are obtained by specifying the function *f*. Multinomial Logit is a choice model of the logit type in which the number of choices is greater than two.

QCM are of use in a number of situations. From the choice of a particular set of routes to go to work, to purchasing a specific consumer product, the options faced by a decision-maker can usually be defined to meet the restrictions mentioned above.

B. The Logit Model

The logistic function, the S-curve, or Sigmoid curve how is also called some times, was invented in the 19th century as a result of an investigation of the growth of populations and chain reactions by Pierre-François Verhulst, and published between 1838 and 1847. Later, the function was re-discovered independently by Pearl and Reed in 1920 in another model of population growth. As shown below, the initial part of the curve shows an exponential growth, and after the inflection point, the curve shows a logarithmic growth.

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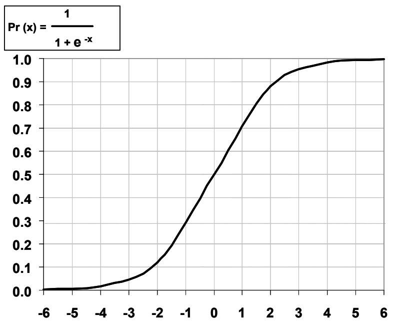

Fig. 1. Standard Logistic function (Sigmoid Curve)

The outcome is not a prediction but a probability of belonging to a particular state. Logit models become popular in 1974 after Daniel McFadden's derivation of the conditional Logit analysis used on his theory of discrete choice [6], which was the subject of his Nobel Prize in Economics in the year 2000. Even though Logit models were first introduced by Joseph Berkson in 1944, McFadden's derivation of the model was entirely new and was immediately recognized as a fundamental breakthrough. McFadden's theory of discrete choice emanates from microeconomic theory, according to which each individual chooses a specific alternative that maximizes his utility. However, as the researcher cannot observe all the variables affecting individual choices, he perceives a random variation across individuals with the same observed characteristics. On the basis of his new theory, McFadden developed micro econometric models that can be used, for example, to predict the share of a population that will choose different alternatives [7]. Suppose that each individual in a population faces a number, say *J,* of alternatives. Let *X* denote the characteristics associated with each alternative that the researcher can observe in his data. In a study of the choice of transportation mode, for example, where the alternatives may be car, bus or subway, *X* would then include information about time and costs. It could also include another different variable, S, to represent individual's characteristics such as age, income and education, but since we are considering only one type of customer, to simplify the problem, the variable S will be ignored. Other differences among individuals and alternatives, besides *X* and S, are unobservable to the researcher, and also determine an individual's utility-maximizing choice. Such characteristics are represented by random "error terms." McFadden assumed that these random errors have a specific statistical distribution in the population. Under these conditions, plus other technical assumptions, the probability or purchase likelihood (*PL*) that individual *i* will choose alternative *j* can be written as:

$$P_{ij} = \frac{e^{Uij}}{\sum_{j \in J_n} e^{U_{ij}}} \qquad (2)$$

Where,
U_{ij} is the utility of alternative j to customer i
J_n is the set of alternatives offered to customer n

In this multinomial Logit model, *e* is the base of the natural logarithm. In the data, the researcher can observe the variables *X*, as well as the alternative the individual chooses. The basic multinomial logit model assumes a particular form for the deterministic component of the utility U_{ij} for alternative j, for example linear in attributes, $U_{ij} = \beta^T x_{ij}$, where β is a vector of parameters and *x* is a vector of attributes for alternative j such as price of product. This form is then used in the calculation of choice probabilities according to the logit model. For a particular example of 2 products and one-type of customer, the linear utility functions associated to each product will be:

$$U_1(x_i) = \alpha_1 + \beta_1^1 P_1 + \beta_2^1 P_2 + \varepsilon_1 \qquad (3)$$

$$U_2(x_i) = \alpha_2 + \beta_1^2 P_1 + \beta_2^2 P_2 + \varepsilon_2 \qquad (4)$$

Where, Ui = Utility associate to consumption of product i; Pi= Price of product i; α, and β are parameters, and ε is the random error term. Then the probabilities that individual will choose each alternative can be written as:

$$\Pr(No\ Buy = 0 \mid x_i) = \quad P_{i0} = \frac{1}{1+e^{(U_1(x_i))}+e^{(U_2(x_i))}} \qquad (5)$$

$$\Pr(Buy = 1 \mid x_i) = \quad P_{i1} = \frac{e^{(U_1(x_i))}}{1+e^{(U_1(x_i))}+e^{(U_2(x_i))}} \qquad (6)$$

$$\Pr(Buy = 2 \mid x_i) = \quad P_{i2} = \frac{e^{(U_2(x_i))}}{1+e^{(U_1(x_i))}+e^{(U_2(x_i))}} \qquad (7)$$

C. Problem Description

A very simple situation will be simulated in order to have a better understanding of the theory and the practical problem involved by using QCM. Let's consider a situation where a single type of customer faces three choices for some specific product. For example consider a college student of certain age and income, who is shopping for RAM memory to upgrade his/her personal computer. The customer accesses an internet website specialized on this product, and after browsing for the memory required for a particular laptop, found out that he could decide between the following alternatives: (a) buy a memory chip of 128 MB for Price 1, (b) buy a memory chip of 256 MB for a price 2, or (c) no purchase. It is important to be aware of that the problem we are trying to focus on, is not from the perspective of the customer who is trying to select the best alternative according to his/her preferences and budget constraints, but from the seller's perspective who wants to select the best combination of prices that maximize his benefits. Consider also that the customer can make exactly one of the choices specified above, that is, they can purchase one unit of the product 1, product 2 or not buy anything. Then the idea is to model this situation as a qualitative choice problem using Logit models to formulate an optimization problem. The solution to this problem should specify the prices to be offered for each product for every customer. In other words, the customers select a product according to their preferences (utility), which depends on attributes of the alternatives (such as prices) and the objective is to obtain the set of prices that maximize the total expected revenue (sales) for the seller, which is given by:

$$\text{Max } Z = (PL_1 \times P_1 + PL_2 \times P_2) \quad (8)$$

PL_i is the Purchase likelihood of buying prod. i.

II. PROCEDURE IMPLEMENTATION

First and in order to compare results from three different approaches: Multinomial Logit, Simple Logit, and Two Stage or Logit-Logit, it is important to define a "theoretical optimum" which will be based on a given utility function. This theoretical optimum will provide the maximum benefit achievable for a combination of prices.

Appropriate values of parameters to be used in the utility function must be chosen by experimenting with the logit function. This is done by plotting the curves of purchase likelihoods (PLs) for both products based on different prices and using fixed values of alphas and betas. This experiment also will help to determine appropriate price ranges for the products. Using these parameters selected, we formulate the objective function to be maximized and solve it to estimate the optimal expected revenue.

Then Multinomial Logit Regressions are applied to sales data simulated (series of prices and decision made by the customer) to estimate the parameters of the utility function and thus the purchase likelihoods for both products. Based on these new parameters, the optimum is recalculated and compared with the theoretical one. The new optimum is computed for the models describe below:

1. Multinomial Logit Model: This model was previously described and under this approach, the equation (8) is optimized, based on utility functions and purchase likelihoods computed from (3) to (7).

2. Binary or Simple Logit Model (buy/no buy): In this case the dependent variable could any of the following two values (binary):

0 = No Buy, or 1 = Buy Product 1 or 2

The objective function to be optimized is the same as (8) and the utility function and purchase likelihood for each product are obtained using:

$$U(x_i) = \alpha + \beta_1 P_1 + \beta_2 P_2 + \varepsilon \quad (9)$$

$$\Pr(Buy) = \frac{e^{(U(x_i))}}{1 + e^{(U(x_i))}} \quad (10)$$

$$\Pr(No\ Buy) = \frac{1}{1 + e^{(U(x_i))}} \quad (11)$$

3. Two Stage Model or Logit-Logit: In this case the following two stages are applied:

Stage 1: Solve Logit model considering categorical values for Buy (either product 1 or 2), and No Buy. This stage should give the same results than the previous model of Simple Logit.

Stage 2: Given that they are buying, consider only the Buy sample and applied Logit model with categorical values for Buy Product 1, and Buy Product 2.

The objective function is as (8) and the purchase likelihoods are obtained using the following for each product:

$$PLi = PLi\,(\text{stage1}) \times PLi\,(\text{stage2}) \quad (12)$$

Different size data sets are considered to study the behavior of the three different models. The data sets are generated by a C++ simulation program described later. The optimization problem is solved using Jensen's add-in [8].

Theoretical Parameters Selection

Since there is just one-type of customer, then customer segmentation is not needed and therefore we only have to define a single set of parameters for each product to represent the customer. The process of determining this set of parameter values is essential, because these values will build the equations for purchase likelihood of each product. Having a reliable set of parameters will give a good representation of the customer behavior when prices of the products vary. By using a program specially designed for this purpose, we can observe the variation of the PLs of each product as the parameter values change. The calculation of PL is made by a Microsoft Excel macro that reads the beginning and ending values defined by the user for the parameter of interest. Starting from the beginning parameter value; the macro gathers all the parameters of the utility functions and steps through a loop which calculates a new set of purchase likelihood for each product. After recording the set of purchase likelihoods, the macro increments the parameter value for the next loop. The loop runs until the parameter of interests reaches the ending value set by the user. With a set of recorded purchase likelihoods, the macro creates a graph illustrating how the purchase likelihoods change when the user decides to vary a certain parameter or the price of a product. In the example below, we varied β_{11} from -0.0613 to 0.0107 to examine purchase likelihood. From the resulting graph for purchase likelihood against β_{11}, we decided to use a value for β_{11} between -0.0493 and -0.0333 because the purchase likelihood decisions are not too far apart.

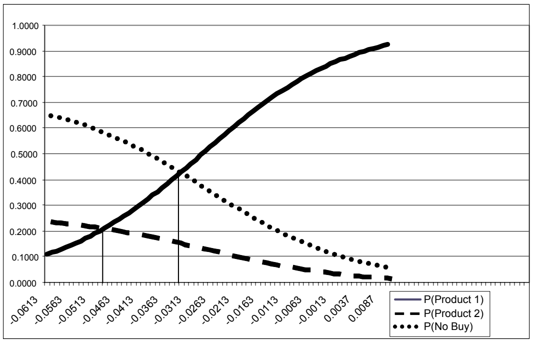

Fig. 2. Purchase likelihood against β_{11}

After we have the initial set of parameters, we proceed to check whether the PL made sense if we vary the selling price of the product. For example, we varied the price of the superior product, Product 1 from $5 to $115 while keeping the price of the inferior product, Product 2 fixed at $37. The resulting graph below shows that when Product 1 is priced below $79, the customer has a higher preference for Product 1 than for Product 2. However as the price of Product 1 continues to rise, the customer switches to Product 2 even though his choice of product is less superior than Product 1. Also, when the price of Product 1 was very low, even the probability of not buying was extremely low because the offer was too hard to resist.

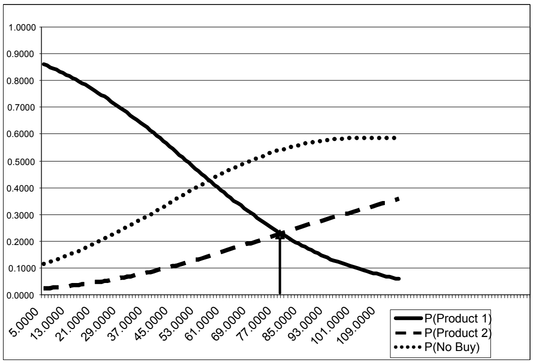

Fig. 3. Purchase likelihood against price of Product 1

We eventually came up with a set of parameters to characterize our customer. These parameter values are shown in the table below. Negative values corresponding to β_{11} and β_{22} imply that the customer reacts negatively to price increases for a product.

TABLE I
Parameter values with corresponding product

	α_1	β_{11}	β_{12}
Product 1	1.99	-0.039	0.0056
	α_2	β_{21}	β_{22}
Product 2	1.28	0.01	-0.079

III. SIMULATION METHODOLY

In this version of the simulation we have simply implemented the problem described above in C++. The purpose of this is to generate ***sales data*** which can be used on an initial model.

User Input:

a. Number of observations
b. Number of products
c. The min and max price of each product. The prices are modeled as a uniform random variable in each time period. Each customer faces a different set of prices for each alternatives
d. The values of the parameters

Control Logic:

a. In this program we have used the random number generator written by Pierre L'Ecuyer of Canada [9]. The output of the program is a Uniform (0, 1) random variable which is then transformed to the required distribution.

b. Once the user specifies the required input, the generation process is started.

c. The user input is through Excel. We have required that the user input any negative parameter with sign separately. This was due to an implementation problem that we encountered.

d. The Excel file is then saved as a text file and the program reads this data from the text file. The problem that we encountered was that the program would read the correct value of the parameter but not the sign. This was the reason we had to enter the sign separately.

e. For each customer the prices offered are different. They are generated as uniform random variates between min price and max price both of which are user specified for each product. The function that does this is *actprices*.

f. The function *compute PL* calculates the purchase likelihood or the choice probabilities for each product. Let us for example suppose that there are three choices corresponding to two products and no buy. The probability that each customer chooses, say, product 1 is calculated as:

$$P_1(S) = \frac{e^{utility(1)}}{e^{utility(1)} + e^{utility(2)} + e^{utility(0)}}$$

g. The utilities are calculated as shown in the previous section. Since the parameters are known and the price offered have been generated it is no problem to compute this for each customer.

h. These probabilities vary from customer to customer because each customer sees different prices. They are calculated for each customer inside the same loop. They are then written to a file where they are stored. Since in this case there are only two products three values are written to the file, $P_1(S)$, $P_2(S)$, $P_3(S)$. These values get overwritten each time the loop is executed. In other words at any point in the execution of the program only the choice probabilities of the current customer are stored.

i. The customer's choice is then decided by a uniform random number generated form a separate stream. If this value is between 0 and P1(S), the customer choice is 1. If it is between $P_1(S)$ and $P_1(S) + P_2(S)$, the choice is 2. If is between $P_1(S) + P_2(S)$ and 1, then the choice is no buy.

j. This function also writes the following fields to a text file. This text file can then be opened using any statistical analysis package for regression analysis. The fields written are: Customer Number, Prices of different choices, choice probabilities (optional), Buy/no buy and Actual Choice.

This function writes data for each customer and is called once for every customer. A sample of the output data is provided below.

TABLE II
Sample of Sales Data Simulated with C++

Trial	Price 1	Price 2	Decision (purchase)
1	84.766	42.183	0
2	79.062	37.873	0
3	73.543	34.273	1
4	74.430	44.609	1
5	65.902	38.933	0
6	85.291	31.921	0
7	79.154	36.452	1

IV ANALYSIS AND RESULTS

Forty data sets were generated for statistical analysis. Ten different seeds were used in the random number generator for four different sizes of data sets: 100, 500, 1000, and 20000 observations. This was done to compare the difference between the optimal solutions obtained by the different methods with increase in available data. The parameters obtained by logistic regressions over the 10 different replications for each data set were averaged to compute the optimum. All the logistic regressions and statistical analysis was carried out using the software SYSTAT. The regression results for multinomial logit model are shown below in Fig 4 and 5.

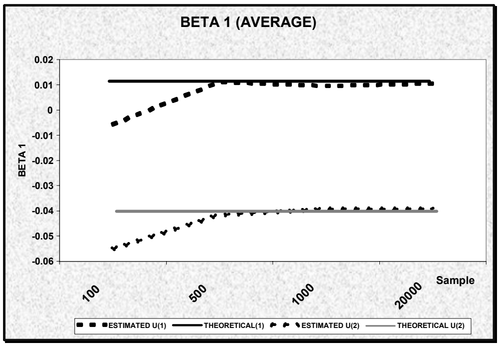

Fig. 4: Average of Alpha Values for Both Products

The figures represent the averages of the parameter values obtained from regression analysis for each data set. As can be observed the average of the parameter values tend to the actual parameter values as the size of the data set increases.

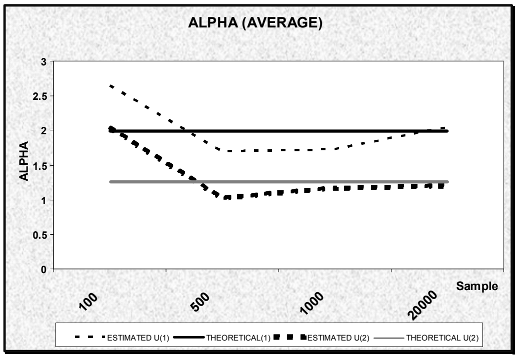

Fig. 5: Average of Beta1 Values for Both Products

A. Optimization Procedure.

We used Jensen's Excel Solver for Non-Linear Program to obtain the optimal expected revenue based on the prices of Product 1 and Product 2. Considering the parameter values obtained from a regression analysis of a particular sample size and seed, we ran an optimization procedure to obtain the optimal prices. We plugged the optimal prices into the multinomial logit model which contains the original parameters and we observe how the set of optimal prices perform with respect to the theoretical expected revenue. For each sample size, we took an average of the expected revenue when the optimal prices were plugged into the multinomial logit model with original parameters.

A total of 120 optimization procedures were computed considering four types of sample sizes, ten different data set of observations from each sample size, and three optimization models. The optimization problem for each approach can be stated as follow:

Decision Variables: $Price_1$ and $Price_2$

1. Multinomial Approach:
Objective: Max E (Revenue) = $Price_1 * PL_1 + Price_2 * PL_2$

Constraints:

$$PL_1 = \frac{e^{utility(1)}}{e^{utility(1)} + e^{utility(2)} + e^{utility(0)}}$$

$$PL_2 = \frac{e^{utility(2)}}{e^{utility(1)} + e^{utility(2)} + e^{utility(0)}}$$

$$p_l \leq price \leq p_u$$

Where, *Utility (1)* = $\alpha_1 + \beta_{11} * Price_1 + \beta_{12} * Price_2$,
Utility (2) = $\alpha_2 + \beta_{21} * Price_1 + \beta_{22} * Price_2$ and
Utility (0) is treated as the reference and set to 1.
P_l and P_u are vectors of upper and lower limits of prices for each product.

2. Binary Logit:

$$\text{Objective:Max E (Revenue)} = \left(\frac{\text{Price}_1 + \text{Price}_2}{2}\right)\frac{e^U}{e^U + 1}$$

$$\text{Constraints: } U = \alpha + (\text{Price}_1)\beta_1 + (\text{Price}_2)\beta_2$$

$$p_l \leq price \leq p_u$$

3. Two Stage or Logit-Logit:

Objective: Max E (Revenue) =

$$(\text{Price}_1)\frac{e^{U_1}}{e^{U_1}+1}\ \frac{e^{U_2}}{e^{U_2}+1} + (\text{Price}_2)\frac{e^{U_1}}{e^{U_1}+1}\ \frac{1}{e^{U_2}+1}$$

$$\text{Constraints: } U_1 = \alpha_1 + (\text{Price}_1)\beta_{12} + (\text{Price}_2)\beta_{12}$$

$$U_2 = \alpha_2 + (\text{Price}_1)\beta_{21} + (\text{Price}_2)\beta_{22}$$

$$p_l \leq price \leq p_u$$

V. CONCLUDING REMARKS

The following graph summarizes and compares the main results for each approach. It can be seen that the Multinomial Logit Model is the approach that produces the best results since it is the closest to the theoretical optimum, and its accuracy increases with the number of observations. Based on these results for this simple situation with two products and one-type of customer, we verified that in general discrete choice models can be implemented by using multinomial logistic regression techniques in order to produce coefficients and likelihood percentages to estimate choice probabilities. The model's output are utility scores that can be used to produce the desired optimal pricing. This initial approach offers plenty of scopes for further research as the problem to be considered approximates better to the true scenario of a customer choice.

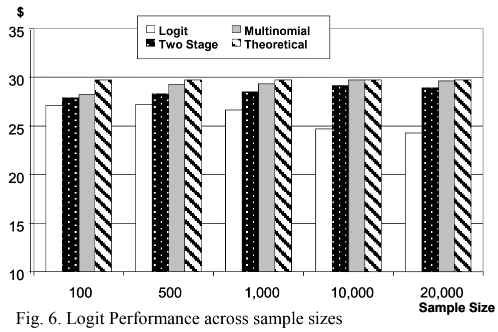

Fig. 6. Logit Performance across sample sizes

Some aspects of the problem that merit further research are: a) consider other attributes of products, and cross product of prices in regression model, b) Effect of Display Order, and effect of suggestions such as 'Most Bought' in the display page, c) Extend the model to several products and different customer segments, and d) Compare with continuous models such as linear regression.

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Distributed Recursive Algorithm for Auto Calibration in Drift Aware Wireless Sensor Networks

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Abstract - The purpose for wireless sensor networks is to deploy low cost sensors with sufficient computing and communication capabilities to support networked sensing applications. The emphasis on lower cost led to sensors that are less accurate and less reliable than their wired sensor counterparts. Sensors usually suffer from both random and systematic bias problems. Even when the sensors are properly calibrated at the time of their deployment, they develop drift in their readings leading to biased sensor measurements. The drift in this context is defined as a unidirectional long-term change in the sensor measurement. We assume that neighboring sensors have correlated measurements and that the instantiation of drift in a sensor is uncorrelated with other sensors. As an extension of our results in [1], and inspired by the resemblance of registration problem in radar target tracking, we propose a distributed recursive Bayesian algorithm for auto calibration of wireless sensors in the presence of slowly varying drifts. The algorithm detects and corrects sensor drifts and improves the reliability and the effective life of the network.

1. Introduction

Recently, Wireless Sensor Networks (WSNs) have emerged as an important research area [2]. This development has been encouraged by the dramatic advances in sensor technology, wireless communications, digital electronics and computer networks, enabling the development of low cost, low power, multi-functional sensor nodes that are small in size and communicate at short distances [3]. This led to a wide spectrum of possible military and civilian applications, such as, battlefield surveillance, home automation, smart environment forest fire detection, etc. A wireless sensor network can be defined as a collection of heterogeneous intelligent sensors distributed logically, spatially, or geographically over an environment and connected through a high speed network. The sensors may be cameras as vision sensors, microphones as audio sensors, ultrasonic sensors, infrared sensors, humidity sensors, light sensors, temperature sensors, pressure/force sensors, vibration sensors, radio activity sensors, seismic sensors [4], etc. The sensors collect measurement data from their respective environments. The collected data is processed by an associated processing unit that transmits it through an interconnected communication network. The information gathered from all parts of the sensor network is integrated using data-fusion strategies [5]. The information becomes useful for inference about the environment in which the sensors are deployed.

Sensor networks represent a significant improvement over traditional sensors. The wired sensors often require careful engineering to be deployed [6]. They are expensive and require frequent calibration. They have to be deployed in highly controlled monitoring environments, and they do not always provide real time data. They suffer from issues such as scalability. This makes them unsuitable for a large number of applications which demand greater temporal spatial resolution, lower cost and little effort in deployment and maintenance. On the contrary, the new sensors are cheap, light weight components that, as a group, can accomplish far more complex tasks and inferences than individual super node [7], [8]. On the down side, these wireless sensors are usually left unattended for long periods of time, which makes them prone to failure. This is due to either a running out of energy or to harsh environmental conditions surrounding them. These cheap sensors also tend to develop drift as they age. This poses a great problem from the end application point of view as the data from the network becomes progressively useless. An example of that is when an array of sensors is deployed on a bridge or other infrastructure to monitor stress. Other similar situations are aquatic environment monitoring, tracking and detection of targets, land mine detection etc. An early detection of such drift is essential for the successful operation of the sensor network. To address this problem we use the fact that neighboring sensors in a network observe correlated data. The measurements of one sensor are related to the measurements of its neighbors. Furthermore, the physical phenomenon that these sensors observe, follow laws with linear/nonlinear spatial correlation. Hence, in principle, it is possible to predict the data of one sensor using the data from other closely situated sensors [9], [1]. This predicted data provides a suitable basis to correct anomaly in a sensor's reported information. The early detection of anomalous data enables us not only to detect drift in sensor readings, but also to correct it. In this process, the sensors, which otherwise would have been deemed unusable, can continue to be used, thus prolonging the effective life span of the sensor network and optimizing the cost effectiveness of the solutions.

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Another common problem faced in large sensor networks is sensors suffering from bias or systematic errors. These errors have a direct impact on the effectiveness of the associated decision support systems. Calibrating these sensors to account for these errors is a costly and time consuming process. For example, applications like agriculture and aquatic environment monitoring, pH measurements and other water quality parameters usually suffer from bias errors. Traditionally such errors are corrected by site visits where an accurate, calibrated sensor is used to calibrate other sensors. The process is manually intensive and is only effective when the number of sensors deployed is small and the calibration is infrequent. In a large sensor network, constituted of low cost sensors, there is a need for frequent recalibration. Due to the size of such networks, it is impractical and cost prohibitive to manually calibrate them. Hence, there is a significant need for auto-calibration [1].

The problem of bias correction has been thoroughly studied in the context of multi-radar tracking problem. In target tacking literature the problem is usually referred to as the *Registration problem* [10], [11]. When the same target is observed by two sensors (radars) from two different angles, the data from those two sensors can be fused to estimate the bias in both sensors. In the context of image processing of moving objects, the problem is referred to as *Image Registration*, which is the process of overlaying two or more images of the same scene taken at different times, from different viewpoints, and/or by different cameras. It geometrically aligns two images, the reference and sensed images [12]. Image registration is a crucial step in all image analysis tasks in which the final information is gained from the combination of various data sources like in image fusion [13]. That is, in order to fuse two sensor readings, in this case two images, the readings must first be put into a common coordinates systems before being fused. The essential idea brought forth by the solution to the registration problem is the augmentation of the state vector with the bias components. In other words, the problem is enlarged to estimate not only the states of the targets, using the radars measurements for example, but also the biases of the radars. This is the approach we map in the case of sensor networks. Target tracking filters, in conjunction with sensor drift models are used to estimate the sensor drift in real time. The estimate is used for correction and feedback to the next estimation step. The presented methodology is a robust framework of auto calibration of sensors in a WSN.

2. Network Structure and Problem Statement

Consider a network of a large number of sensors distributed randomly, but evenly, in a certain area Fig. 1. The sensors are grouped in clusters (sub-networks) according to their spatial proximity. Each sensor measures its ambient temperature or any other parameter of interest such as a chemical concentration, a noise, an atmospheric parameter etc. The temperature, say, is considered to vary from one cluster to another but it is assumed to be approximately constant with distance within the cluster and it can vary with time. i.e. the temperature within the cluster is only function of time.

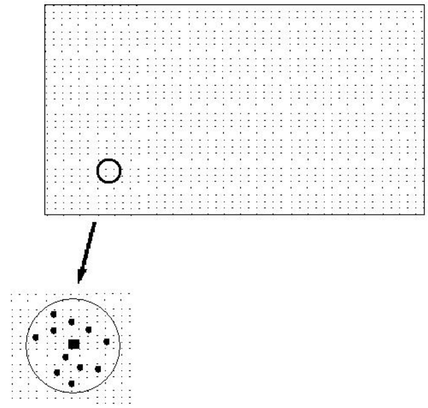

Fig. 1. Wireless Sensor Network Area with encircled sub-network

A cluster example is represented by a circle in Fig. 1. The sensors within the circle are considered to be able to communicate with each other, so each node collects the readings of all the nodes in the cluster. We also assume that the sensors are localized (know there coordinates). As time passes, some nodes will start to develop drift in their readings. If these readings were provided as such to the nodes, it would cause the network to accept erroneous conclusions. After some level of unreliability, the network inferences would not be trusted. At this turning point, the network would be deemed useless since it is impractical and unfeasible to manually recalibrate the sensors. Sensors locations are usually unknown to us or they are uneasy to be reached. To cope with the drift problem, the network nodes should be able detect their own drifts and correct them using the feedback they get from their neighboring nodes. This is based on the fact that the data of all nodes within the cluster are correlated and the faults or drifts instantiations are likely to be uncorrelated. The ability of the sensor nodes to detect and to correct their drifting will make the network's effective life longer. In addition to the drift problem, we also consider the inherent bias that may exist within some sensor nodes. There is a difference between the two errors as the former changes with time and often becomes accentuated, whereas the latter, is considered to be a starting error due possibly to a manufacturing defect or a faulty calibration.

3. Drift Correction

The sensor sub-network under consideration consists of N sensors deployed randomly in a certain area. Without loss of generality we chose a temperature sensor network even though this is applicable to all other types of sensors that suffer from drift and bias errors. Let T be the temperature. T varies with time, so we consider its values $T_1, T_2, \ldots, T_k$ at discrete times 1, 2, ... This time series is not necessarily smooth. At each step of the solution, T will be estimated, rather than predicted through the use of a mathematical model. At each time instant k, each node i in the sub-network makes a reading $r_{i,k}$ of T_k, $k = 1 \ldots$. It then reports a corrected value $x_{i,k}$ to its neighbors. If all nodes were perfect, $r_{i,k}$ would equal T_k for all nodes $i = 1, 2, ..., N$ and the reported values would be the readings, $x_{i,k} = r_{i,k}$. Each node computes the average value $\bar{x}_k = \sum_{i=1}^{N} x_{i,k} / N$. In a perfect situation, $\bar{x}_k = T_k$. Each reading has an associated reading error (noise), and drift $d_{i,k}$. This drift may be null or insignificant for a while, depending on the state of each node. The problem, for each node, is how to account for the drift, using $\bar{x}_k$, so that the reading $r_{i,k}$ is corrected and reported as $x_{i,k}$.

The drift we consider in this work is smooth as it is usually a slowly process, probably linear or exponential. It doesn't have sudden changes, surges or sharp peaks. Besides its dependence on the environmental conditions, it is strongly related to the manufacturing process of the sensor. This is what makes the instantiation of drift different from one sensor to another as we know that it is unlikely that two electronic components be the same unless they were in the same IC. Fig. 2 shows a variety of theoretical drift models that are smooth enough to be captured by a probability model such as the one we propose. So we use these drift models in our simulations. In an upcoming report, we will introduce probabilistic models to capture drifts and errors of a different nature that have surges and sudden escalations. We will also consider the solution to problems with drifts that are correlated to the sensor value.

3.1. *Iterative Drift Correction for Smooth Drift*

The solution to the smooth drift problem consists of the following iterative steps. At stage k, a reading $r_{i,k}$ is made by node i. Rather than sending that value automatically to its neighbors, the node is aware of its drift, and has an estimate for it at this stage. It is a projected value from an estimate of the drift made at the previous stage. Using this estimate of the drift, the node sends a corrected sensor value $x_{i,k}$ to its neighboring nodes. Each node then collects all the neighborhood values $\{x_{i,k}\}_{i=1}^{N}$, and computes the average $\bar{x}_k = \sum_{i=1}^{N} x_{i,k} / N$.

To estimate the drift of a node, the mathematical model of (1) is used. Assuming smoothness in the way the drift changes, the following model is adopted at first:

$$d_{i,k} = d_{i,k-1} + v_{i,k} \quad v_{i,k} \sim N(0, Q_k) \tag{1}$$

Equation (1) is a model that tracks a movement slow enough to be captured effectively. In Target tracking, (1) is called the target dynamics equation, a mathematical model that shows the target's dynamical behavior. In our case, the drift of the node is the target, and the purpose of the methodology is to track that drift's amplitude over time.

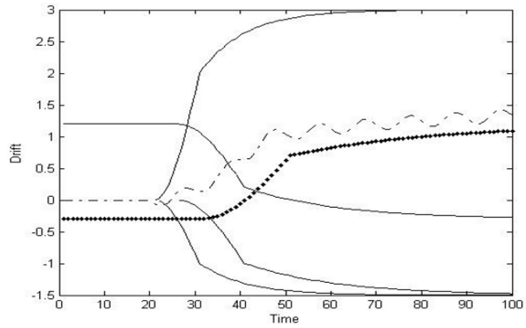

Fig .2. Examples of smooth drifts

A source of information is needed to provide input to a statistical model such as (1). In target tracking, a measurement equation is established to model observations made over time. These measurements provide the statistical information needed for the estimation procedure. In this case, the preferred source of information would be the real value of T, the quantity being sensed. For example, if T_k was available after the reading $r_{i,k}$ is made by node i, then the drift would be assessed exactly, it being $r_{i,k}$ - T_k. However, the whole purpose of the sub-network is to assess T, and so T_k will never be known. Only an estimate of it is available, and in this case, it is the average of the collected sensor reports, $\bar{x}_k$. Using this average as a sensible estimation, an approach used by [9], measurements $\{y_{i,k}\}$ are obtained, where

$$y_{i,k} = r_{i,k} - \bar{x}_k \tag{2}$$

This is based on the assumption that not all sensors will start drifting together, but more likely one sensor at a time will start drifting. This, with the addition that the nodes are self-correcting, makes $\bar{x}_k$ a good statistic for T_k. Equation (2) is derived from the relationship

$$r_{i,k} = T_k + d_{i,k}$$

Often, there is an error associated with the reading of a node, $w_{i,k} \sim N(0, R_k)$ leading to $r_{i,k} = T_k + d_{i,k} + w_{i,k}$. We can now write the measurement equation as

$$y_{i,k} = d_{i,k} + w_{i,k} \quad w_{i,k} \sim N(0, R_k) \tag{3}$$

Equations (1) and (3) form a Kalman Filter (KF) tracking set of equations. The solution to this problem is well known and has been widely applied to many real world problems. Here, it is adapted to the wireless sensor network technology, and to the drift problem of the sensors. This solution is implemented in a distributed iterative procedure i.e. it is run in each node to estimate its drift and then using this estimation; the following reading is corrected and so on. The procedure is summarized as follows:

Distributed Smooth Drift Algorithm

For each node i

- At time step k a predicted drift $\tilde{d}_{i,k}$ is available.
- Each node i obtains its reading $r_{i,k}$.
- The corrected reading is calculated, $x_{i,k} = r_{i,k} - \tilde{d}_{i,k}$ and then transmitted to the neighboring nodes.
- Each node computes $\bar{x}_k = \sum_{i=1}^{N} x_{i,k} / N$.
- The measurement $y_{i,k} = r_{i,k} - \bar{x}_k$ is obtained.
- The one step KF equations for each node i are

$$d_{i,k} = d_{i,k-1} + v_{i,k} \quad v_{i,k} \sim N(0, Q_k)$$
$$y_{i,k} = d_{i,k} + w_{i,k} \quad w_{i,k} \sim N(0, R_k)$$

resulting in the drift estimate $\hat{d}_{i,k}$, at time step k.

- The projected drift $\tilde{d}_{i,k+1} = \hat{d}_{i,k}$ is obtained and the algorithm reiterates.

In 1960, Rudolf Kalman published a landmark paper [14] on a recursive solution to an important class of estimation problems. This led to the well known Kalman Filter algorithm. KF is one of the most important estimation theory results of the twentieth century. It is the most widely applied algorithm for solving problems involving non-stationary random processes. It has been successfully used for the navigation of space vehicles, satellite orbit calculations, radar tracking, process control, statistical inference, econometric studies and a range of other applications. It is particularly well suited for tracking the state of a target. Its solution is well known and can be found in any textbook. In our scenario, it leads to the probabilistic solution, where the drift $d_{i,k} \sim N(\hat{d}_{i,k}, P_k)$ with a mean

$$\hat{d}_{i,k} = \hat{d}_{i,k-1} + \frac{P_{i,k-1} + Q_k}{P_{i,k-1} + Q_k + R_k} \quad (y_{i,k} - \tilde{d}_{i,k})$$

And variance

$$P_{i,k} = (P_{i,k-1} + Q_k)(1 - \frac{P_{i,k-1} + Q_k}{P_{i,k-1} + Q_k + R_k})$$

$\tilde{d}_{i,k}$ is the predicted drift at the beginning of stage k, before the correction. In this case, $\tilde{d}_{i,k} = \hat{d}_{i,k-1}$, a straightforward prediction given by the KF solution. Note that $(y_{i,k} - \tilde{d}_{i,k}) = x_{i,k} - \bar{x}_{i,k}$. The variances Q_k, R_k, $P_{i,k-1}$ and $P_{i,k}$ are numbers, and therefore the solution is easy to compute. Once $\hat{d}_{i,k}$ is found, it is used as the predicted drift $\tilde{d}_{i,k+1}$ for the next stage. This allows for the correction of reading $r_{i,k+1}$.

4. SIMULATION RESULTS

We simulate a small sub-network of 10 nodes measuring the temperature in a certain area. We assume that 2 sensors are developing drifts of the forms mentioned above. The tracking system performs well. Fig. 3 shows the reading of node 1, the actual temperature and the corresponding estimated value. Fig.4 shows the actual drifts we induced in nodes 1 and node 2 and shows the estimated drifts in both nodes. It is clear that our algorithm is capable of tracking the actual drifts and eventually estimating the actual temperature.

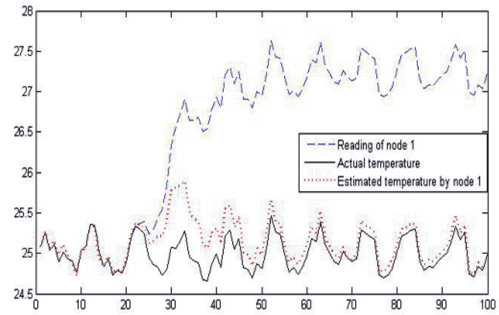

Fig .3. Tracking Actual temperature, sensor reading and corrected reading

We conducted several simulation scenarios and observed that the method works as long as not all sensor drifts start together, which agrees with our initial assumption that drifts and sensor errors are not correlated. Fig 5 shows the actual drift and the estimated drift in sensor 1 when 7 out of 10 sensors have developed drift. It is clear that the system can track drift even when 70% of the sensors are drifting. But comparing fig 5 of the 7 drifting sensors with fig 4 of the 2 drifting sensors we see that the error increases as the percentage of the number of drifting sensors increases. Our simulation results also show that this distributed recursive algorithm has reduced the measurement error to 93% when 2 sensors out of 10 are drifting and by 75% when 7 sensors are drifting. Therefore; the intuitive idea of correcting sensor errors immediately and in a distributed procedure, prolongs the life of the network.

5. CONCLUSIONS AND FUTURE WORK

In this proposed solution we have introduced a formal statistical procedure for estimating sensor errors in a WSN. Our proposed solution exploits the notion that measurements errors due to faulty equipment are likely to be uncorrelated. As in the sensor registration problem in [10] and [11], the state is augmented with the bias and tracked using a filter.

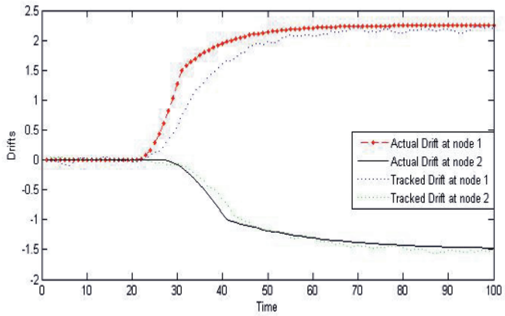

Fig .4. Actual and estimated drifts in nodes 1 and 2 with 2 drifting nodes

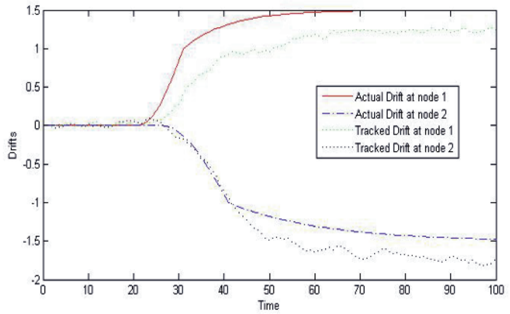

Fig .5. Actual and estimated drifts in nodes 1 and 2 with 7 drifting nodes

The solution is computationally simple using an averaged sensing value as in [9] and the Kalman Filter iterative procedure, allowing its implementation in a WSN. The proposed solution is completely distributed – each node only needs to obtain information from its neighboring sensors and then apply the Kalman Filter iterative procedure.

We consider at first a smooth drift model. The simulation shows, as expected, a high performance of the model. The simulation conducted showed that, with our distributed recursive algorithm, sensors readings errors can be reduced by more than 75 percent when 70 percent of the sensors in the sub-network are drifting and more than 95 percent when fewer sensors are developing drift.

In introducing this solution, our aim is to initiate methodical research in the drift of WSN nodes. The problem is significant in practice, but not much has been done to address it. We are currently conducting research on drift models which will join the presented methodology in starting to address drift in wireless sensor networks.

In an upcoming report, we will introduce probabilistic models to capture drifts and errors of a different nature that have surges and sudden escalations. We will also consider the solution to problems with drifts that are correlated to sensors value. We will also address the problem when temperature in the sub-network varies with distance and time.

ACKNOWLEDGEMENT

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A Multi-Camera Active-Vision System for Dynamic Form Recognition

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***Abstract* – A novel agent-based sensing-system reconfiguration methodology is proposed for the surveillance of time-varying geometry objects/subjects (*targets*). The methodology advocates the use of a multi-camera active-vision system for improved performance via the selection of optimal viewpoints along a time horizon – i.e., maximize the visibility of target's time-varying form as it moves through a cluttered dynamic environment. Extensive simulated experiments have clearly shown the potential performance gains.**

I. Introduction

Surveillance is commonly defined as the collection and analysis of data for object/subject pose (position and orientation) estimation, as well as identification. Form recognition can also be categorized as a surveillance task. A tactic frequently employed in surveillance is *sensing-system planning* (also known as *sensor planning* or *sensing-system reconfiguration*). Sensing-system reconfiguration is defined as the practice of selecting the number, types, locations, and internal parameters of sensors employed in the surveillance of objects/subjects (*targets*) through a formal method (e.g., [1]-[2]). The objective is to improve data quality and, thus, reduce the uncertainty inherent in the sensing process.

An effective real-time surveillance system must be able to cope with the presence of multiple static or dynamic (maneuvering) targets and obstacles as part of its sensing solution (e.g., [3]). As such, viewpoints are differentiated in terms of the useful target data that can be extracted with a given amount of effort.

In the above context, the focus of this paper is sensing-system reconfiguration, as restricted to the on-line modification of camera poses in a multi-camera active-vision system, for the real-time surveillance of time-varying-geometry targets (such as humans), in dynamic, cluttered environments.

A. Sensing-System Planning

Earlier works in sensor planning have typically focused on determining the configuration of a given set of sensors (with known capabilities) for static environments as well as targets with only fixed (time-invariant) geometries. In the survey paper [1], planning methods are characterized as either *generate-and-test* or *synthesis*. A generate-and-test method evaluates possible configurations with respect to task constraints, discretizing the domain to limit the number of configurations that must be considered. An established example of such a method is the HEAVEN system [4], which uses a discretized virtual sphere around the Object of Interest (OoI) to determine un-occluded, achievable poses for a single sensor.

Synthesis methods, on the other hand, characterize task requirements analytically, and determine sensor poses by finding a solution to the set of constraints presented. These systems are often application-specific, as in [5], where the planner synthesizes a region of viewpoints by imposing 3D position bounds from task constraints. In [6], points on the outer edge of an OoI form a virtual box, which the camera must be positioned to view while minimizing the local distortion in the image. In [7], an agent-based system is proposed to allow intelligent feedback between multiple sensors, reducing data redundancy and improving surface coverage of the OoI. The scene is static and individual sensors use a generate-and-test method to evaluate potential poses.

A natural extension to the static environment sensing problem is the consideration of moving OoIs, obstacles, and sensors, as well as the possibility for continuous reconfiguration of sensor poses – *active sensing*. For example, in [8], an 11-camera system was used to examine the effects of viewpoint on recognition rates for human gait. It was determined that the traditional static, off-line single-camera placement methods may lead to poor performance for multi-camera cases [9]. Recently, agent-based planning methods were also applied to the on-line sensing-system reconfiguration problem (e.g., [3], [10], and [11]).

In fully dynamic environments, the obstacles themselves may also be moving. In [12], for example, a method for controlling a team of mobile robots equipped with cameras was presented to optimize data quality. The goal was addressed as an optimization problem with consideration to the *Next-Best-View* (NBV) aspect. Attention-based behavior (e.g., [13]-[15]), where the attention of the system is focused on a single target until the vision task is performed, can also be used to reduce a multi-target problem to a single-target observation.

B. Form and Action Recognition

The recognition of time-varying geometry targets includes the identification of both static forms and sequences of

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movements. In current research, the vast majority of such time-varying geometry targets are humans, although a few papers have focused on simple artificial shapes (i.e., geometric primitives). Identification of even a single static form requires an existing database of characteristic data for known poses. Thus, past works have focused on merely reconstructing the model of the unknown object. For example, in [16], an active-vision system is used to explore a static scene to reconstruct the shape of a geometric primitive. Furthermore, most existing works in time-varying geometry object-form recognition have considered sensor poses as an unchangeable constraint – no formal planning, either off-line or on-line, is applied.

Many objects also exhibit specific, repeatable sequences of form (motions or actions) that one might wish to recognize. In template matching, for example, input images are compared directly to stored templates and multiple pose matches over time form a sequence template (e.g., [17]).

Semantic approaches are analogous to template matching, except the data used in the templates is high-level object configuration data. These are model-based approaches, in that a high-level representation of the OoI may be constructed. In [18], the geometry of a human body is recovered over several frames and in the presence of occlusions.

Statistical approaches attempt to reduce the dimensionality of matching through statistical operations on the template database. For instance, in [19], the authors seek to identify what particular information is most important in identifying a positive match to a template. Analysis of Variance (ANOVA) is used to identify features that highlight differences between subjects. Then, Principal Component Analysis (PCA) is used to reduce the data set to a lower dimensionality for matching.

In the above works, sensing-system reconfiguration was not considered as part of the methodology – input data was taken to be fixed, with no opportunity for quality improvement.

C. Time-Varying Geometry Objects

The use of sensing-system reconfiguration for time-varying geometry objects introduces additional complications to the process, such as non-uniform importance, partial occlusions, self-occlusions, and continuous surveillance. While the constraints do not invalidate the use of past work on sensing-system reconfiguration for time-varying geometry objects, they do necessitate a novel framework designed specifically to cope with these constraints. The goal of this paper is, thus, to demonstrate that (1) sensing-system reconfiguration techniques can tangibly benefit recognition performance for time-varying geometry objects, and that (2) factors not addressed by majority of past works must be addressed for improved recognition.

II. Problem Formulation

From the literature review above, one can conclude that an active-vision system using multiple, mobile cameras may provide autonomous surveillance of a single, dynamic Object of Interest (OoI), as it moves within the confines of a workspace on an initially unknown path (e.g., [15]). The workspace may also be cluttered with multiple, dynamic obstacles (with unknown movement paths) that can impede surveillance of the target. As such, a system that can perform the following tasks, given such a dynamic, multi-object environment, is necessary:

- *Detection*: All objects in the scene must be detected and categorized as either the OoI or obstacle upon entering the workspace.
- *Tracking*: Each object must be tracked, and an estimate of its future pose maintained.
- *Reconfiguration*: Given historical, current, and predicted data about the OoI and obstacles, an achievable set of poses for all sensors that minimizes uncertainty for the surveillance task at hand must be determined.
- *Recognition*: Data from all sensors must be fused into a single estimate of the object's current geometry. A further estimate must reconcile this geometry data with historical data to determine the current action of the target.
- *Real-time operation*: All operations must be limited in computational complexity and depth, such that real-time operation of the system is not compromised.
- *Robustness*: The system must be robust to faults, and the likelihood of false identification or classification must be minimized.

The performance of a surveillance system can be characterized by the success of the vision task in recognizing the target form and its current action. This task depends primarily on the quantity and quality of the sensor data that is collected, characterized herein by a visibility metric, V. This metric in turn depends on the current form of the OoI, the poses of obstacles, the poses of the sensors, and the pose of the OoI. However, the only variables that the sensing system has direct control over are the poses of the cameras. The visibility metric for the i^{th} camera at the j^{th} demand instant, t_j, is expressed herein as a function of $\boldsymbol{p}^j_{S_i}$, the pose of the i^{th} sensor, S_i, at the j^{th} instant:

$$V_i^j = f_i^j(\boldsymbol{p}^j_{S_i}), \text{ and} \quad (1)$$

$$\boldsymbol{p} = [x \quad y \quad z \quad \varphi \quad \psi \quad \theta], \quad (2)$$

where pose is defined as a 6D vector representing position (x, y, z) and orientation (φ, ψ, θ).

This paper proposes a global formulation of the reconfiguration problem for a sensing-system with n_{sens} sensors, n_{obs} obstacles, and with prediction over the time horizon ending at the m^{th} demand instant:

For each demand instant, t_j, j=1…m, perform the following:

For each sensor, S_i, i=1…n_{sens}, solve the following:

Given:

$\boldsymbol{p}^0_{S_i}$, $\boldsymbol{p}^0_{OoI}$, $\boldsymbol{u}^0$, $\boldsymbol{p}^0_{obs_k}$; $k = 1$ to n_{obs}

Maximize:

$$Pr = g\left(V_i^l\right);\ l = 1 \text{ to } j \quad (3)$$

Subject to:

$\boldsymbol{p}_{S_i}^1 \in P_i$

$\boldsymbol{p}_{S_i}^1 \in A_i^1$

$V_i^l \geq V_{min}$; $l = 1$ to j

End of loop.

Continue while: $t_{proc} < t_{max}$,

where $\boldsymbol{p}_{OoI}^j$ is the OoI pose at the j^{th} demand instant, $\boldsymbol{p}_{obs_i}^j$ is the pose of the i^{th} obstacle at the j^{th} demand instant, $\boldsymbol{u}^j$ is the feature vector of the OoI at the j^{th} demand instant, P_i is the discretized set of feasible sensor poses for the i^{th} sensor, A_i^j is the discretized set of achievable sensor poses for the i^{th} sensor at the j^{th} demand instant, V_{min} refers to a user-defined threshold of minimum visibility, t_{proc} is the time spent processing data, and t_{max} is the maximum time before a final pose set must be chosen. The two sets, P_i and A_i^j, are governed by:

$$A_i^j \subseteq P_i \tag{4}$$

$$\boldsymbol{p} \in P \text{ iff. } \boldsymbol{P}_{low} \leq \boldsymbol{p} \leq \boldsymbol{P}_{upp} \tag{5}$$

where $\boldsymbol{P}_{low}$ and $\boldsymbol{P}_{upp}$ are the lower and upper movement limits of the sensor motion. The determination of the subset $A_i^j \subseteq P_i$ depends on the model of motion used, and is not specified.

The proposed objective function, *performance*, or *Pr*, depends on the visibility metric of each sensor at all demand instants in the horizon. It is a measure of success in achieving the sensing objective [15]. Overall, the proposed formulation seeks first to maximize visibility of the OoI at the immediate future demand instant, t_1, for all sensors. If sufficient time remains, the system seeks to maximize expected visibility at t_1 and t_2, then, t_1, t_2, and t_3, and so on. As such, a higher overall metric value may be achieved at later demand instants, at the expense of near-future visibility. This trade-off can be controlled externally by adjusting the minimum desired visibility and the sensor assignment at each demand instant. More weight can be assigned to nearer demand instants in order to minimize exposure to future uncertainties in estimated poses. Computational complexity is bounded, though the determination of poses at each future instant does depend on the poses determined for each previous instant, back to the current time, t_0.

III. Proposed Methodology

An agent-based approach is proposed for sensing-system reconfiguration for the surveillance of time-varying geometry objects, Fig. 1.

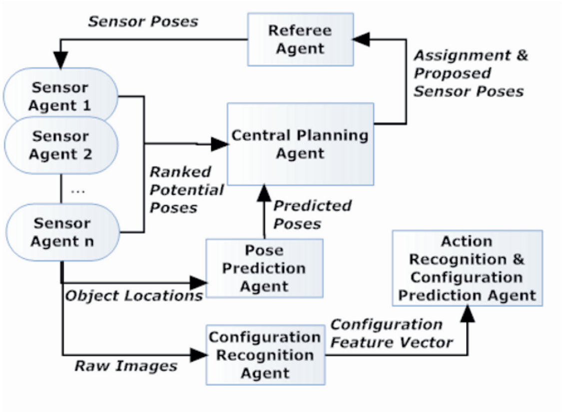

Figure 1. Overview of sensing-system reconfiguration methodology used for the simulations

At the lowest level, each sensor agent is associated with a single sensor. The exact configuration (in terms of number and composition of the sensor set) can be determined through a number of established methods (e.g., [1], [4], [5]). For our system, it is assumed that each sensor is reconfigurable in terms of pose (position and orientation) and that each is limited in capability by positional and rotational velocity and acceleration:

$$t_d = t_1 - t_0 \tag{6}$$

$$\boldsymbol{L}_{min} < \boldsymbol{x}_1 < \boldsymbol{L}_{max} \tag{7}$$

$$\boldsymbol{x}_{L-} < \boldsymbol{x}_1 < \boldsymbol{x}_{L+} \tag{8}$$

$$\boldsymbol{x}_{L-} = f(\boldsymbol{x}_0, t_d),\ \boldsymbol{x}_{L+} = f(\boldsymbol{x}_0, t_d), \tag{9}$$

where $\boldsymbol{x}_0$ is the initial position, $\boldsymbol{x}_1$ is the final position, t_0 is the initial time, t_1 is the final time, t_d is the total time between the demand instants, and $\boldsymbol{L}_{min/max}$ are the outer limits of the motion axis, respectively. A similar set of equations can be used to determine the rotational limits in terms of angular velocity and acceleration. This position space is discretized into n_{pos} possible final positions, where $n_{pos} \propto (t_1 - t_0)$, to bound computational complexity. The visibility metric is evaluated at each discretized sensor pose.

In this paper, all known obstacles and the OoI are modeled as elliptical cylinders. A clipped projection plan is established and all objects projected onto this plane. The visibility metric is defined as:

$$V \propto \sum_{i=1}^{n_{area}} (A_i) \tag{10}$$

where n_{area} is the number of distinct un-occluded regions and A_i is i^{th} distinct subsection of the projection of the OoI, visible on the projection plane. This is effectively measuring the amount of surface area of a virtual cylinder around the OoI that is visible. The effect of foreshortening is not removed; this is intentional, as the effective surface area visible decreases as the angle from camera focal direction to the cylinder tangent-plane normal increases.

Such a system tends to *look-ahead* and assign sensors to improve future visibility whenever necessary. Poses are ordered according to: the largest object surface area visible, the largest possible view of this same area, and the least distance traveled.

IV. Experimental Results

In order to validate the proposed sensing-system reconfiguration methodology in a single-target, dynamic environment with a time-varying geometry OoI, a set of controlled experiments was carried out.

A. Experimental Set-up

A total of four sensor agents were implemented. For each sensor, the proposed local optimization problem is solved by the agent, and the best solutions ranked and presented to a central planner. For the experimental implementation, a simple set of rules is used by the central planner to select a subset of the sensors to service the OoI at the current demand instant:

1. Sensors with a visibility metric less than the minimum, V_{min}, at all poses, are unassigned.
2. The three highest-visibility sensors are assigned to this instant.
3. The sensor with the lowest metric is asked to re-evaluate the visibility metric for an additional demand instant and is assigned to the next instant.
4. For assigned sensors, a weighted sum of metrics is evaluated.

In order to provide synthetic images under carefully controlled conditions, a simulation environment was designed. This environment can produce images from any of the virtual cameras, and includes a segmented model that approximates the human form. A simple walking action was used in all the experiments.

Form recognition is provided by a model-based algorithm. First, the location of a reference point in the world-coordinate system is found through robust estimation of the head center in 2D images. Contiguous regions of the OoI are then identified, and key-points determined. For the synthetic images, these points can be uniquely identified using color cues, so robust estimation is subsequently used to find the 3D world coordinates of as many points as possible. A fitting method is then used to form the feature vector via four form-invariant reference points on the OoI:

1. Subtract the reference from each interest point location to remove dependence on world position.
2. Remove the effects of rotation and scaling from the object data using a change of basis utilizing vectors formed by three other points around the reference point.
3. Merge all points with distance $|\boldsymbol{x}_1-\boldsymbol{x}_0|<K_1$ into a single point, where $\boldsymbol{x}_0$ and $\boldsymbol{x}_1$ are the two points.
4. Start with an empty feature vector. For each point, $\boldsymbol{x}$, and each model location, $\boldsymbol{x}_m$, if $\|\boldsymbol{x}-\boldsymbol{x}_m\| < K_2$, place $\boldsymbol{x}$ in the blank feature vector at the position corresponding to $\boldsymbol{x}_m$. For all remaining points, place them in the feature vector location corresponding to $\min(\|\boldsymbol{x}-\boldsymbol{x}_m\|)$, subject to $\|\boldsymbol{x}-\boldsymbol{x}_m\|< K_3$, where $K_3 >> K_2$.

A measure of the uncertainty of the fit is defined as:

$$E \propto C_1 \sum\left(\|\boldsymbol{x}-\boldsymbol{x}_m\|\right) + C_2 n_{ua} + C_3 n_{ms}\,, \qquad (11)$$

where C_1, C_2, and C_3 are proportionality constants, n_{ua} is the number of unassigned points, n_{em} is the number of missing points, and the sum is over all assigned points. This method of fitting is performed on each model in the database and a minimum uncertainty fit determined (subject to an upper limit for recognition).

In order to recognize the current OoI action, a post-process method is used. Using *time normalized* input data, a metric of distance from the library data is formulated for each database set that contains both two distinct *start* and *end* forms.

B. Simulation Results

An example test that was conducted over 100 frames of a walking motion to determine the effect of sensing-system reconfiguration on the recognition performance is presented below. Three runs were performed. Static sensor poses were used for the first. The remaining two runs were conducted using a velocity-constrained and ideal (unconstrained) reconfiguration system, respectively. For each frame, the error metric is calculated from the recovered feature vector. The results for each of the three runs are shown in Fig. 2. An upper threshold of recognition is also defined, as shown on the graph.

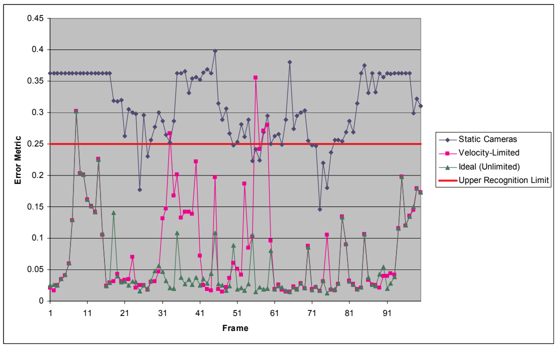

Figure 2. Comparison of error metric over three trials of 100 frames each, with walking action performed by the target.

For each trial, the target maintained a constant velocity of 10 mm/frame on a straight-line path through the center of the workspace, Fig. 3. Although the simulation was performed in a quasi-static manner, a virtual frame-rate of 10 frames-per-second was assumed. For the velocity limited system, the maximum velocity is given as 45 mm/frame, with a maximum acceleration of 90 mm/frame^2. The ideal system is considered to have unlimited maximum velocity and acceleration. The static-camera system uses a standard off-line sensor placement common in many other works, such as [19]. The four cameras are placed in a square configuration, each with its focal line placed on a normal to the path of the OoI. For the two final trials, the initial camera poses were the same as the static-camera poses, but two of the cameras were given a linear-translation axis, and all cameras could pan up to ±90˚ from their initial pose.

An upper limit for the error metric of 0.25 was selected for these simulations. Any frame with an error value above this limit is not considered to have a positive match from the form-recognition process. This value was determined through statistical analysis of multiple previous runs, resulting in at least 95% true positive matches for un-rejected frames.

The results showed a clear overall reduction in error values with the use of sensing-system reconfiguration.

1st Run: For the case of static cameras, performance was poor in Frames 1 to 20 and 80 to 100, since the object is near the limits of the workspace. Frames 32 to 45 showed that an obstacle would also cause most of the object form to be unrecoverable.

2nd Run: The use of constrained sensing-system reconfiguration lowered overall error, as the majority of frames are now considered to be positively recognized. Significant error still remained, for example in Frames 31 to 41, where the system was unable to move to its 'best' sensor positions due to movement-time constraints. We also noted that the use of reconfiguration is not guaranteed to produce improved results if movement is constrained – Frames 56 to 59. For these frames, the system chooses the best pose from those achievable, but the previous reconfiguration choices have resulted in a poor set of choices - poses too far from the ideal set for the system to achieve them before several demand instants have passed. As a result, the algorithm cannot recover some portions of the model for these frames.

3rd Run: The 'ideal' run showed strong correlation to the velocity-limited case, when the best pose for a given frame was achievable by the velocity-limited system. For frames where previous reconfiguration choices or motion time constraints limited the achievable poses to a poor set of alternatives, there is a definite improvement – Frames 30 to 41 and 56 to 59. However, there were still some frames where sub-sections of the object could not be recognized due to missing data (e.g., Frames 34, 44, 54, and 59), resulting in a higher than average error metric.

C. An Example Frame

Let us consider Frame 85, where a positive form match was determined by the ideal algorithm, but significant error in the recovered model still exists. This is the result of the system rejecting incomplete data on one or more model sub-sections and, thus, not recovering that portion of the model. Specifically, as shown in Fig. 3, the left arm of the subject is not visible.

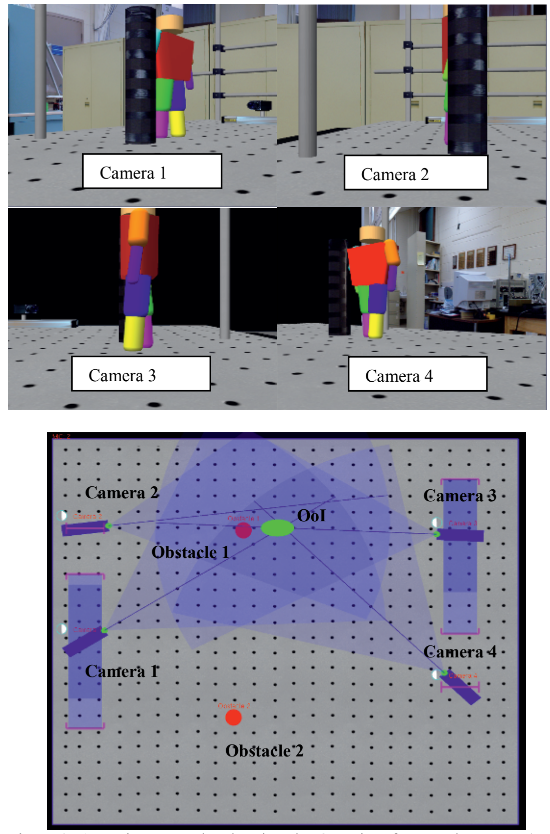

Figure 3. A motion example, showing the OoI view from each camera (top), and current system configuration (bottom).

Using the proposed algorithm, each sensor uses a weighted combination of three metrics to rank its desired poses: the area of the target bounding cylinder that is visible, the angle to the target, and the distance to the target. A plot of these metrics, plus the weighted combination metric, is shown for Camera 1 in Fig. 4.

One can note that if Camera 1 (and Camera 3, as well) were to select a non-optimal pose under the current rules, then, the left arm of the subject would be visible and, thus, it would be recovered. This pose would likely have a translational d value closer to the maximum of 1.6. However, such poses cannot be distinguished from other, less informative poses using any of the current metrics, Fig. 4. In this specific case, the current target distance metric would act against such poses. The addition of a fourth metric, which can differentiate viewpoints that contain unique information from those that do not, could potentially solve this problem. The only way to determine (without *a priori* knowledge of the target's actions) which views are likely to contain unique data is to predict the form of the object at a future instant. As such, feedback from the form-prediction process could be used to improve this sensing-system reconfiguration algorithm for time-varying geometry object/action recognition.

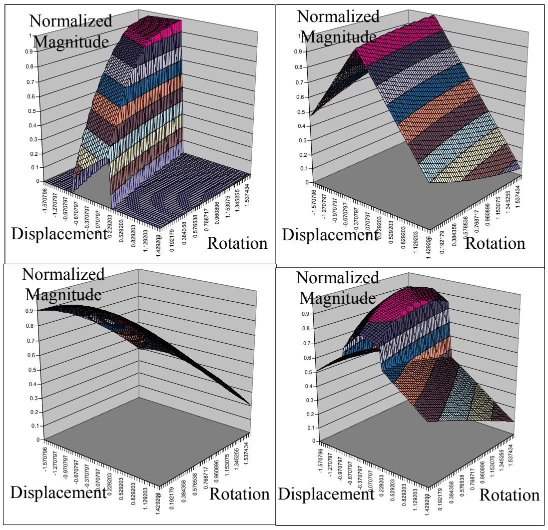

Figure 4. (Upper Left) Visibility Metric (Upper Right) Angle Metric (Lower Left) Distance Metric (Lower Right) Combined Metric.

VI. Conclusion

In this paper, sensing-system reconfiguration is proposed for the surveillance of time-varying geometry objects. A tangible improvement in form-recognition performance has been shown through controlled experiments. A proportionate relationship between the reconfiguration ability of the system and the recognition performance is also shown.

Acknowledgments

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Satellite Image Enhancement by Controlled Statistical Differentiation

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Abstract- **This paper presents a new method of statistical differencing for the enhancement of contrast images. Our approach controls the sharpening effect using two constants in such a way, that enhancement occurs in intensity edges areas and very little in uniform areas. It has been proven that our method is superior to similar existent and can be applied to pre-process satellite images.**

I. Introduction

Physical and psychological experiments have shown that photographs or images with edges enhanced are more visual satisfying than an exactly reproduction. But the graphical purposes of image processing are not the only field where such techniques are required. The results of recognition stages in computer vision are improved when suitable pre-processing like edge enhancement is applied [1].

The classic unsharp masking technique [9] is widely used and it is based on a highpass filter. Being simple and effective, the method suffers from several disadvantages: it is highly sensitive to noise and produces undesirable artifacts, particularly in uniform areas.

Some approaches have been proposed in the direction of noise sensitivity reduction. An adaptive filter [4] has been used to emphasize the medium-contrast details in the input image more than large-contrast details such as abrupt edges to avoid artifacts in the output image.

An overview of enhancement techniques can be found in [5] and [9]. In the purpose of non-photorealistic edge enhancement for 3D computer graphics, Nienhaus and Doellner [2] are using the edge map as a locality information. A multiresolution approach is given by Wang, Wu, Castleman and Xiong [3] in the aim of chromosome image enhancement, using differential wavelet transforms. Other multiresolution methods are using a Lapacian pyramid [6] or non-linear filtering [13].

In this work, we are introducing a new method for image enhancement, suitable for satellite images, as it was tested in the aim of optical flow detection [12]. Starting from common statistical differentiation methods, we have developed a method that offers a high enhancement in edges while presenting a low effect on homogenous areas.

The rest of the paper is organized as follows, section II presents the actual statistical differencing methods, section III describes the proposed method and section IV specifies an analytical investigation of the enhancement capability of our method. The concluding remarks are made in section V.

II. Statistical Differentiation Image Enhancement

Statistical differentiation has been first proposed by [9] and implies the division of original pixels $F(j,k)$ by their standard deviation $S(j,k)$:

$$G(j,k)=\frac{F(j,k)}{S(j,k)} \tag{1}$$

Where

$$S(j,k)=\frac{1}{W^2}\sum_{m=j-w}^{j+w}\sum_{n=k-w}^{k+w}[F(m,n)-M(j,k)]^2 \tag{2}$$

is the standard deviation computed for every pixel on a WxW window and W=2w+1. $M(j,k)$ represents the estimated mean value for the pixel having coordinates (j,k) and computed on a same sized window:

$$M(j,k)=\frac{1}{W^2}\sum_{m=j-w}^{j+w}\sum_{n=k-w}^{k+w}F(m,n) \tag{3}$$

The enhanced image $G(j,k)$ has a significant increase in magnitude for pixels that are different from neighbors and a decrease of magnitude for similar pixels. This process has some resemblance with automatic gain control in electronics.

Lee [8] proposed the following method for enhancement:

$$G(j,k)=M(j,k)+A(F(j,k)-M(j,k)) \tag{4}$$

with A a constant influencing the degree of enhancement, having current values in the range of [0.2, 0.7]. Wallis [11] has first extended (4) to:

$$G(j,k)=M_d+\frac{S_d}{S(j,k)}(F(j,k)-M(j,k)) \tag{5}$$

employing a desired mean value M_d and a desired standard deviation S_d.

The mentioned author also suggested a generalization of the differenting operator, in which, the enhanced image is forced

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 32–36.

to a specific form, with desired first-order and second-order moments:

$$G(j,k) = [F(j,k) - M(j,k)]\left[\frac{AS_d}{AS(j,k) - S_d}\right] + [rM_d + (1-r)M(j,k)] \tag{6}$$

where M_d and S_d represents the desired mean and standard deviation, A is a gain factor which prevents lager values when $S(j,k)$ is too small and r is a mean proportionality factor controlling the ratio between the edge and the image background.

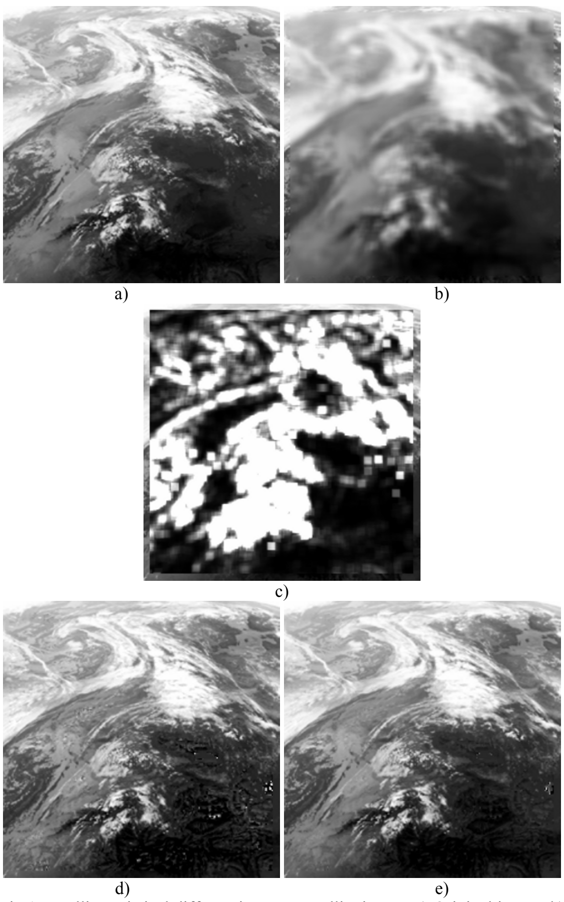


Fig.1. Wallis statistical differencing on a satellite image. a) Original image, b) mean image, c) standard deviation, d) enhanced image for r=0.1, A=4, M_d=0.5, S_d=0.33, e) r=0.8, A=4, M_d=0.5, S_d=0.33

Figure 1 presents an example of Wallis enhancement method, for two different desired mean and standard deviation factors, using a W=7 window.

III. A New Method for Differential Enhancement

Enhancement techniques are principally based on a high-pass filter convolution mask or on the statistical properties of neighborhood pixels as in the Lee (4) and Wallis enhancement method. One can observe that in (5), the variance of the pixels is used to enhance the edges by a weighted addition (inverse to variance) of the difference between the original image and the mean image. In this case, for small values of variance (neighbor pixels are alike) the weight will be important and consequently image noise will be enhanced.

Figure 2 shows the influence of the variance, using two extremes and opposite values on the enhancement of satellite images.

From (5) and also from the resulting images, it can be noticed that, using the mean value, uniform areas are also enhanced. Replacing the mean value by the original value of the pixel in the same equation, the enhancement effect will be improved, but the effect on uniform areas will remain, as shown in figure 3.

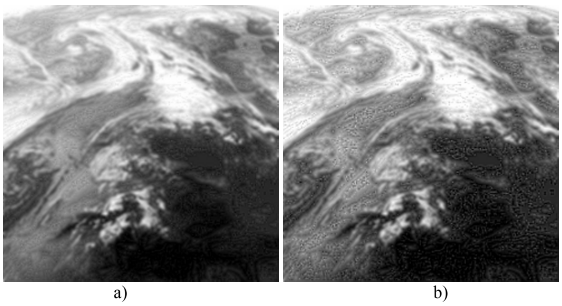


Fig.2. Satellite image enhancement using the Wallis method for desired variance a) S_d=20 and b) S_d=60.

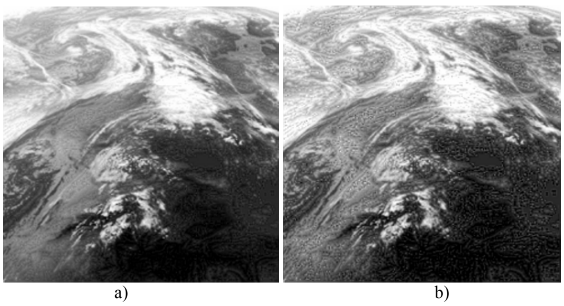


Fig.3. Satellite image enhancement using the modified Wallis method for desired variance a) S_d=20 and b) S_d=60.

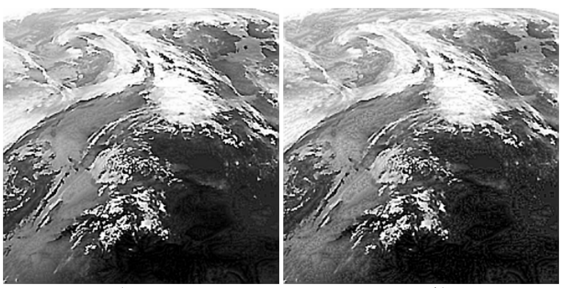
Satellite images present a high variance of pixels, due to the nature of phenomenon implied and to the integration effect of sensor (ground resolution). After testing several adaptive methods in view of cloud motion detection, none of them were satisfactory. Based on our experience [12] in processing this particular type of images, we are proposing a new method adapted to the context.

In order to perform an adaptive and distinct enhancement of homogenous and gray level discontinuity areas, we have developed empirically, the following filtering method:

$$G(j,k) = F(j,k) + \left(F(j,k) - M(j,k)\right)\frac{S(j,k)}{A} + \left(F(j,k) - M(j,k)\right)\frac{B}{S(j,k)} \quad (7)$$

where A and B are constants influencing the enhancement of edges and uniform areas, respectively. In the case of satellite images, optimal values determined for A are in the range [50, 150], and for constant B in [10, 40].

The windows size for variance estimation $S(j,k)$ and mean value of pixels is WxW, having W=2w+1 and $w \in \{1,2,3,4\}$. Figure 4 shows the results of the enhancement process, using different values of the two constants and w=1.



a) b)

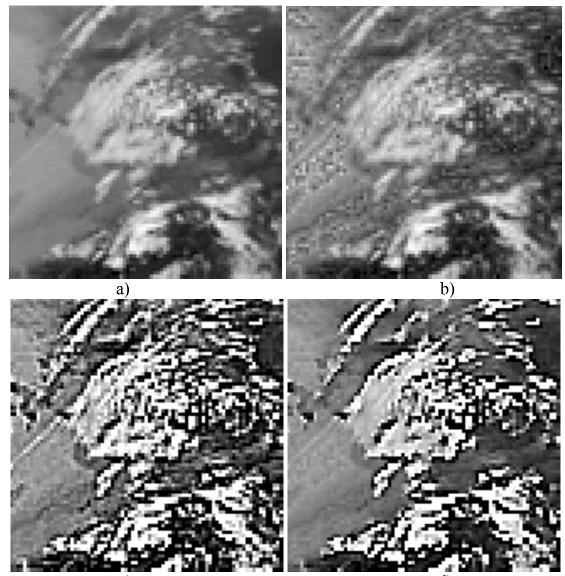
Fig.4. Results of the proposed enhancement method using different values of the constants a) A=50, B=20 and b) A=100, B=50.

In the purpose of visual comparison, we are presenting in figure 5, a zoomed subimage of the original one shown in figure 1 a) and enhanced subimages using different methods. One can observe the differential enhancement effect produced by our method together with the improvement in processing uniform areas. Parameter A is controlling the enhancement of high gradient areas, while B is responsible with the enhancement of homogenous regions.

IV. Experimental Investigation

In the aim of an analytical investigation of the enhancement capability of our proposed method in comparison with other techniques using the statistical differentiation, we have carried out several tests. First, we have established the conditions for a proper comparison using the constants in (4), (5) and (7), setting the enhanced image variance to a value of 80. In the case of the Lee method the constant A has a value of 4, the Wallis constant S_d was determined as 80, and for our method, A=200 and B=40. The window size was w=1 (3x3 pixels). The relative high value of variance is creating a high order enhancement effect, supporting the scope of the analysis.

First of all, we have tested the capability of differential enhancement effect on uniform areas, by subtracting the original image form the enhanced one and forming the error image. The gray-scale value of 128 represents a null variance, while negative and positive values are corresponding to black respectively white side. From figure 6, it can be observed a smaller dispersion of errors in the case of our method compared with the other investigated.



a) b) c) d)

Fig.5. Comparison between different enhancement methods; a) 2x zoom of the image in fig. 1a) within coordinates (60,110) and (140,190); b) Wallis enhancement method; c) enhancement using a classic convolution mask [7]; d) our method.

In the case of the Wallis statistical differencing, one can observe the high gain in uniform areas, also visible as a high error rate. As expected, the classical unsharp masks have a global effect, while in the case of our method, error histogram shows that a large number of pixels remain unchanged. Also, the enhanced pixels present a small difference, smaller than other methods. As an issue, the resulting image has no artifacts and the edges have no distortion.

TABLE 1
COMPARISON BETWEEN ENHANCEMENT METHODS

Method	Variance of enhanced image	Gray level mean value	Variance of difference image	Unaltered pixels
Unsharp mask	89.87	121.30	28.36	4230
Wallis	78.28	124.87	21.76	2682
Lee	80.01	124.44	23.11	4983
Our method	79.61	124.41	20.12	5056

In table 1, a comparative analysis of the four methods is represented, using approximately the same variance value of the enhanced image. The error and the number of unaltered pixels are showing that our method performs better. The initial value of the variance for the test image was 74.02, and the gray level mean value 124.76.

TABLE 2
ERROR DISPERSION FOR DIFFERENT GAUSSIAN SMOOTHING

Method / Gauss	σ=0.75	σ=1	σ=1.4	σ=1.6	σ=2
Lee	10.83	4.60	7.06	8.41	10.04
Wallis	12.30	12.27	13.15	13.68	14.68
Our method	10.54	7.16	9.44	10.37	11.59

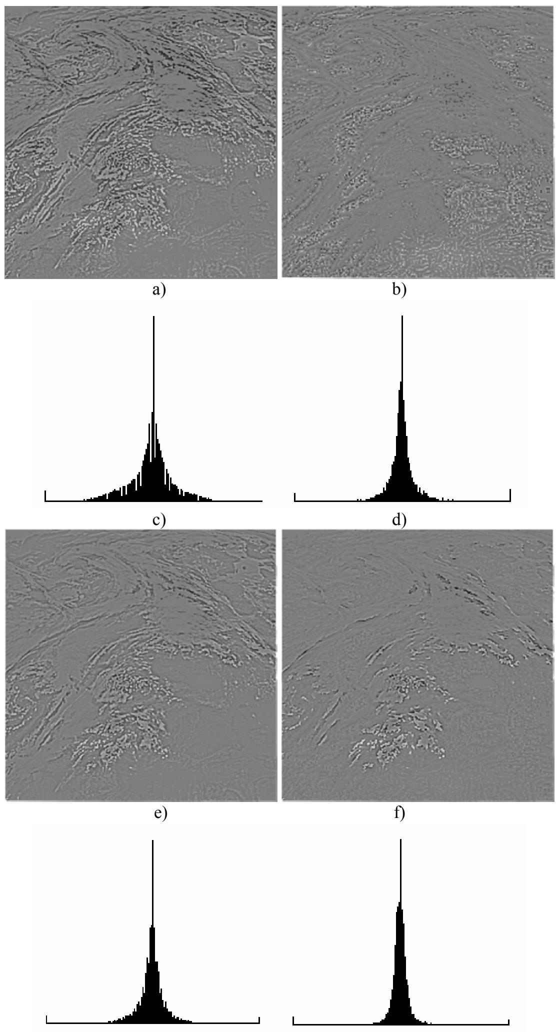


Fig.6. Error image and histogram for different enhancement methods; a) classical unsharp mask [9] and c) corresponding histogram; b) Wallis method and d) histogram; e) Lee and g) histogram; f) our method and the corresponding h) histogram.

In order to prove the satisfactory results obtained (visually and analytically), we have also tested the enhancement effect on gaussian smoothed images. The motivation of such a test consists on the potential edge recovery and enhancement after a smoothing degradation. Three statistical differencing methods were tested, Lee, Wallis and the proposed one, using the same initial conditions and returning approximately the same enhancement variance. The smoothing was performed by a 5x5 gaussian convolution mask and several dispersion values σ. After the enhancement process, based on the computed error, a histogram was created and the error dispersion represented as shown in table 2. It can be seen that our method is superior to Wallis and close to Lee, in enhancing the smoothed edges.

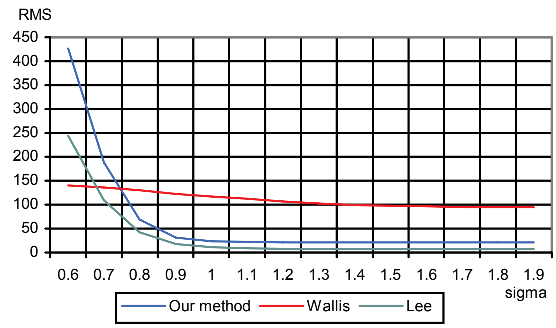


Fig.7. Evolution of RMS for different enhancement methods

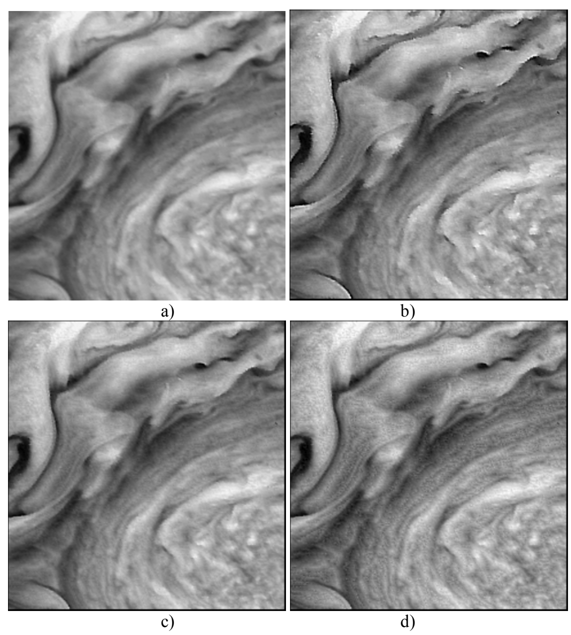


Fig. 8. Planet Venus satellite image a) and the enhanced images using the following parameters: b) A=50, B=10; c) A=150, B=10; d) A=200, B=30

The RMS error was determined for different values of σ ranging in [0.6, 2] with a step of 0.1. RMSE is decreasing while smoothing increase and our method performed equal to Lee. If constants *A* and *B* where restored to the previous optimal determined values, our method displayed smaller errors. Figure 7 presents the evolution of RMS with gauss parameter σ.

As it can be seen from figure 8., a planet Venus probe image has been enhanced for different values of the two parameters. The *A* weight regulates the enhancement of edges, smaller the values, higher the enhancement effect. Parameter *B* has the same significance, but over the homogeneous areas, increasing the effect proportional with his value. In this case, the best results were obtained using A=50 and B=10. It can be seen that for high values of *B*, the noise from relative uniform areas is also accentuated.

V. Conclusions

Based on several tests and results, the proposed method for statistical differencing enhancement was superior to all techniques encountered and investigated. If the differential enhancement capability is take into account, our method could be the only one adapting locally to uniform and non-uniform areas based on its intrinsic adapting capacity.

We also believe that having the possibility to adjust the degree of enhancement for edges and uniform areas offers a high efficiency, also shown for satellite images [12].

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Optimization of a Hexapod Micro Parallel Robot Using Genetic Algorithms

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***Abstract* - In this paper a mono-objective optimum design procedure for a six-degree of freedom parallel micro robot is outlined by using optimality criterion of workspace and numerical aspects. A mono-objective optimization problem is formulated by referring to a basic performance of parallel robots. Additional objective functions can be used to extend the proposed design procedure to more general but specific design problems. A kinematic optimization was performed to maximize the workspace of the mini parallel robot. Optimization was performed using Genetic Algorithms.**

Index Terms – optimization, hexapod, 6 degree of freedom, Genetic Algorithms.

I. Introduction

The workspace of a robot is defined as the set of all end-effector configurations which can be reached by some choice of joint coordinates. As the reachable locations of an end-effector are dependent on its orientation, a complete representation of the workspace should be embedded in a 6-dimensional workspace for which there is no possible graphical illustration; only subsets of the workspace may therefore be represented.

There are different types of workspaces namely constant orientation workspace, maximal workspace or reachable workspace, inclusive orientation workspace, total orientation workspace, and dextrous workspace. The constant orientation workspace is the set of locations of the moving platform that may be reached when the orientation is fixed. The maximal workspace or reachable workspace is defined as the set of locations of the end-effector that may be reached with at least one orientation of the platform. The inclusive orientation workspace is the set of locations that may be reached with at least one orientation among a set defined by ranges on the orientation parameters. The set of locations of the end-effector that may be reached with all the orientations among a set defined by ranges on the orientations on the orientation parameters constitute the total orientation workspace. The dextrous workspace is defined as the set of locations for which all orientations are possible. The dextrous workspace is a special case of the total orientation workspace, the ranges for the rotation angles (the three angles that define the orientation of the end-effector) being $[0,2\pi]$.

In the literature, various methods to determine workspace of a parallel robot have been proposed using geometric or numerical approaches. Early investigations of robot workspace were reported by Gosselin [1], Merlet [2], Kumar and Waldron [3], Tsai and Soni [4], Gupta and Roth [5], Sugimoto and Duffy [6], Gupta [7], and Davidson and Hunt [8]. The consideration of joint limits in the study of the robot workspaces was presented by Delmas and Bidard (1995). Other works that have dealt with robot workspace are reported by Agrawal [9], Gosselin and Angeles [10], Cecarelli [11]. Agrawal [12] determined the workspace of in-parallel manipulator system using a different concept namely, when a point is at its workspace boundary, it does not have a velocity component along the outward normal to the boundary. Configurations are determined in which the velocity of the end-effector satisfies this property. Pernkopf and Husty [13] presented an algorithm to compute the reachable workspace of a spatial Stewart Gough-Platform with planar base and platform (SGPP) taking into account active and passive joint limits. Stan [14] presented a genetic algorithm approach for multi-criteria optimization of PKM (Parallel Kinematics Machines).

Most of the numerical methods to determine workspace of parallel manipulators rest on the discretization of the pose parameters in order to determine the workspace boundary [15, 16]. In the discretization approach, the workspace is covered by a regularly arranged grid in either Cartesian or polar form of nodes. Each node is then examined to see whether it belongs to the workspace. The accuracy of the boundary depends upon the sampling step that is used to create the grid. The computation time grows exponentially with the sampling step. Hence it puts a limit on the accuracy. Moreover, problems may occur when the workspace possesses singular configurations. Other authors proposed to determine the workspace by using optimization methods [14]. Numerical methods for determining the workspace of the parallel robots have been developed in the recent years.

In this paper, the optimization workspace index is defined as the measure to evaluate the performance of a six degree of freedom parallel micro robot.

Another contribution is the optimal dimensioning of the six degree-of-freedom parallel micro robot of type Hexapod for the largest workspace.

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II. Workspace Evaluation

The workspace is one of the most important kinematic properties of manipulators, even by practical viewpoint because of its impact on manipulator design and location in a workcell [17]. A general numerical evaluation of the workspace can be deduced by formulating a suitable binary representation of a cross-section in the taskspace. A cross-section can be obtained with a suitable scan of the computed reachable positions and orientations **p**, once the forward kinematic problem has been solved to give **p** as function of the kinematic input joint variables **q**. A binary matrix P_{ij} can be defined in the cross-section plane for a crosssection of the workspace as follows: if the (*i*, *j*) grid pixel includes a reachable point, then $P_{ij} = 1$; otherwise $P_{ij} = 0$, as shown in Fig. 1. Equations (1)-(5) for determining the workspace of a robot by discretization method can be found in Ref. [18].

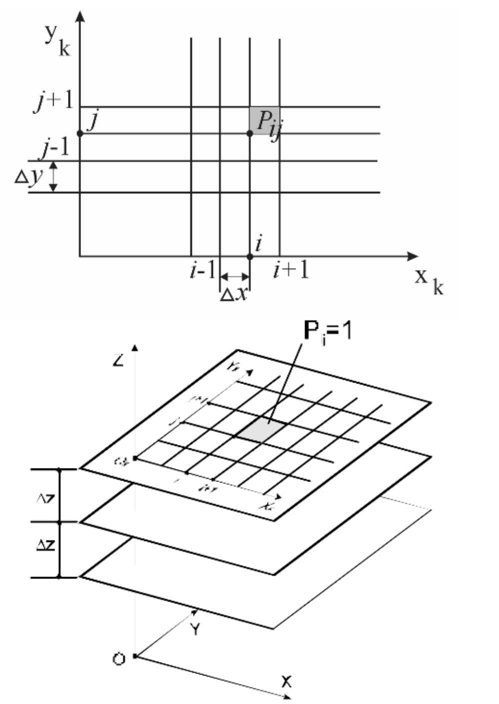

Fig. 1. The general scheme for binary representation and evaluation of robot workspace.

After that it is computed *i* and *j*:

$$\mathrm{i} = \left[\frac{\mathrm{x}+\Delta\mathrm{x}}{\mathrm{x}}\right]; \mathrm{j} = \left[\frac{\mathrm{y}+\Delta\mathrm{y}}{\mathrm{y}}\right] \tag{1}$$

where *i* and *j* are computed as integer numbers. Therefore, the binary mapping for a workspace cross-section can be given as:

$$P_{ij} = \begin{cases} 0 & if \;\; P_{ij} \notin W(H) \\ 1 & if \;\; P_{ij} \in W(H) \end{cases} \tag{2}$$

where W(*H*) indicates workspace region; ∈ stands for "belonging to" and ∉ is for "not belonging to". In addition, the proposed binary representation is useful for a numerical evaluation of the position workspace by computing the sections areas *A* as:

$$A = \sum_{i=1}^{i_{\max}} \sum_{j=1}^{j_{\max}} \left(P_{ij} \Delta x \Delta y\right) \tag{3}$$

and finally, we can computer the workspace volume V as:

$$V = \sum_{z=z_{\min}}^{z=z_{\max}} A_z \Delta z \tag{4}$$

In general, the parallel robots performances depend very much on their geometry, thus it became of great interest the optimization problems of parallel robots.

A key aspect of the performances of the parallel robots is that they are very dependent on the topology and dimensioning of the robot. It became of a real interest for developing parallel robots for fulfilling the work tasks, thus introduction of the performance indices like or optimization criteria used to characterize the robot has become compulsory. In design of a robot, the workspace is of primary importance. Maximizing the workspace of the parallel robot is one of the main goals of the optimal design of parallel robots.

III. Optimal design of the Hexapod

A. Six DOF micro parallel robot

The micro parallel robot is a six degrees-of-freedom parallel manipulator comprising a fixed base platform and a payload platform, linked together by six independent, identical, open kinematic chains (Fig. 2). Kinematics of this structure is presented in Refs. [14].

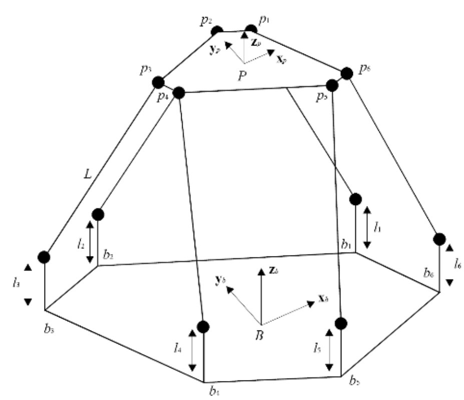

Fig. 2. Six degrees-of-freedom micro parallel robot

B. Workspace index

One of the most important issues in the process of design of robot is their workspace. For parallel robots, this issue may be more critical since parallel robots will sometimes have a rather limited workspace. Closed loop nature of the parallel robots limits their workspace. Also, in the context of design, the workspace determination procedure should be simple enough to be included in an optimization algorithm.

Because of this, applications involving these parallel robots require a detailed analysis and visualization of the workspace of these robots. The algorithm for visualization of workspace needs to be adaptable in nature, to configure with different dimensions of the parallel robot's links. The workspace is discretized into square and equal area sectors. A multi-task search is performed to determine the exact workspace boundary. Any singular configuration inside the workspace is found along with its position and dimensions. The volume of the workspace is also computed. A type of parallel robot, namely Hexapod-type six-degree of freedom robot is considered to demonstrate the effectiveness of the algorithm.

Workspace is another significant design criterion for describing the kinematics performance of parallel robots. Parallel robots use volume to evaluate the workspace ability. However, is hard to find a general approach for identification of the workspace boundaries of the parallel robots. This is due to the fact that there is not a closed form solution for the direct kinematics of these parallel robots. That's why instead of developing a complex algorithm for identification of the boundaries of the workspace, it's developed a general visualization method of the workspace for its analysis and its design.

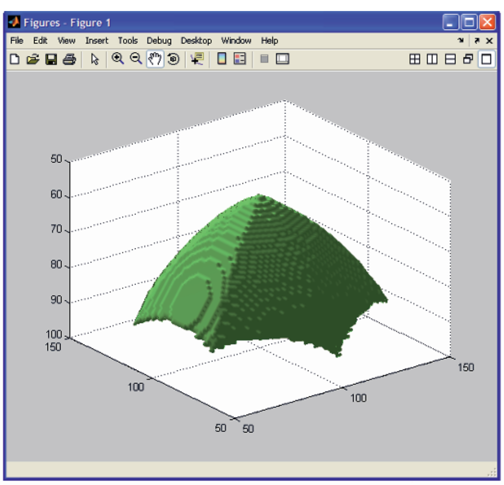

Fig. 3. Workspace views of the HEXAPOD micro parallel robot with six degrees-of-freedom

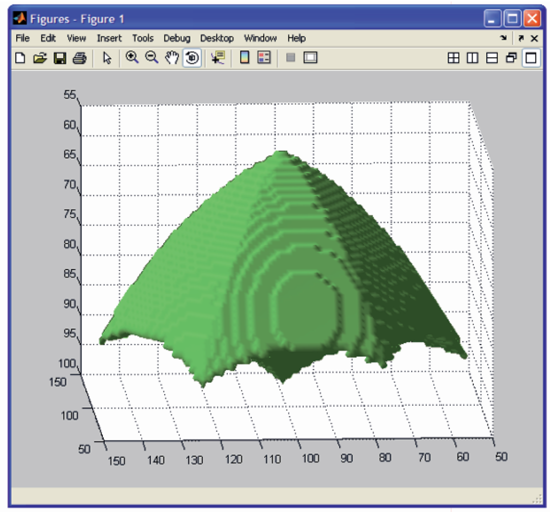

Fig. 4. Workspace views of the HEXAPOD micro parallel robot with six degrees-of-freedom

The possible workspace of the robot is of a great importance for optimization of the parallel robots. Without the ability to solve the workspace is impossible to state that the robot can fulfill any work task. The general analysis of the workspace consists in workspace determination using the described discretization method.

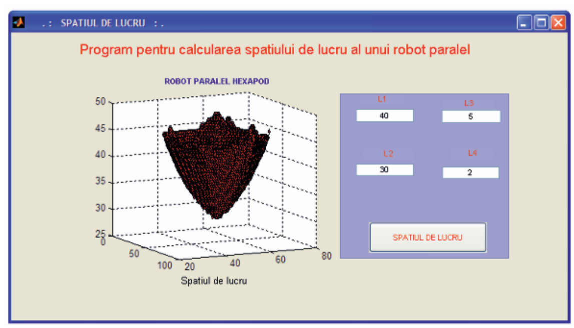

Fig. 5. Graphical User Interface for determining the shape of the HEXAPOD micro parallel robot with six degrees-of-freedom

The workspace is the volume in the space case where the tool centre point (TCP) can be controlled and moved continuously and unobstructed. The workspace is limited by *singular configurations*. At singularity poses it is not possible to establish definite relations between input and output coordinates. Such poses must be avoided by the control.

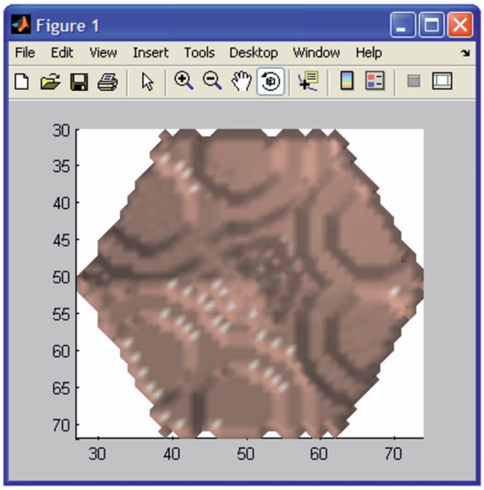

Fig. 6. Top view of the workspace of the HEXAPOD micro parallel robot with six degrees-of-freedom

The robotics literature contains various indices of performance [19], [20], such as the workspace index *W*.

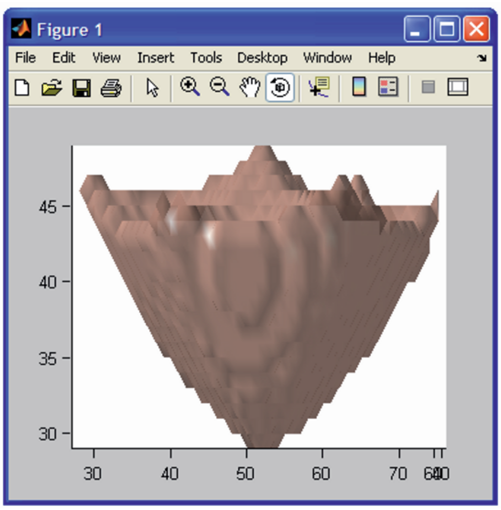

Fig. 7. Lateral view of the workspace of the HEXAPOD micro parallel robot with six degrees-of-freedom

C. Performance evaluation

Beside workspace which is an important design criterion, transmission quality index is another important criterion. The transmission quality index couples velocity and force transmission properties of a parallel robot, i.e. power features [21]. Its definition runs:

$$T = \frac{\|E\|^2}{\|J\| \cdot \|J^{-1}\|} \tag{3}$$

where E is the unity matrix. T is between 0<T<1; T=0 characterizes a singular pose, the optimal value is T=1 which at the same time stands for isotropy [22].

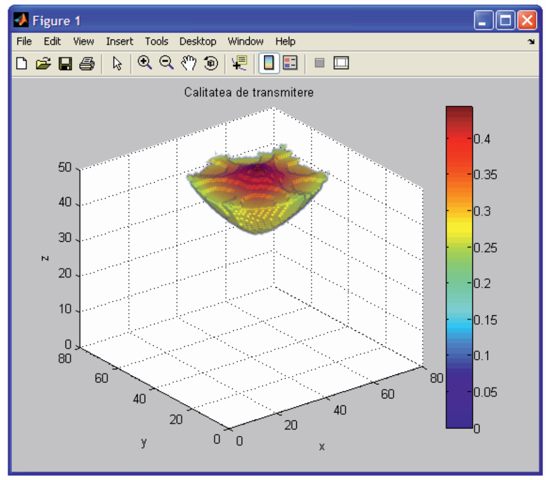

Fig. 8. Transmission quality index for HEXAPOD micro parallel robot with six degrees-of-freedom (3D view)

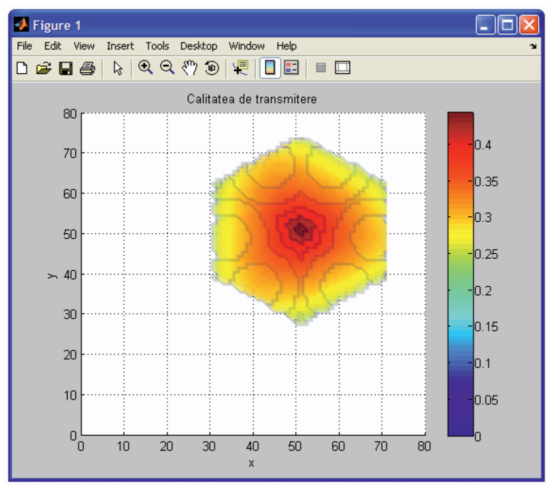

Fig. 9. Transmission quality index for HEXAPOD micro parallel robot with six degrees-of-freedom (top view)

As it can be seen, the micro parallel robot presents better performances in the middle of its workspace, as presented in Fig. 6-7.

D. Design optimization

The design of the robot can be made based on any particular criterion. The paper presents a genetic algorithm approach for workspace optimization of six-dof parallel micro robot. For simplicity of the optimization calculus a symmetric design of the structure was chosen.

In order to choose the robot dimensions L, q_{1min}, q_{1max}, q_{2min}, q_{2max}, q_{3min}, q_{3max}, q_{4min}, q_{4max}, q_{5min}, q_{5max}, q_{6min}, q_{6max}, we need to define a performance index to be maximized. The chosen performance index is W (workspace). One objective function is defined and used in optimization. It is noted as W, and corresponds to the optimal workspace. We can formalize our design optimization problem as the following equation:

$$Goal\_function = \max(\mathrm{W}) \tag{5}$$

Optimization problem is formulated as follows: the objective is to evaluate optimal link lengths which maximize (5). The design variables or the optimization factor is the ratios of the minimum link lengths to the base link length b, and they are defined by:

$$L \tag{6}$$

Constraints to the design variables are:

$$20<L<60 \tag{7}$$

$$\begin{gathered} q_{1min}=q_{2min}=q_{3min}=q_{4min}=q_{5min}=q_{6min}, \\ q_{1max}=q_{2max}=q_{3max}=q_{4max}=q_{5max}=q_{6max}, \\ q_{1max}=1{,}6q_{1min},\ q_{2max}=1{,}6q_{2min},\ q_{3max}=1{,}6q_{3min}, \\ q_{4max}=1{,}6q_{4min}, \\ q_{5max}=1{,}6q_{5min},\ q_{6max}=1{,}6q_{6min}, \end{gathered} \tag{8}$$

For this example the lower limit of the constraint was chosen to fulfill the condition $L \geq 30$. For simplicity of the optimization calculus the upper bound was chosen the length $L \leq 60$.

During optimization process using genetic algorithm it was used the following GA parameters, presented in Table 1. A genetic algorithm (GA) optimization method is used because of its advantages like robustness and good convergence properties.

TABLE I. GA PARAMETERS

1	Population	50
2	Generations	100
3	Crossover rate	0,08
4	Mutation rate	0,005

The GA approach has the clear advantage over conventional optimization approaches in that it allows a number of solutions to be examined in a single design cycle. The traditional optimization methods searches optimal points from point to point, and are easy to fall into local optimal point.

Using a population size of 50, the genetic algorithm was run for 100 generations. A list of the best 50 individuals was continually maintained during the execution of the genetic algorithm, allowing the final selection of solution to be made from the best structures found by the genetic algorithm over all generations.

We performed a kinematic optimization in such a way to maximize the workspace index W. It is noticed that optimization result for Hexapod when the maximum workspace of the 6 DOF micro parallel robot is obtained for L=60 mm. The used dimensions for the 6 DOF micro parallel robot were: q_{1min}=0 mm, q_{1max}=100 mm. Maximum workspace of the micro parallel robot with six degrees-of-freedom was found to be W= 45493 mm^3.

And the shape of the optimized workspace of the parallel micro robot is shown in Fig. 10. The results show that GA can determine the architectural parameters of the robot that provide an optimized workspace. Since the workspace of a parallel robot is far from being intuitive, the method developed should be very useful as a design tool.

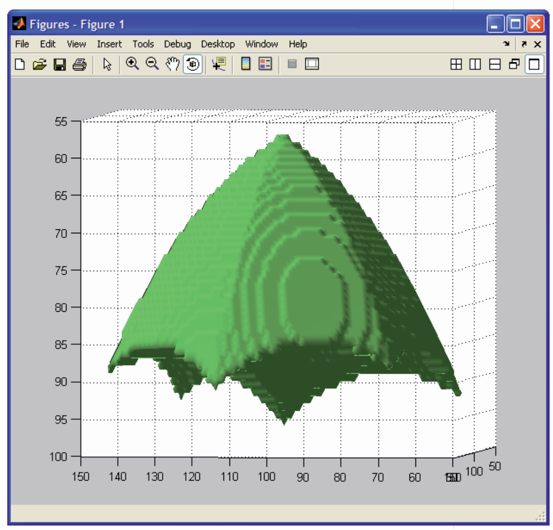

Fig. 10. Workspace of the optimized parallel micro robot with six degrees-of-freedom.

However, in practice, optimization of the parallel robot geometrical parameters should not be performed only in terms of workspace maximization. Some parts of the workspace are more useful considering a specific application.

Indeed, the advantage of a bigger workspace can be completely lost if it leads to new collision in parts of it which are absolutely needed in the application. However, it's not the case of the presented structure.

IV. Conclusions

In this paper a mono-objective optimum design procedure for parallel robot was outlined by using optimality criterion of workspace and numerical aspects. A mono-objective optimization problem was formulated by referring to a basic performance of parallel robots. A kinematic optimization was performed to maximize the workspace of the 6 degrees-of-freedom micro parallel robot. Together with other optimization oriented toolboxes from MATLAB, the GAOT Toolbox provides a uniform environment for the mechanical engineer to experiment with and apply GAs to problems in optimization of parallel robots.

Acknowledgment

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A Method of Real-time Moving Vehicle Detection for Bad Environments Using Infrared Thermal Images

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***Abstract-* We propose a method of real-time moving vehicle detection using infrared thermal images. It can detect moving vehicles robustly even for bad environments compared with conventional vehicle detection methods using visible light cameras. It can also measure the size of each vehicle around the clock. The algorithm we propose for this detection is designed for a high-speed processing without complicated calculations and also designed for a real-time vehicle detection by using a general-purpose personal computer. Experimental results of our method by use of infrared thermal traffic images both at daytime with vehicular shadows and at nighttime in darkness show that the vehicle detection accuracy is 100%.**

I. Introduction

It is a pressing matter how to develop the vision-based traffic monitoring systems in the field of ITS (Intelligent Transport Systems). They have the advantage of measuring what cannot be measured by conventional vehicular detectors: vehicular positions, the gaps between two vehicles, vehicular lengths, and the trajectories of moving vehicles on the multilane. Therefore, by use of the vision-based traffic monitoring systems, we can make up a new traffic control strategy on the one hand designed to reduce traffic jams and an automatic traffic investigation system on the other designed to find traffic incidents without time delay.

The present research level of the vision-based traffic monitoring systems is high enough to recognize vehicles robustly around the clock [1][2]. However, there are some defects as follow.

Many of the conventional methods detect the bodies of vehicles only in the daytime, and nighttime detection is adopted in much fewer cases. Yoneyama et al. [2] pointed out that most of the daytime detection methods lose their accuracy when they are directly applied to nighttime detection. Therefore, generally adopted methods are those of detecting the headlights or the taillights of vehicles at nighttime and of preparing two algorithms separately for daytime and nighttime detection [1][2]. It is generally difficult to measure the vehicular lengths at nighttime. In the method of detecting taillights [2], the vehicular lengths can be measured only at the limited camera angles because the vehicular length is estimated by the triangle of a pair of taillights and the vehicular front edge detected as the gray level difference between the head of the vehicle and the headlight reflection area on the road. What is most important for both traffic signal control and traffic simulation is to be able to measure the vehicular lengths and to specify the number of large-sized vehicles in the detected area around the clock. It is especially true because the saturation flow rate used as one of the fundamental values for them is largely influenced by the rate of large-sized vehicle mixing.

Vehicular shadows at daytime impair the vehicle detection of the adjoining lane. Therefore, a method of eliminating vehicular shadows has already been proposed [3]. However, it has the disadvantage of restricting the elimination of the vehicular shadows to limited camera angles.

In the conventional methods of vehicle detection by use of visible light cameras, it is difficult to detect vehicles with high accuracy in bad weathers such as fog, snow, and heavy rain. However, traffic accidents and traffic jams are more likely to happen under such circumstances. Therefore, it is a pressing matter for us to develop a method of detection designed to recognize vehicles with high accuracy under all circumstances.

In this paper, we propose a new method of moving vehicle detection that uses a infrared thermography camera instead of a visible light camera. The infrared thermal images of vehicles are expected to detect the shapes of vehicles robustly regardless of any environments around the clock.

First, we refer to our simulation experiment in Section II which shows how clear visions of vehicles can be obtained even in an artificial fog environment. Second, the algorithm developed in our method is explained in Section III. Third, our experiments at daytime with vehicular shadows and at nighttime in darkness are explained. Then we refer to the experimental results which show that the vehicle detection accuracy is 100% in Section IV. Finally, we have the conclusion in Section V.

II. Advantages of Infrared Thermal Images in the Vehicle Detection for Bad Environments

The infrared thermography camera used in our detection is TVS-200 [4]. The frames of the infrared thermal image are transmitted to a notebook personal computer with the 1/60 seconds interval through the IEEE1394 interface.

We have confirmed through our experiment that the

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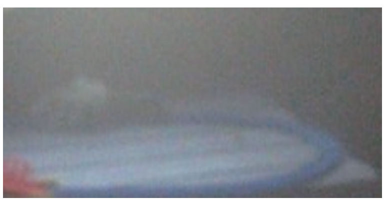
obtained infrared thermal images are clear enough to detect the shapes of vehicles in bad environments. We have obtained real images on the road both at daytime with vehicular shadows and at nighttime in darkness. However, we could not obtain the images of vehicles in the still worse environments because this research was made in summer. So, we have done a simulation experiment by using an artificial deep fog environment.

A glove made of vinyl was filled with the hot water of 42 degrees centigrade. The temperature is similar to the average one in the real moving vehicles. A model train that carries the glove moved on an oval rail in an artificial deep fog environment produced with the steam of a bath room.

Fig. 1 (a) shows an image of the moving model train taken from a visible light camera. Fig. 1 (b) shows its image taken from an infrared thermography camera. Fig. 1 (b) shows that we can see the outline of the glove more clearly. The visible light camera shows only an indistinct image compared with that of the infrared thermography camera.

The infrared thermography camera, therefore, can be effectively used for object detection in a bad weather such as deep fog.



(a) A visible light image

(b) An infrared thermal image

Fig. 1. Two samples of vehicular image taken in an artificial deep fog environment.

III. An Algorithm for Moving Vehicle Detection Using Infrared Thermal Images

We have developed an algorithm for moving vehicle detection using infrared thermal images with Visual C++ 2005 and the computer vision library OpenCV [5]. The

1) Two continuous image frames are taken.

2) The optical flow is calculated.

3) The points of velocity vectors that exceed the threshold value are totaled along the vertical direction.

4) Moving averages of the numbers along the *x* axis direction are calculated.

5) The horizontal line candidates of rectangles are extracted.

6) The points of velocity vectors that exceed the threshold value are totaled along the horizontal direction.

7) Moving averages of the numbers along the *y* axis direction are calculated.

8) The vertical line candidates of rectangles are extracted.

9) The average magnitude and the average direction of the velocity vectors in each rectangular candidate are calculated.

10) If the average magnitudes and the average directions of the velocity vectors among adjoining candidates of rectangles are similar to each other, they are integrated as a rectangle.

11) The rectangles are displayed.

Fig. 2. Flowchart of the moving vehicle detection.

proposed algorithm is explained through the flowchart of Fig. 2. The numbers below correspond to the ones in Fig. 2.

1) Two continuous image frames with 1/60 seconds interval are taken.

2) Optical flow between the two frames is calculated. Out of several algorithms available for calculating optical flow, we chose the Lucas-Kanade algorithm [6] because it can make more rapid calculation than any other ones.

3) The points of velocity vectors that exceed the predetermined threshold value are totaled along the vertical direction. Specifically, the points with the same x coordinate are assumed to be a unit, so the number of points in each unit can be counted.

4) Moving averages of the numbers along the x axis direction are calculated. The purpose of this process is to prevent the area of moving objects from dividing into several parts.

5) The horizontal line candidates of rectangles which encircle moving objects are extracted. Fig. 3 shows their vertical projection processes.

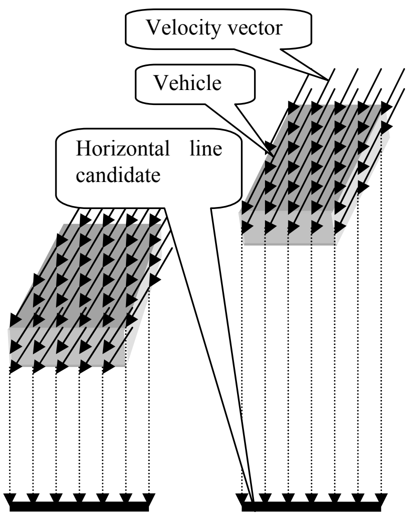


Fig. 3. Vertical projection processes.

6)-8) We repeat the processes 3)-5) in 6)-8) to the horizontal direction, and obtain the vertical line candidates of the rectangles. Fig. 4 shows their horizontal projection processes.

9) The candidates of rectangles are generated from the extracted horizontal and vertical line candidates, and the average magnitude and the average direction of the velocity vectors in each candidate of rectangles are calculated.

10) The average magnitudes and the average directions of the velocity vectors among adjoining candidates of rectangles are compared with one another. Then if the average magnitudes and the average directions of the velocity vectors are less than or equal to the predetermined threshold values, they are regarded as the candidates of rectangles in the same vehicle, so they are integrated as a rectangle.

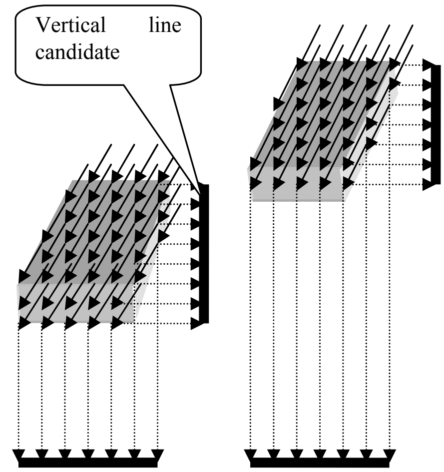


Fig. 4. Horizontal projection processes.

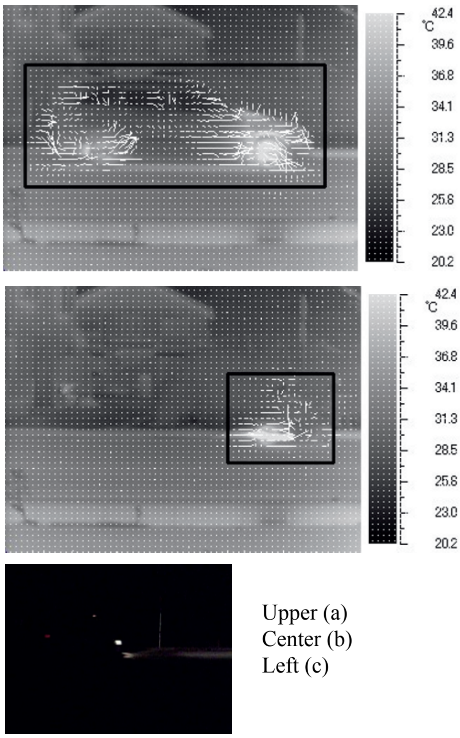
11) The rectangles that encircle the moving objects are displayed.

IV. Experimental Results

We have confirmed the validity of the proposed method by using traffic flow images at two spots of detection in Kumamoto city, Japan. At one spot of detection with a single lane, we took vehicular side views through the infrared thermography camera and a visible light camera both installed on the sidewalk. We call this spot Point A below. At the other spot of detection with multilane, we took vehicular front views through the infrared thermography camera and the visible light camera both installed on the crosswalk. We call this spot Point B below.

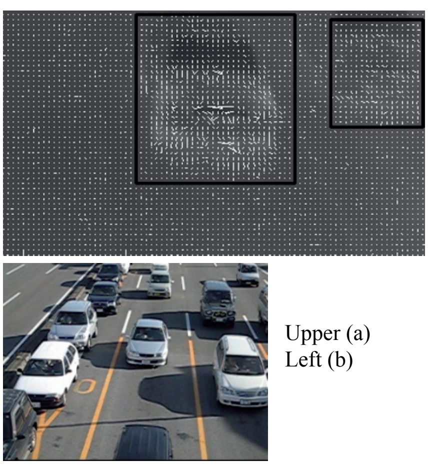
The frame size of collected images was 412×240 pixels. The optical flow was calculated five-pixel intervals in both x and y coordinates, and each moving average for the number of the points was calculated with the values of seven pixels.

Fig. 5(a) (b) are detection results of infrared thermal images of a moving vehicle and a moving motorcycle taken at Point A around 8 p.m. in darkness. Fig. 5(c) is a visible light image of a moving vehicle. We can obtain the head light positions from the visible light image but cannot measure the vehicular length. On the other hand, we can recognize the type of moving vehicles by processing the infrared thermal images in darkness.



Upper (a)
Center (b)
Left (c)

Fig. 5. Detection results for nighttime images in darkness.



Upper (a)
Left (b)

Fig. 6. Detection results for daytime images with vehicular shadows.

Fig. 6(a) is detection result of an infrared thermal image of moving vehicles taken at Point B. Fig. 6(b) is a visible light image taken at the same time. The influence of the vehicular shadows is remarkable in this image while there is no vehicular shadow in the infrared thermal image. The white lines in the rectangles on the infrared thermal images in Figs. 5 and 6 show the velocity vectors.

In our experiments, the proposed algorithm has performed a high-speed processing without complicated calculations, and a real-time vehicle detection by using a general-purpose personal computer.

V. Conclusions

We have proposed a method of real-time moving vehicle detection using infrared thermal images. The experimental results show that the vehicle detection accuracy is 100%. We used the infrared thermal images both at daytime and nighttime. In the two environments, the shapes of moving vehicles can be obtained clearly enough to recognize their types. Since the rate of large-sized vehicles can be calculated around the clock regardless of the camera angle, the proposed method is effective for both traffic signal control and traffic simulation as a vehicle detection sensor system. By using our method, a reliable automatic traffic investigation system can be also designed.

The algorithm for vehicle detection is expected to be advanced further in the future so that we can detect stopped vehicles as well. Then the vehicle detection in the still worse weather conditions such as a deep fog and snow will be made possible.

Acknowledgment

I wish to express my gratitude to my colleague Prof. Shuhei Takada, a researcher of American culture & literature, for having carefully corrected my sentences in this paper.

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Computational Simulation on An Energy Saving Regenerative Refrigerating System

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Abstract

A new type of energy-saving regenerative refrigerating system has been designed by the author and the prototype has been proved functioning properly. In this paper, the principle of the system is analyzed and the model is simulated by the computational fluid dynamics. The major advantages of this system include: the functioning efficiency is higher, the structure is simple and compact with less vibration, and frictional losses are very small between free piston and cylinder.

Nomenclature

F, F' = free piston

Ff = frictional force between free piston and cylinder

G = fixed plate with a small hole on it

H, H' = piston of compressor

L = Vk / Veo

Mc, M'c = mass in compressive cavity

Me, M'e = mass in expansive cavity

Mk, M'k = mass in clearance volume

Mt = total mass of gas in the refrigerating system

N = Vcmax / Veo

Pc, P'c = pressure in compressive cavity

Pe, P'e = pressure in expansive cavity

Ph, Pl = high and low pressure

Pwa = mixed pressure in fluctuation cavity

Q = Tc / Te

R = gas constant

S = cross section area of free piston

Tc, Te, Tk = temperature in compressive, expansive cavities and clearance volume

V = Vc + Ve

Vc, V'c = compressive cavities

Vcmax = maximum value of Vc

Ve, V'e = expansive cavities

Veo = maximum value of Ve

Vk = volume of clearance

Vmc, V'mc = middle fluctuation cavity

Vt = Vc + Ve + Vk

W = Tc / Tk

Ye = moving distance of free piston

Yo = maximum moving distance of free piston

m = mass of free piston

w = speed of turning angle of crank

α, β = turning angles of crank in different time

Introduction

The cryogenic refrigerating system has been widely applied in many industrial fields and more researches have been done to improve its performance. In this paper, the author introduces a new type of energy-saving regenerative gas refrigerating system with the theoretical analysis and computational simulation. The simplified structure and working cycle of this system are

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shown in Fig. 1 and 2. From the figures, we can see that this new mechanism makes the structure of this refrigerating system more simplified. Compared with Solvay refrigerating system, the middle fluctuating cavity Vme(V'me) is simplified with less components and more compact in this refrigerating system. In addition, part of compressive work in middle fluctuating cavity of this new mechanism can be transformed into motive work compared with Solvay system that no energy can be retrieved from middle cavity in which the total compressive work has been exchanged to heat discharged to the air. Moreover, the working efficiency of the system is improved since the refrigerating capacity can be output by two ends of cylinder. The vibration and shock are reduced because of its symmetrical structure. The phase angle of various functional curves can be easily adjusted by selecting proper parameters of structure, such as width of inlet and outlet of free piston and size of small hole on the plate which is fixed in the middle of cylinder. Because there is no need to adjust the tension of elastic ring in order to change the phase angle of functional curves in this new machine, the frictional losses are significantly reduced.

Analysis of System Functioning

The working principle of this system can be analyzed with reference of Fig.1 and 2. There are two compressive cylinders Vc (V'c) and two expansive cylinders Ve (V'e) that are symmetrically installed in the machine. Also there are two same free pistons F (F'), one fixed plate G with a small hole on it in the cylinder and two middle fluctuation cavities Vme (V'me). Due to its symmetrical setup, we only analyze the working process in right side, namely Vc, Ve and Vme, H and F. The initial condition of machine is indicated in Fig.1. We assume that the steady working condition is reached at its initial stage of the cycle, piston H is at the bottom position in the compressive cylinder and free piston F is at the right end of expansive cylinder. Because some gas with high pressure (P=Ph) enters Vmc through a small hole at the end period of last working cycle, gas with low pressure (P=Pl) in Vmc is mixed with high pressure gas. The mixed pressure in Vmc is Pwa. When H begins to move upward in Vc, gas in Vmc is assumed not flowing into V'mc because the moving speed of the piston is fast and the diameter of hole is small. Since the left side pressure of F is higher than right side pressure of F (Pmc = Pwa >Pe), F does not move and Pe reaches to Ph and Ve increases from zero to V2 as H continuously moves up. This process is shown by curve 1-2 in Fig.2. Then gas in Vmc flows into V'mc. In this period, pressure at the left and right sides of F is same and F moves to the left with no change of speed. When angle of crank α nears 180°, Ve increases from V2 to V3 and the process is represented by line 2-3. In the earlier stage that H moves downwards in Vc, F does not move because gas in V'mc is assumed not entering Vmc. Pe reduces from P3 to P4 and corresponding course is 3-4. As H continuously moves downwards in Vc, gas in V'mc flows into Vmc through small hole and F moves to the right end in the expansive cylinder with same speed and corresponding process is 4-1. The full working cycle consists of all the above processes.

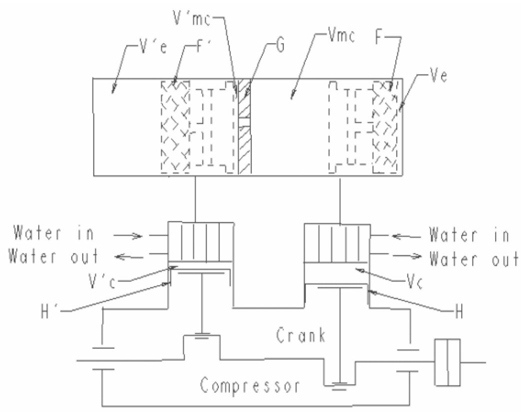

Fig.1 Simplified Structure of New Refrigerating System

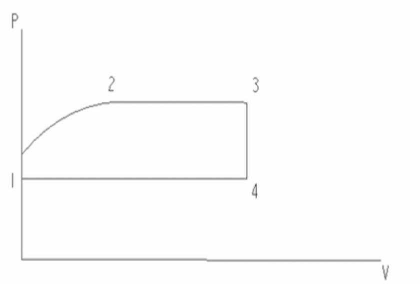

Fig.2 P-V Diagram of System Cycle in Expansive Cavity

Computer Aided Simulations

The prototype of this new refrigerating system has being tested and the results show that this system is feasible in working principle. The computer and numerical simulations have been demonstrated to verify the function of the system. By numerical method, the variations of all functional parameters can be found based on the mechanical analysis of free piston.

Because the compressive piston H moves in sine law, we can get

$$Vc(\alpha)=(1/2)*N*Vco*[1+\cos(\alpha)] \qquad (1)$$

If Vk is the volume of clearance, then

$$V=Vk+Vc+Ve \qquad (2)$$

$$P=R*\sum Mi/[(Vk/Tk)+(Ve/Te)+(Vc/Tc)] \qquad (3)$$

When compressive piston H moves up with crank turning to some angle β,

$$(Pe - Pme) * S = Ff \qquad (4)$$

$$P(\beta)=R*\sum Mi/[(Vk/Tk)+0+(Vc(\beta)/Tc)] \qquad (5)$$

Assume:

$W=Tc/Tk$, $L=Vk/Veo$, $Q=Tc/Te$

Combine (3) and (5):

$$P=P(\beta)*[L*W+(Vc(\beta)/Veo)]/[L*W+(Ve/Veo)*Q+(Vc(\beta)/Veo)] \qquad (6)$$

From equation (6), the solution of $P = P(\alpha)$ can be found after relation between Ve and α being determined. The differential equation of motion for free piston can be shown, based on Newton's second law, as follows:

$$m*(d^2Ye/dt^2)=(Pe-Pmc)*S-Ff \qquad (7)$$

Combine equations (6) and (7):

$$d^2[Ve(\alpha)/Veo]/d\alpha^2=P(\beta)*[(L*W+Vc(\beta)/Veo/[m*w^2Yo/S(L*W+Q*Ve(\alpha)/Veo+Vc(\alpha)/Veo*S*(Pl+Ph)/(2*m*w^2*Yo)] \qquad (8)$$

Here

$P(\beta)$ can be calculated by the following equation

$$d^2(Vc(\alpha)/d\alpha^2=0 \qquad (9)$$

and $Vc(\beta)$ can be found by formula (5).

Simulated by numerical analysis, $Vc(\alpha)$ in (8) and $P(\alpha)$ in (6) can be determined. On the basis of these results, we can have variations of all functional parameters indicated in Figs 4, 5, 6 and 7 that verifies the proper function of this energy-saving regenerative refrigerating system.

The refrigerating temperature can reach down to 180°K and refrigerating capacity is 224 Btu/hr. The working gas is nitrogen and turning speed of crank is about 140 R/min. The refrigerating temperature as a time function is shown in Fig.3. The other variations of the functional parameters are also resolved by simulation. The relation that P/Pmax changes with α is shown in Fig.4. The indicator cards of expansive cavity and full system are shown in Fig.5 and 6. The distributions of masses such as Me in expansive cavity, Mc in compressive cavity and Mk in clearance volume are indicated in Fig.7. From these figures, we can see that this regenerative refrigerating system functions properly because the variations of pressure and mass in the system are ideal and enclosed area by cycle curves in expansive cavity is acceptable.

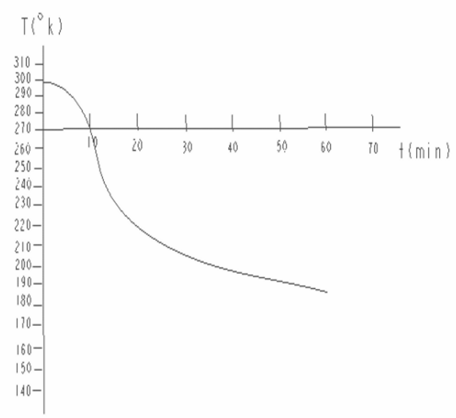

Fig.3 Refrigerating Temperature as a Function of Time

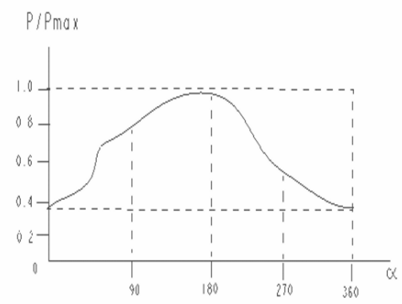

Fig.4 Pressure Ratio P/Pmax as a Function of Crank Angle α

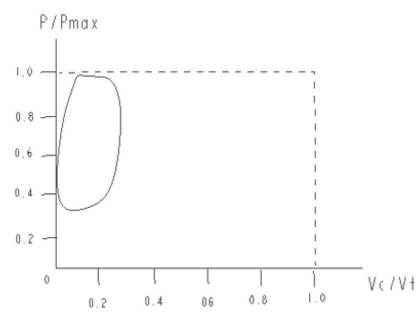

Fig.5 Pressure Ratio P/Pmax as a Function of Volume Ratio Ve/Vt in Expansive Cavity

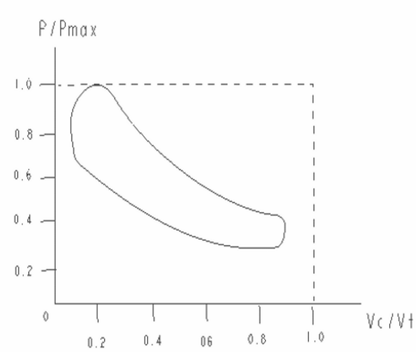

Fig.6 Pressure Ratio P/Pmax as a Function of Volume Ratio V/Vt in Total Cavity

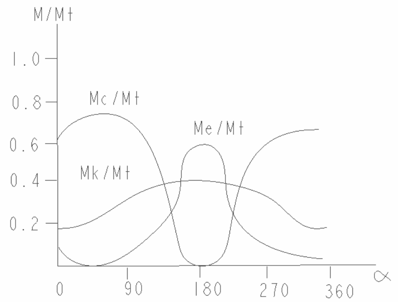

Fig. 7 Mass Ratio Mi/Mt as the Function of Crank Angle ∝ in Compressive Cavity, Expansive Cavity and Clearance Space

Conclusion

The feasibility of this new energy-saving refrigerating system has been verified on the basis of thermodynamic study, theoretical and numerical analysis. Its proper function has also been proved by the prototype of this energy-saving system. The major advantages of this system are that the system is simple and compact, vibration is significantly reduced due to its symmetrical structure, functioning efficiency is higher because its refrigerating capacity can be conveyed out through both ends of the cylinder.

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Iterated Local Search Approaches to Maximin Latin Hypercube Designs

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***Abstract*- The problem of maximizing the minimal distance in Latin Hypercube Designs (maximin LHD) calls for arranging *N* points in a *k*-dimensional grid so that no pair of points share a coordinate and the minimal distance between all the pairs of points is as large as possible. Such problem is particularly relevant in designing computer experiments.**

In this paper we propose two Iterated Local Search (ILS) heuristics for this problem and show through some computational experiments that the proposed algorithms compare very well with different heuristic approaches in the established literature.

I. Introduction

Consider a set of N points in a uniform k-dimensional grid $\{0,1,\ldots,N\text{-}1\}^k$. A configuration

$$X = \begin{pmatrix} x_1 \\ \vdots \\ x_N \end{pmatrix} = \begin{pmatrix} x_{11} & \cdots & x_{1k} \\ \vdots & \cdots & \vdots \\ x_N & \cdots & x_{1N} \end{pmatrix} \qquad (1)$$

with all $x_{ij} \in \{0,\mathrm{K}\,,N-1\}$ is a Latin Hypercube Design (LHD) if each column has no duplicate entries, i.e., no pair of points share a coordinate. Given a point-to-point distance metric $d(x_i, x_j), i, j \in S$, the *maximin* LHD problem is to find a LHD such that the minimum point-to-point distance occurring in such configuration is as large as possible.

LHDs, first introduced in [1], are important in the design of computer-simulated experiments (see, e.g., [2]), where the chosen *design points* are often required to be evenly spread in the parameter space, and *non-collapsing* --- i.e., non-overlapping even when projected on a single axis of the parameter space. The max-min objective, first introduced by Niederreiter [3], captures the requirement of having evenly spread points. Different definitions for the distance $d(x_i, x_j)$ are considered in literature; in this work we consider d to be the Euclidean distance, which is one of the most frequently used in applications (see [1]). The definition of optimal LHDs through the maximin criterion has been proposed in [5].

For our purposes, we discriminate among LHDs as follows; let $D_1(X) = \min\{d(x_i, x_j) : i \neq j\}$ be the minimum distance occurring in X, and let $J_1(X)$ be the number of occurrences of such distance in X. We consider X to be a LHD better than Y if

$D_1(X) < D_1(Y)$, or

$D_1(X) = D_1(Y)$ and $J_1(X) < J_1(Y)$. (2)

We note that, following [6], in the case where different LHDs share the same D_1 and J_1 values, one could further discriminate by maximizing lexicographically, D_2, the second smallest distance in X, and, if equality still holds, by minimizing J_2, the number of occurrences of D_2, and so on. In [6] all the above measures have been combined into a unique objective function Φ_p to be minimized and tests have been performed with a simulated annealing approach (see the reference for details).

From the point of view of combinatorial optimization, the complexity of the maximin LHD design problem (with Euclidean distance) is, to the authors' knowledge, open (not known to be NP-hard, neither polynomially solvable), but very likely NP-hard. Different methods have been presented in the literature to detect maximin LHDs. In [7] a class of algorithms based on column pairwise exchange has been proposed to build supersaturated designs.

In [8] an exchange algorithm for finding approximate maximin LHDs has been presented with the further restriction to Symmetric LHDs (SLHDs). In [8] general formulae for maximin LHDs with k=2 are given for the 1-norm and infinite norm distances, while for the Euclidean distance (approximate) maximin LHDs up to N=1000 design points are obtained by (adapted) periodic designs, while, using a branch-and-bound algorithm, exact solutions have been obtained for N up to 70. Inspired by [9], Husslage et al. [10] proposed (adapted) periodic designs and simulated annealing to extend the known results and construct approximate maximin Latin hypercube designs for k up to 10 and N up to 100. All these designs are available at `http://www.spacefillingdesigns.nl`. In [10] it has been shown that the periodic heuristic tends to work better when the number N of design points gets above some threshold which depends on the dimension k of the design (more precisely, such threshold increases with k).

We remark that the maximin criterion is not the only one used in the literature. Other criteria are the maximum entropy [11], the integrated mean squared error [5], the minimum

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correlation between components [13],[14] and a mixed criterion involving both maximin distance and correlation [15]. For more details we also refer to the book [16].

In this paper we present an Iterated Local Search (ILS) for the maximin LHD problem with Euclidean distance. ILS is a well known metaheuristic approach for solving combinatorial optimization problems (for a review see, e.g., [17]). Its basic idea is simple but at the same time quite effective. The efficiency of ILS depends on a clever choice of Local Search procedures as well as perturbation techniques. Our preliminary experiments show that ILS is able to outperform existing methods for the maximin LHD problem.

In Section II, we give the details of ILS heuristics to detect maximin LHDs; in Section III, we present some computational experiments. Conclusions are drawn in Section IV.

II. ILS FOR MAXIMIN LHD

A Local Search procedure for an optimization (maximization) problem --- from now on we explicitly assume that the problem is maximin LHD --- generates a sequence of solutions X^1, X^2,…, X^t, where each X^i is better than X^{i-1} and is obtained by applying to X^{i-1} a *local move* operator; X^i is also said to belong to a neighbourhood of X^{i-1}. The procedure stops at a local maximum X^t, from which the local move is not able to further generate better solutions. Once a local search procedure is defined, it is a crucial issue how to select a good starting solution for the local search; in general, this is not obvious at all, and a single run of local search usually returns a poor solution. An obvious improvement can be attained by multiple runs of the local search procedure, each starting from randomly generated solutions. This leads to the so called *MultiStart* (MS) approach. But this way each local search is completely independent from all the previous ones, so that even when previously detected local maximizers are good ones, the next run cannot get any advantage from this. ILS exactly tries to avoid that: in ILS the next starting point is generated by a perturbation (usually slightly larger than those performed during the local search steps) of the current local maximizer. The rationale behind this idea is the following: if we are currently in a good local maximizer, it is not a good strategy to completely disrupt its structure because, very likely, there are some of its portions that are already "optimized" and should not be changed; therefore, we do not generate a completely new solution but we (slightly) perturb the current one and this is used as the initial solution for the next Local Search. In the following subsections we will detail the key components of our ILS heuristic for maximin LHD, namely: the choice of the initial solution, the local move operator used in the local search, and the perturbation operator.

Initial solution. The first initial solution is randomly generated. In particular, the generation is as follows. For each component $h\in\{1, ..., k\}$ a random permutation $(v_1, \dots, v_n)$ of the integers 0, 1 ,..., N-1 is generated and we set

$$x_{rh} = v_r \quad \forall\, r \in \{1,\Lambda,N\}$$

Local search. In order to define a local search procedure, we need to introduce the set of possible *local moves* with respect to a given LHD $X=(x_1,\dots,x_N)$, with $x_i \in \{0,\dots N-1\}^k$, $i\in\{1, ..., N\}$. The local move employed here is a quite natural one already proposed, e.g., in [10]. First, we need to introduce the notion of *critical point*. We say that i is a critical point for X, if $\min_{j:i\neq j}\{d(x_i,x_j)\} = D_1(X)$.

We denote by $I(X)\subseteq\{1,\dots,N\}$ the set of indices of the critical points in X. Given $i\in I(X)$, $j\in\{1,\dots,N\}$, and a component $l\in\{1,\dots,k\}$, the local move employed here simply swaps the l-th component of the two points and returns the resulting LHD denoted by Y. More formally, the resulting LHD Y is defined as follows

$$y_{rh} = \begin{cases} x_{rh} & \text{if } r \neq i,j \quad \text{or} \quad h \neq l \\ x_{ih} & \text{if } r = j \quad \text{and} \quad h = l \\ x_{jh} & \text{if } r = i \quad \text{and} \quad h = l \end{cases} \tag{3}$$

The local search procedure based on such local move is the following

```
repeat
  for  i = 1,Λ ,N   do
    for  j = 1,Λ ,N do
     If  i ≠ j and  {i, j} ∩ I(X) ≠ φ ,  then:
     Let  Y' = X
     for  l = 1,Λ ,k  do
      let  Y_l =the LHD obtained by (3)
      if  Y_l  is better than  Y'  then
             set  Y' = Y_l .
     end for
     set  X = Y'  .
    end for
  end for
until <No improvement is detected>;
return  X
```

We remind the reader that the meaning of "Y better than X" is that established by (2). Note that with this definition it perfectly makes sense not to consider pairs $\{i,j\}$ such that both points do not belong to $I(X)$. Indeed, according to (3), any swap involving two non critical points is not able to improve the current LHD. We note that a crude re-evaluation of the objective function after a swap would lead to a computationally expensive procedure — this would take at least $O(N^2)$ time. Instead, by storing the distance matrix $\mathbf{D}$, whose entries are the distances $d(x_i,x_j)$, and incrementally compute the modified distances for entries

$\{i,h\}$, $\{j,h\}$, $h \neq i,j$, we have a much cheaper linear-time procedure.

Finally, we note that in the inner loop we always update the current LHD X with the largest improvement (if any) after trying to swap all the components. We also tested the version where we update X as soon as a swap leads to an improvement. However, the two strategies have not shown significative differences.

Perturbation. Perturbation is the key operator in ILS. Basically, a perturbation might be seen as a local move like the ones described for the local search procedure, but it is somehow *less local*, or, more precisely, it is a move within a neighbourhood larger than that employed in the local search. In this paper we propose two different kinds of perturbations defined below.

Our first perturbation operator is a Cyclic Order Exchange (COE) upon a single component (column) of a randomly selected portion of the design points (rows). First, we randomly choose two distinct rows/points, say x_i and x_j, such that $i<j$ and $j-i \geq 2$, in the current LHD X^*. Then, we randomly choose a component/column, say l. Finally, we swap in cyclic order the value of component l from point x_i to point x_j. Note that we require $j-i \geq 2$ because otherwise the perturbation would be a special case of the local move employed in the local search procedure. We illustrate how COE works by an example. Assume we have, with N=6 and k=8, the current LHD

$$X^* = \begin{pmatrix} x_1 \\ x_2 \\ x_3 \\ x_4 \\ x_5 \\ x_6 \end{pmatrix} = \begin{pmatrix} 0 & 1 & 2 & 3 & 4 & 5 & 5 & 4 \\ 1 & 2 & 3 & 4 & 5 & 0 & 4 & 3 \\ 2 & 3 & 4 & 5 & 0 & 1 & 3 & 2 \\ 3 & 4 & 5 & 0 & 1 & 2 & 2 & 1 \\ 4 & 5 & 0 & 1 & 2 & 3 & 1 & 0 \\ 5 & 0 & 1 & 2 & 3 & 4 & 0 & 5 \end{pmatrix} \qquad (4)$$

Now we randomly choose two rows/points, say x_2 and x_5, and we randomly choose the component/column l=4. Then, after the COE perturbation we get the following LHD X'_{COE} (changes are emphasized in bold)

$$X'_{COE} = \begin{pmatrix} x_1 \\ x_2 \\ x_3 \\ x_4 \\ x_5 \\ x_6 \end{pmatrix} = \begin{pmatrix} 0 & 1 & 2 & 3 & 4 & 5 & 5 & 4 \\ 1 & 2 & 3 & \mathbf{1} & 5 & 0 & 4 & 3 \\ 2 & 3 & 4 & \mathbf{4} & 0 & 1 & 3 & 2 \\ 3 & 4 & 5 & \mathbf{5} & 1 & 2 & 2 & 1 \\ 4 & 5 & 0 & \mathbf{0} & 2 & 3 & 1 & 0 \\ 5 & 0 & 1 & 2 & 3 & 4 & 0 & 5 \end{pmatrix}.$$

Our second perturbation is Single Pair Crossover (SPC) upon a randomly selected pair of rows/points. First, we randomly choose two different rows/points, say x_i and x_j, such that $i \neq j$ in the current LHD X^*. Then, we randomly choose a component say l, $l \in \{2,\ldots,k-1\}$, with respect to which the crossover operation will be performed . Finally, we exchange the portion made up by the first l components of x_i (i.e., coordinates 1 to l) with the same portion of x_j. Note that for k=2, such perturbation reduces to a single swap of the local search procedure, so that it makes sense to apply this perturbation only for k>2. As an example, starting from the current LHD (4) we can randomly choose two rows/points, say x_1 and x_4, then we randomly choose component l, i.e., the number of column, say l=3. Then, after the SPC perturbation we get the following LHD X'_{SPC}

$$X'_{SPC} = \begin{pmatrix} x_1 \\ x_2 \\ x_3 \\ x_4 \\ x_5 \\ x_6 \end{pmatrix} = \begin{pmatrix} \mathbf{3} & \mathbf{4} & \mathbf{5} & 3 & 4 & 5 & 5 & 4 \\ 1 & 2 & 3 & 4 & 5 & 0 & 4 & 3 \\ 2 & 3 & 4 & 5 & 0 & 1 & 3 & 2 \\ \mathbf{0} & \mathbf{1} & \mathbf{2} & 0 & 1 & 2 & 2 & 1 \\ 4 & 5 & 0 & 1 & 2 & 3 & 1 & 0 \\ 5 & 0 & 1 & 2 & 3 & 4 & 0 & 5 \end{pmatrix}.$$

Note that COE as well as SPC only slightly modify the current LHD X^* but this exactly follows the spirit of ILS, where the perturbation should keep unchanged large portions of the current solution and should not completely disrupt its structure.

Details of the ILS procedure. The ILS procedure executes successive runs of local search, and keeps track of the best known local maximizer returned by the local search. The starting point for each run is obtained by applying a perturbation to the current best known local maximizer. When a stopping condition is met, this best maximizer is returned as the result of ILS. For the stopping condition, we introduced an integer parameter called `MaxNoImprove` and stopped the heuristic if the best maximizer did not change for `MaxNoImprove` iterations of the heuristic, i.e., if no improvement was detected in the last `MaxNoImprove` iterations. In what follows we will denote by ILS (COE) and ILS (SPC) the versions of ILS employing, respectively, the COE and SPC perturbation.

III. Results and Discussion.

We performed different computational experiments to test our ILS heuristic and compared it with results offered by known algorithms in the state of the art literature. The details are given in Tables I and II. The results for Periodic Design (PD) and Simulated Annealing (SA) algorithms are from [10] (also available at `http://www.spacefillingdesigns.nl`. Table III compares the total times of Multistart (MS), ILS(COE) and ILS(SPC) spent on the whole batches used for Tables I and II.

We considered LHDs with N=3, 4, ..., 25 and k=3, 7, 9,10. For each pair of these (N,k) values we ran ILS 100 times with `MaxNoImprove=1000`. Columns ILS(SPC) and ILS(COE) report the best *squared* maximin distance obtained over the

TABLE I :

COMPARISON AMONG DIFFERENT APPROACHES

	k=3					k=7				
N	PD	SA	MS	ILS COE	ILS SPC	PD	SA	MS	ILS COE	ILS SPC
2	3	3	3	3	3	7	7	7	7	7
3	3	6	6	6	6	7	13	13	13	13
4	6	6	6	6	6	16	21	21	21	21
5	6	11	11	11	11	16	32	32	32	32
6	14	14	14	14	14	29	47	47	47	47
7	14	17	17	17	17	31	61	61	61	61
8	21	21	21	21	21	46	79	78	79	79
9	21	22	22	22	22	47	93	91	93	93
10	21	27	27	27	27	68	110	108	**111**	108
11	24	30	30	30	30	69	128	128	**131**	128
12	30	36	36	36	36	95	150	148	**153**	152
13	35	41	41	41	41	95	174	168	**180**	179
14	35	42	41	42	42	119	204	188	**215**	213
15	42	48	44	48	48	129	211	210	221	**223**
16	42	50	46	50	50	155	238	231	241	242
17	42	53	51	53	53	161	256	257	**266**	263
18	50	56	54	56	56	186	281	279	**292**	288
19	57	59	56	59	59	195	305	304	317	**321**
20	57	62	61	62	62	226	332	329	**349**	344
21	65	66	65	**69**	**69**	236	361	356	**375**	374
22	69	69	68	70	**72**	270	384	385	**408**	406
23	72	74	72	**75**	**75**	273	410	416	**441**	434
24	76	78	75	78	78	308	444	444	468	**471**
25	91	81	78	83	**86**	350	467	471	500	**504**

TABLE II.

COMPARISON AMONG DIFFERENT APPROACHES

	k=9				k=10			
N	SA	MS	ILS COE	ILS SPC	SA	MS	ILS COE	ILS SPC
2	9	9	9	9	10	10	10	10
3	18	18	18	18	19	19	19	19
4	28	28	28	28	33	33	33	33
5	43	43	43	43	50	50	50	50
6	61	61	61	61	68	68	68	68
7	80	79	80	80	89	89	89	89
8	101	101	**103**	101	114	114	114	114
9	126	124	**127**	126	141	140	**142**	141
10	154	150	**156**	155	172	170	**173**	172
11	178	173	**180**	178	206	198	**207**	206
12	204	200	**208**	205	235	228	**237**	235
13	232	227	**239**	235	267	262	**271**	269
14	265	260	**273**	270	298	294	**309**	305
15	296	291	**310**	312	337	333	**352**	346
16	330	324	**351**	347	378	371	**394**	391
17	367	360	**399**	391	415	411	**444**	435
18	398	395	**446**	**446**	458	454	**494**	484
19	438	433	**467**	462	498	496	**537**	530
20	472	470	**504**	500	542	543	590	**596**
21	517	508	**542**	541	592	589	629	**631**
22	555	554	**586**	583	643	639	**684**	681
23	596	593	**629**	627	685	686	**732**	728
24	639	636	**678**	676	739	738	**785**	781
25	688	681	**725**	**725**	792	793	**840**	**840**

100 runs, for ILS algorithms embedding SPC and COE perturbations respectively. The column labelled MS reports results obtained by a pure Multi Start strategy embedding our local search procedure. See below for details of MS testing. All the experiments ran on a Opteron 252 1 GHz CPU machine with 4 GB RAM running Linux. We stress that the figures in the tables are best-found solutions, so even a small (in percentage) increase in the value is a significantly better result.

Comparison with Multi Start. In order to understand the impact of the perturbation moves in ILS when applied to LHD optimization, it is important to compare it with the simplest method based on multiple local searches, i.e., Multi Start. We recorded the best result obtained by ILS(SPC) in 100 runs and the total number of local searches performed. After that, for this experiment, we ran MS with exactly the same number of local searches performed by ILS. From Tables I-II we observe that MS is never better and often worse, especially when the number of points and the dimension are large. Moreover, the running time for ILS is clearly inferior (see also Table III below). This is due to the fact that the starting points of local searches in ILS are only slight perturbations of local maximizers, so that the local search procedure gets to a (possibly) new local maximizer in fewer iterations with respect to the case where starting points are completely random ones.

Comparison with existing literature. Now we compare our results produced by the algorithms MS, ILS(COE) and ILS(SPC) with those reported for the Periodic Design (Column `PD`) and Simulated Annealing (Column `SA`). All such results are reported in Tables I-II. In bold we indicate the best obtained result for all the tested (N,k) cases.

Both ILS(COE) and ILS(SPC) are always at least as good as the SA approach and it is interesting to note that the improvements tend to become very large as the dimension k increases. Both ILS(COE) and ILS(SPC) are also often (much) better than PD but inferior for N=25,k=3. This is similar to what observed in [10]. In that paper it has been observed that there is a critical number of points above which PD tends to outperform SA. Such critical number quickly increases with the dimension k and, e.g., for k≥6 such value is certainly above N=100. Also considering further experiments that we have done for N>25, we observed that both ILS(COE) and ILS(SPC) allow to increase the critical value where PD becomes better than these ILS versions, but for N large enough both versions get dominated by PD for small k values.

Although the quality of the results obtained by the two ILS versions appears to be quite good, it is important to see whether such results have only been obtained by brute force, i.e., at the cost of a very large computational effort. Collecting all the results for ILS required a few hours on our machine (see Table III).

We do not have the details for the computation times of the SA and PD approaches discussed in [10]. Therefore, we performed the following experiment. For each pair (N,k) of tested values we stopped ILS (for this experiment we only tested `ILS(COE))` when it reached a solution with at least the same quality of SA. It turned out that the best results of

TABLE III. TIME COMPARISON (IN SECONDS) BETWEEN MULTISTART AND BOTH VERSIONS OF ILS.

k	MS	ILS(COE)	ILS(SPC)
3	663.2	334.7	259.7
7	4192.7	2150.7	1222.5
9	5609.6	3969.0	2490.1
10	9466.8	4771.7	3027.2

SA are often reached within fractions of a second and in any case in a time which never exceeds 30 seconds.

Note that in some cases even MS returns results better than SA, but we suspect that the computation times for MS are definitely much larger than those for SA.

We observe in Tables I-II that though both versions are comparable, yet ILS(COE) on most instances gets better results than ILS(SPC) with respect to solution quality. However, with respect to the computational cost, Table III suggests that ILS(SPC) is somewhat better than ILS (COE). Finally, we remark that our results are always at least as good and in some cases improve the few (N,k) cases tested in [6]. In particular, improvements have been obtained for the four cases (11,3) ; (12,6); (7, 7); (9,9).

IV. CONCLUSIONS

In this paper we considered the problem of finding maximin Latin Hypercube Designs and we attacked such problem by Iterated Local Search heuristics. In our experiments we used two different kinds of perturbation, COE and SPC. The computational experiments have shown that both versions of ILS are particularly efficient for this problem and allow to improve (in some cases considerably) the known results in the literature, in particular as the dimension k of the problem increases. We point out that here we restricted our attention to heuristics which are quite easy to implement but we are confident that more sophisticated versions would allow to further improve the obtained results.

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Fragmentation in Distributed Databases

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Abstract. **The design of distributed database is an optimization problem and the resolution of several sub problems as data fragmentation (horizontal, vertical, and hybrid), data allocation (with or without redundancy), optimization and allocation of operations (request transformation, selection of the best execution strategy, and allocation of operations to sites). There are some different approaches to solve each problem, so this means that the design of the distributed databases is become hard enough. There are many researches connected to the dates fragmentation and they are presented both in the case of relational database and in the case of object-oriented database. In this paper is presented the implementation of a heuristic algorithm conceived before that uses an objective function who takes over information about the administrated dates in a distributed database and it evaluates all the scheme of the database vertical fragmentation.**

1. INTRODUCTION IN VERTICAL FRAGMENTATION OF DATABASES

There exits three fragmentation types: vertical, horizontal and hybrid. Vertical fragmentation consists of subdividing a relation into sub relations that are projections of the original relation according to a subset of attributes. The horizontal fragmentation divides a relation into subsets of tuples based on selection operations. The hybrid fragmentation consists of dividing a relation horizontally, and then splitting vertically each of the obtained horizontal fragments or vice-versa.

Vertical fragmentation is used in order to increase transaction performance. The more obtained fragments are close to transaction requirements, the more the system is efficient. The ideal case occurs when each transaction matches exactly a fragment, i.e. it needs only this fragment. If some attributes are always used together, the fragmentation process is trivial. However, in reality applications are rarely faced with such trivial cases. For relations having tens of attributes, it is necessary to develop systematic approaches for vertical partitioning. If a relation has m attributes, it can be partitioned following B(m) different ways, where B(m) is the m^{th} Bell number which is almost m^m (Hammer and Niamir [8]).

Since the beginning of the 80's, many works have adressed the database vertical partitioning problem.

(Hoffer and Severance, [9]) have developed the attribute affinity concept. This metric measures the frequency of accesing simultaneously a couple of attributes. The attributes having high affinity are grouped together by using the Bond Energy Algorithm.

(Hammer and Niamir, [8]) have proposed a heuristic where the input is a set of blocks corresponding each one to an attribute. This is the initial candidate partition. At each step of the search, several modifications of the partitions are generated and then submitted to a cost evaluator. If one of the modified partitions becomes the current candidate partition and the search continues until no modification is possible. Modifying a partition may be obtained in two different ways: by grouping two blocks or by regrouping an attribute i.e. by removing it from one bloc and inserting it into another one.

(Navathe, *et al.,* [13]) extend the work of (Hoffer and Severance, [9]). The authors use an attribute affinity matrix that they order by using Bond Energy Algorithm as proposed in (Hoffer and Severance, [9]). However, determining the vertical fragments is done automatically, whereas it was let the subjectif judgement of the designer in (Hoffer and Severance, [9]). There are two steps in the partitioning algorithms. In the first step, the fragmentation is obtained by appying iteratively a binary partitioninf algorithm. At this step, no cost factor is considered. At second step, estimations of cost reflecting the physical environmemt, are included in order to optimise the initial fragments.

(Cornell and Yu, [5]) proposed a vertical partitioning algorithm which minimizes the number of disk accesses. The algorithm is based on integer programming methods. The partititoning of a relation requires the knowledge of several parameters concerning the relation (length, selectivity and number of attributes) and transaction types and behaviour (their frequency and the attributes they access).

(Ceri, *et al.* [2]) propose two tools for vertical fragmentation: "DIVIDE" and "CONQUER". The tool "DIVIDE" performs only data fragmentation and allocation; it implements the partitioning algorithm proposed in (Navathe, *et al.,* [13]). The tool "CONQUER", in addition to data fragmentation and allocation, ensures the optimisation and allocation of operations.

Navathe and Ra proposed in [15] a graphical technique of partitioning. The attibute affinity matrix

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is considered as a complete graph where nodes represent attibutes and edges' weights represent the affinty values. The algorithm, by successively adding edges, generates all the fragments in one iteration by considering a cycle as a fragment.

(Lin *et al.* [11]) extend the work of (Navathe and Ra, [15]) on graphical partitioning. The input to the algorithm is the affinity graph. They proposed searching a subgraph of at lest two nodes for which affinity values are greater than those of each incident edge.

(Chakravarthy, *et al.* [3]) have develop a partition evaluator which evaluates the partition quality by using two costs: the access cost to the irrelevant local attributes (present on the execution site of the transaction but not used by the transaction), and the access cost to the irrelevant remote attributes (not present on the execution site of the transaction but necesary for its execution).

Several authors have approached the generalization of fragmentation techniques to complex value and object oriented data models. For instance, horizontal fragmentation is discussed in (Bellatreche, Karlapalem and Simonet [1], Ezeife and Barker [5], Ma [12], Schewe [16]), and vertical fragmentation in (Chinchwadkar and Goh [4], Ezeife and Barker [6], Malinowski and Chakravarthy [13]).

2. STRUCTURE OF THE PAPER

In this paper we present the implementation of the partition evaluator which was describe in [4].

We delimit our discussion to one of the data fragmentation problems, namely the vertical partitioning problem. Vertical Partitioning (also called attribute partitioning) is a technique that is used during the design of a database to improve the performance of transactions. In vertical partitioning, attributes of a relation R are clustered into non-overlapping groups and the relation R is projected into fragment relations according to these attribute groups. In distributed database systems, these fragments are allocated among the different sites. Thus the objective of vertical partitioning is to create vertical fragments of a relation so as to minimize the cost of accessing data items during transaction processing. If the fragments closely match the requirements of the set of transactions provided, then the transaction processing cost could be minimized. Vertical partitioning also has its use in partitioning individual files in centralized databases, and dividing data among different levels of memory hierarchies etc. In the case of distributed database design, trans-action processing cost is minimized by increasing the local processing of transactions (at a site) as well as by reducing the amount of accesses to data items that are not local. The aim of vertical partitioning technique (and in general data partitioning techniques) is to find a partitioning scheme which would satisfy the above objective.

3. CONTRIBUTION

In this paper we are using the approach of formulating an objective function (named Partition Evaluator) before developing (heuristic) algorithms for the partitioning problem. This approach enables us to study the properties of algorithms with respect to an agreed upon objective function, and also to compare different algorithms for "goodness" using the same criteria. The objective function formulated in this paper is a step in this direction. Moreover, the objective function derived in this paper can be easily extended to include additional information (e. g., query types - retrieval/update, allocation information about the partitions, remote processing cost, and the transaction usage pattern at any particular site).

4. DEFINITIONS AND NOTATIONS

A *partition* (scheme) is a division of attributes of a relation into vertical fragments in which for any two fragments, the set of attributes of one is non-overlapping with the set of attributes of another. For example, the partition {(1,3) (2,4) (5)} defines a collection of fragments in which attributes 1 and 3 are in one fragment, 2 and 4 are in another and 5 is in a separate fragment. The following are used in the derivation of the Partition Evaluator.

n: Total number of attributes in a relation that is being partitioned.

T: Total number of transactions that are under consideration.

q_t: Frequency of transaction t for $t = 1, 2, \ldots, T$.

M : Total number of fragments of a partition

n_i : Number of attributes in fragment i

n_{ikt}^{r} : Total number of attributes that are in fragment k accessed remotely with respect to fragment i by transaction t.

f_{tj}^{i} : Frequency of transaction t accessing attribute j in fragment i. Note that f_{tj}^{i} is either 0 or q_t.

A_{ij} : Attribute Vector for attribute j in fragment i. *t-th* component of this vector is f_{tj}^{i}.

R_{itk} : Set of relevant attributes in fragment k accessed remotely with respect to fragment i by transaction t; these are attributes not in fragment i but needed by t.

$| R_{itk} |$: Number of relevant attributes in fragment k accessed remotely with respect to fragment i by transaction t.

Some algorithms such as Bond Energy [5], and [15], use affinity matrix as the input. The attribute affinity is a measure of an imaginary bond between a pair of attributes. Because only a pair of attributes is involved, this measure does not reflect the closeness or affinity when more than two attributes are involved. Hence the algorithms which use attribute affinity matrix are using a measure (that is an ad hoc extrapolation of pair wise affinity to cluster affinity) that has no bearing on the affinity as measured with respect to the entire cluster. As a consequence, we believe, it was difficult to show or even characterize affinity values for the resulting clusters having more than two attributes.

As we wanted to obtain a general objective function and a criterion for describing affinity value for clusters of different sizes, our approach does not assume an attribute affinity matrix. The input model that we consider is a matrix which consists of attributes (columns) and the transactions (rows) with the frequency of access to the attributes for each transaction, as the values in the matrix. With this input model we overcome the limitations that are inherent to approaches based on attribute affinity matrix.

The objective function used by one algorithm is not suitable for evaluating the "goodness" of other algorithms. Thus we do not have a common objective function to compare and evaluate the results of these partitioning algorithms, or in general evaluate the "goodness" of a particular partitioning scheme. Hence we need a partition Evaluator to compare and evaluate different algorithms that use the same input in the database design process. Since attribute usage matrix is the most commonly used input available during the initial design stage, we first design an Evaluator which can be used to evaluate the "goodness" of partitions arrived at using this input. This Partition Evaluator can be used as a basis for developing algorithms to create fragments of a relation. With this approach, there is hope that admissibility aspects of algorithms can be shown. In addition, this Partition Evaluator has the flexibility to incorporate other information, such as type of queries (retrieval/updates), allocation information about the partitions, remote processing cost (transmission cost) and the transaction usage pattern at any particular site.

In any practical database application, a transaction does not usually require all the attributes of the tuples of a relation being retrieved during the processing of the transaction. When a relation is vertically divided into data fragments, the attributes stored in a data fragment that are irrelevant (i.e., not accessed by the transaction) with respect to a transaction, add to the retrieval and processing cost, especially when the number of tuples involved in the relation is very large. In a centralized database system with memory hierarchy, this will lead to too many accesses to the secondary storage. In a distributed database management system, when the relevant attributes (i.e., attributes accessed by a transaction) are in different data fragments and allocated to different sites, there is an additional cost due to remote access of data. Thus one of the desirable characteristics of a distributed database management systems that we wish to achieve through partitioning is the local accessibility at any site. In other words, each site must be able to process the transactions locally with minimal access to data located at remote sites.

4.1. IRRELEVANT LOCAL ATTRIBUTE ACCESS COST

For the first component we use square-error criterion as it was presented in (Jain and Dubes, 1988)

The general objective is to obtain that partition which, for a fixed number of clusters, minimizes the square-error.

Let us assume that n attributes have been partitioned into M fragments ($P_1, P_2, \ldots, P_m$) with n_i attributes in each fragment. Thus $\sum_{i=1}^{M} n_i = n$. The mean vector V_i for fragment i is defined as follows.

$$V_i = \frac{1}{n}\sum_{j=1}^{n_i} A_{ij} \qquad 0 < i \le M$$

This mean vector represents an average access pattern of the transactions over all attributes of fragment i. For an attribute vector A_{ij}, $(A_{ij} — V_i)$ is called the "difference vector" for attribute j in fragment i. The square-error for the fragment P_i - is the sum of the squares of the lengths of the difference vectors of all the attributes in fragment i. It is given by

$$e_i^2 = \sum_{j=1}^{n_i} (A_{ij} - V_i)^T (A_{ij} - V_i) \qquad 0 < i \le M \tag{3}$$

If $A_{ij} = V_i$ then e_i^2 will be zero. This will occur for the trivial case when there is a single attribute in each fragment or for the case when all the attribute in each fragment are relevant to all the transactions that access that fragment. It is the latter case that we are interested in and to avoid the former case, we will use the second component.

The square-error for the entire partition scheme containing M fragments is given by

$$E_M^2 = \sum_{i=1}^{M} e_i^2 \tag{4}$$

4.2. RELEVANT REMOTE ATTRIBUTE ACCESS COST

Now we will include the second component which would compute a penalty factor that computes the function. Given a set of partitions, for each transaction running on a partition compute the ratio of the number of remote attributes to be accessed to the total number of attributes in each of the remote partitions. This is summed over all the partitions and over all transactions giving the following equation. The second term is given by:

$$E_R^2 = \sum_{t=1}^{T} \Delta_{i=1}^{M} \sum_{k \ne i} \left[q_t^2 * |R_{itk}| \frac{|R_{itk}|}{n_{itk}^r} \right] \tag{5}$$

Here Δ^2 is an operator that is either an average, minimum or maximum over all i. These different choices of the operator give rise to average, optimistic and pessimistic estimates of the remote access cost. If specific information is available regarding transaction execution strategies, then we can determine for each transaction *t,* the remote fragments accessed by the transaction and the remote access cost can be refined accordingly. In our experimental investigation, we use the optimistic estimate for illustration.

Partition Evaluator (PE) function is given by:

$$PE = E_M^2 + E_R^2 \quad (6)$$

5. ANALYSIS OF THE PARTITION EVALUATOR

The final form of Partition Evaluator is given in equation 6. For analize and testing evaluator behavior, we implement an C++ program who produce all possible combinations of attribute with an number of fragments. We testing this program in three cases: case 1 - an 10 attributes and 8 transactions matrix; case 2 - an 5 attributes and 5 transactions, and case 3 – an 6 attributes and 4 transactions (1 to 10 fragments for case 1, 1 to 5 fragments for case 2, 1 to 4 fragments for case 3) partition evaluator was computed, and for minimum values, partitions scheme was stored and write.

Program we used is composed from 2 algorithms, one (called **PE algorithm**) for computed value on a given partition scheme and an number of fragments, and the other algorithm (called **GEN_PE** algorithm) computed the minimal value of the PE from all partition schemes generated in a backtracking mode.

The algorithm on which base we implemented the evaluator of parts is presented below.

The outgoing dates consist of the value for the local cost of access at irrelevant local attribute cost $E^2_{M,}$, the access cost on the distance of the relevant attributes E^2_R, respective the value of the fragmentation evaluator - *EP*

First we implemented the algorithm EP based on the formula from equation 6 calculates the value of EP, for a given fragmentation scheme so we used an entrance date: the matrix used for attributes - A; the lots of fragments on which it calculated the value of EP, the relation -R.

Algorithm PE
Input: *A* = attribute usage matrix;
R = Relation ; *F* = fragments set
Output: E^2_M: irrelevant local attribute cost;
E^2_R: relevant remote attribute cost;
EP : partition evaluator value
Begin
$E^2_M=0$
for i **from** 1 **to** number_of_fragments **do**
begin
$e_i=0$
for j **from** 1 **to** number of attributes from i fragment **do**
{ $X_{ij} – V_i$ – mean vector for j attribute form i fragment }
$e_i = e_i + (X_{ij}-V_i)^T*(X_{ij}-V_i)$
end_for
$E^2_M = E^2_M + e_i$
end_for
$E^2_R=0$
for t **from** 1 **to** number of transactions **do**
begin
minim =maxint
for i **from** 1 **to** number of fragments **do**
for k **from** 1 **to** number of fragments **do**
begin
if k $\neq$ i **then**
begin
{ R_{itk} – set of relevant attributes in fragment k accessed remotely with respect to fragment i by transaction t }
{ n^{remote}_{itk} – number of relevant in fragment k accessed remotely with respect to fragment i by transaction t }
if exist attribute in matrix A who is from k fragment **then**
$E^2_R= E^2_R+ (f^t_k)^2 * | R_{it} | * | R_{it} | / n^{remote}_{itk})$
end_if
if $E^2_{R\ min}$ < minim **then**
minim = $E^2_{R\ min}$
end_if
end_for
$E^2_R = E^2_R + E^2_{R\ min}$
end_for
end_for
$EP = E^2_M + E^2_R$
End.{Algorithm PE}

The second algoritm is presented below:

Algorithm GEN_PE
Input: A = attribute usage matrix;
Output : lowest PE value
partition scheme coresponding to the lowest EP value
Begin
minim=maxint
for frag **from** 1 **to** number_of_fragments **do**
{ one partition scheme is generation for frag }
pe = **call** PE(A, frag, F)
if pe<minim **then**
minim = pe
number_fragment = frag
G = F{set G is one copy of set F for corresponding value of minim}
end_if
end_for
write number_fragment, set G and PE value
End. {Algorithm GEN_PE}

For the execution of one transaction, we know that if a transaction could be run at one fragment and that fragment haven't one single attribute accessed by that transaction, then transaction not be run on that fragment.

For the first test we used a matrice of attributes use with ten attributes accesed by eight transactions.

Tranzactions \ Attributes	1	2	3	4	5	6	7	8	9	10
T1	25	0	0	0	25	0	25	0	0	0
T2	0	50	50	0	0	0	0	50	50	0
T3	0	0	0	25	0	25	0	0	0	25
T4	0	35	0	0	0	0	35	35	0	0
T5	25	25	25	0	25	0	25	25	25	0
T6	25	0	0	0	25	0	0	0	0	0
T7	0	0	25	0	0	0	0	0	25	0
T8	0	0	15	15	0	15	0	0	15	15

We present in Figure 1 values for each number of fragments and values for E_M^2, E_R^2 and EP.

Total number of fragments evaluates was 115975. Optimal value (minimum) is obtained for 3 fragments – fragment I (1,5,7), fragment II (2,3,8,9) and fragment III (4,6,10).

The program to generate all the combinations of ten attributes accessed by eight transactions offers three solutions (for five fragments) and two solutions (for eight fragments), having the same value for EP. However, the project of distributed database can choose which scheme of partition wishes to use it.

Number of fragments	Partition scheme	E_M^2 values	E_R^2 values	EP values
1	(1,2,3,4,5,6,7,8,9,10)	15085	0	15085
2	(1,4,5,6,7,10) (2,3,8,9)	7091	1366	8457
3	(1,5,7) (2,3,8,9) (4,6,10)	3312	2508	5820
4	(1,5) (2,3,8,9) (4,6,10) (7)	2078	3950	6028
5	(1,5) (2,3,8,9) (4,6) (7) (10) (1,5) (2,3,8,9) (4,10) (6) (7) (1,5) (2,3,8,9) (4) (6,10) (7)	2078	4800	6878
6	(1,5) (2,3,8,9) (4) (6) (7) (10)	2078	5650	7728
7	(1) (2,3,8,9) (4) (5) (6) (7) (10)	2078	6900	8978
8	(1) (2,8,9) (3) (4) (5) (6) (7) (10) (1) (2,3,8) (4) (5) (6) (7) (9) (10)	1386	10308	11694
9	(1) (2,8) (3) (4) (5) (6) (7) (9) (10)	0	14000	14000
10	(1) (2) (3) (4) (5) (6) (7) (8) (9) (10)	0	18350	18350

Fig. 1. Results for first test

For the second test we used a matrice of attributes use with five attributes accesed by five transactions.

Transactions \ Attributes	*A1*	*A2*	*A3*	*A4*	*A5*
T1	0	30	0	30	30
T2	15	15	15	0	15
T3	40	0	0	40	40
T4	0	10	10	0	0
T5	15	15	15	0	0

We present below values for each number of fragments together with the accordingly value opting for E_M^2, E_R^2 and EP.

Number of fragments	Partition scheme	E_M^2 values	E_R^2 values	EP values
1	(1,2,3,4,5)	3477	0	3477
2	(1,4,5) (2,3)	1369	770	2139
3	(1,4,5) (2) (3)	791	1470	2261
4	(1) (2) (3) (4,5)	144	3192	3336
5	(1) (2) (3) (4) (5)	0	5836	5836

Fig. 2 Results for second test

We can notice that for a number of two fragments - the fragment I (1,4,5) and the fragment II (2,3) we obtain the lowest value for EP.

Transactions \ Attributes	*A1*	*A2*	*A3*	*A4*	*A5*	*A6*
T1	0	105	105	0	105	105
T2	5	5	0	5	0	0
T3	14	0	0	14	14	0
T4	0	0	86	0	0	86

For the third test we used a matrice of attributes use with six attributes accesed by four transactions.

Number of fragments	Partition scheme	E_M^2 values	E_R^2 values	EP values
1	(1,2,3,4,5,6)	24895	0	24895
2	(1,4) (2,3,5,6)	7565	55	7620
3	(1) (2,3,5,6) (4)	7565	276	7841
4	(1) (2) (3,5,6) (4)	5063	11336	16399
5	(1) (2) (3,6) (5) (4)	0	22492	22492
6	(1) (2) (3) (4) (5) (6)	0	40913	40913

Fig. 3. Results for third test

We present below values for each number of fragments together with the accordingly value opting for E_M^2, E_R^2 and EP.

We can notice that for a number of two fragments - the fragment I (1,4) and the fragment II (2,3,5,6) we obtain the lowest value for EP.

6. CONCLUSIONS

In this paper is presented a general approach of the vertical fragmentation issue of the dates from a distributed database. Using an objective function used on the group models we obtained the implementation of an evaluator of partitions that can be use in the verification of some scheme of the dates fragmentation. Using this evaluator it's easier to project the heuristic algorithm or other nature for the partition of databases.

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Accessibility to the Disabled: A Service Oriented Architecture Approach

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***Abstract*-As today technology has witnessed remarkable advancement in almost all branches, still much attention needs to be directed to helping those disadvantaged from essential facilities. Specifically, much attention must be paid to the virtual world as it is the main theme of today's technology that makes a big world small. The virtual world is of much help to people with physical impartment. We need not forget those who are in the most need to such technology advancement, the handicapped people. In this paper, I propose a novel solution called "Adaptive Facility Middleware" – a Service-Oriented Architecture-Based middleware- to be incorporated in virtual educational environments enabling the join of handicapped students who wish to earn their degrees despite of their disability. This middleware will not require any virtual learning entity to change its general structure; however, it can be smoothly integrated to the current virtual learning environment to extend its operation to include the handicapped students. The proposed middleware adheres to the Service Oriented Architecture (SOA) which supports the integration of independent services. This makes the middleware extensible and interoperable.**

I. INTRODUCTION:

Nowadays technology is not only advancing in the real world, but it is also advancing in the virtual one opening a mean for innovative globalization that narrows down the world and make it a small village. However, no sufficient efforts have been directed toward people who will directly benefit from the virtual world; they are the physically handicapped people. Today's much efforts to help handicapped people is inclusive to establishing private schools or organization for those people putting them in more alienation from the rest of the world. Those organizations and/or associations yet offer a primarily education enough to get those people able to live a basic life while the rest of healthy people are requested to put pledges to aid those organizations. This makes the physically handicapped people like a burden on the healthy ones while they possess the same mental capabilities and can contribute to the world. What I am trying to stress in this paper is that handicapped people are mentally equal to the healthy ones and can be doctors, engineers, or accountants if they received the proper means to allow them access to professional education. This will not only solve the handicapped own problems, but it will solve our whole world's economical problem by getting those people participate in the advancement of the economy instead of being a burden on it. It will also increase services because those people when they become useful people, they will offer significant services to all kind of people. According to the InfoUse Project funded by the National Institute on Disability and Rehabilitation Research "An estimated 19.4% of non-institutionalized civilians" in the U.S only have a disability [1]; and it grows.

The proposed system which is called "Adaptive Facility Middleware" is a middleware to be placed between the real physical environment such as a university or a learning center and the virtual handicapped students at their homes or special organizations. The middleware acts as a transformation layer that receives the learning information from one end and converts it to proper representation according to the student's type of disability. The system is interoperable with both the educational environment and the student's environment. Also, the system adheres to the Service Oriented Architecture (SOA) methodology which allows the ability to add or remove independent services without affecting any other interacting parties or involved environments.

II. RELATED WORKS:

There have been many movements in the world trying to address the problem of accessibility to disabled people. The Americans with Disabilities Act (ADA) launched at 2000 called for considering disabled people in the web development [2] in its section 508 that disabled people

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shall have access to electronic information and the digital world without discrimination [3]. A considerable flaw to this act is that it proposes ethical guidelines to be followed in the development world to allow access to disabled people. However, abiding by those guidelines and rules is not guaranteed. Study has shown that only 59 percent of universities basic websites are accessible to disabled people [3]. This rate of providing access to disabled people to basic websites infer that the percentage of websites that enable interaction with disabled students for education or knowledge acquisition is far less this number.

Another more technological oriented endeavor has to deal with people of audio disability. The system proposed was called "Polygon Mesh" that uses Water's abstract-muscle based model that detects the facial expressions and jaws movements in order to generate liable audio expressions [4]. The system uses OpenGL C++ in order to simulate the vocal expressions of the disabled. Nevertheless, the proposed system imposes implementation efforts. The implementation of such system needs heavy integration with biomedical engineering and calls for heavy investment in terms of time and research efforts. Also, the seamless integration of such system with an e-learning infrastructure and the reliability of its simulated output are at question.

A notable study was presented at the proceedings of the fifth international ACM conference on Assistive technologies studied some virtual reality systems aiming at modeling educational systems for disabled students allowing them access to education and communication through virtual reality techniques. One studied system that can be used by blind students called "VirtualAurea" that is uses spacial sounds to convey a virtual world to students; for example, by embodying the structure of the classroom and the school map through sounds and it can customized by parents and teachers to accommodate other objects; another system for deaf students called "Mehida" that uses some techniques to enhance communication such as: finger spelling, hand gestures, and lip reading [5].

One major limitation in all systems discussed is that they do not address the difficulty of movement disabled people face. They jump to the conclusion of providing assistive means for the disabled to help them communicate and interact with the world around them; however, they do not address the problem from its early root which is that disabled people face the problem of commuting from one place to another. Another major limitation is the complexity of integrating such technologies with already existing educational online infrastructure. Such proposes systems did not show the handle of interoperability and compatibility issues associated with them.

III. THE ADAPTIVE FACILITY MIDDLEWARE:

The proposed middleware as a novel solution to the interaction needed between handicapped people and educational institutions is built on a Service-Oriented Architecture (SOA) methodology to allow the seamless extensibility and both flexible implementation and interoperation. I have adapted the SOA as an architecture to the middleware in order to address the diversity of disability that requires more service elaboration, the expected advancement in virtual reality technologies that has to be accommodated with minimal cost, and to reduce the cost of implementing educational system for the disabled for an online educational institution that want to offer education to the disabled students; hence, encouraging them to help the disabled.

A. SOA as an Architecture:

The SOA as an architectural organization aiming at treating system components as independent services that uses message passing to communicate with one another, it is extensible because as more services needed to be added later on to upgrade the virtual education system, it can be attached to the middleware easily without reimplementation efforts; components-off-the-shelf (COTS) and web services can be integrated to enhance the service of the middleware. The SOA enables platform independence feature for services implementation. This is a highly desirable feature of a system where learners behind different operating systems can have access to the educational system; thus, assuring accessibility. The implementation independence of programming languages offered by SOA will facilitate more integration and system enhancement as many developers can develop their own services and offer it to the system; it encourages global collaboration in software development to solve the accessibility problem of the disabled people. Services and applications can be integrated as a reusable code which reducing the implementation time and effort while assuring wide accessibility [6].

B. Adaptive Facility Middleware Services:

The proposed middleware is composed of different services which are used to deliver convenient means for the disabled students to access the virtual e-learning environment and interact with it. Essentially, more and more services can be added and different technologies can be adopted as provided by the SOA.

1. Blind Students:

The blind students are deprived from the ability to see things. However, they possess high sensitivity to hearing and sensing senses. The learning materials for blind students like books are available for them in Braille's method of writing. Thus, there is no problem for them to have access to hardcopy learning materials. However, what they are deprived from is the direct interaction in the real learning process. They are impaired with two obstacles regarding that. Firstly, there is a mobility problem because blind people have difficulty moving from place to place unless there is a guide which creates a dependency problem. Thanks to virtual technology, today there are e-learning environments provided by many respectable universities worldwide. Secondly, there is an interaction and involvement problem because blind people cannot get involved in the learning activity in classrooms. They will not able to see what's written on the virtual whiteboard or see the virtual handouts distributed in class which is considered a significant impairment especially for students wishing to pursue studies in science and engineering.

The adaptive facility middleware is to be placed between the blind students' environment and the virtual classroom. The middleware is provided with a text-to-speech service which transforms any written material into audio. This includes what's written on the virtual whiteboard by the instructor, all handouts distributed in class time, and assignments. Also, if the lecture entails diagrams or drawings done on the virtual whiteboard, the text-to-speech service can audibly describe the drawing; this can be achieved by incorporating a pattern recognition algorithm that can track the pattern of the drawing and convey it in an audio format. The transformed text into audio is done in real time while the lecture is being conducted to enable the blind students to participate actively in the lecture. Also, they are being recorded to be available for them to download the lectures for their later reference.

2. Mute Students:

Mute students are considered a different case from the latter two because they do not have any problem hearing the lecture, viewing the writing, or any learning material. However, they lack expressing themselves. This is crucial for the learning process because students need to interact with the instructor through asking questions or participating in class discussion. This problem can be solved in two techniques one of them is considered to be primitive and other is considered more sophisticated and automated. The first technique is to allow the mute students to type to the instructors in the lecture real time in a chat manner. The instructor and colleagues shall be notified upon receiving a message from the mute students. The second technique is to provide the mute students with a webcam at their end which will capture snapshots for their hand gestures as they express themselves normally with their sign language. There will be stored pictures of each sign in the sign language database by which the system will compare the taken snapshots of the mute student's hand gestures with the stored pictures in the database; then, displays the equivalent word to the instructor and colleagues in text or audio.

3. Deaf Students:

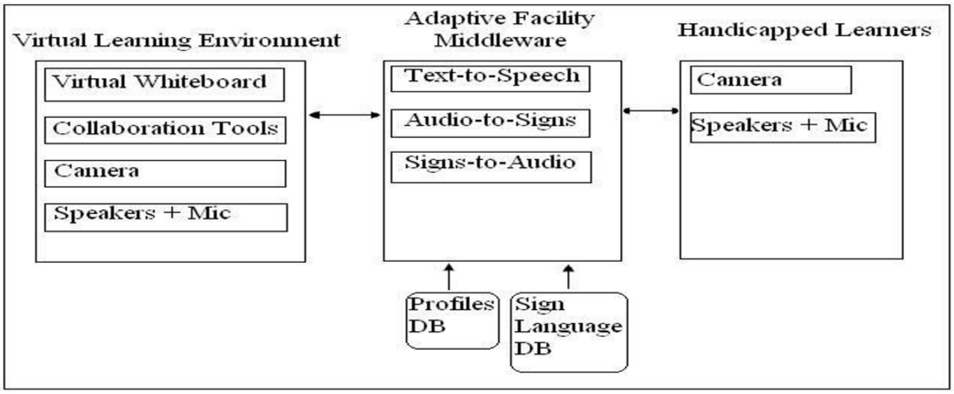
Deaf students on the contrary have the problem of hearing the live lecture conducted by instructors. Basically, deaf students can read lecture notes and study from books with no problem; however, live lectures are very significant for the learning process and hearing the instructors' explanation impact the level of learning for students.

The middleware is provided with a service that translates spoken words into sign language for the deaf people to interpret. As one of the goals of the middleware is to reduce efforts, cost, and easily incorporation of the disabled students, it is not feasible to conduct training courses for instructors in order for them to learn the sign language to teach for the disabled students. Also, we are integrating the disabled to the normal learning facilities; thus, healthy students are existent in the learning environment. Two types of spoken-to-sign-language services can be incorporated in the middleware.

Firstly, the spoken-to-sign-language service can be provided in an algorithm form that takes the audio spoken word as an input, generate corresponding code for it, match it against the sign-equivalent code stored in a database, and display the corresponding sign to the disabled student. This type of service will require a high computational performance in order to carry out the service efficiently although some delay must be considered. This requirement is not problematic since processing technologies are advancing rapidly. A notable advancement is the Intel's announcement published at the *ACM TechNews* of a new family of microprocessors to be released at the first quarter of 2008; these new chips called "Penryn" are made up of 16 processors to be manufactured in Intel Core 2 and Xeon microprocessors and marked as Intel Core 2 Extreme in notebook PC's [7]. Such example of advancement shall tackle the performance issue of the translating algorithm.

Secondly, another compatible spoken-to-sign-language service that can be incorporated in the middleware is proposed by IBM researchers who developed a system they call SiSi short for "Say-it-Sign-it" that is capable of converting spoken language into sign language and can be extended to different languages. The system which uses speech recognition to animate digital character has been approved by the Royal National Institute for Deaf people (RNID) [8]. Such developed system can be seamlessly integrated into the middleware and help deliver functionality to deaf people.

IV. OVERALL ARCHITECTURE & WORKING:



The Adaptive Facility Middleware as shown in the figure is to be placed between the handicapped learners and the virtual classroom conducted at the university. This middleware will be acting as a transformation layer between the virtual learning environment and the handicapped learners by which all learning means shall be transformed into a convenient representation addressing the type of student's disability.

A. Virtual Learning Environment:

This layer consists of all the e-learning facilities such as the whiteboard which will be used by instructors for conducting a lecture. It will contain also other normal collaboration tools such as a mailing facility, grading system, discussion board. The video component of that layer is to live broadcast the instructor while conducting the lecture. It is also used for recorded tutorials or other videos posted by the instructor. The Audio component is to transmit the instructor's voice while lecturing through a microphone attached to the physical environment where the instructor is located. It is also used for pre-recorded lectures posted by the instructor.

B. Adaptive Facility Middleware:

The Adaptive Facility Middleware consists of a "Text-to-Speech" service component which will be responsible for transforming the text written on the virtual whiteboard, the text of handouts or assignment to audio for the blind students, an "Audio-to-Sign" service component for deaf students to transform spoken words to deaf sign language, and a "Sign-to-Audio" service component for mute students to transform their sign language as they interact to audio for instructors and colleagues. Those components are explained at the previous section.

i. Profiles Database:

The middleware is connected to a "profile database" containing all handicapped students profiles. As part of the information written in each student's profile is his/her disability type, upon the student's login to the system using his/her unique username and password the system can check in that database the disability type of the logged students in order to adapt itself dynamically according to the student's need. This allows the system to efficiently and appropriately serve the students.

ii. Sign-Language Database:

The sign language database is crucial for the involvement of deaf and mute students. It contains two critical forms of data. Firstly, it contains pictures of each sign of the sign language. Secondly, it contains equivalent code for the sign language itself. The system will query to that database in order to match the taken snapshots of the mute students' sign language and also in order to compare the generated code for the spoken words with the equivalent sign code to display the appropriate sign to the deaf students.

The involved two parties of the whole system which are the handicapped students party and the virtual classroom party are interacting with the middleware by providing the necessary input through the attached cameras and microphones and retrieving the services output from the middleware.

V. CHALLANGES:

The implementation of the SOA Adaptive Facility Middleware poses a unique research direction that has its own challenges on the conceptual and the implementation level. The conceptual challenge is represented in the extensive research that has to be conducted and the research questions that have to be answered in order to implement a robust system. The implementation challenge is represented mostly in performance issues rather than feasibility issues. The proposed middleware and overall system do not carry any hindrance to the available technologies and science; the question is a about the performance of such middleware. I have tried to address some of the foreseen challenges under those two categories.

A. Conceptual Challenges:

The implementation of the middleware and such type of a system requires exhausted research in many branches of computing. It involves a data management research in order to decide on how the sign symbols will be stored in a database and to address the question of querying the symbols from the database in the transformation process which has to be efficient. Another important area of research is related to the basic artificial intelligence field regarding the pattern recognition; however, this challenge can be of minor effect since many workarounds can be made other than having an artificial intelligent algorithm.

B. Performance Challenges:

The most crucial areas of research are on the electronics technology field regarding the processing efficiency and on the algorithm field regarding the algorithm efficiency. Extensive research has to be undertaken in the electronics technology because in order for the middleware to perform efficiently in timely manner, it has to be places on capable machine acting as a server. Although electronics technology is advancing and Intel is expected to penetrate the market with super speed microprocessors, a research is still due in this area. On the other hand, extensive analysis and research must be undertaken regarding the executing algorithms. These performance issues are crucial to such systems because it is an educational system and timely manner response is a must. The system shall aid the education process and shall not exhibit delaying features that may waste the lecture time, cause difficulty to some students to connect, or be relatively slow especially in exams or important lectures.

VI. CONCLUSION:

This disabled-friendly e-learning application is meant to allow disabled students to have equal access to different fields of learning and from different universities around the world. Not only this system will solve disabled students' educational problems, but it will also solve their psychological problems. Those people feel disadvantaged from the world which causes them constant depression. They will no longer feel the sense of pity from other people because they will be self-independent and will be able to reach high levels of professional education which will allow them to produce and even add new inventions to the world instead of sitting waiting for pledges. They will not try to hide from society as now they will be presented in it as leaders not as a burden and they will have self-worth and will feel a high sense of dignity which will not be present without the aid of such system. The system will also narrow the gap between disabled people and normal ones as they will be equally receiving opportunities. Those people have need for achievement in order to defeat their disability and such system will be the instrument.

VII. FUTURE WORK:

This middleware sets new research directions as it brings our attention to the marginalized world of disabled students. Disabled people are always put in the last item in our thinking; however, it is due time to start thinking of them because it is our responsibility as healthy educated people to plan and do something useful to help those people. An intensive field of research can be made on two areas that complement each other. One area is the field of e-learning itself; by researching the current technologies involved in that trend and whether they are suitable to be adapted or we still need to find other technologies that boost the standard of distant learning as we are entering the era of wireless communication. The other area is the disabled people themselves; we need to research those people's current condition, compare and contrast them with those healthy people or disabled people who had their way in success. We need to research the psychological conditions due to disability especially newly disabled people. We need to investigate and probe their needs and their interpretation of the future; they are full of brilliant ideas. By conducting those two branches of research, we will be holding two solid pieces of information; the tool and the audience of that tool. These will gives us a great opportunity to start doing something for humanity.

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TMSE Optimization of Bandwidth of Tuned Electronic Circuits

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***Abstract*-As transfer functions of realizable electronic devices differ from the ideal Π-shaped one, at the design stage an optimal choice of bandwidth is required. Total mean square error (TMSE) criterion allows balancing linear distortions of the signal and remaining noise at the device's output. It is shown that the former term should be determined based on an easily calculated constant level but not necessarily 1 as used by the TMSE. Relative TMSE criterion was derived to address the above and some other shortcomings. It is shown by examples (bandpass RLC circuit, low pass Butterworth filter) that using relative criterion could lead to substantially lower output error spectra comparing to the TMSE.**

I. INTRODUCTION

A linear electronic filter should pass the signal to its output with minimal distortions whilst eliminating the noise as much as possible. The optimal transfer function for wide sense stationary (WSS) signal and noise that minimizes the total means square error (TMSE) was determined by N.Wiener [1]. His results were communicated by H.Bode and C.Shannon in a form which was more accessible to electronic engineers [2]. The following equations hold [2]:

$$W_{in}(\omega) = W_s(\omega) + W_n(\omega);$$

$$W_{out}(\omega) = |H(\omega)|^2 W_{in}(\omega) = |H(\omega)|^2 W_s(\omega) + |H(\omega)|^2 W_n(\omega);$$

$$output = \int W_{out}(\omega)d\omega = \int |H(\omega)|^2 (W_s(\omega) + W_n(\omega))d\omega\,, \quad (1)$$

where W denotes power spectrum (alternatively called power spectral density), indices *s, n, in, out* relate to the signal, noise, input and output respectively, $H(\omega)$ is the transfer function of the filter, and *output* refers to the power at the output. TMSE at the output consists of two terms representing the linear distortion of the signal and the remaining noise [2]:

$$E = \int |H(\omega) - \exp(j\alpha\omega)|^2 W_s(\omega)d\omega + \int |H(\omega)|^2 W_n(\omega)d\omega\,, \quad (2)$$

where α is the desired (or allowed) signal delay. The minimum of the first term is obtained if the filter provides the delay α, and thus the phase response is specified by this delay [2]:

$$\arg(H) = \alpha\omega\,. \quad (3)$$

As far as all the frequency components of the WSS process remain independent, the determination of the optimal $|H|$ values can be done independently at each frequency. It yields:

$$E = (|H| - 1)^2 W_s + |H|^2 W_n\,, \quad (4)$$

where the dependence of the variables on frequency is omitted hereafter for clarity.

The optimal value can be found by differentiation of E with respect to H:

$$\frac{dE}{dH} = 0 \quad \Rightarrow \quad |H| = \frac{W_s}{W_s + W_n}\,. \quad (5)$$

This is the classical solution for the Wiener filter ([1], eq.3.46, [2], eq.18). The value of the output error is obtained by backward substitution of (5) into (2), which yields ([1], eq.3.445, [2], eq.19):

$$E = \int \frac{W_s W_n}{W_s + W_n} d\omega\,. \quad (6)$$

Direct application of the equation (5) to the design of electronic systems is complicated by particular constraints imposed by realizable circuits on the continuum of their transfer functions. For example, a realizable circuit can not have its transfer function constant at any finite interval thus making implementation of, *e.g.*, Π-shaped transfer function, impossible.

Additionally, inconsiderate application of the TMSE criterion could lead to some contradictions [3].

Optimization of the quality factor of electronic circuits using the TMSE is considered in this paper. It is shown that inconsiderate application of the TMSE leads to substantial overestimation of the linear distortion error. The correct level of this error can be found by considering non-unity gain for the signal. This consideration leads to the relative TMSE criterion that does not depend on the device's gain; has a clear physical limit of 1; avoids contradictions associated with the TMSE itself; can allow better optimization of electronic filters as shown by examples.

II. OVERESTIMATION OF LINEAR DISTORTIONS FOR AN RLC CIRCUIT

In this section linear distortions imposed on a pure input signal by an ideal second order RLC circuit [4], with a unity gain at its central frequency ω_0, without phase distortions, are

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calculated. The transfer function of such an ideal system is defined by the following expression

$$|H(\omega)| = \frac{1}{\sqrt{1+\left(\frac{\omega-\omega_0}{\Delta\omega}\right)^2}}, \tag{7}$$

where $\Delta\omega$ equals to half of the device's bandwidth at –3 dB. Values of ω_0=2 s^{-1} and $\Delta\omega$=1 s^{-1} are assumed for numerical convenience. If the signal power spectrum is uniform and equal to 1 W*s within [ω_0-$2\Delta\omega$; ω_0+$2\Delta\omega$] and the noise is absent, formula (2) gives E=0.44 W as the error here equals to the linear distortions. This value seems too high when compared to

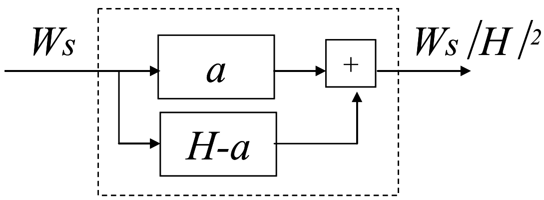

Fig. 1. Decomposition of the system that aids the determination of the power in the linear distortion.

the overall output power of 2.21 W. A correction to this figure can be obtained using the following reasoning. Let us consider the system presented in fig.1 where $a(\omega)$=$const$>0, $a \in R$ and there is no noise. (This definition could be amended to the case where a also incorporates a signal delay.) Its frequency response is equal to H and it is thus equivalent to the system discussed above. The output power spectrum of this system at any frequency can be viewed as the sum of two terms:

$$W_s(\omega)\left|H^2(\omega)\right| = a^2 W_s(\omega) + \left|H(\omega)-a\right|^2 W_s(\omega) \quad . \tag{8}$$

The first term represents the scaled power spectrum of the input signal and is thus free of linear distortion. In contrast, the second term describes how the deviations of H from the constant level a affect the output and therefore represent the linear distortion. The overall linear distortion is found by integration over the signal band

$$linear\_distortion = \int \left|H(\omega)-a\right|^2 W_s(\omega) d\omega \quad . \tag{9}$$

The TMSE corresponds to the case a=1, but this value does not work well here because $H(\omega)$=1 at the central frequency only and $H(\omega)$<1 elsewhere. Results of numerical calculation of $E(a)$=$linear\_distortion(a)$ are shown in fig.2 (a Matlab™ script used for calculations is presented in the appendix, the integration here and thereafter was carried out numerically using rectangles on a dense grid). The error reaches its minimum (0.13 W) at a=0.72 - a value which corresponds to the average value of H within the signal band. This result is illustrated further in fig.3, where the overall linear distortions represent areas under the curves fig.3,bottom.

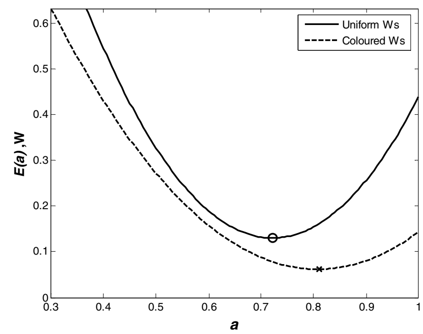

Fig. 2. The power of linear distortion at the output of the RLC filter.

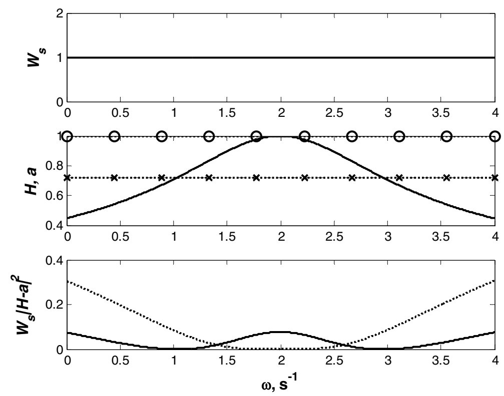

Fig. 3. Calculation of linear distortion: the uniform signal power spectrum (top). (middle) The magnitude response of the system (solid line), TMSE level (a=1, circles), relative TMSE level (a=0.72, stars). (bottom) Resulting power spectra of linear distortion (solid line – relative TMSE, dotted line - TMSE).

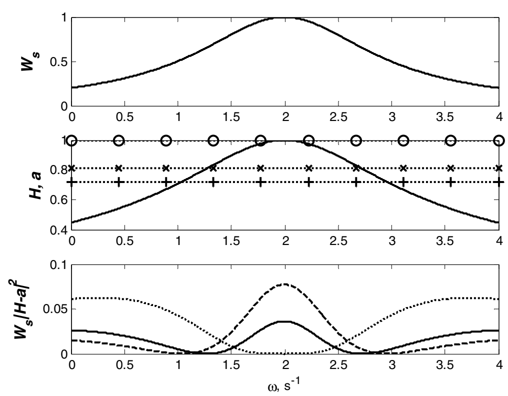

Fig. 4. Calculation of linear distortion: the colored signal power spectrum (top). (middle) The magnitude response of the system (solid line), TMSE level (a=1, circles), relative TMSE level (a=0.81, stars), average magnitude response level (a=0.72, pluses). (bottom) Power spectrum of the linear distortion (solid line - RTMSE, dashed line - average magnitude response level, dotted line - TMSE).

If W_s is not uniform the optimal level of a is not necessarily equal to the average value of the magnitude response. Fig.4 (top) presents a non-uniform W_s applied to the input of the considered filter. Fig.4 (middle) shows its frequency response along with three constant levels that can be used for determining the linear distortions. These levels result in different power spectra of the linear distortions presented in fig.4, bottom. The least area under the curve (i.e., the power of the linear distortion) is bounded by the curve obtained for a=0.81. This level provided an error of 0.062 W, whilst the average H (0.72 as before) resulted in the error of 0.079 W. The error power for the TMSE ($a=1$) was 0.14 W, significantly exceeding both other powers.

This example shows that the fixed unity level employed by the TMSE criterion can lead to undue over-determination of the level of linear distortion. Therefore both the error estimate and the optimization result might not be adequate if there is a particular constraint on the realizable magnitude response.

III. Derivation of the relative TMSE criterion

Let us consider a system shown in fig.1 when the signal and noise are both applied, and determine the signal power at the output. The output of the filter given by equation (1) contains the linear distortion (9) and the noise

$$noise = \int W_n(\omega)\left|H(\omega)\right|^2 d\omega \quad . \qquad (10)$$

The difference between the overall output power and these two error components represent the power of the signal at the output thus:

$$\begin{aligned} signal &= S = output - linear\_distortions - noise = \\ &= \int \left|H\right|^2 (W_s + W_n)d\omega - \int \left|H-a\right|^2 W_s d\omega - \int \left|H\right|^2 W_n d\omega = \\ &= 2a\int \left|H\right| W_s d\omega - a^2 \int W_s d\omega \end{aligned} \qquad .(11)$$

(Here the delay caused by the device is ignored, and its transfer function is assumed to be purely real. This simplification can be valid, for example, for FIR digital filters.) The condition for maximizing the output signal power can be found by differentiating S with respect to a:

$$\frac{dS}{da} = 0 \quad \Rightarrow \quad a_{opt} = \frac{\int \left|H\right| W_s d\omega}{\int W_s d\omega} \quad \Rightarrow$$

$$S_{opt} = 2\frac{\int \left|H\right| W_s d\omega}{\int W_s d\omega} \times \int \left|H\right| W_s d\omega - \left(\frac{\int \left|H\right| W_s d\omega}{\int W_s d\omega}\right)^2 \times \int W_s d\omega =$$

$$= \frac{\left(\int \left|H\right| W_s d\omega\right)^2}{\int W_s d\omega} = a^2 \int W_s d\omega \qquad . \qquad (12)$$

Hence the output signal power depends on the squared value of the level a that determines linear distortion. This result appears to be quite reasonable if the equivalent structure of the filter (fig.1) is considered.

Consequently, the error power becomes:

$$\begin{aligned} error = output - signal = \int \left|H\right|^2 (W_s + W_n) d\omega - \\ \left[\int \left|H\right| W_s d\omega\right]^2 / \int W_s d\omega \end{aligned} \quad . \qquad (13)$$

Equation (13) can be used as an optimization criterion for a linear system design, but it is still dependant on the device's gain that would still lead to some contradictions [3]. For this reason let us consider an alternative optimization criterion, taking a ratio of *signal* to *output* :

$$\frac{signal}{output} = SOR = \frac{\left[\int \left|H\right| W_s d\omega\right]^2}{\int W_s d\omega \times \int \left|H\right|^2 (W_s + W_n) d\omega} \to \max \le 1, \qquad (14)$$

This criterion is independent both on the gain of the filter, and of the level chosen for determination of the linear distortion as it is inherently optimized. It represents the ratio of the signal power at the output to the overall output power, and can be considered as a relative TMSE.

IV. Optimization of an RLC circuit

Let us consider an optimization of the second order RLC

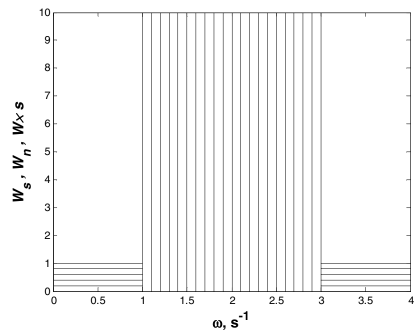

Fig. 5. Signal (vertical lines) and noise (horizontal lines) power spectra for optimization of the RLC filter (section IV)

circuit system described by (7) in terms of its bandwidth $\Delta\omega$ for the signal and noise power spectra shown in fig.5. Again, a Π-shaped filter would be optimal here, but the device used restricts the continuum of the responses realizable.

Both TMSE and relative TMSE values for different $\Delta\omega$ in the range from 0.7 to 1.4 were calculated using a Matlab™ script presented in the appendix, and have been plotted in fig.6 (top) and fig.6 (middle) respectively. The optimal values are distinct at 1.15 for the TMSE and 0.95 for the relative TMSE, and result in notably different optimal transfer functions (fig.6 (bottom)). The TMSE criterion demands a narrower bandwidth that is beneficial for the noise suppression. However this feature compromises the linear distortion.

Fig.7 (bottom) represents the power spectra of the output error for both devices. These values are smaller for the relative

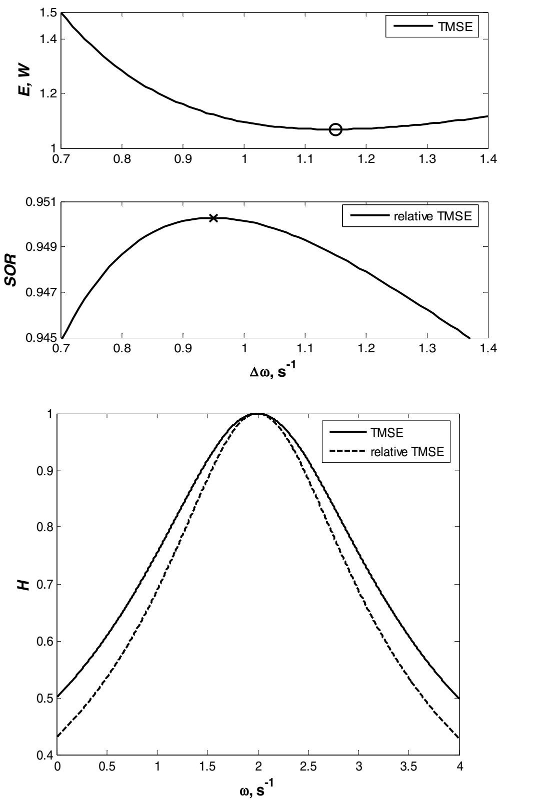

Fig. 6. Optimization of the bandwidth of the RLC circuit: (top) TMSE curve with an encircled optimal value *1.15*, (middle) relative TMSE curve with a crossed optimal value 0.95, (bottom) the associated transfer functions.

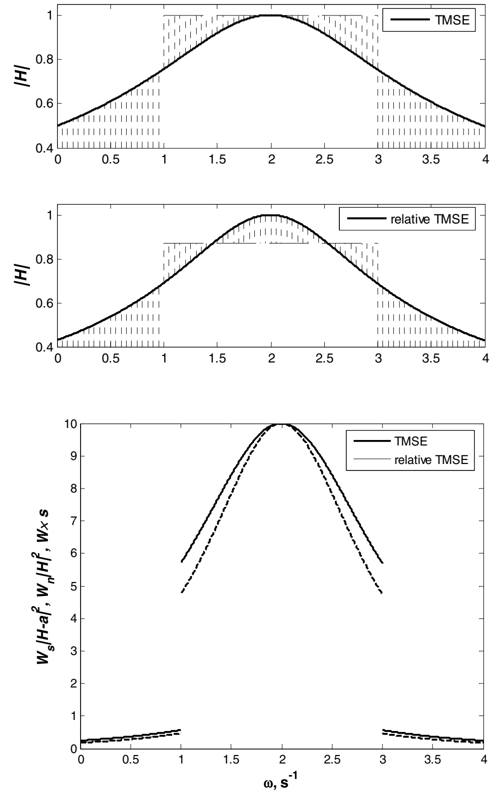

Fig. 7. Calculation of the error power spectra for the RLC circuit: (top, middle) transfer functions with shown deviations from the ideal Π–shaped filter; (bottom) squared deviations multiplied by the relevant spectra

TMSE filter both in the stopband (as expected from smaller bandwidth) and in the passband. The latter happened because the relative TMSE optimization refers to a non-unity gain as it is shown in fig.7, middle. In contrast, the TMSE criterion refers to the universal unity gain that is not feasible here (fig.7, top). (The error would be even worse if the device to be optimized had a peak value of its transfer function of less than 1.) On the other hand, both criteria would lead to the same filter if the peak value of the transfer function was 1.09. This fact shows both the validity of the TMSE for non-constrained optimization, and its inconvenience for practical applications as this level was determined here by trial and error running the above mentioned Matlab™ script.

V. Optimization of a Low Pass Butterworth Filter

Butterworth filters are frequently used in electronic circuits because of maximal flatness of their magnitude response that equates [5]:

$$|H(\omega)| = \frac{1}{\sqrt{1+(\omega/\omega_c)^{2n}}}, \tag{15}$$

where ω_c is the cut off frequency of the filter, and n is its order. The low pass Butterworth filter of the first order can be implemented conveniently using a resistor and a capacitor. It

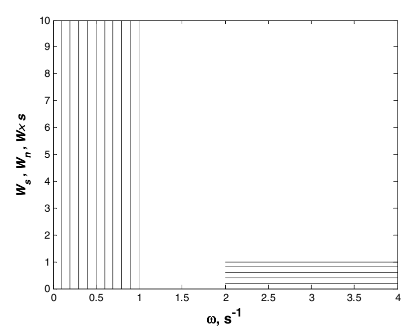

Fig. 8. Signal (vertical lines) and noise (horizontal lines) power spectra for optimization of the Butterworth filter (section V)

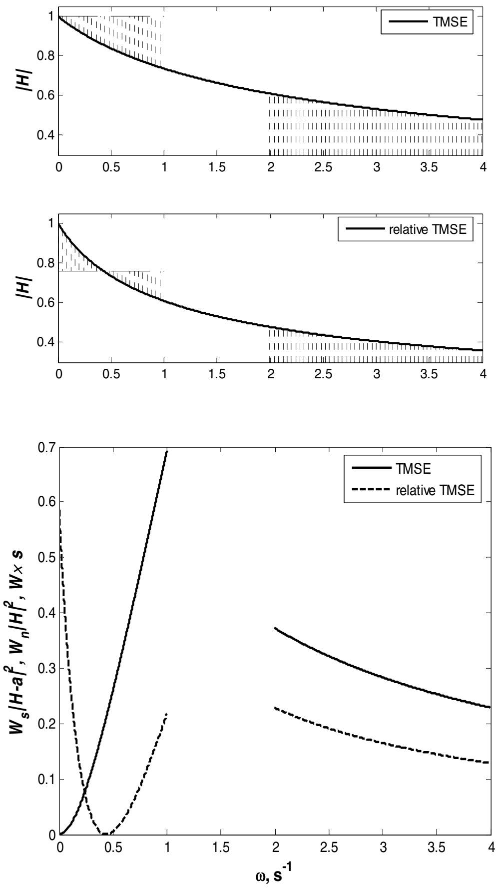

Fig. 9. Optimization of the bandwidth of the low pass Butterworth filter: (top, middle) magnitude responses of optimal filters (ω_c=1.09 for TMSE, and ω_c=0.77 for relative TMSE), (bottom) the associated error power spectra

presents the easiest option for an electronic filter that aims to reduce aliasing [6].

The optimization was conducted to the signal and noise power spectra shown in fig.8 using a slightly modified Matlab™ script from the appendix. Phase distortions introduced by a realizable device were not considered. The TMSE and relative TMSE optimizations resulted in quite distinct magnitude responses that are presented in fig. 9. Again, relative TMSE design resulted in much lower error power spectra.

VI. Quantitative Analysis Using Relative TMSE

The above considered examples showed that the related TMSE ensured lower error power spectra. Due to this criterion was chosen in this section for quantitative comparison of the optimized designs.

Numerical evaluation of the designed filters is greatly simplified if the SOR is calculated at the filter's input. For this calculation the magnitude response can be set to unity, and equation (14) reduces to

$$SOR_{in} = \frac{\int W_s d\omega}{\int (W_s + W_n) d\omega} \leq 1 \quad . \tag{16}$$

This defines the given input SOR, which is to be improved by the filter. The input and output SORs for all the designs considered are presented in table 1. It also shows the improvement of the SOR provided by the filter. It is obvious that the wider bandwidth of the signal assumed for the RLC circuit optimization (fig.5) resulted in higher input SOR than that assumed for the Butterworth filter (fig.8). However the former SOR was more difficult to improve because of the absence of the transition band. Table 1 quantifies these observations, and indicates that the achieved improvement in both cases was less than 10%. Relative TMSE optimization outperformed the TMSE optimization by approximately the same margin. Therefore the relative TMSE design can be considered worthwhile when comparing to its TMSE counterpart.

TABLE I
SOR and its Improvement

	Input SOR	Output SOR		SOR improvement	
		TMSE	Related TMSE	TMSE	Related TMSE
RLC circuit	0.909	0.945	0.950	3.6%	4.1%
Low-pass Butterworth filter	0.833	0.919	0.926	8.6%	9.3%

VII. Conclusions

The TMSE criterion is a proven tool for unconstrained optimization of linear systems in the frequency domain; however its inconsiderate application can lead to contradictory results when some constraints do exist. These contradictory results are related to the use of absolute, not related, error power and the rigid unity level for determining the linear distortion. The amended criterion, the relative TMSE, derived from the TMSE, avoids these shortcomings. It can be distinctly preferable when the optimized device imposes some constraints on the continuum of realizable transfer functions. For example, it allows convenient optimization of linear phase FIR filters in the frequency domain [7] that generalizes signal-to-noise ratio optimization of these devices [8].

Acknowledgment

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Anonymous FTP: ftp://128.243.76.144/TMSE/index.htm

APPENDIX

```
% Matlab script for section II
% range of level a considered
      al=0.3:0.005:1; lal=length(al);
% range of angular frequencies
      do=4; om0=2; step=0.0001; om=0:step:do;
% frequency response of the device
      h=1./sqrt(1+(om-om0).^2);
% PSDs
      ws=h.^2;                          %  coloured signal spectrum
      ws1=ones(1,length(om));           %  uniform signal spectrum

% calculation of the linear distortions
% for all considered levels
      for i=1:lal
         err(i)=sum(ws.*(al(i)-h).^2)*step;
         err1(i)=sum(ws1.*(al(i)-h).^2)*step;
      end

      [mx,im]=min(err); [mx1,im1]=min(err1);
      aopt=al(im), aopt1=al(im1)
```

```
% Matlab script for section IV
      dom=0.7:0.01:1.4;                       % range of bandwidths
% range of angular frequencies for the consideration
      om0=2; step=0.0001; om=0:step:4;
      oms=1:step:3; ls=length(oms);           % signal
      omn1=0:step:1-step; ln1=length(omn1);   % noise low frequency part
      omn2=3+step:step:4; ln2=length(omn2);   % noise high frequency part
      amps=10;                                % multiplier for the signal

      for i=1:length(dom)
         h=1./sqrt(1+((om-om0)/dom(i)).^2); %  *1.09  % to show equivalence of both
%%%%%%%%%%%%%%%%%%%%%%% TMSE %%%%%%%%%%%%%%%%%%%%%%
         e1=step*1*sum(h(1:ln1).^2);                % low frequency noise
         e2=step*1*sum(h(ln1+ls+1:ln1+ls+ln2).^2);  % high frequency noise
         e3=step*amps*sum( (1-h(ln1+1:ln1+ls)).^2); % linear distortion TMSE
         err(i)=e1+e2+e3;                           % complete TMSE
%%%%%%%%%%%%%%%%%%%%%%% relative TMSE %%%%%%%%%%%%%%
         n1=step*amps*sum(h(ln1+1:ln1+ls));         % numerator
         d1=step*amps*ls;                           % denominator - first term
         e3=step*amps*sum(h(ln1+1:ln1+ls).^2);      % output power in the passband
         s2o(i)=n1^2/(d1*(e1+e2+e3));               % complete relative TMSE
      end

      [mx,mi]=min(err), [mx1,mi1]=max(s2o)
```

A Fuzzy Scheduling Algorithm Based on Highest Response Ratio Next Algorithm

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***Abstract*—The field of fuzzy systems and control has been making rapid progress in recent years. Fuzzy logic in a wider sense is a nonlinear mapping from the inputs to the output of the system. It has a variety of applications. In this paper, we have applied it for process scheduling. We propose a fuzzy logic process scheduler in CPU, which uses the idea of HRN (Highest Response Ratio Next) scheduling algorithm as its method of inference and decides based on this manner. Because of the approximate nature of the Service Time of processes, a fuzzy logic algorithm would help us set priorities to processes in a more appropriate and fair manner. At the end, we compare our work with the existing conventional process scheduling algorithms and give the comparison graphs for our evaluation. Our algorithm has precise priority values and does not need any other scheduling algorithm concurrently. It also has a better evaluation output compared with other existing algorithms.**

I. Introduction

The main objective of multiprogramming is to have some process running all the times, to maximize CPU utilization. In the uni-processor system, there is only one process running at a time while others wait until CPU is free. If the process being executed requires an I/O then in that time period processor remains idle, resulting is CPU's waste of time. With multiprogramming, we try to use this time productively. Several processes are kept in the memory. When one process has to wait, operating system can take away CPU from that process and gives it to another process. Scheduling is a fundamental operating system function. Whenever the CPU becomes idle, the operating system must select one of the processes in the ready queue to be executed. The selection process is carried out by the short-term scheduler. The scheduler selects from among the processes in memory that are ready to execute, and allocates the CPU to one of them.

CPU scheduling is the foundation of multiprogramming. The scheduling helps in the reduction of the Waiting Time (the time that a process spends in the ready queue) and Response Time (the time interval between admission of the process in the ready queue and receiving the first response from the scheduler) of the processes. Along with, it also increases the throughput of the system.

When a system has a choice of processes to execute, it must have a strategy for deciding which process to run at a given time. This strategy is known as Process Scheduling Policy. A scheduling policy should attempt to satisfy certain performance criteria, such as Maximizing Throughput, Latency, Preventing Indefinite postponement of Process, and Maximizing Process Utilization. It is the job of the scheduler or dispatcher to assign a processor to the selected process.

Several process scheduling algorithms [2] have been introduced, namely: First Come First Served (FCFS), Round Robin (RR), Shortest Process First (SPF), Shortest Remaining Time Next (SRTN) and Highest Response Ratio Next (HRN). Different process scheduling algorithms have different properties and may favor one class of processes over another and have their own pros and cons.

In this paper, we proposed a fuzzy logic based process scheduling algorithm, which uses the HRN scheduling algorithm's priority calculation formula, in its fuzzy inference engine for decision making. Fuzzy logic [1] has a good adaptability with inaccurate and uncertain inputs and outputs. In a computer system, we do not have an accurate measure of the Service Time (the total time that a process needs the CPU resource), required by every process. Thus, we would have inaccurate priority for every process. The scheduling algorithm in this paper, measures the priority in the form of some fuzzy sets, and this output can be used in the system. We used MS-Windows priority classes for processes, as our output priority fuzzy sets. However, we can use accurate priority value after defuzzification of the result. Our approach may lack efficiency in terms of computation overhead (in the case of defuzzified priority), however, nowadays this small overhead can be ignored. Nevertheless, it has a good decision making approximation. We used

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 75–80.

MATLAB-7.1® [5] fuzzy logic toolbox to generate the fuzzy inference engine and required diagrams.

The organization of this paper is as follows: section II determines the relation of our work with previous works in this subject. Section III describes the idea of our approach. In section IV we define the linguistic values or fuzzy sets for inputs and outputs. Section V elaborates the inference engine used by our fuzzy system. Section VI determines the defuzzification methods for our fuzzy scheduler. Section VII evaluates our approach with other existing approaches and gives the evaluation results. Section VIII concludes the paper.

II. Relation to Previous Work

Scheduling of processes in a computer system is a main concern in the system. HRN is one of the process scheduling algorithms. This algorithm has a decision mode based on the priority of processes, which is computed on-demand. This algorithm corrects some of the weaknesses of the SPF. The SPF algorithm is biased towards the processes with short Service Times. This keeps the longer processes waiting in the ready queue for the longer time, despite of arriving in the ready queue before the short jobs. In the HRN algorithm, priority is a function of not only the Service Time, but also of the time spent by the process waiting in the ready queue in a nonpreemptive scheduling algorithm. Once the process obtains the control of the processor, it completes to completion. The priority is calculated by the formula:

$$\Pr iority = \frac{WaitingTime + ServiceTime}{ServiceTime} \quad (1)$$

In this algorithm too, short processes receive preference. However, longer processes that have been waiting in the ready queue are also given the favorable treatment. Based on this scheme, we build our fuzzy inference engine rule base. We also build the input fuzzy sets and output ones, upon inputs and outputs of HRN algorithm.

III. Our Approach

In Our approach, we used fuzzy logic to decide which process takes the CPU resource, next. Our algorithm is also a nonpreemptive algorithm, which computes the Waiting Time of each process, on-demand. Below, we elaborate each part of our fuzzy logic decision-making algorithm. We used product function for fuzzy and operator and summation function for fuzzy or operator. Fig. 1 illustrates our fuzzy system.

IV. Membership Functions

Fuzzification is defined as a mapping from a real-valued (crisp value) point to a fuzzy set. Our fuzzy decision making algorithm receives the inputs (Waiting Time and Service Time of the process), and delivers the output (Priority). This fuzzification process converts these input values to fuzzy linguistics. All the membership functions in our approach are normal, complete and consistent. For simplicity and reducing the cost of computation, we used triangular membership functions for all membership functions in our system.

For input variable Service Time, we define two fuzzy sets, Small and Long. We also considered an interval between of 0 to 100 points for this variable in our simulation. Nevertheless, this range can be tuned in a real system with the real data. Fig. 2 illustrates the membership functions of the variable Service Time. Since the system is not very sensitive to the variation of this variable, we defined two fuzzy sets for it.

For input variable Waiting Time, we define three fuzzy sets (since the system is more sensitive to the variation of this variable.) The fuzzy sets for this variable are Short, Medium and Long. The names correspond to the values of these sets. We choose an interval of 0 to 1000 points for this variable. In Fig. 3, you can see the member ship functions for fuzzy sets of this variable.

Finally, we define six fuzzy sets for output variable priority, and an interval between 0 and 10. The value of priority ranges in this interval and can be determined in the sets Low (L), Below normal (BN), Normal (N), Above normal (AN), High (H), and Real time (RT). The linguistic value indicates the priority of a process in the ready queue. The higher value determines the higher priority for the process. Fig. 4 illustrates membership functions for the fuzzy sets of Priority variable.

To have more accurate and real result, we defined six sets

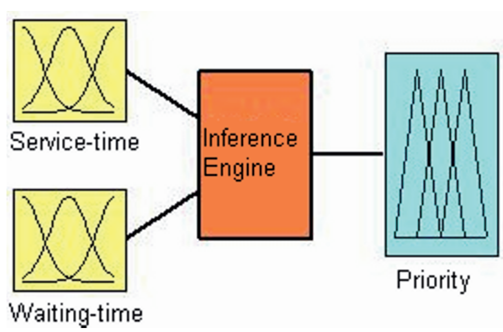


Figure 1. Our fuzzy scheduler

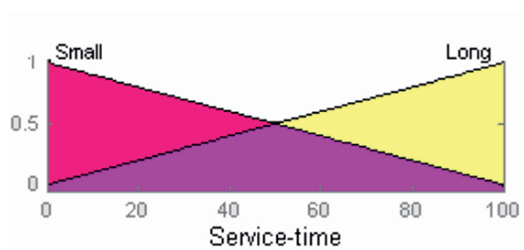


Figure 2. Service-time membership functions

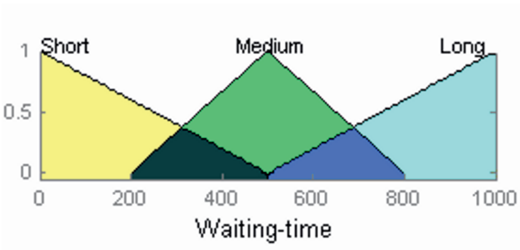


Figure 4. Waiting-time membership functions

for the output variable. Thus, the fuzzy set value of the early result from the inference engine can be used as the priority of the process, without defuzzification. Actually, we can remove defuzzification phase from our system to reduce the computation cost.

V. FUZZY INFERENCE ENGINE

After determining the input and output fuzzy sets, we are ready to design the inference engine for our fuzzy scheduler. First, we have to build the rule-base of our system. We define the IF-THEN rules, with HRN-bias results, for different ranges of inputs. We have six IF-THEN rules, based on the number of input fuzzy sets. All the rules have the same preference. The rule-base is shown in Table I.

Again, for simplicity and having lower computation burden, we choose Mamdani-product inference engine [1]. The output set is computed as follows:

$$\mu_{B'}(y) = \max_{l=1}^{M}\left[\sup_{X\in U}\left(\mu_{A'}(X)\times\prod_{i=1}^{n}\mu_{A_i^l}(x_i)\times\mu_{B^l}(y)\right)\right] \quad (2)$$

Where, A_i^l and B^l are the input set number i, and output set of the rule l, respectively, n is the number of input sets and M is the number of rules in the rule-base of the system. X is the input vector (Service Time, Waiting Time) to the fuzzy scheduler and y is the resulting output set (Priority) from the scheduler.

Since, we have only six rules, which at most four of them fire in the worst case and have only two elements in our

TABLE I. RULE BASE

		Service-time	
		Small	***Long***
Waiting-time	***Short***	AN	L
	Medium	H	BL
	Long	RT	N

Figure 3. Priority membership functions

input vector that minimizes the computation burden for multiplication and supremum function, this computation would be very simple.

If we are relax about accuracy of the priority of a process and only want to classify the processes priorities, we can use the resulting output set as a linguistic priority class for each process. Otherwise, we have to go ahead to the defuzzification phase and get a crisp value for the priority.

VI. DEFUZZIFICATION

Defuzzification is the reverse procedure of fuzzification. In this phase, we design a mapping from a fuzzy set to a crisp value. The resulting output fuzzy set from the inference engine is now changed to a real priority value. There are several defuzzifiers, such as Center of Gravity (Centroid), Center Average, Smallest of Maxima (SoM), Mean of Maxima (MoM) and Largest of Maxima (LoM) defuzzifier [1], that are used in fuzzy systems and each of them has its own pros and cons. We present our fuzzy scheduler by Centroid and Mom defuzzifiers, since each of them can be used in the proper application.

A. Centroid Defuzzifier

Centroid defuzzifier is the most accurate, plausible and continual defuzzifier, nevertheless it does not have computational simplicity. Thus, it is not appropriate for small systems. In this method, the center of area under the output fuzzy sets is considered as the resulting crisp value. Thus, the crisp output value is determined as:

$$y^* = \frac{\int_V y\,\mu_{B'}(y)dy}{\int_V \mu_{B'}(y)dy} \quad (3)$$

Where, B' is the output fuzzy set from the inference engine and is the output crisp value. Since, the output sets are triangular, computation of area under the sets would be very simple. Thus, more suitability is provided for this method. Below, we presented the MATLAB® Surface diagram for our scheduler with Centroid defuzzifier. As you see in Fig. 5, by this method, the diagram is smooth and there is no jump or rupture in the output value. The main advantage of this defuzzifier is its accuracy. In the HRN algorithm, sometimes we get equal priorities for deferent

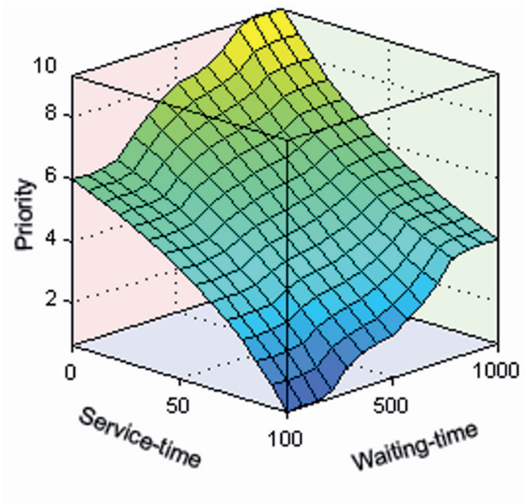


Figure 6. Centroid defuzzifier surface diagram

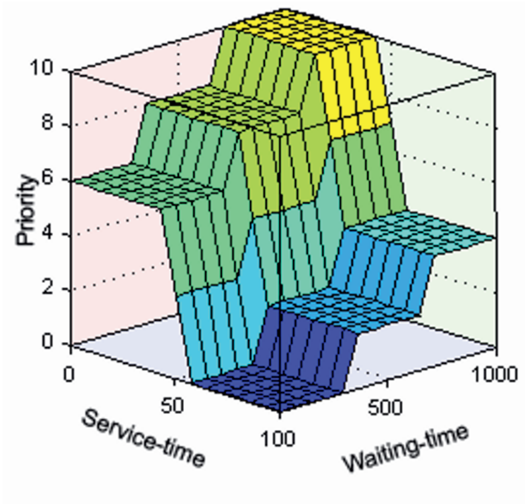


Figure 5. Mean of maxima defuzzifier surface diagram

processes, thus we need to use FCFS algorithm in this case to determine which process takes the CPU first. In our approach, by Centroid defuzzifier we always have exact priority value for each process and do not need any other method concurrently.

B. *MoM Defuzzifier*

Mom Defuzzifier Mean of maxima is a simple and plausible method of defuzzification. Nevertheless, often does not have continuity of the output value. In this method, the mean value of the maximum values of output sets is considered as the result value. The result value in this method is computed as follows:

$$y^* = \frac{\int_{hgt(B')} y dy}{\int_{hgt(B')} dy} \quad (4)$$

Where, is the output set from the inference engine, hgt is the set of all points which have the maximum membership value in the output sets.

In our approach, there are at most four fired rules, thus four output sets and four maximum values. Therefore, we need to compute the mean of at most four values. We have continuity, but still do not have a smooth surface. As it is shown in Fig. 6, there are some flat points, where we have fix priority values which reduce the accuracy of the system and enforce us to use another algorithm (i.e. FCFS) concurrently. In addition, small changes in input values result in large changes in the output.

VII. ALGORITHM EVALUATION

We have programmed a model of the computer system using Software data structures, which represent the major components of the system, discussed above. The Ready Queue and the memory are simulated, using Vectors in which we store objects of class Process. A Process object contains information about the Process, which is also updated when the process runs. In the real system, we call this entity as PCB (Process Control Block).

Ready Queue contains the list of ready processes, which are ready to execute. Ready queue is maintained in a priority order, which depends on the algorithm we are using to calculate the priority. In our simulation, the ready queue has been programmed to serve the processes in the First in First out, Round Robin, Shortest Process first, Fuzzy Highest Response Ration Next (our algorithm), (FHRN), and Shortest Remaining time.

The simulator has a variable representing a clock; as this variable value is increased, the simulator modifies the system state to reflect the activities of the processes and the scheduler. Our system has a function called ProcessReady, which checks which processes are ready to enter the system depending on the current clock. Preemption is performed based on the current clock. Based on the algorithm, if the next process in the ready queue should get the CPU, the current process is pushed into the queue and the next process, based on how the priority of the process is calculated in ready queue, is taken and given the CPU time. We call this in real systems as context switch. We will be providing this overhead a simple variable that we fill add to a process when it is preempted.

The scheduler is an abstract class in which we have defined the basic components, which are needed by the scheduler like ready queue. FIFO, RR, SPF, SRT and FHRN, are the classes, which extend this scheduler class and implement the ready queue based on specific scheduler.

As we run the simulations, the statistics that indicate algorithms performance are collected and printed. The analysis is shown. The data that we are using to drive the simulation is generated using a random-number generator.

The process PCB in our simulation consists of following attributes: Process Id, Process Service Time, Process Arrival Time, Process Finish Time and Process ResponseTime. The same set of processes is inserted into the scheduling algorithm to evaluate the algorithms effect on the processes and CPU. These are initialized for all the processes that we randomly generate. Once the process gets the CPU, its Service Time is updated and if the simulation performs a context switch, which preempts the current running process and puts it at the back of the ready queue, then we save the PCB of the process. After this, the first process in the ready queue is given the block. In the end, the system outputs the Arrival Time, Service Time, Turnaround Time, Waiting Time and Response Time for each process executed by the system.

We have shown the inputs, outputs, and analysis for our algorithm and then compared it with other existing algorithms. The interval for Service Time and Waiting Time is considered between of 0 to 10.

Consider the data set in Table II. In our (FHRN) algorithm, the P0 enters the Ready Queue when no other process is there to compete for the processor. Hence, P0 is given access to the processor and it starts its execution. Up to this point, the algorithm works as FCFS. However, by the time P0 completes its execution, other processes P1, P2, P3 and P4 arrive in the Ready queue. Now FHRN algorithm based upon the Service Time and the Waiting Time of the processes will determine the priority of the processes. The process with highest priority will be given access to the processor to execute. When P0 releases the processor, at that time the value of the clock is six. The calculation to determine the priority of the process in this time is done in the Table III steps:

TABLE IV. SIMULATION DATA SET

Process name	Arrival Time	Service Time
P0	1	5
P1	2	4
P2	3	6
P3	4	2
P4	5	7

As you see, process P1 has the greatest Priority in both Centroid and MOM fuzzy scheduler. Therefore, P1 is selected for execution. After P1 finishes its execution, the value of the clock is ten. Now we have three processes waiting for CPU as shown in Table IV:

TABLE II. PRIORITY CALCULATION AT TIME SIX

	P1	P2	P3	P4
Arrival Time	2	3	4	5
Waiting Time	4	3	2	1
Service Time	4	6	2	7
Centroid Priority	5.41	4.05	5.38	3.05
MOM Priority	8	0	6	0

TABLE III. PRIORITY CALCULATION AT TIME TEN

	P2	P3	P4
Waiting Time	7	6	5
Service Time	6	2	7
Centroid Priority	4.89	6.93	3.80
MOM Priority	4	8	2

Here P3 has the highest priority among all three processes in the ready queue. The scheduler gives the CPU to process P3. After P3 releases the processor, the clock is twelve. Table V shows the priority calculation at this time.

Now with the Centroid defuzzifier scheduler, P2 has the highest priority, but with the MOM defuzzifier scheduler, both processes have the same priority. Here we need another algorithm (i.e. FCFS) to determine which process takes the CPU first. Thus, again P2 will be selected to take the CPU resource. Finally, after P2, P4 which is the last in our simulation, takes the processor and completes its execution.

In Table VI we have calculated the total Waiting Time, Turnaround Time, and Response Time for the four processes in our simulation model.

TABLE V. PRIORITY CALCULATION AT TIME TWELVE

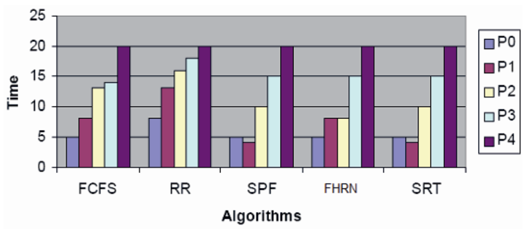
	P2	P4
Waiting Time	9	7
Service Time	6	7
Centroid Priority	5.39	4.40
MOM Priority	4	4

TABLE VI. TOTAL WAITING TIME, TURNAROUND TIME AND RESPONSE TIME CALCULATION

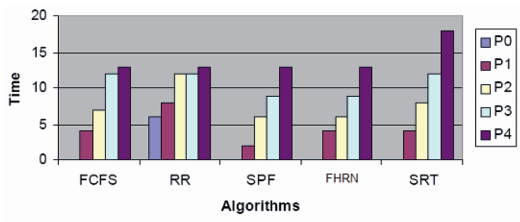
Process	Turnaround Time	Waiting Time	Response Time
P0	5	0	0
P1	8	4	4
P2	15	9	9
P3	8	6	6
P4	20	13	13

In Fig. 7, we present a graphical comparison for various scheduling algorithms by Turnaround Time, total Waiting Time, and Response Time.

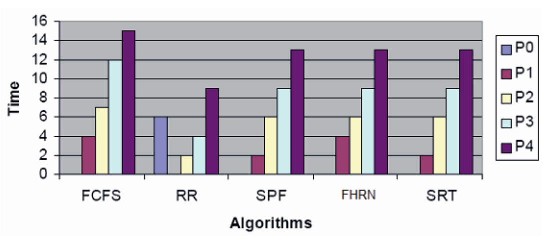
As it is shown, in all three graphs, our algorithm has a noticeable result among other scheduling algorithms. It gives



a) Turnaround Time comparison



b) Waiting Time comparison



c) Response Time comparison

Figure 7. Comparison between our algorithm and other existing algorithms by Turnaround Time, Waiting Time and Response Time metrics

better treatment to the long waiting jobs, thus removes starvation and overcomes the limitations of other algorithms.

In Table VII, we have shown the average values of the three evaluation parameters to compare the algorithms. If we compare the sum of three values as a ranking for the algorithms. SPF will get the first and our algorithm gets the second position. SPF has the least average value amongst the other algorithms, but it is suitable only for interactive systems, since it gives higher priority to short jobs, thus, long jobs should wait a long time. Our algorithm overcomes the limitations of SPF and HRN, and is appropriate for most systems.

VIII. Conclusion

In this paper, we proposed a new fuzzy logic scheduler for process scheduling in CPU. Our approach is based on the idea of HRN algorithm, in which upon Service Time and Waiting Time of the process, the scheduler decides the priority of the process. We defined triangular membership functions for the input fuzzy sets, used Mamdani-product fuzzy inference engine, and introduced two types of defuzzifiers (Center of Gravity and Mean of Maxima) for our scheduler. However, we can ignore the defuzzification phase for simplicity, if we are certain about the accuracy of the priority of the processes and want to classify the processes. Since Service Time of the processes is approximate for the scheduler and we are not aware of the Service Time in advance, a fuzzy algorithm can help us having a better approximation about the priorities of the processes. Another contribution of our algorithm in contrast with HRN is that it does not need any other algorithm for determining the priority of processes, which take the same priority value, by HRN algorithm. Finally, we evaluated our algorithm with other common scheduling algorithm and presented the Turnaround Time comparison, total Waiting Time comparison, and Response Time comparison graphs and average values to rank the algorithms. According to our comparison our approach stands the best especially for long job systems.

TABLE VII. Average Values of the Three Metrics

	Average Turnaround Time	Average Waiting Time	Average Response Time	Summation
SPF	10.8	6	6	22.8
FHRN	11.2	6.4	6.4	24
SRT	10.8	8.4	6	25.2
FCFS	12	7.2	7.2	26.4
RR	15	8.2	4.2	27.4

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Analysis and Design of a Family Reunion Information System

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Keywords: information system, social network

Abstract: **Family reunions are often large events which require advance planning, data processing and feedback analysis. The successful planning, budgeting and organizing of this modernized social gala are a considerable feat. Several desktop solutions exist to aid reunion planners, but at the same time they are limited to use only by the planners. OurFamily247.com is a proposed information system tailored to organizing all types of family reunion gatherings in a distributed fashion. The system will allow planners and family members to be abreast of all information during the planning process, pre and post event. This research uses the software development life cycle to provide a structured analysis of the information needed to economically, socially and sequentially produce a desired and joyous event.**

I. INTRODUCTION

The family reunion is a time-honored method of preserving the extended family and celebrating the legacy of elders. While reunions are most common in African-American families, the benefits of these gatherings would have a positive effect on family unity for any participating group. A typical family reunion is summarized as an event that assembles the survivors of grandparents or earlier ancestors for recreation [9]. This normally includes dining and fun activities, as well as a reflection on family history. When families become united, communities are also strengthened. It is reminiscent of the sentiment "Strengthen a community, and in turn a city, a state, a nation, and the world are strengthened."

As the discussion of information systems begins, it is important to note technology can be of great assistance to the family reunion planning and executing processes. An information system, as defined by Jeffrey Whitten and Lonnie Bentley [8], is "an arrangement of people, data, processes, and information technology that interact to collect, process, store, and provide as output the information needed to support an organization." The main point to be gleaned from this particular definition is an information system is not just the technology, but also includes the people using the technology, as well as their purpose for using it.

Upon analyzing the given definitions of family reunion and information system, the following definition may be derived for a family reunion information system: "an arrangement of people, data, processes, and information technology that interact to collect, process, store, and provide as output the information needed to support an event that assembles the survivors of grandparents or earlier ancestors for recreation." In short, technology may be used to support the reunion and its planning processes.

In order for a software package to be considered a family reunion information system, several characteristics have been identified that must be present. (1) The system should be accessible by all family members. (2) The system should be user-friendly. (3) The system should be secure. (4) The system should support user interaction and feedback. (5) The system should provide innovative and useful services. Finally, (6) the system should be affordable.

This paper has the following key contributions:

1. A set of information system characteristics and how to apply them specifically to a family reunion information system.
2. A comprehensive set of requirements and detailed design work are provided for development of a family reunion information system.
3. An improved method of managing a family reunion via a functional version of a family reunion information system.

II. RELATED WORK

It was concluded in [4], after an extensive search that there are limited resources and room for improvements for an efficient and economically sound family reunion planning system. Technology improvements, specifically related to database and the internet, can make planning and executing a reunion much easier and more compact. More support for this need is presented, as well as

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discussions on the components of planning and the purpose of reunion software in general.

Family reunions are events that, if done well, deliver a sense of well-being and togetherness to everyone involved. A poorly planned and executed reunion, however, may have disastrous effects on family unity. An example of this was given in an interview with Saundra Loving [5], a member of the Helms-Johnson Family Reunion. The reunion planner chose a restaurant for the family banquet that could not accommodate large parties and only provided limited menu choices. The family had to sit at separate tables throughout the restaurant and many members order from the menu. The resulting effects were angry family members and many promises not to participate in future reunions. This demonstrates a necessity that all provisions be made for the planning of the event. In today's society, it is becoming more and more commonplace to use computers and the internet to facilitate most activities. Combine this with an estimated 200,000 reunions held each year in the United States, and immediately the significance of having a tool such as a family reunion information system becomes evident [3].

Often a considerable time constraint is placed on the planning process for a family reunion. For this reason, it takes many family members who are willing to help with the process to make it work [2]. The need to manage family members and their activities are a critical stage in the process.

The relative success or failure of a family reunion can largely be attributed to the types of activities presented. The activities chosen should reflect all age groups and preferences [10]. It is prudent that a mechanism be in place to provide an advance review of activities, as well as an avenue for feedback.

Communication is required in planning almost any type of event, and a reunion event is not an exception by any means. At every stage of planning, some form of communication should exist. Even if a family member is only at the point of considering a reunion. It is ideal to gauge interest among other family members; contacts have to be made [2].

Family reunion planners often struggle with the budgeting aspect of the reunion. This is natural because reunions tend to be expensive, and people want to know where their money is going. It is important for the success of the event to 1) raise enough money to put on the event, and to 2) make the budget transparent [2]. Raising money is often one of the most difficult tasks, so facilitating the fee payment and donation processes with a family reunion information system is critical to the success of the event. Though the funds of the event might be low, the expectations of the family members will remain high. So budgeting is key. Having a transparent budget, or a detailed statement of transactions and account balances, decreases dissension among fee-paying family members, provided the funds are disbursed wisely.

The most difficult task in planning a reunion is the initial collection of everyone's information. However, nothing will take place without this step being executed with some degree of success. Usually the event planner may choose to contact members of each branch of the family to collect this information, either via postal mail, phone, or email [1]. The question becomes "What is the best method of managing and maintaining this information?" to which the obvious answer is a completely functional family reunion information system.

The term "reunion software" is differentiated from the term "family reunion information system" in that reunion software only addresses excerpts of the total information need, while a family reunion information system addresses the entire need. Reunion software tends to deal with only the planning aspects of the reunion. As a result, the software is not useful for the inclusion of all family members or participants. Both entities have the same purpose, which is to provide tools for planners to use in creating a successful event. Reunion software stops at this point, while a family reunion information system also provides tools for the family members themselves to use.

Examples of reunion software are Family Reunion Organizer by Roots Magic Inc. and Reunion Planner Software by ReunionPlanner.com. It is important to note these are desktop applications which would only be available locally to the reunion planner. Any information shared with the family would have to be done by the planner.

These software packages do have many useful features. These include to-do lists, budget spreadsheets, personal information tables, etc… They include correspondence production, such as mailing labels, letters, and forms. Also, these tools provide the ability to create needed documents for the actual event, including name tags and award certificates. Roots Magic [6] goes a step further to allow import of a GEDCOM file, which is a format for exchanging genealogical data. With the purchase of Family Reunion Organizer, a customer is allowed to announce their planned reunion on family-reunion.com [6]. These features in themselves make a helpful tool in planning a reunion, but do not constitute a family reunion information system.

The biggest issues with the previously mentioned software packages are their architecture. This reviles an extreme fault in the foundation of those systems. The architecture of the system must provide a solid instructional base. Desktop software is not as collaborative as a family reunion information system should be. Most of the time only one user may access desktop software at a time. Also, desktop software can only be accessed locally or via remote desktop connection. The easiest remedy for the appropriate architecture is to use a distributed, web-based architecture.

III. REQUIREMENTS ANALYSIS

The purpose of this section is to show in detail the proposed OurFamily247.com (OurFamily) requirements and design. OurFamily is an information system tailored to organizing all types of family reunion gatherings. The system will allow planners and family members to be abreast of all information during the planning process, during the event, and after the event. A few of the system functions are as follows: comprehensive family member database, email communications, information collection, event programming, budget, etc…

A. User Requirements Specification

1. Purchase Family Reunion System account
2. Family Member Database
    a. Create generation groups
    b. Create family groups
    c. Create family member
    d. Edit Associations
    e. Print family member directory
3. Fee Collection
    a. Select financial officers
    b. Add mailing address for paper checks
    c. Accept online payments via PayPal
    d. Donations
4. Communications

   a. Send emails to family members
   b. Host blogs
5. Event programming
   a. Show family members the events planned
      i. Upload pictures
      ii. Display information
6. Budget
   a. Track spending
   b. Toggle display of accounting
7. Printing
   a. Print mailing labels
   b. Print directories
8. Lodging
   a. Show lodging information
9. Travel
   a. Directions to/from the surrounding airport
10. Items for purchase
   a. Enter all items that may be bought
11. Shop online
   a. Enter items into a shopping cart
   b. Check out
      i. Pay online via PayPal
      ii. Pay by mailing check or money order to physical address
12. Polls
   a. Pre-Reunion
   b. Post-Reunion
13. Post-Reunion
   a. Post awards
   b. Post next location
14. Committees
   a. Set up committees
   b. Assign members

Non-functional requirements represent constraints on the system, as well as other characteristics that are more difficult to verify than functional requirements. The following non-functional requirements have been identified:

1. Host shall be able to add, edit, publish, and delete pages without assistance after thirty (30) minutes of video-based training.
2. Financial officers can update and view every member's records.
3. Web pages shall load within three (3) seconds over a high-speed internet connection.
4. Features of the application shall be comparable to industry standard features.
5. Site shall be 508 compliant for handicapped accessibility. This will include descriptive alt tags for any image rendering, as well as the use of text-based linking.

B. Domain Requirements

Host shall be able to produce intact html code with a wysiwyg editor when editing pages. Use of WYSIWYG (what you see is what you get) tools has become standard practice for allowing non-technical users to update web pages.

C. Interface Specification

The completed system of OurFamily.com must interact with several different entities in order to provide its services. These include a database server, an application server, and a web browser. Here we describe the interfaces between these entities and OurFamily.com.

1. Database Server:
   Microsoft Access with ODBC data source which requires Windows 2000/2003/XP Operating system
2. Application Server:
   ColdFusion Application Server which requires Windows 2000/2003/XP Operating system
3. Web Browser:
   Internet Explorer, Firefox, Netscape Navigator, Opera, all of which requires javascript to be enabled

D. Viewpoints

Viewpoints describe the different actors that use or communicate with a system. We deal with two types of viewpoints, interactor and indirect. Interactors are directly related with the development of the system. The indirect viewpoint has a role in the systems, but is primarily in the background. Interactor viewpoints for OurFamily include:

1. Host - responsible for the creation and organizing of reunion
2. Family Member – participant in reunion
3. System Administrator – creates Host accounts

Indirect viewpoints include:

1. Vendors – take orders that the Host submits
2. PayPal – accepts payments from those who wish to order goods and pay online

E. CASE Workbench

CASE (Computer-Aided Software Engineering [6]) tools workbench consists of:

1. Macromedia Dreamweaver for code generation
2. MS PowerPoint for presentations
3. Project Folder for central repository
4. MS Word, Excel, and Access for program documentation/reporting
5. StarUML/Macromedia Fireworks for diagram design

IV. DESIGN SPECIFICATIONS

A. System Context

When developing an information system, it is important to grasp the context of a system in relation to its surroundings. This is best illustrated using a context diagram similar to Figure 1. Arrows denote data flows between the actors and the system in a manner that facilitates the understanding of the total context.

B. Use Cases

Use cases are excellent means of depicting the functionality of a system in non-technical terms. End users can look at use cases and understand what they represent intuitively. A number of use cases have been developed for OurFamily, which give a high-level understanding of what the system will be able to do. The use cases were developed from a collection of scenarios that describe user interaction with the system. Each use case, as shown in Figure 2, has an accompanying narrative.

The use cases were ranked each use case on a scale of 1-5 against six criteria. Those criteria are:

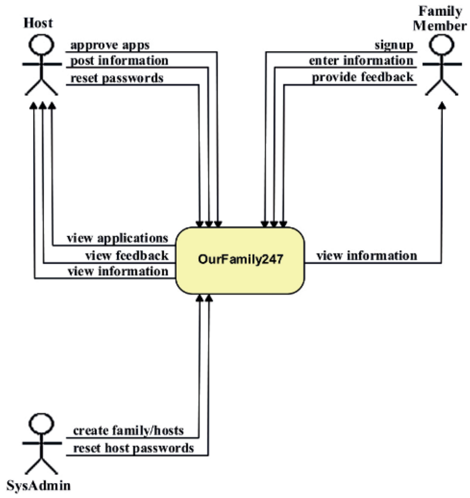

Figure 1: System Context Diagram.

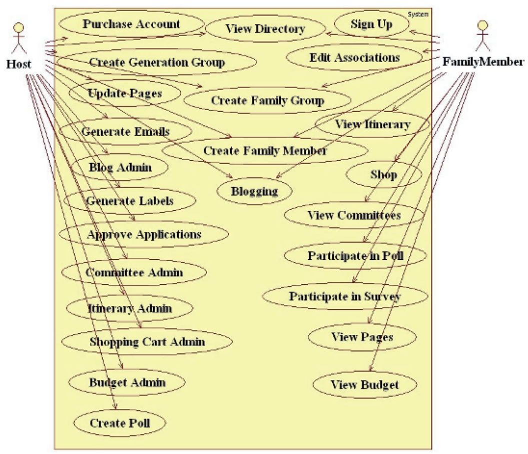

Figure 2: Use Case Diagram.

1. Significant impact on the architectural design.
2. Easy to implement but contains significant functionality.
3. Includes risky, time-critical, or complex functions.
4. Involves significant research or new or risky technology.
5. Includes primary business functions.
6. Will increase revenue or decrease costs.

The totals of the six criteria were then used to prioritize the use cases according to the following schedule: High – 18 +, Medium – 11-17, Low – 0-10.

All use cases of High priority are included in Build 1. Use cases of Medium priority are included in Build 2, and use cases of Low priority are included in Build 3. The results are as listed in Table 1.

Table 1: Use Case Ranking and Priority Matrix.

Use Case	Ttl	P	Cycle
Purchase account	18	H	1
Create family member	22	H	1
Create family group	21	H	1
Create generation group	22	H	1
Update Pages	19	H	1
View Directory	20	H	1
Generate Emails	13	M	2
View Itinerary	22	H	1
Blogging	15	M	2
Blog Admin	15	M	2
Itinerary Admin	19	H	1
Approve Applications	22	H	1
Budget Admin	12	M	2
Shopping Cart Admin	10	L	3
Generate Labels	8	L	3
Committee Admin	6	L	3
Sign Up	26	H	1
Edit Associations	26	H	1
View Budget	8	L	3
Shop	6	L	3
Participate in Survey	8	L	3
Participate in Poll	13	M	2
View Committees	6	L	3
View Pages	18	H	1
Create Poll	13	M	2

C. Design Goals

1. The system should have clear navigation and unambigious.
2. The system should not be resource-intensive.
3. The system should include core functions as listed in requirements.
4. The system should use text based links whenever possible.
5. The system should provide user accounts access and functionality in a timely manner.

D. Design Methods

Standard pictorial models were used in creating the design documentation for this system. A detailed entity-relationship diagram depicts the layout of the database design. Data flow diagrams are used to illustrate information flows and to give general overview of how to implement the described functionality. Sequence diagrams (see Figures 3 and 4) are also used to explain the operations even further.

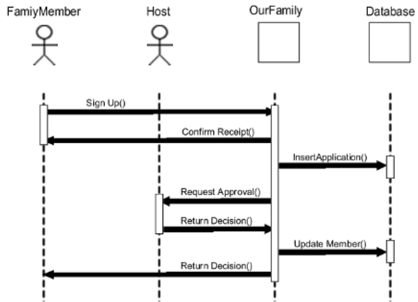

Figure 3: Sequence Diagram – Sign Up Use Case.

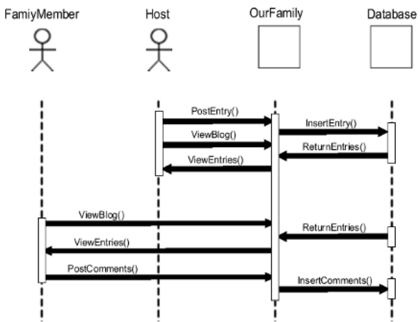

Figure 4: Sequence Diagram – Blogging Use Case.

Also important is the diagram of the system decomposition based on the available use cases. This provides a necessary grouping of related functionality into system components as shown in Figure 5.

E. User Interface Design

The user interface is designed with the following principles in mind:

1. Minimal number of links
2. Explanation of features
3. Protection of users from catastrophic errors
4. Minimal graphics for faster loading
5. Intuitive labels
6. Error messages that detail what happened and the next action to take

The resulting interface prototype consists of a header graphic, a navigation menu on the left side, and context-driven menus to the right side. The color scheme consists of black text, blue links, and gray buttons/highlights on a white background, all within a 1-pixel border.

V. Security Features

In today's society, preventative steps are needed to ensure protection from identity theft. In order to prevent any security concerns, OurFamily utilizes code that affords a level of protection from identity theft. The level and type of security is domain dependent. The term information security is broadly defined as an ongoing attempt to help secure information flow in the systems (network, database, online, etc). A password feature is implemented in the system for each host and family member. The password restricts access to personal information such as home addresses and individual names. If the wrong password is entered more than three times, it will have to be reset by the system administrator. Also another feature that can assist in theft prevention is an email alert that would go to the host or system administrator when a new member is added to the account. This alert will provide the host the right to confirm or deny whether this individual is a family member and can view other member's information.

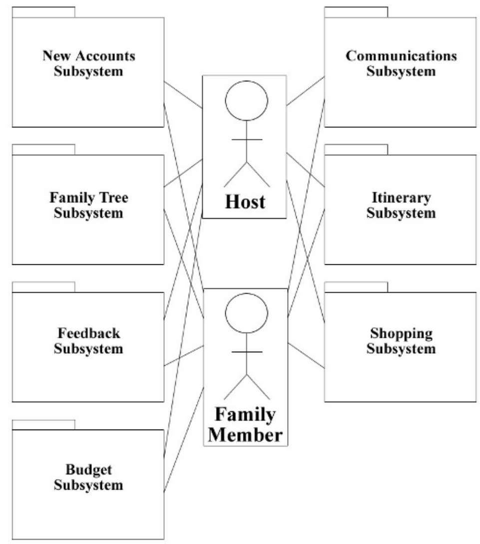

Figure 5: System Decomposition Diagram.

Although, the previous security features have been implemented on the website, there are several other options that would assure members that their privacy is protected.

The database security features can provide a second layer of security to the members of OurFamily reunion website. A database can secure access privileges to user in a hierarchical database. As follows, there will be three types of users: administrative users, financial officers, and family members as shown in Table 2. The administrators have full control over the database. They can add members, delete members and update the member's records. The financial officers can update and view every member's records. The family members can only view their individual records.

Yet, a third layer of protection can be implemented, an encryption protocol such as Secure Socket. This protocol was implemented by Netscape for transmitting private documents via the Internet. SSL uses a cryptographic system that uses two keys which encodes the data:

- public key known to everyone
- private or secret key known only to the recipient of the message

Both Netscape Navigator and Internet Explorer support SSL, and many Web sites use the protocol to obtain confidential user information, such as credit card numbers. By convention, URLs that require an SSL connection start with https: instead of http.

These are just several features have been or will be implemented to reduce the potential for identity theft. The OurFamily system is out – fitted to foil any attempt made by internet savvy criminals.

Table 2: Access Parameters for User Types

User Type	Full	Medium	Limited
Administrator	✓		
Financial Officer		✓	
Family Members			✓

VI. CONCLUSIONS

The focus of this work has been to use research and knowledge of the software development life cycle to identify a solution to a given problem. The problem was the need for family reunions to have a comprehensive system which includes high availability of information to family members, a central point for planning and communication in a secure environment, and ease of use, all at a low cost. This work offers proof of concept for a system to satisfy these needs.

There is a potential for future work. The system needs to be integrated with a payment gateway (PayPal) for online credit card/debit card/e-check transactions. The system should allow for customized reunion websites, as well as multimedia/image sharing among family members.

One particularly interesting area of future work is a concept not covered in the scope of this research. This concept consists of ad-hoc graphical family tree generation based on the family member, generation, and association database tables. It can be suggested the tools for implementation of this concept will be Java and XML, or similar technologies.

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Modified Ternary Optical Logic Gates and their Applications in Optical Computation

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Abstract: Optical logic gates are most important factors for the parallel operations and utilizations in VLSI circuits for the very high speed data processing and communications. This calls for multivalued logic implementation. In this paper we have communicated the trinary optical logic gates with optical implementations and a few applications of it to the half adder and full adder circuits, this may also be extended to multiplexers and de-multiplexers, flip-flops, registers, counters, etc.

Index Terms: Ternary, dibit, polarization, savart plate, MTN, SLM.

1. INTRODUCTION

During the last twenty-five years, digital optical computing has stimulated a great deal of interest. Optical processing offers an attractive approach for designing opto-electronic computer system by exploiting the inherent parallelism, non-interfering communication, 2-D storage capacity and ultrahigh processing speed[1-4]. However, the parallelism of optical beam could not be properly utilized using cascaded single-bit operating units. Signed digit number was initiated with the pioneering works of Avizienis[5]. The parallel nature of such modified binary systems promoted researchers to adopt the modified signed digit algorithm for optical computing[6-10].

The demand of the state-of-the-art is the parallel processing for the very fast operations in the practical fields. The optical gates may be the only solution to meet up the gap in the very grass root level. In the last two decades so many works[16,17] has been reported in this area. Carry and borrow free operations also have been reported to perform the high speed operations. The representation in optical output with state of polarization along with presence and absence of light has extended the activities to multivalued logic[11-15]. Our present method is the extension of our previous work on tri-state logic system in ternary as well as dibit representation to perform logical operations with better possibilities. Carry and borrow operations can also be performed in a simpler way to meet up the needs of data processing.

2. TERNARY REPRESENTATION AND THE LOGIC SYSTEM

The tri-state of the ternary representation are classified as the true, false and contradiction. In this case we have consider these representations explicitly in sign magnitude form of modified ternary number(MTN) as per the Table-I given below:

TABLE I
TERNARY LOGIC SYSTEM

Logical state	Represented by	Dibit representation	State of polarization
True/ Complete information	1	01	Vertical polarization
False/ Wrong information	$\bar{1}$	10	Horizontal polarization
Contradiction/ Partial information	0	11	Presence of both the horizontal & vertical polarization

It is to be noted here that the state 00 is considered as redundant.

3. BASICS OF TERNARY ALGEBRA

The Ternary algebra is on the basis of the following basic operations:

Unary Operations
Binary Operations
Algebraic Laws

3A. Unary Operations

In case of ternary algebra the following unary operations are applicable.

1) Invert/Complement/NOT
2) True Selector (↑)
3) False Selector (↓)
4) Exclusive True Selector (⬆)
5) Exclusive False Selector (⬇)

The above operations are defined as follows:

1) INVERT/COMPLEMENT/NOT**:** This operation simply invert the input. The complement of true is false and vice-versa but the complement of the state contradictory is always contradictory as we cannot come to any conclusion.

2)TRUE SELECTOR(↑):This operation simply select the true value lying in the input information. In the case of contradiction some true information is lying in it so it selects the true value – but in case of false since there is no any true component so it is contradictory.

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3)FALSE SELECTOR(↓):This operation simply select the false value lying in the input information. In the case of contradiction some false information is lying in it so it selects the false value – but in case of true since there is no any false component so it is contradictory.
4)EXCLUSIVE TRUE SELECTOR ($\Uparrow$):This operation select the true value if and only if the input information contain only true value. In other cases the output is contradictory.
5)EXCLUSIVE FALSE SELECTOR ($\Downarrow$):This operation select the false value if and only if the input information contain only false value – otherwise it is contradictory.
The truth tables for the above operations are given in table–II.

TABLE -II

1) Invert/Complement/NOT

A	$\bar{A}$
1	$\bar{1}$
0	0
$\bar{1}$	1

2) True Selector (↑)

A	A↑
1	1
0	1
$\bar{1}$	0

3) False Selector (↓)

A	A↓
1	0
0	$\bar{1}$
$\bar{1}$	$\bar{1}$

4) Exclusive True Selector ($\Uparrow$)

A	A $\Uparrow$
1	1
0	0
$\bar{1}$	0

5) Exclusive False Selector ($\Downarrow$)

A	A $\Downarrow$
1	0
0	0
$\bar{1}$	$\bar{1}$

3B. Binary Operations

The operations for two variables as applicable for the ternary logic system is defined in the Table-III. The logic gates are defined as follows.
1) OR GATE [A$\vee$B]: The output of this gate always gives us the value *true* or *false* depending on that at least one of its input values is either true or false respectively, but in the case for two input values when the one is true and the other is false the output is not determinable i.e, contradictory.
2) AND GATE [A$\wedge$B]: This gate always gives us the value true or false when and only when both the inputs values are simultaneously true or false respectively, but in all other cases the output will be not determinable i.e, contradictory.
3) XOR GATE [A $\oplus$ B]: The output of this gate always gives us the value true or false depending on one of the input values is either true or false– but in the case for which the two input values are simultaneously true or false or one is true and the other is false the output is not determinable i.e, contradictory.

TABLE-III
TRUTH TABLE FOR OR, AND, XOR, NOR, NAND AND XNOR GATES

B	A	OR	AND	XOR	NOR	NAND	XNOR
1	1	1	1	0	$\bar{1}$	$\bar{1}$	0
1	0	1	0	1	$\bar{1}$	0	$\bar{1}$
1	$\bar{1}$	0	0	0	0	0	0
0	1	1	0	1	$\bar{1}$	0	$\bar{1}$
0	0	0	0	0	0	0	0
0	$\bar{1}$	$\bar{1}$	0	$\bar{1}$	1	0	1
$\bar{1}$	1	0	0	0	0	0	0
$\bar{1}$	0	$\bar{1}$	0	$\bar{1}$	1	0	1
$\bar{1}$	$\bar{1}$	$\bar{1}$	$\bar{1}$	0	1	1	0

4) NOR GATE [$\overline{A \vee B}$]: The output of this gate is simply the complemented output of the OR Gate.
5) NAND GATE [$\overline{A \wedge B}$]: This gate always gives us the complemented output as that of AND Gate.
6) XNOR GATE [$\overline{A \oplus B}$]: The output of this gate always gives the complemented output as that of XOR Gate.

3C. Algebraic Laws

The basic algebraic laws – Commutative Law, Associative law, Distributive law and identities are also valid in this case.

(i) $A \vee B = B \vee A$ $\quad A \wedge B = B \wedge A$ Commutative law
(ii) $(A \vee B) \vee C = A \vee (B \vee C)$
$(A \wedge B) \wedge C = A \wedge (B \wedge C)$ Associative law
(iii) $A \vee (B \wedge C) = (A \vee B) \wedge (A \vee C)$
$A \wedge (B \vee C) = (A \wedge B) \vee (A \wedge C)$ Distributive law
(iv) $A \vee \bar{A} = 0$ $\quad A \wedge \bar{A} = 0$ Complements
(v) $A \vee A = A$ $\quad A \wedge A = A$
$A \vee 0 = A$ $\quad 0 \wedge A = 0$ Idempotence

3C.1 Ternary Addition and Subtraction

Rules : Ternery addition and subtraction rules follow as given in table-IV.

TABLE -IV

Addition Rule:

$1 + 1 = (1)\,0$	$0 + 1 = 1$	$\bar{1} + 1 = 0$
$1 + 0 = 1$	$0 + 0 = 0$	$\bar{1} + 0 = \bar{1}$
$1 + \bar{1} = 0$	$0 + \bar{1} = \bar{1}$	$\bar{1} + \bar{1} = (\bar{1})\,0$

Subtraction Rule:

$1 - 1 = 0$	$0 - 1 = \bar{1}$	$\bar{1} - 1 = (\bar{1})\,0$
$1 - 0 = 1$	$0 - 0 = 0$	$\bar{1} - 0 = \bar{1}$
$1 - \bar{1} = (1)\,0$	$0 - \bar{1} = 1$	$\bar{1} - \bar{1} = 0$

Explanation: $1 + 1 = (01 + 01) = 10 = 0$ [carry = 1]
$\bar{1} + \bar{1} = (-01 - 01) = -10 = \bar{1}0$ [carry = -1 = $\bar{1}$]
$1 - \bar{1} = 1 - (-1) = 1 + 1 = (1)0$
$\bar{1} - 1 = \bar{1} + (-1) = \bar{1} + \bar{1} = (\bar{1})0$

The number within bracket ( ) shows carry in case of addition and borrow in case of subtraction.

3C.2 Complements: Ī is a signed bit and so the system acts just like a binary system. The 1's complement of a signed ternary number is simply the invert or complement of the same. Therefore, it is obtain by performing the Invert/Complement/NOT operation.

In case of ternary number:-
The 2's complement = The 1's complement + 1.

Example: The 1's complement of the ternary number 100Ī01Ī is Ī0010Ī1, which we get simply by replacing 1 by Ī, 0 by 0 and Ī by 1.

The 2's complement of the above number is obtained by following the following steps:

Say, any ternary number :	1 0 0 Ī 0 1 Ī
1's complement of the above number :	Ī 0 0 1 0 Ī 1
2's complement of the original number :	Ī 0 0 1 0 Ī 1
	+ 1
is :	Ī 0 0 1 0 0 0

3.C.3 Complemented subtraction

In case of ternary system the complemented subtractions are also valid. The same can be demonstrated by the following example.

Example:

Decimal	Binary	Ternary
7	1 1 1	1 1 1
- 2	- 0 1 0	+ 0 Ī 0
5	1 0 1	1 0 1

N.B.: In case of ternary the complemented operations are not required as we get negative numbers directly within the ternary [as it signed numbers] – so direct addition instead of subtraction yields the result.

3C. 4 Addition Operation

Example :

Decimal	Binary	Ternary
	1 1 1(carry)	1 1 1 (carry)
15	1 1 1 1	1 1 1 1
+10	+ 1 0 1 0	+ 1 0 1 0
25	1 1 0 0 1	1 1 0 0 1

N.B.: In case of ternary positive numbers no signed bits are required – so it is just like the binary operations.

4. ADDERS

The adders can be classified into two categories –Half Adder and Full Adder.

4A. Half Adder

The truth table for half adder is given in the table-V. The Half Adder Sum output is represented by the following equations:

Sum Output (Sum) = XOR gate output
Carry Output (Cout) = AND gate output.

TABLE-V
TRUTH TABLE FOR HALF - ADDER

Input		Output	
B	A	Cout	Sum
1	1	1	0
1	0	0	1
1	Ī	0	0
0	1	0	1
0	0	0	0
0	Ī	0	Ī
Ī	1	0	0
Ī	0	0	Ī
Ī	Ī	Ī	0

4B. Full Adder

The truth table for the full adder is given in the table-VI. It is clear from the truth table that it is accomplished by creating three half adders and multiplexing their output based on the carry-in.

The first half adder is identical to a normal half adder since the carry-in is 0. The second adder with carry-in =1 ,and the third adder with carry-in = Ī.

TABLE-VI
TRUTH TABLE FOR FULL - ADDER

Input			Output	
C_{in}	B	A	Cout	Sum
1	1	1	1	1
1	1	0	1	0
1	1	Ī	0	1
1	0	1	1	0
1	0	0	0	1
1	0	Ī	0	0
1	Ī	1	0	1
1	Ī	0	0	0
1	Ī	Ī	0	Ī
0	1	1	1	0
0	1	0	0	1
0	1	Ī	0	0
0	0	1	0	1
0	0	0	0	0
0	0	Ī	0	Ī
0	Ī	1	0	0
0	Ī	0	0	Ī
0	Ī	Ī	Ī	0
Ī	1	1	0	1
Ī	1	0	0	0
Ī	1	Ī	0	Ī
Ī	0	1	0	0
Ī	0	0	0	Ī
Ī	0	Ī	Ī	0
Ī	Ī	1	0	Ī
Ī	Ī	0	Ī	0
Ī	Ī	Ī	Ī	Ī

5. PARITY BIT

Parity bit is the most important in case of verification of information transmission. Parity generator and checker have a lot of applications in communications. The details are discussed hereunder.

5A. Parity Checker

From the truth table of XOR gate it clear that the for the input combinations of (11), (1 $\bar{1}$), ($\bar{1}$ 1) or ($\bar{1}$ $\bar{1}$) the output of an XOR gate is always 0(zero). We can send the even or odd numbers of 1s or $\bar{1}$s through the system and verify it at the receiving ends by observing the output of the parity checker. It is also important to note here that the $\bar{1}$ is the signed representation of -1 by using true selector we can verify the presence of even or odd number of 1s in the data flow or information and similarly by using the false selector we can easily verify the even or odd numbers of $\bar{1}$ s in the data flow or information. It is to be mention here that for the even number of 1s or $\bar{1}$ s in the data flow or information the output of the parity checker is always LOW(0) and for the odd number of 1s or $\bar{1}$ s in the data flow or information the output of the parity checker is always HIGH(1 or $\bar{1}$).

5B. Parity Generator

Parity generators are two types: Even and Odd. Both can be implemented by using the XOR gate. In case of even or odd parity generators a parity bit is generated at the input depending on the number of bits present in the transmitting system. Therefore, before sending the data stream it is essential to verify whether the transmitting data stream contains even or odd numbers of 1s or $\bar{1}$ s in the data stream. So a parity checker is essential at the input of the transmitting system to verify the same and the output of it is used as the generator of parity bit. In case of even number of 1s or $\bar{1}$s present in the input of the transmitting system the output of the parity checker is 0 (LOW) – so the parity bit is 0 and in the event of odd number of 1s or $\bar{1}$ s present in the input of the transmitting system the output of the parity checker is HIGH(1 or $\bar{1}$)– so the parity bit is 1 or $\bar{1}$. Therefore, considering along with the parity bit when we send it through the transmitting system it is always even parity generator – but in case of odd parity generator we have always to invert or complement the output of the parity generator before sending it through the transmitting system.

6. IMPLEMENTATION OF TERNARY LOGICS USING OPTO-ELECTRONICS SYSTEMS

In optics, two orthogonal states of polarization as well as that of absence and presence of light may express the binary states. Using both the properties at a time we can generate four-state logic system using dibit representation. In this paper we have considered the 00 state as the redundant state. In the two previous papers Basuray and others[11,12] have used such representation to ternary logic systems. In this report the logic used has been extended to incorporate modified ternary logic systems.

6A. The Basic Building Block

Light from a laser source polarized at an angle 45 degrees with respect to the two crystal axes incident on the savart plates S_1 and S_2 as shown in figure 1(a). The light incident on S_1 will be splitted into two orthogonal components and comes out of S_1 with a spatial shift between them. The electrically addressed negative SLMs - P_1 and P_2 are then used for the two inputs. The nature of the negative SLM is such that it is transparent when there is no electric voltage applied on it and it becomes opaque when an electric voltage is applied on it. The property of positive SLM is just reverse. Hence the input may be considered as in the form of dibit (two bits) representation.

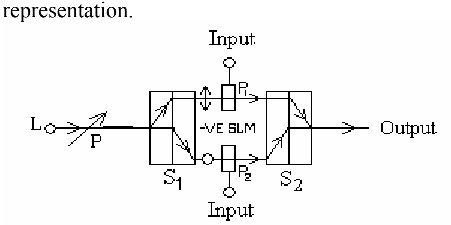

Fig. 1(a): Basic Building Block

The second savart plate S_2 is then used to re-unite the two polarized beam for further operations. Different logical gates may be implemented by combining this very basic module. The truth tables for the logic gates in ternary system for some common gates are defined in Table-II and Table-III.

6B. The Invert/Complement/NOT Gate

The output obtained from the gate is simply the complement of the inputs. The savart plate S_1 decomposes the incident beam into two orthogonal polarized beams. The inputs are applied to the SLMs P_1 and P_2 and accordingly the essential components of the polarized beams are present in light coming out of the savart plate S_2 and they are again allowed to pass through the savart plate S_3. Here, the optical to electrical converter (O/E) is used in the path of the ray to convert the light into electric voltage and feed it to the

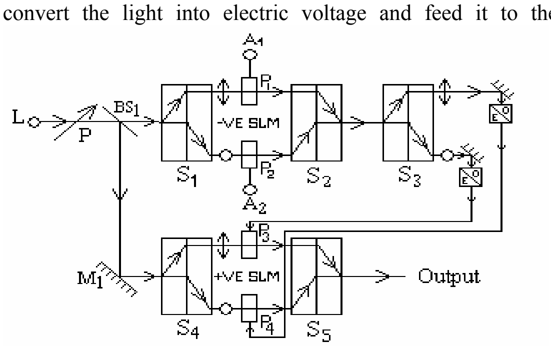

Fig.1(b): Invert/Complement/NOT Gate

positive SLMs P_3 and P_4 to control the polarized components of the beam coming out of the savart plate S_4. The savart plate S_5 is used to re-unite the final components present and to get the final output. Here, the output as we get follows the truth table as per the Invert/Complement/NOT Gate.

6C. True Selector (↑)

This is a special type of selector circuit – which selects only the true components present in the input information. Here, four number of savart plates S_1, S_2, S_3 and S_4 and five number of negative SLMs P_1, P_2, P_3, P_4 and P_5 are used to implement the circuit for the logic. The input is applied to A (A_1 and A_2) and B is set as B_1=0 and B_2=1.

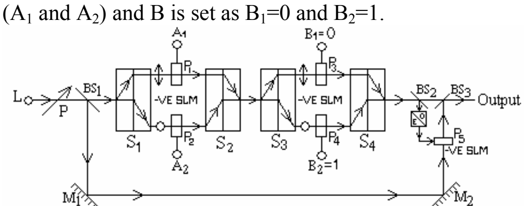

Fig. 1(c): True Selector

6D. False Selector (↓)

False selector is used to select only the false components

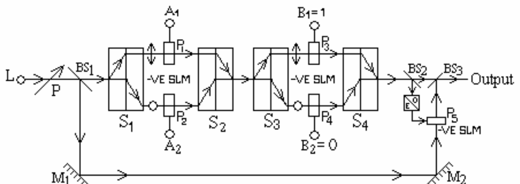

Fig. 1(d): False Selector

present in the input information. Here, also four number of savart plates S_1, S_2, S_3 and S_4 and five number of negative SLMs P_1, P_2, P_3, P_4 and P_5 are used to implement the circuit for the logic. The input in applied to A (A_1 and A_2) and B is set as B_1=1 and B_2=0 to get the required output

6E. Exclusive True Selector (⯅)

The circuit to implement the exclusive true selector is shown in the figure 1(e) below. Here we get the exclusive true output when and only when the input is true.

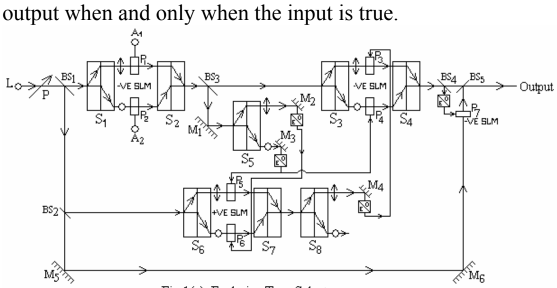

Fig. 1(e): Exclusive True Selector

6F. Exclusive False Selector (⯆)

The circuit to implement the exclusive false selector is shown in the figure 1(f) below. Here we get the exclusive false output when and only when the input is false.

6G. OR Gate

In this case the polarized parallel beam from the laser source L is incident on the beam splitter BS_1 - where it is splitted into

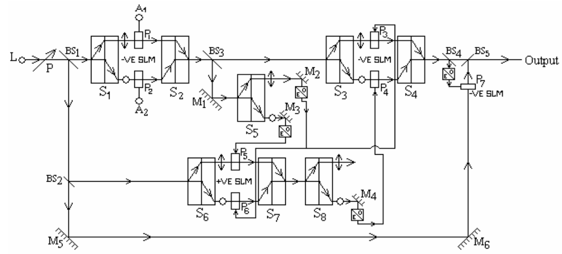

Fig. 1(f): Exclusive False Selector

two directions [Fig.1(g)]. One part is incident on the savart plate S_1 and the other part on the mirror M_1. The savart plate S_1 splits the beam into two orthogonal components–the p-polarization and the s-polarization. The input A (combination of A_1 and A_2) controls the –ve SLMs P_1 and P_2 and accordingly the p and s-polarizations come out of P_1 and P_2. Then they recombine by the savart plate S_2 and incident on the next savart plate S3 and by similar process it is also spatially modulated by the negative SLMs P_3 and P_4 depending on the input B (combination of B_1 and B_2). Then the rays re-united by S_4 and incident on the beam splitter BS_2. One part from BS_2 incident on the optical to electrical converter (O/E) which is connected to the negative SLM P_5 to control the light coming from the mirror M_2 and the other part directly to the output. The output follows the truth table for OR gate.

For example, say A=0 (i.e.A_1=0 and A_2=0), the components of beam splitted by S_1 will not be obstructed by P_1 and P_2 and hence the output of S_2 will contain both the

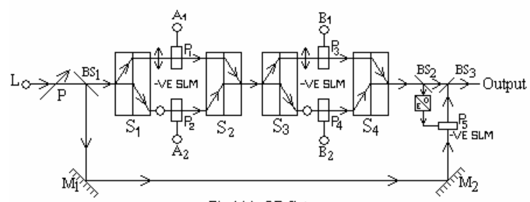

Fig.1(g): OR Gate

p and s - polarizations. Say, the second input B=1 (i.e, B1=0 and B2=1) then the p –polarization will pass out through P_3 but the s–polarization will be obstructed by P_4 and hence there will be only p-polarization from S_4, as a result the optical to electrical converter (O/E) will generate the electrical signal and the negative SLM P_5 will obstruct the light coming from the mirror M_2. Hence only the p-polarization will be present in the output – which is equivalent to 1. So the Truth Table for OR gate follows. Similarly, it is valid for other input combinations also as per the truth table.

6H. AND Gate

The circuit for the AND gate is shown in Fig.1(h). The polarized beam coming from the source L is incident on the beam splitter BS_1 and splitted into two parts. One part is passes through the first savart plate S_1 and splitted into two orthogonal components with spatial shift and modulated by the negative SLMs P_1 and P_2 by applying the input at A (combination of A_1 and A_2) and they reunited again at S_2. The other part of the ray coming out from the beam splitter BS_1

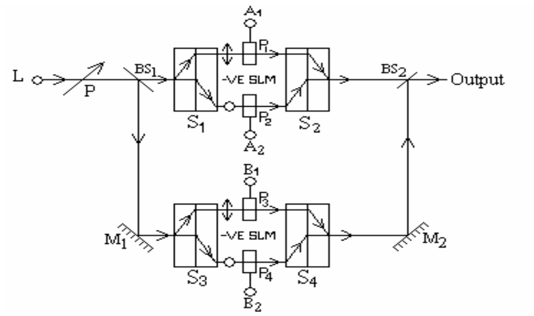

Fig.1(h): AND Gate

and mirror M_1 incident on the savart plate S_3 of the second building block. The beam is splitted and modulated by the negative SLMs P_3 and P_4 as per the other input at B (combination of B_1 and B_2). Depending on A and B the output will occur according to the AND truth table as shown in Table-III.

6 I. Exclusive-OR (XOR) Gate

The circuit diagram for the exclusive-OR gate is shown in the figure 1(i). The combination as shown follows the truth table for Ex-OR (XOR) gate. The two inputs are A (combination of A_1 and A_2) and B (combination of B_1 and B_2).

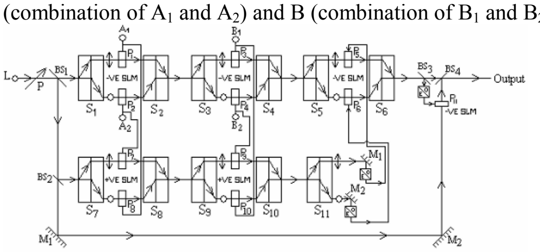
Fig 1(i): XOR Gate

6 J. NAND Gate

The circuit diagram for the NAND gate is shown in the figure 1(j). The combination as shown follows the truth table for NAND gate.

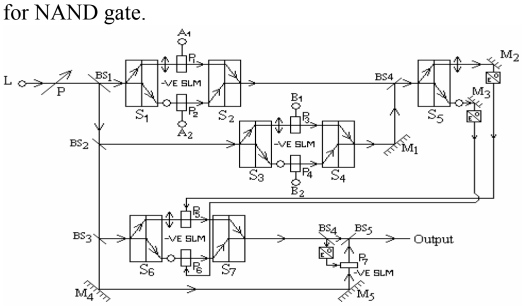
Fig 1(j): NAND Gate

6K. NOR Gate

The circuit diagram for the NOR gate is shown in the figure 1(k). The combination as shown follows the truth table for NOR gate.

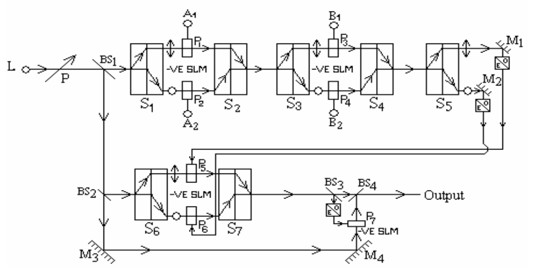
Fig.1(k): NOR Gate

7. CONCLUSION

We have discussed in this report the very basic ternary logic systems and their practical implementations by using the opto-electronic devices for the fast operation. Due to signed bit implementation the mathematical operations are also very simple. This tri-state logic find its applications in gray image processing, cellular automata, fuzzy logic systems, fractals and other emerging areas where fast operations are needed.

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Globalization and Parallelization of Nelder-Mead and Powell Optimization Methods

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***Abstract* — Optimization problems in engineering involve very often nonlinear functions with multiple minima or discontinuities, or the simulation of a system in order to determine its parameters. Global search methods can compute a set of points and provide alternative design answers to a problem, but are computationally expensive. A solution for the CPU dependency is parallelization, which leads to the need to control the sampling of the search space. This paper presents a parallel implementation of two free-derivative optimization methods (Nelder-Mead and Powell), combined with two restart strategies to globalize the search. The first is based on a probability density function, while the second uses a fast algorithm to uniformly sample the space. The implementation is suited to a faculty network, avoiding special hardware requirements, complex installation or coding details.**

***Index Terms*— global optimization, parallelization, free-derivative methods, Nelder-Mead, Powell.**

I. Introduction

MANY engineering problems involve the minimization of a cost function corresponding to performance measures of a given system. Very often, such problems present two characteristics that make it harder to find a solution: the non-convexity of the objective function and the computational complexity of its numerical evaluation [1].

Characteristics as non-convexity and discontinuity constraint the choice of numerical procedures, as well as the setting of parameters and the performance of the optimization. Termination criteria like final tolerance and limit of function evaluations are difficult to tune in advance but have a decisive impact on the quality of results. The presence of multiple minima generally require the computation of a set of feasible solutions, in order to provide engineers with the possibility to chose qualitatively between different designs.

The evaluation cost depends on several factors as the size of the search space, the required accuracy, or stability issues that lead to the use of very fine grids. In many cases the cost function involve results obtained by simulating a complex system along a given trajectory, as for example in fluid dynamics. These tasks are highly time consuming; it is not rare that calculation times of the order of several hours are considered tolerable in a given scenario.

Globalization techniques may be grouped in three broad categories: population-based, stochastic and multistart-based. Population-based globalization computes a set of solutions, which are combined according to certain rules in order to generate new candidate points [2]. Examples are evolutionary [3] and memetic algorithms [4]. Stochastic methods like perturbed gradient [5] and simulated annealing [6] repeatedly compute one individual and adjust parameters to continue the search [7]. Multi-start globalization is based on the repeated use of minimization algorithms like gradient descent, generally coupled with an heuristic to sample the search space.

Population-based and multistart techniques are well suited for parallelization on a computer network. For instance, parallel genetic algorithms can be divided into single and multiple population [8], each approach working better in different hardware architectures. Multistart algorithms are easily implemented with the single population approach, where a master computer is responsible for assigning new starting points and the slave machines run a standard optimization method [9]. Another possibility is to modify the base method itself to exploit the available parallelism [10, 11].

The main difficulty with the multi-start approach is to balance economy of computational resources with an appropriate sampling of the search space. This article describes two globalization strategies, applied in conjunction with the parallelization of two local free-derivative minimization methods: Nelder-Mead and Powell. The software computes a set of solutions, which may be selected, for example, according to qualitative criteria. The implementation described runs on a standard PC/Ethernet network and prioritizes the easy of use, requiring only an ANSI C++ compiler.

II. Local Optimization Algorithms

There is a variety of optimization algorithms available; the selection between them depends on factors as the mathematical properties of the cost function, like discontinuity, and the convergence rate for a given problem.

The Nelder-Mead and the Powell algorithms are zero-order

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering*, 93–98.

methods, also known as direct search methods. The relative inefficiency of such algorithms is compensated by their capability to handle problems where the gradient is not available, their robustness and easy of use [12].

A. *Nelder-Mead Method*

The Nelder-Mead method is probably the most known direct search method. It is based on the comparison of the function values at the n+1 vertices x_i of a polytope (usually called the simplex). The initial simplex is determined from a starting point $\mathbf{x}_0$ (the initial guess) and its vertices generally form a hypercube.

As the optimization proceeds the vertices are moved by operations named reflection, expansion and contraction. At each step the algorithm choses the next operation so that the function value decreases. The cumulative effect of the operations is, roughly speaking, to stretch the simplex along the descent directions and to zoom around local optima. As the volume of the simplex decreases, the algorithm becomes local. The termination criteria can be chosen as a test for flatness, max [f($\mathbf{x}_0$), ... ,f($\mathbf{x}_N$)] – min [f($\mathbf{x}_0$), ... ,f($\mathbf{x}_N$)] < ε; a minimal standard deviation of the function values on the vertices; or a limit on the number of iterations.

There are two situations where the method may fail to converge to a local minimum. First, it may wrongly avoid to explore a region that would be a basin of attraction, if the simplex is too large. Second, the simplex may collapse into a subspace, $\mathbf{x}_i = \mathbf{x}_j$, $i \neq j$. This case must also be tested as a termination criterion.

The original Nelder–Mead algorithm was conceived for unbounded domain problems, since the reflexion and expansion operations may move a vertex outside a bounded domain. This situation can be handled by defining a projection operation that maps outside points back to the boundaries of the search space. Care must be taken not to make the simplex collapse into the subspace of the saturated variables.

B. *Powell Method*

The Powell Method, also called Conjugate Directions Method, performs one-dimensional minimizations following conjugated directions. Two vectors (or directions) $\mathbf{v}_i$ and $\mathbf{v}_j$ □ $\mathbf{R}^n$ are said to be conjugated with respect to a positive definite symmetric matrix **A** if $\mathbf{v}_i^T \mathbf{A}\, \mathbf{v}_j = 0$.

The rationale behind the Powell method is the property that the minimum of a quadratic function f in a n-dimensional space can found in n one-dimensional successive minimizations, following conjugated directions (see proof in [13]). The one-dimensional minimization along each direction can be performed using line search algorithms, like dichotomous (binary) or golden section search .

Generally the unit vectors $\mathbf{e}_0$, $\mathbf{e}_1$, ... , $\mathbf{e}_n$ are taken as the first set of directions. The algorithm modifies this set according to the results of the search. For example, the vector formed by vertices with highest and lowest function values replaces the worst direction of the previous step [14].

III. Globalization by Restart

Restart is the simplest globalization strategy, but may also be the more expensive. Since our objective is to compute a set of solutions possibly containing the global optimum, multistart is a reasonable choice provided that the search space be explored as uniformly as possible. In order to do this, new points must be generated in regions not yet inspected.

Using a client-server network architecture, the server computer may be assigned the task of recording the points evaluated and generating new candidates for optimization. Although ideally the server should store all points evaluated by the clients, in practice we limited the record to the start and end points of each optimisation run. This strongly reduces the network traffic and speeds-up the clients tasks, by avoiding time-consuming I/O operations. The restriction was used in the implementation of the two sampling solutions developed.

One last requirement for the quality of results is the random number generator. The implementation here described makes use of the Mersenne Twister algorithm [15], which has a low computational cost and good statistical properties.

A. *Probability-density based sampling*

The first restart solution was first described in [16] using a sequential implementation. It is based on an estimate of the probability of a point being selected, assuming a Gaussian distribution.

The estimate is calculated using a Parzen-windows approach [17]. Given the set **X** of all previous start points $\mathbf{x}_{0..N}$, a set **R** of random points $\mathbf{r}_{0..K}$ is generated and for each $\mathbf{r}_i$ it is computed

$$p_i = \left(N\sqrt{2^n \pi^n \det \mathbf{M}}\right)^{-1} \sum_{j=0}^{N} e^{v_j} \qquad (1)$$

where n is the dimension of the space, **M** is the diagonal matrix $m_k = \sqrt{\alpha(\mathbf{x}_k^{\max} - \mathbf{x}_k^{\min})^2}$, α is parameter of the distribution and $v_j = (\mathbf{r}_i - \mathbf{x}_j)^T \mathbf{M}^{-1} (\mathbf{r}_i - \mathbf{x}_j)$. The point with minimum p_i is chosen for a new start.

Each m_k corresponds to the extension of the search space along each dimension; the value of α changes the shape of the associated Gaussian and allows to control the contribution of distant points $\mathbf{x}_j$ to the value of p_i. For example, by choosing α=0.01, one standard deviation from the mean covers 20% of the domain.

The Nelder-Mead and Powell algorithms are optimization methods for unbounded problems. However, the random generation of points in an unbounded domain is not adequate for practical engineering problems, which deal with limited quantities as energy or deformation. For this reason the elements of **R** are randonly selected from a n-dimensional hypercube defined by the user.

The process of generating a new point and evaluating (1) involves three nested loops, totalizing $K□n□N$ iterations. In order to obtain an uniform sampling of the search space, a necessary condition is $K □ \sqrt[n]{N}$. This has the undesirable effect of increasing the computational cost of the sampling as the global search proceeds and N gets larger. As a result, the client machines will spend increasingly longer idle times waiting for the server to compute new starting points.

B. *Grid based sampling*

The second restart procedure arises from the fact that minimizing (1) corresponds to finding a point that occupies the less densely populated area in the search space. In other terms, this point maximizes the distance to its nearest neighbour. Such a distribution can be obtained by placing points in a regular grid; Figure 1 illustrates the process in two dimensions. The distance between points is given by □ = 2^{-k} for a grid with $2^{n^k} + 1$ points.

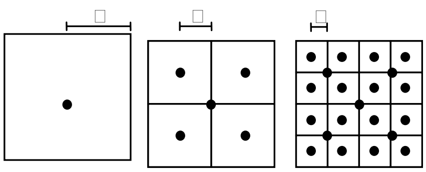


Fig. 1. Recursive sampling in two dimensions. The distance □ decreases gometrically while the number os points increases exponentially.

In higher dimensions, filling exhaustively the search space with a lattice is in principle unfeasible. Another alternative is to generate a random point and test the distance to its nearest neighbours in **X**. The minimum acceptable distance is given by □ and can be computed using the maximum norm, which is computationally faster than the Euclidean norm.

This solution induces the new problem of searching nearest neighbours in an arbitrary dimension space [18]. Since in the present case the maximum norm was choosen, it is possible to use a simple and very efficient implementation, based on binary search. In this work an adaptation of the solution developed by [19] was used, allowing the inclusion of points on the fly.

The set **X** of points already used to start the minimization is kept sorted along the first dimension. Given a new random candidate $\mathbf{r}_j = \langle r_0, r_1, \ldots, r_N \rangle$, a binary search in **X** constructs a new set **S**, whose elements $\mathbf{s}_j = \langle s_0, s_1, \ldots, s_N \rangle$ are such that $|s_0 - r_0| <$ □. Next, points **s** are removed from **S** if $|s_i - r_i| \geq$ □, $i = 1..n$. If **S** becomes empty during this process, **r** do not have any □-neighbours: it is accepted as a new starting point for optimization, included in X and sent to a client computer at the network. Otherwise, a new **r** must be randomly generated and tested.

The algorithm ensures the obtainment of evenly distributed points, making the set **S** much smaller than **X**. In average, **S** contains $\sqrt[n]{N}$ points. For example, in a 100-dimensional space containing 1 000 000 points, this value is only 1,149. In the worst case the algorithm will compare the random candidate **r** with all points in **S**. The average algorithmic complexity is below $O(n \cdot \sqrt[n]{N})$ per candidate, which is much better than $O(n□N)$ for the PDF-based approach. Finally, the quality of the distribution does not depend on any parameter, like K that appears in (1).

IV. Parallel Implementation

There are several software libraries available for the implementation of parallel algorithms. Among the most used it can be cited PVM, MPI, CORBA, DCOM and more recently, the .NET framework. While all these options provide useful functionalities that include synchronization, error handling and naming, they also require a considerable learning effort from the user. Another complicating factor is the need to install and configure additional software.

One requirement for the implementation was to keep the solution as simple as possible, so that a basic knowledge of C language would suffice to employ the software. To avoid complex dependencies and installation, all network communications are handled using an open source socket library, portable between Windows and Linux. All the exchanges are done in plain ASCII, obeying to a very simple protocol. The first character of each message identifies its purpose. The principal messages exchanged are:

- Dim, sent by the first client to configure parameters on the server;
- F, sent by clients to ask the server for their first starting points;
- R, used by the client to register a solution;
- P, used to register a solution and ask for a new point.

The server computer runs an infinite loop where new points and the corresponding function values are recorded. Each iteration deals with one client, receiving its data, computing and sending a new starting point. Each client computer also runs an infinite loop, receiving points, running the optimization and sending back results. The Table 2 depicts the algorithms.

TABLE II
SERVER AND CLIENT ALGORITHMS.

Server	*Client*
Loop wait for message if "Dim", accept config messages or report already done if "F", provide new point if "R", register data if "P", register data and provide new point	send "Dim" message if server not configured, send information about the problem, like search space bounds send "F" and receive new point Loop run optimization send "R" with starting point send "P" with end point receive new point

The server runs only one thread. Each incoming call is handled as a new connection, so that clients may quit – or crash – and reconnect at any time. When a client is started, the user must provide the IP address of the server.

The software was written in ANSI C++ and the user is required to provide only the essential code, related to the optimization problem. This is done by writing only two functions in C:

1. `int search_space_dimension (void)`: returns the dimension of the space;
2. `double objective_function (double*)`: receives the address of an array containing the coordinates of a point and returns the function value.

The code that hadles the communications and the optimization algorithms are packed in a library which is linked to the two user functions. This simplifies the whole process of project setup, compilation and linking.

The server records an ASCII file with the set of points evaluated and the respective function values; this allows the implementation of a procedure to restart a previous optimization session. The software allows the user to limit the number of optimization cycles and the running time. In principle the only practical limit is the server memory, where are stored the points evaluated (the set $\mathbf{X}$). The space needed (in bytes) is given by $|\mathbf{X}| \times (n+1) \times 8$, where $|\cdot|$ stands for the cardinality of the set. As an example, 8 Mbytes can hold more than 5000 points in a space of dimension 200; this corresponds to 5 machines computing one optimization per minute for 16 hours.

V. Numerical Tests

Three optimization problems were selected to evaluate the implementation: the minimization of Griewank and Rosenbrock functions; and the training of a physically constrained neural network.

The Griewank function is a well known test case for optimization; it is given by

$$f_G(\mathbf{x}) = \frac{1}{400n}\sum_{j=1}^{n} x_j^2 - \prod_{j=1}^{n}\cos\left(\frac{x_j}{\sqrt{j}}\right)$$

and it has a minimum at $\mathbf{x}$ = <0,0, ... ,0>. The dimension of the function was chosen as 30 and the vertices of the hypercube that delimits the search space were given the values $x_{j=1..n}$ = ± 3.0. The presence of multiple minima makes this function very hard to minimize, even with a moderate dimension and a restricted search space.

Since the grid based globalization uses the center of the hypercube as starting point, the global minimum would be immediately found. To avoid this the minimum was translated to $\mathbf{x}$ = <1,1, ... ,1>.

The Rosenbrock function is given by

$$f_R(\mathbf{x}) = \sum_{i=1}^{N-1}[(1-x_i)^2 + 100(x_{i+1} - x_i^2)^2]$$

and it has a minimum at $\mathbf{x}$ = <1,1, ... ,1>. The same space dimension and limit vectors were used.

The third optimization problem consisted of the trainning of a neural network to simulate the heat equation,

$$\frac{\partial u}{\partial t} = \frac{\partial^2 u}{\partial x^2} \quad ,$$

under the initial and boundary conditions

$$\begin{aligned} &u(x,0) = 1 \qquad 0 < x < 1 \\ &\left.\begin{aligned} u(0,t) = 0 \\ u(1,t) = 0 \end{aligned}\right\} \quad t \geq 0 \end{aligned} .$$

Under the physically constrained approach, the analytical model is used as constrains in the neural network trainning equations [20]. The optimization problem in this case is to minimize f_H given by

$$f_H = \int_0^1\int_0^1 \left(\frac{\partial h}{\partial t} - \frac{\partial^2 h}{\partial x^2}\right)^2 dxdt \ ,$$

where h(x,t) is given by $(x^2\text{-}x)\cdot\varphi(x,t)$, and φ is a one-hidden layer perceptron. The term $(x^2\text{-}x)$ accounts for the boundary condition and facilitates the search.

The tests were conducted in a small Ethernet network with a server and five client machines, Pentium 4, 2.8GHz, all with 512 MBytes of RAM and running Windows 2000.

The optimization algorithms on the clients were configured to stop after 10000 iterations or when the function decrease was less then 10^{-7}. The server was programmed to stop after 1800 seconds. In all tests, the parameter K for the *pdf* was made equal to the problem dimension.

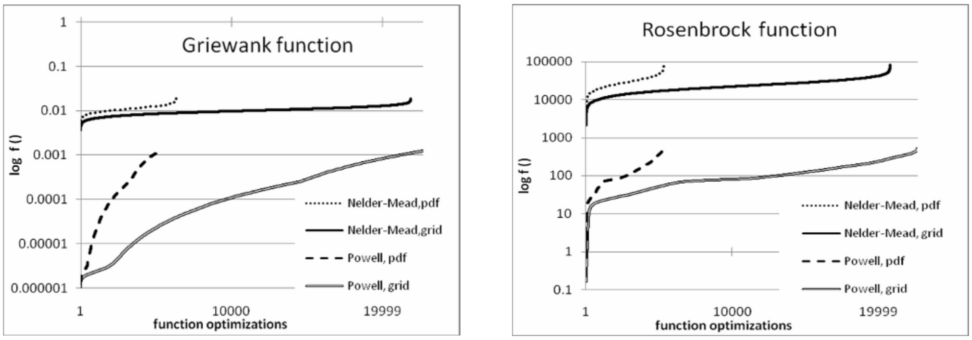
The results for the Griewank function appear in Table III.

TABLE III
GRIEWANK FUNCTION OPTIMIZATION.

Method	*Sampling*	*Optimizations*	*Best value*
Nelder-Mead	pdf	6359	0.004558747
	grid	21904	0.00367421
Powell	pdf	5261	1.26330×10^{-6}
	grid	22702	1.00537×10^{-6}

The results confirm that the grid-based sampling is faster than the *pdf*-based one, allowing a greater number of optimizations in the same processing time. To illustrate this point, the last optimization cycle using the *pdf* sampling with the Nelder-Mead approach required the generation of $K \times n \times N$= 30× 30× 6358 = 5 722 200 random coordinates. The total number of iterations executed by the server is equal to $KnN^2/2$ and the cumulative effect of clients waiting for this process is the increase of idle CPU time over the whole network. While reducing the value of K may ameliorate these figures, this also negatively affects the quality of the sampling, which is an essential characteristic for the global search.

The Powell method performed better than Nelder-Mead in the tests; however, interpreting the results requires some care, since they depend on several factors.



(A) (B)

Fig. 2. Comparison of the optimization algorithms and sampling methods for the same processing time.

Two parameters of the Nelder-Mead method affect its performance: the size of the initial simplex and the stopping criteria. The length of the vertices of the initial polytope were set empirically to 0.1. The best value for this parameter depends on the shape of the function being minimized. The number of iterations must, in principle, be greater for the Nelder-Mead algorithm due to the slower convergence rate. The interested reader may find a detailed analysis in [21].

The Powell method showed a better relation between CPU effort and results obtained. Besides computing the best result in each test, the algorithm also generated a better set of optimum points. This is illustrated for the Griewank function in Fig. 2-A, using a logarithmic scale; the curve represents all the function values found in each experiment. It indicates that for a given number of function evaluations, the Powell method tend to be more effective.

The grid based sampling proved to be effective, reducing the idle time and maximising the use of computing resources. This effect is more important in those cases where the evaluation time of the cost function is comparable to the evaluation time of the *pdf*.

The results for the Rosenbrock function appear in Table IV and are illustrated in Fig. 2-B. In this case, one of the solutions found with the Nelder-Mead algorithm and the grid based sampling was not included in Table IV. The result was produced by the optimization of the first candidate generated with the grid, which is the point located at the center of the grid ($\mathbf{x} = \mathbf{0}$). It was considered not representative of the capability of the method to explore the function shape, despite its value ($f_R(\mathbf{x}^*) = 28.99$).

TABLE IV
ROSENBROCK FUNCTION OPTIMIZATION.

Method	*Sampling*	*Optimizations*	*Best value*
Nelder-Mead	pdf	5330	4914.454259
	grid	20797	2115.296047
Powell	pdf	5271	0.228975496
	grid	22674	0.165871658

The test with the Rosenbrock showed again a better performance of the Powell algorithm and of the grid-based sampling.

The coordinates of the best point of f_R were close to the global minimum and illustrate the difficulty of the optimization in a 30-dimensional space. The coordinates were:

0.994802095 0.993429902 0.993381884 0.999555591
1.002428176 0.996894309 0.996108027 0.992484929
0.992834763 0.992399527 0.997715555 0.995275
0.993499928 0.994664936 0.998413607 0.992770923
0.993074047 0.990389948 0.991416271 0.997486168
0.999584865 0.996557238 0.991751864 0.991598001
0.994436143 0.986518829 0.971530571 0.939781392
0.885739381 0.794033868.

For the third optimization test the neural network utilised was a one-hidden layer perceptron with 25 neurons, for a total of 101 optimization parameters. The dimension of the problem makes this a very difficult case.

The software was configured as follows:

- the search space was sampled with the grid-based approach;
- the stop criteria for the optimization algorithms was set to 5×10^4 iterations or a minimum function improvement below 10^{-7};
- the total calculation time was limited to 3 hours.

The same network with six computers was used. The test was run using the Powell optimization method. The best value obtained was 0.0662 and the worst value was 38.0438 from a total of 1034 optimization runs. In order to fix the evaluation time of the cost function f_H, it was used a trapezoidal rule with a 20×20 mesh.

CONCLUSIONS

Global search methods are powerful tools, but require significative amounts of CPU workload to be effective. In order to deal with this problem, a numerical implementation can use at least two strategies.

The first one is the parallelization, a tendency of the computer industry clearly illustrated by the recent advent of multi-core CPUs. Computer networks constitute a stock of computational resources and are widely available in Universities and Research Institutes. However, the use of parallel optimization solutions is rather rare; the most likely explanation to this paradox is the complexity of programming network applications based on libraries like CORBA.

The second point to be addressed is the effectiveness of the globalization method. Even if a computer network may invite a brute-force approach, a careful design of algorithms and data structures may enhance performances in a dramatical way. The comparison between the two sampling approaches used in this article illustrates this point.

The solution presented here opens up some opportunities for further improvement.

The cost associated with the *pdf*-sampling may be reduced by selecting more than one point from **R** at each cycle. The interest on the *pdf* is the possibility of tuning parameters like the gaussian spread and altering the behaviour of the sampling, in ways not possible with the grid-based approach.

A weighting scheme could be coupled with the sampling procedure, so that regions with higher function values would be less densely explored. The mathematical formulation of this solution and the evaluation of its feasibility for particular cases as non-Lipschitz functions remain to be studied.

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Software Development Methodology of the Book Retrieval Robot

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***Abstract*-The robot programming and software development methodology for the cognition and computational intelligence of the book retrieval robot is presented in this paper. The book retrieval robot is a prototype, Linux-based, autonomous mobile robot developed to assist humans retrieve a specific book from a bookshelf. A generic software development methodology related to the control architecture and computational intelligence of the developed robot is presented utilizing the object oriented approach to robot programming.**

I. Introduction

Notably a key design component involved in the production of an intelligent robot is the software development for the artificial and computational intelligence of the robot. Artificial Intelligence is the inter disciplinary field of computer science and software engineering dealing with applications such as speech and handwriting recognition, game playing, natural language understanding and as related to the concept discussed in this paper, robot programming [1]. In this respect, the intelligence of a robot is the means of operationalizing the behavior of a robot and its design and development encompasses multi-faceted robot programming considerations ranging from system and software design and architecture, inductive logic, reasoning methods machine learning and deductive problem solving [2]. Once it has been programmed, the reasoning methods, intelligent behavior, sensing and decision making and learning and adaptation capabilities of the robot come into play. As it is, these functionalities are essential requisites of the cognitive functioning of a robot if it is to do any useful task [3]. Conversely, a robot which is not programmed is just functionless hardware.

With robots having a significant role in modern industrial and manufacturing systems, robot programming for intelligence remains an important design phase for the flawless operation and utilization of the robot. Moreover, at times the demanding nature of the application, with respect to issues such as parallel or distributed execution, multiple and retroactive split-second decision making tasks the robot is subjected to, has the implication that the software development for the robot cognition and intelligence is a greater feat compared to hardware design and development. Moreover, the ever increasing ubiquity of computer integrated robotic devices such as domestic, office and industrial service robots with embedded software also emphasizes the importance of software engineering for the overall development of robots.

In addition, a commercially developed robot is usually engineered to meet some requirements that are most often attained in the software rather than in the hardware design. This may include features such as cross platform deployment which refers to operational compatibility on multiple operating systems, device driver development, and parallel or distributed processing. It is because of these reasons that the software engineering for artificial intelligence remains the frontier of the design phases of the entire development lifecycle of a robot. Discussions in literature of advancements in robotics hardware imply the need for an equal literature discussion in the techniques and advancements in the field of robot programming. Key issues such as system architecture design are some examples of the challenging applications that have to be attained in software.

In retrospect and within the context of mobile robotics and related subjects, the major focus of attention and discussion of most literature works reporting the development of mobile robots has been the hardware implementation [4] [5]. This paper conversely presents a comprehensive overview of the software design and implementation for the artificial intelligence of a robot to demonstrate the process of programming a robot for real time automation. The genre of robot on which this paper is based is a service robot, more specifically an automatic book retrieval robot. However it is noteworthy that the process presented is in fact a generic model for robot development and may be extrapolated for use for any object retrieval. This therefore has direct relevance in industry where retrieval of objects is executed on a stupendous scale and is henceforth usually automated.

Section II provides an overview of the book retrieval robot while the software design and implementation is presented in Section III. Concluding remarks and some experimental results are given in Sections IV.

II. The Book Retrieval Robot

The book retrieval robot is an autonomous robot prototype intended to eliminate the human component in the book retrieval process by autonomously navigating to and retrieving a specific book from a bookshelf [6]. A salient feature of this robot is the employment of a novel book detection and identification technique, bar coding technology, in the formulation of a time-optimal and computationally

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elementary method in contrast to vision-based techniques that have been used by researchers in similar work [7][8]. A synopsis of the operation of the robot is given next.

The robot is in principle a mobile differentially-driven base encompassing a gripper and custom designed book retrieval mechanisms. The principal controller of the robot is a laptop computer onboard the robot running on the Linux operating system, Red Hat distribution of version 9.0. The user enters the call number of a desired book through the laptop and the robot navigates to the appropriate shelf and begins searching for the target book.

The laptop interfaces to the robot hardware through the Enhanced Parallel Port (EPP). A custom-made bidirectional, software selectable, input and output EPP interfacing card provides the laptop-robot interfacing medium. All sensors, transducers and other electronic circuits are interfaced to either the input or output port of the parallel port interfacing card. A microcontroller board, acting as a secondary controller is also interfaced to the laptop through the interfacing card.

The control electronics encompasses direct current (dc) motor controller circuits, infra red sensors for obstacle detection, line tracer sensors for navigation, force sensors for the provision of force dexterity, limit switches and the respective interfacing circuits. All inputs to these circuits are Transistor to Transistor Logic (TTL) levels. In simpler terms, the inputs to the circuits are simply either a logic high (1 or 5V) or a logic low (0 or ground). This is very easily generated in software and this is what makes the software-hardware integration straightforward.

The robot motion is controlled by the laptop parallel port using a 2-bit code sent to the motor controller circuits via the microcontroller board, via the parallel port interfacing card. All other motor controller circuits for the actuation of the various mechanical systems are also controlled in a similar fashion, by virtue of the fact that the rotational direction of a dc motor may be controlled by changing the polarity of the applied voltage. All sensory information from the transducers are interfaced to the input port of the parallel port interfacing card and the instant a high signal is received from the sensor, the corresponding event handler for the event executes the appropriate behavior of the robot. A barcode scanner provides the input data for the robot to realize the identities of books while a force sensor determines the minimum force with which the gripper suffices to grip the book without slip. The barcode scanner scans the barcode of books to match the scanned book call number with the user entered book call number and an affirmative match results in a positive identification and the robot consequently executes a systematic chain of mechanical book retrieval maneuvers. Conversely, a negative match implies the robot persists with its existing state of scanning. More details of the robot operation in a book retrieval operation are given in [9] and [10].

The following sections provide an exposition of the software development methodology for the artificial intelligence of the book retrieval robot described above.

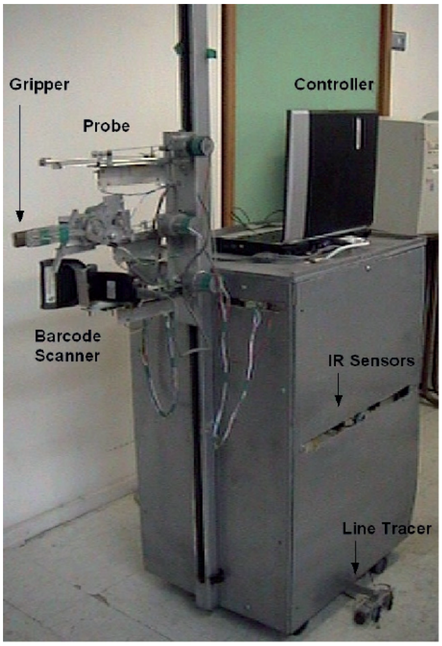

Fig. 1. The book retrieval robot

Programming was done in the C++ programming language in Linux platform and a strict objective of the software development was to utilize codes that were part of standard C++ libraries for cross platform compatibility. The exhibit of Fig. 1 illustrates the book retrieval robot.

III. Software Design Paradigm And Nomenclature

Notably embedded systems designs encompass concurrent and iterative designs of hardware and software. Therefore the incremental model [11] may be the best model for creating the software for a hardware embedded application. This model characterizes the iterative nature of development where initial iterations attain basic functionality and subsequent iterations implement more advanced functionalities.

The overall software developed for the book retrieval robot is in principle an object oriented program and is a three-tiered application in its entirety. The traditional and common method of robot programming is referred to as functional programming embodying a behavior-based system where the different behaviors of the robot are represented as user defined functions in software. Invocation of a function from the main program supplying the correct input arguments, if any, then attains the corresponding behavior. However recent years have seen an emerging paradigm shift towards object oriented programming methodology [12].

An object-oriented methodology was therefore used for the programming to attain the overall objective of object-oriented programming, namely modularity and reusability. In addition object oriented programming also provides abstraction and

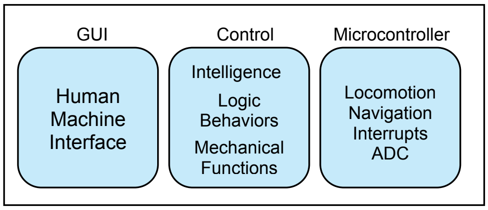

Fig. 2. Nomenclature of robot software

more details on its implementation are given in later sections. There are three major parts of the software. First is the graphical interface part, the second the control part and the third the microcontroller software. The block diagram depicting the software for the book retrieval robot is given in Fig. 2.

The user interface program is solely responsible for receiving user input, specifically the desired book call number. The control software contains the intelligence and logic behaviors of the book retrieval maneuvers while robot locomotion and interrupt events are controlled by the micro controller software. These three separate programs are then linked together to work in unison and should there be any changes carried out to either of the three programs, it will have minimum effect on the functionality of the overall software as any change is prevented from being cascaded throughout the overall software, by virtue of the fact that there are three separate components.

A. Human Machine Interface

The human-machine interface (HMI) is the point of physical interaction between the human operator and the robot [13]. The HMI of an application can assume any form ranging from a relatively simple graphical user interface to a highly sophisticated instrumentation control panel.

The design of a HMI can be easily performed by carrying out a needs-analysis followed by the implementation of the desired interface. For the book retrieval robot, the only input parameter required is the desired book call number. By implication, a very simple graphical user interface (GUI) with a suitable text box for text input and clickable button for submission and program execution commencement suffices. For the GUI development Kylix software was used.

Kylix® is a visual component-based Rapid Application Development (RAD), Integrated Development Environment (IDE). Kylix offers visual component based development featuring a Visual Components Library (VCL) consisting of a toolbox that encompasses user-interface components with which interfaces may be built at design time. An additional merit of using Kylix is that it boasts cross platform compatibility through the use of the Cross Platform Toolbox (CLX). Use of the CLX toolbox guarantees inter-platform deployment. The GUI developed for the book retrieval robot simply features two visual components: a text box for the user input and a button to submit the desired call number and start activation of the mechanical book retrieval maneuver. There is one event handler for the event when the user has entered the desired book call number and clicked the button. The event handler stores the user entered book call number as a string in a text file, generates and transmits a 1-bit signal (a logic high) through the parallel port to a specified pin of the microcontroller, hereafter referred to as the motion pin. Configured as an interrupt pin, a logic high signal on the motion pin instantaneously triggers the robot locomotion by supplying the necessary signals to the wheel motor controller circuits. Activation of the robot motion causes the robot to trace the navigation routes which lead the robot up to and along the bookshelf. As the robot moves along the bookshelf, the barcode scanner scans the barcodes on the book spines.

B. Barcode Scanner Programming

Absolutely imperative in the process of book identification, the programming of the barcode scanner is a fairly straightforward process as explained in this section. Barcode scanners are commonly available as serial devices, interfaced to the serial port of the personal computer, USB devices or alternatively as PS2 devices with a PS2 male connector to be interfaced to the PS2 female port of the computer. The latter is utilized here.

The PS2 port is a communications specification for input devices originally developed by International Business Machines (IBM). It is noteworthy that the programming of a barcode scanner with a PS2 connector shares a high degree of commonality with the programming of a keyboard with a PS2 type connector, as essentially, only the data written at the PS2 port needs to be accessed by the computer program irrespective of whether the device is a keyboard or barcode scanner. By implication, any software that is designed to read in data from a keyboard also reads the data that a barcode scanner returns. Thus a simple program using standard C++ routines was written to read in data from a keyboard and it was then extrapolated to read in the data of the barcode scanner. A more technical explanation is given next.

The software developed for the barcode scanner is fundamentally a half duplex, interrupt driven software implementation of PS2 communications. There are a number of routines of retrieving data from the PS2 port, however these routines were either platform-specific or machine-dependant and thus they were ignored as cross platform compatible and hardware independent software was desired.

Fortunately there exists a high-level method of retrieving the scan codes and working out the corresponding key that does not require low-level implementation considerations of either platform or hardware. Reception of a byte written at the PS2 port by the barcode reader has been performed by using the `cin` object defined in the `iostream.h` header. The `cin` is an input stream object with its definition and operations specified in the file `iostream.h` and is conventionally used to read keyboard input but the net effect of its usage is that reads the data at the PS2 port which is generated by some input device; in this case the input device is a barcode scanner and a read operation is just as possible, as essentially, the barcode reader emulates a keyboard. Being part of standard C++ language, it guarantees that the code used to retrieve data from the PS2 port will be platform-independent and thus compile under either Windows or Linux platform while simultaneously having the merit of being machine-independent. Thus it can be seen that the software retrieval of data scanned by the barcode reader uses elementary programming knowledge and hence does not constitute computation overhead as is the case with using computer vision for book identification as done by [4] and [5]. When the experimental results of [4] and [5] are considered, it is noted that it takes a minimum of 6 seconds and a maximum of 11 seconds (1 second to localize a label from an image and an additional 5-10 seconds for the image processing and optical character recognition phase) to scan and realize the identity of books using vision-based techniques. With this magnitude of duration, the scanning and localization of an entire bookshelf will no doubt incur a much higher processing time and render the robot inefficient, compared to the time it takes a human to identify a book. Conversely, from experimentation results, it takes the barcode scanner an average time of 0.45 seconds to scan and realize the identity of a book. The barcode scanner software was developed using the Linux GNU C++ compiler g++ and it is part of the main control software.

Barcodes for providing digital signatures to books have been printed using free trial edition software called TbarCode5 ActiveX that takes alphanumeric strings, converts them to their respective scan codes and creates the corresponding barcodes with a combination of bars and spaces of varying width. Installation of this software embeds an ActiveX component into MS-Word®. With this, a user simply enters an alphanumeric string of arbitrary length into a text box; the barcode ActiveX component then creates the corresponding barcodes. The barcodes can then be printed using any standard printer. There are different formats of barcode encoding available but with a multiple decoder barcode scanner, the software retrieval of the data can be treated in oblivion to the barcode formats.

C. Microcontroller

The secondary hub of operations and intelligence of the book retrieval robot is a PICmicro® microcontroller, specifically the Microchip Technology PIC16F877 eight bit CMOS microcontroller with built in EPROM. The microcontroller is available as a 40-pin DIP package containing a central processor, EPROM, RAM, timer(s), and TTL / CMOS compatible or user defined input and output lines [14]. The use of a microcontroller as a secondary controller also provides a method of distributed computing.

This microcontroller is responsible for coordinating and receiving sensory information from the force sensors, line tracers and IR sensors and in addition it also controls the wheel motors of the book retrieval robot. The decision to allocate the PIC microcontroller to control the line tracers and IR sensors instead of the onboard computer was made based on the fact that these entities directly determine the actions of the wheel motors. Since the microcontroller controls the wheel motors, it made sense for it to also control the line tracers and IR sensors. In addition, the microcontroller offers use of interrupts for the reading of inputs.

Interrupts provide immediate detection of a change of state in input. In this way, whenever there is a change in the state of the IR sensors or line tracer sensors, the software immediately controls the wheel motor appropriately. Therefore certain pins of the microcontroller boot loader board were configured as interrupt pins using PIC C Compiler software and the IR and line tracer sensors were interfaced to these interrupt pins. Therefore whenever the book retrieval robot veers off the guide path or whenever the IR sensors detect an obstacle, the software immediately changes the actions of the wheel motors appropriately. In a similar fashion, whenever the laptop computer generates a low signal as a halt command on the motion pin, particularly in the event the comparison of the barcode scanner output with the user entered book call number returns a true result, the software immediately detects this and thus halts the motor. Interrupts offer the merit of high performance computing which may not be attainable using polling. Henceforth it guarantees the robot will stop precisely in front of the target book. Results derived from experimentation also augment this connation; recorded results indicate trivial deviations in the braking distances of the robot once it detects a target book.

Another supplementary benefit of using the PIC167877 microcontroller is that offers an inbuilt, 8-bit, self clocking and successive approximation Analog to Digital Converter (ADC) port which is capable of receiving and processing analog signals. This was also the reason for its selection as a secondary controller as had the onboard laptop computer been used, the purchase of an ADC IC would have been mandatory.

The ADC port of the PIC reads the output voltage of the force sensors, coverts it to an alias eight (8) bit numerical quantity and henceforth makes processing more feasible. The eight (8) bit digital representation can then be used with elementary comparison and decision structures to determine the correct magnitude and application of the forces that suffice to retrieve (pry) and grip a book.

D. Overall Robot Intelligence

The byte read from the barcode scanner is then compared to the book call number that was previously entered by a user and stored in a buffer. If the comparison returns a positive result, it indicates a match and the software immediately generates a code via the parallel port to a secondary controller (which controls the motor controller circuit) to halt its movement. Following this the robot executes a systematic chain of mechanical maneuvers that retrieves the subject book out of the shelf. Conversely if the comparison returns a negative result, the scanned book call number differs from the desired book call number and the robot proceeds with its scanning motion until it attains a match.

Therefore by simple decision making and invocations of mechanical functions, the book retrieval robot has identified and retrieved the target book. All behaviors (mechanical actions) of the robot are represented as functions and all attributes are represented as data variables. Logically related functions and data are then grouped together into one structure and have been implemented in a class which implies Object Oriented (OO) methodology with the functions and variables becoming the methods and variables of the class, respectively. For further exemplification of object orientated robot programming, consider the operation of the gripper of the book retrieval robot. There are four (4) behaviors or actions of the gripper:

(i) Gripper fingers opening
(ii) Gripper fingers closing
(iii) Gripper moving forward
(iv) Gripper retracting back

These four mechanical actions or behaviors of the gripper have corresponding C++ functions written for them:

```
void Gripper_FMotorCW();
void Gripper_FMotorACW();
void Gripper_MotorCW();
void Gripper_MotorACW();
```

It is easily observed that these four functions are logically related to one another, by the virtue of the fact that they all govern the operation of the same device, the gripper. Hence a class Gripper is then declared and these four functions now become the methods of this class. The code segment is given:

```
class Gripper {

   private :
     // Attributes (data)
     // In this case, there are none.

   public:
     // Methods of the Gripper class
     void Gripper_FMotorCW();
     void Gripper_FMotorACW();
     void Gripper_MotorCW();
     void Gripper_MotorACW();
} ;
```

The invocations of any of these functions then first require the instantiation of the Gripper class. For example, to close the gripper fingers, the following lines of code would be used.

```
Gripper gripper;
gripper.Gripper_FMotorCW();
```

This is how functions have been grouped into classes signifying object-oriented methodology. A listing of all the principal classes and corresponding methods used for the operation of the book retrieval robot in the retrieval of books is given in Table 1. Invocations of these functions (which correspond to mechanical actions) in their correct sequences lets the robot pry the detected book out of a shelf. A detailed picturesque illustration and narrative description of the book retrieval operation is given in [6].

TABLE I
LISTING OF SOFTWARE FUNCTIONS OF BOOK RETRIEVAL ROBOT

Class : : Function	Description
Robot::Forward()	Robot base moves forward
Robot::Right()	Robot base turns right
Robot::Left()	Robot base turns left
Robot::Back()	Robot base reverses
Robot::Halt()	Robot base stops
Barcode::BCode_MotorCW()	Move back/retract barcode reader
Barcode::BCode_MotorACW()	Extend barcode reader
Gripper::Gripper_MotorCW()	Retract gripper back
Gripper::Gripper_MotorACW()	Extend gripper forwards
Gripper::Gripper_FMotorCW()	Open gripper fingers
Gripper::Gripper_FMotorACW()	Close gripper fingers
Probe::Probe_MotorCW()	Extend Probe into bookshelf
Probe::Probe_MotorACW()	Retract Probe from bookshelf
Rack::Rack_MotorCW()	Move Rack upwards
Rack::Rack_MotorACW()	Move Rack downwards

The final step in the software is to link the GUI software with the control software such that when the user clicks the button on the GUI, the control software is launched or executed. For this remote and automatic execution, the control software had to be first configured as a daemon process.

The daemon function, when applied to particular software, detaches the software from the console terminal and lets it run in the background as a system daemon. Since the control software was written using the console and g++ compiler, this was a necessary step before an attempt could be made in remotely launching or executing the control software from the GUI software. Once it is a daemon, the control software can then be invoked from the GUI software using the system command. This enables the linking of the two programs.

IV. CONCLUSION

The software design paradigm and development methodology using an object-oriented approach for the intelligence of a service robot has been presented. The process involved is the culmination of key tasks covering human-machine interface development for the user interface, barcode reader and microcontroller logic programming. It was demonstrated how a seemingly complex task, the automatic retrieval of library book using an autonomous robot, can be attained using elementary programming with a barcode scanner for book localization. The software architecture was developed using the C++ programming language, is based on the Linux platform and is cross platform compatible. The object oriented methodology allows for encapsulation and abstraction thereby implying reuse.

From current experimental results, the robot is capable of correctly detecting and identifying books with 99% accuracy at a speed of 0.6m/s. It takes an average time of 0.45 seconds for the robot to detect, scan and realize the identity of a single book. These results verify the efficacy of the software developed.

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An Approach to Cadastre Map Quality Evaluation

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***Abstract*- An approach to data quality evaluation is proposed, which is elaborated and implemented by State Land Service of the Republic of Latvia. The approach is based on opinion of Land Service experts about Cadastre map quality that depends on its usage points. Quality parameters of Cadastre map objects identified by experts and its limit values are used for evaluation. The assessment matrix is used, which allow to define Cadastre map quality that depends on its usage purpose. The matrix is used to find out, of what quality a Cadastre map should be in order to be used for the chosen purpose. The given approach is flexible, it gives a possibility to change sets of quality parameters and their limit values as well as to use the approach for other type data quality evaluation.**

I. Introduction

Scientific literature identifies several aspects of quality: data quality has several components such as accuracy, relevance, timeliness, completeness, trust, accessibility, precision, consistency, etc. [1], [2]. There are currently two main research streams, which are addressing the problem of ensuring a high level of data and information quality. One is a technical, database-oriented approach, while the second is a management and business-oriented approach. Engineering of information system brings both streams together and addresses issues related to the design and modeling of information systems [3].

This research deals with identification of the parameters important to geographical data quality for specific goals. Geographical data are data describing an object's spatial location and various properties. Conceptually a geographical database (GDB) may be thought of as consisting of two databases (DB) - one being a common attribute DB and the other is a coordinate DB describing the objects' global locations and dimensions. High quality geographical data will include space location and object properties at given times (where-what-when) [4].

Data quality is the degree to which data meet the specific needs of specific customer. Note that one customer may find data to be of high quality (for one use of the data), while another finds the same data to be of low quality (for another use) [5]. What features do experts* working with geographical data use to judge the quality of data? The authors are not aware of any published studies in this area to date. This paper present an approach to the evaluation of the quality of Cadastre map that caters for the differing levels of quality required of various parameters in order to meet different goals.

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∗ An expert is an employee of organization who works with geographical data, e.g., inputs information, draws maps, supervises maps, etc.

The subjective assessments of experts in geographical data processing are sought to determine the factors which impact most upon the quality of geographical data. When these assessments are evaluated, freed of subjective elements and classified, it becomes possible to specify parameters for the evaluation of data quality, their values and the required levels of quality. The result of this is a matrix for quality assessment which can be used to determine the data quality level that is necessary for specific purposes or, alternatively, the specific goals for which data at a specific level of quality may be used.

This paper describes the method that is to be taken in preparing the quality assessment matrix and how this approach is used for Cadastre map evaluation in State Land Service (SLS) of the Republic of Latvia.

II. An Approach To Data Quality Evaluation

The discussion of quality must begin with identification of the objects of interest. Every object will have a number of quality parameters (QP1, QP2, etc.) (Fig.1.). Each quality parameter QPn has values taken from one or more sets of values QPnVSk (TABLE 1), where QPnVS1 may contain the best values. QPnVS2 contains the second best values for some particular goal, etc. [6]

The quality of the object is based upon several or all quality parameters. For instance, an object can belong to the highest level of quality if all of the estimated values of the relevant quality parameters belong to the best sets of values. It belongs to the second level of quality if the values of the relevant quality parameters belong to the second best sets of values, etc.

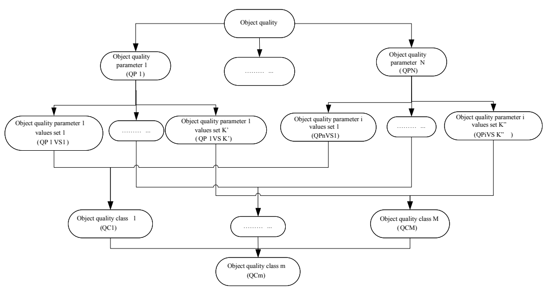

Fig. 1. Object quality, parameter and value and object quality class

TABLE 1
QUALITY PARAMETER VALUE SET

Quality parameter (QP)	Quality parameter value set (QPVS)			
	QPnVS1 (high)	QPnVS2	...	QPn VS K (low)
QPn	from-until	from-until	...	from-until

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 105–110.

TABLE 2
QUALITY ASSESSMENT MATRIX

Object quality class (QC)	Quality parameter/ Quality parameter value set			
	QP1	QP2	...	QP N
QC1 (high)	QP1VS1	QP2VS1	...	QP N VS1
QC2	QP1VS2	QP2VS2	...	QP N VS2
...	...	...	...	...
QC M (low)	QP1VS K’	QP2VS K”	...	QP N VS K’”

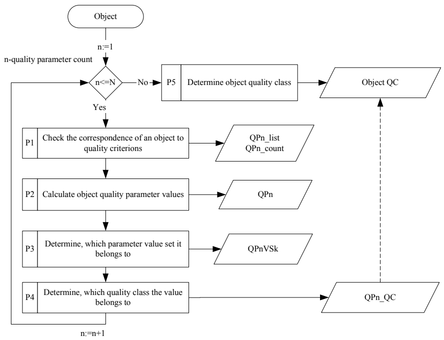

Fig. 2. Object quality class

As a result the object quality assessment matrix (TABLE 2) is obtained, which is used to determine, to which quality class the object belongs, as well as to determine, which should be quality parameter values so that the object would correspond to the chosen aim of use.

Quality parameter quality class (QP_QC) depends on a quality parameter value set, to which belongs the quality parameter value (1).

$$QPn\_QC = 1, \text{ if } QPn \in QPnVS1;\ 2, \text{ if } QPn \in QPnVS2, \ldots,\ M, \text{ if } QPn \in QPnVS\ K,\ n=\{1...N\},\ k=\{1...K\} \qquad (1)$$

In its turn, object corresponds to the lowest quality parameter quality class (2).

$$QC = \text{lowest}\ (QP1\_QC, QP2\_QC, \ldots, QPN\_QC) \qquad (2)$$

The aim of object quality evaluation is to determine, which quality class the object belongs to and which aims it can be used for. In order to evaluate an object (Fig. 2.): first, check the correspondence of an object to quality criterions, obtain the list or the number of items not corresponding to the quality criterions, second, calculate object quality parameter values, obtain QPn, third, determine, which parameter value set (TABLE 1) it belongs to, obtain QPnVSk, fourth, determine, which quality class the value belongs to (TABLE 2), obtain quality parameter class QPn_QC and fifth, determine object quality class (2), obtain object QC .

This approach is implemented in SLS of the Republic of Latvia for Cadastre map evaluation and is based on the defined by field experts quality parameters, which describe the usage purpose of a certain Cadastre map.

III. CADASTRE MAP QUALITY EVALUATION IN THE REPUBLIC OF LATVIA

In the Republic of Latvia, Cadastre map (CM) is created in Latvian coordinate system LKS-92 in Transverse Mercator (TM) projection. Following elements are represented in CM: land parcels -boundaries of parcels and their Cadastre designations; buildings - outlines of buildings and their Cadastre designations; encumbrances - areas occupied by encumbrances of right to use real property and their designations; parts of land parcels- leaseholds and their Cadastre designations; boundaries of Cadastre territories and Cadastre groups. The CM is used to locate Cadastre objects with precision so that any changes in boundaries for administrative or other purposes may be accurately described and to describe the relationships between objects for the purposes of environmental and town planning and for various reports. The principles and content of the CM are established by Regulation, which is an ordinance of the SLS of Latvia. The Cadastre IS databases consist of two parts, the textual part (TP) and the graphical part, which includes the CM in vector graphics form [7].

CM quality depends on the quality of each object, whereof the CM is made. CM can consist of such objects land parcel, building, encumbrance and part of land parcel. Therefore, in order to evaluate CM quality, firstly, it is necessary to evaluate qualities of land parcel, building, encumbrance and part of land parcel – wherewith the approach described above (Fig. 1.) has to be applied for each CM object.

A. Cadastre map objects quality parameters, value sets and quality classes

In this article an approach to CM quality evaluation is proposed, which is based on experts’ opinions about CM quality that depends on its usage points. Expert opinions are obtained from more than 50 expert interview surveys. Having summarized the results of surveys, such quality criterions are obtained: the CM meets the legal regulation requirements, CM objects are topologically correct, coordinates of CM land parcels are precise, CM objects (land parcels, building, encumbrance and part of land parcel) are in both Cadastre databases and the data is the same – in the TP and in the CM. Quality criterions are given in TABLE 3

Experts’ opinions about CM quality are subjective and therefore have to be structured and, according to normative acts and existing IT solutions in SLS, we obtain Cadastre object quality parameters (QPn) (Fig. 1.) – for land parcel (LP) 5 quality parameters are defined (LP_QPn, n=1…5), for building (BD) – 4 quality parameters (BD_QPn, n=1…4), for encumbrance (EB) – 2 quality parameters (EB_QPn, n=1…2), for part of land parcel (PLP)– 3 quality parameters (PLP_QPn, n=1…3) (TABLE 4).

In collaboration with experts and in the result of experiments, sets of quality parameter values are defined.

TABLE 3
CADASTRE MAP QUALITY CRITERIA

Code	Title
C1	CM meet the legal regulation requirements
C2	CM objects are topologically correct
C3	Coordinates of CM land parcels are precise
C4	Object data in the TP and the CM are identical:
C4.1	A Cadastre object (land parcels, building, encumbrance and part of land parcel) has to be in both Cadastre databases – in the TP and in the CM:
C4.1.1	the object marked in a CM has to be in the TP
C4.1.2	the object in a TP has to be marked in the CM
C4.2	Cadastre object data in both Cadastre databases:
C4.2.1	the surveying type for land parcel has to be the same in CM and TP
C4.2.2	cadastre surveyed land parcels' and parts of land parcels' legal area (indicated in the documents) and area defined by graphical methods (marked in the Cadastre map, further in the text – geographical area) cannot be larger or smaller than the acceptable space difference defined in the Regulations
C4.2.3	a building, in both databases, has to be attached to one and the same land parcel

There are three sets of values for all the parameters: excellent, good and bad values. In Fig. 1. QPnVSk, k=1...3.

Parameter values of excellent quality are such ones, which describe that an object meets quality criterions, values of good quality are such ones, which do not overrun the defined acceptable error rate, but values of bad quality are such ones, which overrun the defined rate (TABLE 5). Parameter value of excellent quality to any quality parameter (except for land parcels and buildings) is 0%, but to the surveyed land parcels and buildings – 100%. Value of good quality to any quality parameter (except for land parcels and buildings) is from 0.01% to 5%, but to the surveyed land parcels and buildings – from 99.99% to 10%. Value of bad quality to any quality parameter (except for land parcels and buildings) is from 5.01% to 100%, but to the surveyed land parcels and buildings– from 9.99% to 0%.

Theoretically, object quality parameters and sets of values can be chosen in thousands of variants, but practically, suitable is only such variant, where parameters are defined by field experts that depends on what object (in this case – a CM) will be used for.

LP_QP1, LP_QP2, BD_QP1, BD_QP2, EB_QP1, EB_QP2, PLP_QP1 and PLP_QP2 characterize Cadastre map objects completeness in Cadastre IS TP and CM databases.

LP_QP3 and BD_QP3 characterizes consistency between TP and CM. LP_QP3 characterize land parcels survey type consistency between in TP and CM. BD_QP3 characterizes building land parcel attachment consistency between TP and CM (in both databases the building has to be attached to one and the same parcel).

LP_QP4 and PLP_QP3 characterize trusted of area. LP_QP4 characterize trusted land parcels area. In accordance with the Regulations for CM, the graphical area of a surveyed land parcel listed in the CM (which is calculated on the basis of coordinates) can possibly differ from the legal area of the land parcel shown in the TP (which is declared in legal documents) but within prescribed limits. The admissible level of variation is determined by Regulation. PLP_QP3 characterize trusted part of land parcels area. The purpose of this parameter is the same as that of quality parameter LP_QP4.

TABLE 4
CADASTRE MAP QUALITY PARAMETERS

Code	Description	Value High – low	Quality criteria
LP_QP1	Describes how much (%) of CM land parcels are missing in the TP	0%-100%	C4.1.1
LP_QP2	Describes how much (%) of TP land parcels are not marked in the CM	0%-100%	C4.1.2
LP_QP3	Describes how much (%) of CM land parcels surveying type differs from TP surveying type	0%-100%	C4.2.1
LP_QP4	Describes, how much (%) of CM Cadastre surveyed land parcels' geographical area is larger or smaller than the acceptable space difference of TP legal area	0%-100%	C4.2.2
LP_QP5	Describes, how much (%) of CM land parcels are Cadastre surveyed	100%-0%	C3
BD_QP1	Describes how much (%) of CM buildings are missing in the TP	0%-100%	C4.1.1
BD_QP2	Describes how much (%) of TP buildings are not marked in the CM	0%-100%	C4.1.2
BD_QP3	Describes how much (%) of CM buildings have different land parcel cadastre designation in TP, to which the building is attached	0%-100%	C4.2.3
BD_QP4	Describes, how much (%) of CM buildings are cadastrally surveyed	100%-0%	C3
EB_QP1	Describes how much (%) of CM encumbrances are missing in the TP	0%-100%	C4.1.1
EB_QP2	Describes how much (%) of TP encumbrances are not marked in the CM	0%-100%	C4.1.2
PLP_QP1	Describes how much (%) of CM parts of land parcels are missing in the TP	0%-100%	C4.1.1
PLP_QP2	Describes how much (%) of TP parts of land parcels are not marked in the CM	0%-100%	C4.1.2
PLP_QP3	Describes, how much (%) of CM cadastral surveyed parts of land parcels' geographical area is larger or smaller than the acceptable space difference of textual part legal area	0%-100%	C4.2.2

LP_QP5 and BD_QP4 characterize accuracy of co-ordinate. LP_QP5 characterize accuracy of land parcels co-ordinate. The database which includes the graphic component of the Cadastre register includes graphic data to various levels of accuracy. The database of land parcels includes data at three different levels of data accuracy – surveyed land parcels, allocated land parcels, and designed land parcels. The coordinates of the surveyed land parcels are obtained by surveying the relevant parcel with the appropriate instruments. Coordinates of allocated land parcels may have been obtained with older measuring instruments that are no longer in use (field compasses, tape measures), or through conversion from other co-ordinate systems which differ from the specified LKS-92 TM coordinate system. The coordinates of designed land parcels are approximate, because they are usually obtained from ortophoto maps, photo plans or other materials. These coordinates are not based on direct land measurement. BD_QP4 characterize accuracy of building co-ordinate. The database which includes the graphic component of the Cadastre register includes graphic data to various levels of accuracy. The database of building includes data at three different levels of data accuracy – surveyed building, stereo vectorized building, and vektorized building. The coordinates of the surveyed building are obtained by surveying with the appropriate instruments. A stereo vectorized building contour is marked by a stereo tool, but a vectorized building – by scanned material, the building is not surveyed.

Taking into account the purpose of a CM and collaborating with experts, three quality classes of objects are defined (TABLE 6): high, medium and low. In Fig. 1 QCm, m=1…3.

B. Quality assessment matrix

Having summarized quality parameter sets of values and quality classes, an object quality assessment matrix (TABLE 7) is obtained. According to quality parameter values, object quality is: High (QC1), if quality parameter value is excellent – appertains to the set of values QPnVS1. Medium (QC2), if quality parameter value is good – appertains to the set of values QPnVS2. Low (QC3), if quality parameter value is bad – appertains to the set of values QPnVS3.

The main principle of using the quality evaluation matrix – an object corresponds to its quality class, which the worst quality parameter value belongs to.

For example, land parcel quality class 'LP_QC' (Fig. 3.) depends on the lowest quality parameter quality class (3).

$$LP\_QC = MAX(LP\_QPn\_QC) \quad (3)$$

Land parcel quality parameter quality class (4) acquire using the sets of values for quality parameters (TABLE 5) and the quality assessment matrix (TABLE 7)

$$LP\_QPn\_QC = 1, \text{ if } LP\_QPn \in QPnVS1;\ 2, \text{ if } LP\_QPn \in QPnVS2;\ 3, \text{ if } LP\_QPn \in QPnVS3,\ n=\{1..5\} \quad (4)$$

In its turn, quality parameter LP_QPn, n=1...5 is calculated according to define formulas.

Building, encumbrance and part of land parcel quality classes are evaluated in a similar way as the quality of land parcels.

TABLE 5
CADASTRE MAP QUALITY PARAMETERS VALUES SETS

Quality parameters	Quality parameter values sets		
	QPiVS1 excellent	QPiVS2 good	QPiVS3 bad
- LP_QP1, LP_QP2, LP_QP3, LP_QP4, - BD_QP1,BD_QP2, BD_QP3, - EB_QP1, EB_QP2, - PLP_QP1,PLP_QP2, PLP_QP3	0%	0.01-5.00%	5.01-100%
- LP_QP5, - BD_QP4	100%	99.99%-10%	9.99%-0%

TABLE 6
CADASTRE MAP QUALITY CLASSES

Quality class		Description
High	1st quality class (QC1)	A CM can be used for making decisions and other activities, where information from the CM is needed
Medium	2nd quality class (QC2)	A CM can be used for making decisions, but it is necessary to be sure about quality of a certain object, which is used for making the decision
Low	3rd quality class (QC3)	A CM cannot be used for making decisions, it can be used to get primary information

TABLE 7
CADASTRE MAP OBJECT QUALITY ASSESSMENT MATRIX

Object quality class		Quality parameters value set
High	1st quality class (QC1)	QPnVS1
Medium	2nd quality class (QC2)	QPnVS2
Low	3rd quality class (QC3)	QPnVS3

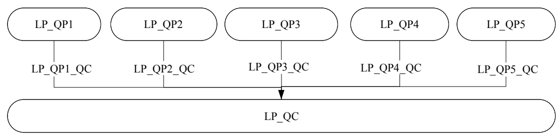

Fig.3. Land parcel quality class

IV. CASE STUDY OF CADASTRE MAP QUALITY ASSESSMENT

Let's evaluate quality of land parcels in the chosen CM (Fig. 4.). We have: five quality parameters for land parcels LP_QPn, n=1…5 (TABLE 4), three sets of values for quality parameters LP_QPn_VSk, n=1…5, k=1…3 (TABLE 5) and three land parcel quality classes – high, medium, low LP_QCm, m=1…3 (TABLE 7), CM and TP data, which are given in TABLE 8.

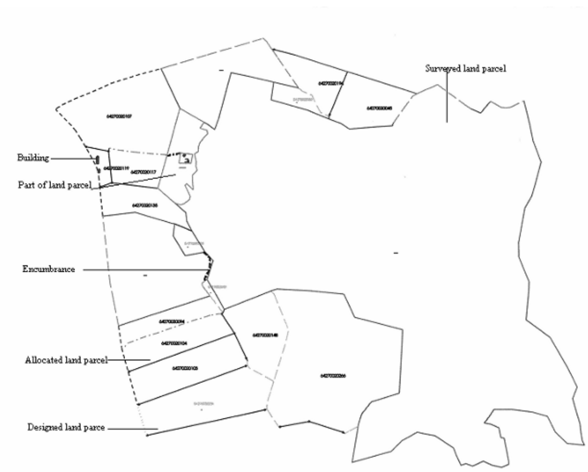

Fig.4. Detail from the *Durbe* country Cadastre map

TABLE 8
DURBE LAND PARCELS CM AND TP DATA

CM				TP			
Nr	Cadastre number of land parcel	Survey type	Graphical land area m2	Nr	Cadastre number of land parcel	Survey type	Legal land area m2
1	64270020045	allocated	73349	1	64270020045	allocated	82000
2	64270020094	allocated	43925	2	64270020094	allocated	51000
3	64270020103	allocated	91950	3	64270020103	allocated	91000
4	64270020104	allocated	65236	4	64270020104	allocated	59000
5	64270020107	allocated	163022	5	64270020107	allocated	158000
6	64270020117	allocated	40520	6	64270020117	allocated	38000
7	64270020119	allocated	12563	7	64270020119	allocated	15000
8	64270020135	allocated	54089	8	64270020135	allocated	64000
9	64270020146	surveyed	192035	9	64270020146	surveyed	192100
10	64270020148	allocated	81174	10	64270020148	allocated	82000
11	64270020151	surveyed	121532	11	64270020151	surveyed	121600
12	64270020189	designed	19453	12	64270020189	designed	18000
13	64270020190	designed	12905	13	64270020190	designed	13000
14	64270020191	designed	4411	14	64270020191	designed	4000
15	64270020194	allocated	49874	15	64270020194	allocated	53000
16	64270020200	surveyed	2114825	16	64270020200	surveyed	2115500
17	64270020251	designed	119254	17	64270020251	designed	119000
18	64270020266	allocated	322332	18	64270020266	allocated	320000
19	64270020317	surveyed	2690	19	64270020317	surveyed	2700

TABLE 9
DURBE BUILDING CM AND TP DATA

CM				TP		
Nr	Cadastre number of building	Survey type	Cadastre number of land parcel	Nr	Cadastre number of building	Cadastre number of land parcel
1	64270020119001	Vectorized	64270020119	1	64270020119001	64270020119
2	64270020119002	Vectorized	64270020119	2	64270020119002	64270020119
3	64270020119003	Vectorized	64270020119	3	64270020119003	64270020119
4	64270020119004	Vectorized	64270020119	4	64270020119004	64270020119
5	64270020195001	Vectorized	64270020317	5	64270020195001	64270020317
6	64270020195002	Vectorized	64270020317	6	64270020195002	64270020317
7	64270020195003	Vectorized	64270020317	7	64270020195003	64270020317

TABLE 10
DURBE ENCUMBRANCE CM AND TP DATA

CM			TP		
Nr	Cadastre number of land parcel	Encumbrance code	Nr	Cadastre number of land parcel	Encumbrance code
1	64270020200	050301 001	1	64270020200	050301 001
2	64270020146	050301 003	2	64270020146	050301 003

TABLE 11
DURBE PART OF LAND PARCEL CM AND TP DATA

CM			TP		
Nr	Cadastre number of part of land parcel	Graphical land area m2	Nr	Cadastre number of part of land parcel	Legal land area m2
1	642700202008001	58766	1	642700202008001	55800

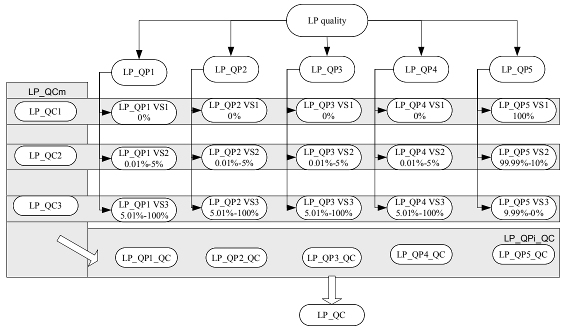

Fig.5. Land parcel quality evaluation

Evaluation of a land parcel consists of the following steps: 1st step – acquire the number of land parcels in the chosen CM the number of CM land parcels is 19, CM_LP_count=19. Also in the TP the number of land parcels for the chosen region is 19, TP_LP_count=19. 2nd step - acquire how many land parcels do not comply with the proposed criterions, the result is 'LP_QPn_count' or 'LP_QPn_list', n=1...5. Then calculate LP_QPn, how many percents it is and using the sets of values for quality parameters (TABLE 5) and the quality assessment matrix (TABLE 7), acquire quality parameter quality class LP_QPn_QC, n=1...5. Finally, get LP_QC (Fig. 5.).

For example, LP_QP1_QC acquisition (Fig. 6.) a)check, how many land parcels are not in the TP (TABLE 8). After the check let us make sure that all land parcels in the CM are also in the TP, therefore LP_QP1_count=0, b) calculate the rate LP_QP1=LP_QP1_count/CM_LP_count*100= =0/19*100=0%. Using the TABLE 5 we see that LP_QP1 value appertains to the set of values LP_QP1_VS1 and using the TABLE 7, the value corresponds to the High class LP_QC1, we acquire that LP_QP1_QC=1.

Qualities of the other land parcel quality parameters class are evaluated in a similar way as the quality of LP_QP1_QC. Qualities classes are given in TABLE 12.

Finally**,** land parcel quality depends on the lowest quality class in every quality parameter: LP_QC=MAX (LP_QPi_QC), i=1…5 (Fig. 3.) and it is Medium class LP_QC=2 - CM (taking into account land parcel quality only), it is permitted to use it for making decisions (TABLE 12), by making sure that land units, which were not surveyed, do not influence the decision. However, if CM usage purpose is not connected with it or land parcels are surveyed (do not take into account LP_QP5, therefore it is not a necessary requirement to be surveyed), then quality of the CM is already High class – LP_QC=1.

If CM usage purpose is connected with involvement of all the objects, then it is necessary to evaluate quality of the other objects. Quality of the other objects is evaluated in a similar way as the quality of land parcels. Quality evaluation of all the objects is given in TABLE 12.

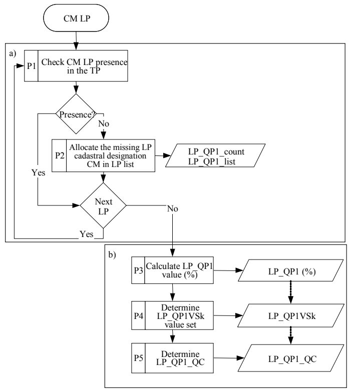

Fig.6. Quality parameter LP_QP1 quality class

TABLE 12
CM *DURBE* QUALITY CLASSES

Land parcel		Building		Encumbrance		Part of land parcel	
LP_QPi_QC		BD_QPi_QC		EB_QPi_QC		PLP_QPi_QC	
LP_QP1_QC	1	BD_QP1_QC	1	EB_QP1_QC	1	PLP_QP1_QC	1
LP_QP2_QC	1	BD_QP2_QC	1	EB_QP2_QC	1	PLP_QP2_QC	1
LP_QP3_QC	1	BD_QP3_QC	1			PLP_QP3_QC	1
LP_QP4_QC	1	BD_QP4_QC	3				
LP_QP5_QC	2						
LP_QC	2	BD_QC	3	EB_QC	1	PLP_QC	1

Evaluation for the chosen CM is acquired taking into account the lowest quality class of each object: CM_QC= MAX(LP_QC, BD_QC, EB_QC, PLP_QC).

As a result we obtain that quality class of the given CM (taking into account quality of all the objects) is the Low class – CM_QC=3 and it cannot be used for making decisions, it can be used to get primary information.

The evaluation method is based on object usage purpose and first, if CM usage purpose does not depend on whether a building is surveyed (quality parameter BD_QP4 is not taken into account), then CM quality is of Medium class – CM_QC=2 and it can be used for making decisions, second, if CM usage purpose does not depend on survey of land parcels and buildings (quality parameters LP_QP5 and BD_QP4 are not taken into account), then quality class is High class - CM_QC=1 and the CM and be used for any purpose.

V. CONCLUSION

The described approach can be applied to any CM. Quality assessments can be obtained not only for CM of small territories but also for big areas, e.g., cities, regions. The example given in this paper is an assessment of a portion of the Latvian country *Durbe* and reveals where the weaknesses of the map may be.

The insights gained from this analysis are varied. For example, lists of land parcels for which data quality is poor and where data quality needs to be improved in order to useful for given purposes. In particular, approximate calculations can be done estimate the time and financial commitment required to bring a CM to a desired quality. For example, to carry out border adjustments in particular territories.

The elaborated method can be used for quality evaluation of any type objects and the main steps of the methods are: firstly, from experiments obtain subjective opinion about object quality descriptive parameters, which value depends on object usage purpose. Secondly, perform structuring of expert subjective opinion and define object quality parameters and their values, according to object binding normative documents and existing IT solutions in company. Thirdly, together with experiments define object quality classes depending on object usage purposes and what quality parameter values create each quality class, consequently, obtain object quality evaluation matrix, which is used to evaluate the use of an object for the chosen purpose.

In order to make everyday use of a Cadastre map easy and simple, support software (Data Quality Evaluation Software - DQES) is elaborated for calculating values of quality parameters and for quality class determination as well as for obtaining charts to analyses data and to elaborate a plan for improving data quality. If without DQES data quality evaluation of one regional unit (i.e. *South Kurzeme* regional unit) required 2-3 days, now the needed time is 1-2 hours. DQES data quality evaluation algorithms tested in practice can be used for supplementing in existing IT solution of SLS.

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XBRL in Public Administration as a Way to Evince and Scale the Use of Information

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Abstract - This study aimed at demonstrating why public organizations need to standardize and flexibilize their way to release accounting information to provide administrators with improved quality, timeliness, reliability, and, above all, a reduction in their costs to evince financial information. In this sense, we analyzed a standard formatting language called e*Xtensible Business Report Language* (XBRL), with the purpose of verifying its applicability to evince and scale information in public administration. It is qualitatively confirmed that XBRL is a formatting language capable of standardizing and flexibilizing information for its users, and also of providing a series of management analyses when combined with *Business Intelligence*.

I. INTRODUCTION.

The public sector has its limitations in providing information for citizens and regulatory agencies, which causes duplicate and unnecessary work, due to different sources of data and to the time consumed of accounting and management personnel. Based on this reflection, we studied the formatting language XBRL (*eXtensible Business Report Language*), with the objective of verifying how it could be used to scale and evince accounting information in the public sector. To support this proposal, we analyzed the case of the Municipality of Coqueiros do Sul, in the State of Rio Grande do Sul, which adopts a **S**pecific **S**ocial **S**ecurity **R**egime (**SSSR**, -Regime Próprio de Previdência Social -, in Portuguese), and has to distribute information to several agencies and entities to comply with legal provisions concerning transparency and accountability.

To achieve our objective, we analyzed the legislation that regulates the public sector activity to identify which are the reports and information that must be frequently released by municipalities, and what characterizes accountability, transparency, and possibility of scaling information in public administration, through the *Business Intelligence* (BI) system.

II. EXTENSIBLE BUSINESS REPORT LANGUAGE (XBRL).

XBRL is a standard formatting language to release financial statements on the Internet. It is also a free tool to standardize, flexibilize, and make released information more transparent and broadly accessible to all users, either for managerial or legal purposes (MOREIRA, 2005 and IFAC, 2002). Ward (2004) says that **IFAC** (International Federation of Accountants) considers XBRL as an ideal tool to make information easier, faster, more transparent and accessible. Thus, financial information would also be more reliable along the whole information chain in organizations.

For Ray (2001, p. 3), a markup language can be defined as a "set of symbols that can be included in the text of a document to mark and label its parts". It is an evolution of the markup languages produced by studies carried out in the 1960s, and its main differential feature is that data are independent and standardized in all formats (DEBRECENY and GRAY, 2004), which allows programs to interact. Consequently, users can obtain the information they need without having to ask for customized or re-typed reports. According to Riccio, Sakata and Moreira (2005), XBRL has a significant presence in the world, mainly in Germany, Australia, Belgium, Canada, Korea, Denmark, Spain, United States, France, Netherlands, England, Ireland, Japan, New Zealand, and Sweden.

Lack of standardization in information published on the Internet makes it difficult for users to access and interact with it. This is the reason why many have mentioned XBRL as one of the alternatives to standardize and flexibilize information, and to reduce its cost. XBRL has already been approved by major regulatory agencies, so it is enough to promote it and check how it will be accepted by different countries and accountable agencies, as well as by the public companies and agencies that release their information. To prove this assertion, we cite Ward (2004), who states that IFAC's mission is to serve the public interest. For this reason,

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 111–115.

this author supports the use of XBRL, it serves the public interest because it is market-oriented and makes more accurate and rapid information available for decision making, besides reducing the associated costs.

This is possible because the XBRL was conceived with a structure that gives flexibility to the information. Its structure and functionality are based on the Inter-relationship of three components: the Taxonomy, *the Instance Document* and *the Style Sheet*, being that this last one is considered a complement of the two first ones.

- *The Taxonomy*: basically is the study of the principles of scientific classification. According to MOREIRA 2005, p. 28, taxonomy is "as a dictionary, built from its structure of hierarchies, to supply standard definitions to the information of the financial reports".
- *The Instance Document:* in a definitive moment it will be informed to the application which are the value these elements possess. From the reading and interpretation of these two elements (taxonomy and *instance document*), the XBRL application will generate, in a XML file format, the information about the financial reports, MOREIRA (2005, p. 26).
- *Style Sheet:* basically is a sheet that summarizes editorial guidelines relating to and used when preparing text for publication. According to MOREIRA 2005, p. 27, style sheet makes possible to format the visualization of the reports results from arising from the two first components. Its function allows to transform the information for any desired format, such as the files format: XML, PDF, HTML, printed etc.

III. TRANSPARENCY IN PUBLIC ADMINISTRATION AND THE EXCESS OF REPORTS.

The **F**ederal **C**ouncil of **A**ccountants (**FCC**, - Conselho Federal de Contabilidade-, in Portuguese) (2003) points out that accountants are made publicly liable when they are requested to release accounting statements that will inform decision making by several information users, such as the society – represented by taxpayers –, the Department of Justice, the **S**ecretariat of the **N**ational **T**reasury (**SNT**, Secretaria do Tesouro Nacional – in Portuguese), the government of a State, the Federal Ministries, the **N**ational **I**nstitute of **S**ocial **S**ecurity (**NISS**, - Instituto Nacional de Seguridade Social, - in Portuguese), etc.

To ensure transparent and effectively controlled public activities and a reliable rendering of accounts, several laws were implemented to set standards for transparency, among them we point out: (1) Federal Law 9.755/98, also called Public Accounts Law; (2) Supplementary Law 101/2000, known as **F**iscal **R**esponsibility **L**aw (**FRL**, - Lei de Responsabilidade Fiscal -, in Portuguese). Both make it mandatory the publication of a series of reports and fiscal statements in media for broad public access, including the Internet. Moreover, the FRL pursues an accountable fiscal management, supported by Federal Law 10.028/2000, which alters the Brazilian Penal Code and defines the crimes against public finances, and the punishment for such crimes and for the infringement of the FRL.

The Public Accounts Law disposes of the creation of homepages on the Internet and affects all agencies and entities belonging to the three spheres of government, with the purpose of making available on their websites data and information related to public accounts.

As for the Specific Social Security Regime (SSSR), it represents a social security fund, with or without its own legal entity. It applies, exclusively, to incumbent civil servants of the Union, the States, the Federal District, and the Municipalities, active and inactive, and pensioners. The SSSR must guarantee to civil servants, at least, the benefits of retirement and pension, as established in the article 40 of the Federal Constitution; its implementation depends on a law that disposes explicitly of such benefits (LIMA, 2005). According to this author, the **S**ecretariat of **S**ocial **S**ecurity (**SSS**, - Secretaria da Previdência Social-, in Portuguese), an agency of the Ministry of Social Security, is responsible for formulating the social security policy, for supervising the programs and actions of associated entities, and for proposing general rules to organize and maintain the SSSR. For this reason, it needs bimonthly reports with information to follow up the collection and correct investment of funds regularly.

The FRL defined as instruments for transparency in fiscal management:

- The **S**ummary **B**udget **E**xecution **R**eport (**SBER**, - Relatório Resumido da Execução-, in Portuguese), established by article 52, which must be released within thirty days after the end of each bimonthly period.
- The **F**iscal **M**anagement **R**eport (**FMR**, -Relatório de Gestão Fiscal-, in Portuguese), established by article 54, which must be released within thirty days after the end of each four-month period, or each six-month period for municipalities with less than 50,000 inhabitants.

Prado and Loureiro (2005) point out that the statements required by the FRL and the Public Accounts Law, concerning information quality and intelligibility, must follow a pattern already determined in the mentioned legislation. They also state that the information required by both laws is repeated over and over, that is, the information is basically the same, but report formats and titles are different. Aware of this fact, we think that it is worth emphasizing the usefulness of XBRL to release information on public accounts, aiming at complying with both laws and providing the transparency they require. Based on the concepts presented previously, we understand that XBRL is flexible enough to allow users to retrieve the desired information from the database.

Analyzing the content of the laws mentioned, we concluded that there is an excess of reports and information to be released by municipal accounting departments, aiming at complying with several laws and rules established by several different entities, which are eventually looking for the same information, and this is being released in different formats. The SNT and the **A**udit **C**ourt of the **S**tate of Rio Grande do Sul (**ACS**/RS, -Tribunal de Contas do Estado do Rio Grande do Sul-, in Portuguese) require the same reports (SBER and FMR), at the same intervals, but in different formats and from different databases. Therefore, reports have to be re-typed, which consumes time and resources that could be allocated to other tasks. Using XBRL, both entities would be able to retrieve the desired reports and find the information they need without requiring rework from municipal accountants, allowing them to use their time for managerial accounting.

In the municipal sphere, a mayor must render consolidated accounts (from the executive, the legislative, and the indirect administration) to the ACS, which will give its opinion and publicize it widely. Besides the ACS, the system of internal control and the municipal legislative body are also responsible for inspecting the municipality's budget, finance, and property management, directly or with the aid of the ACS.

4. XBRL AND INFORMATION IN PUBLIC ADMINISTRATION

XBRL fits perfectly with the needs of the public sector, since today Brazilian municipalities such as Coqueiros do Sul are required to release their financial statements; for example, the SBER bimonthly, and the FMR every six months. These reports must be sent to the SNT, the ACS, the municipal legislative, published in electronic media, newspapers etc. As a consequence, several reports containing the same information are sent to several users, but in different formats, without following standard language and contents. This requires retyping, conference, and reconfiguration of the integrated management system, generating unnecessary costs and consuming time.

By adopting the concepts recommended in this study, public accountants would release accounting information in XBRL, and the user would be able to retrieve what he wanted in the desired format, without reworks, since data are independent of the application in which they are created, allowing multiple users to retrieve information directly from the Internet and format it according to their needs. For Moreira (2005), XBRL – through the three elements that compose it: taxonomy, instance document and style sheet – allows information to be retrieved by other programs that will process it later or, even, configure it to be viewed in any desired format (PDF, XLS, DOC, HTML and others).

Analyzing the previous information and relating it to the public administration context, we perceive that XBRL is custom-made to suit the needs of the public service. This instrument is appropriate to release information in compliance with legal requirements and to fulfill the economicity principle in the public service, since there will be considerable time and effort savings in the collection, formatting, and consolidation of data to generate reports for different entities.

With the use of XBRL, information will be released based on the accounting system employed by the municipality and will be retrieved from the datawarehouse in a predefined format, observing the mandatory taxonomy. Through XBRL style sheet, the information becomes independent and every user can import it and convert it into the most appropriate reading format (PDF, Word, Excel etc.), which saves time and reduce costs considerably in the activities of analysts. Later, it is also possible to gather information from other public entities and make comparisons. And, to make things even easier, it is possible to use BI for economic and financial analyses. Ideally, in according to Balloni (2006), could also be possible utilizing the BAM – Business Activity Monitoring -, a tool that provides real time access to the critical indicators of any business performance. In broader level, the BAM proposal deals, in real time, with the operational convergence from BI and Enterprise Application Integration (EAI), guided for the business objective.

Another issue that deserves to be highlighted is that all users of public information, such as SNT, ACS, NISS and the States, must also forward or distribute the information they obtained, besides releasing analyses of such information or creating and publishing new information for other users. Consequently, XBRL will be used again in the same circumstances, and this is possible only because any software can use it, through applications that import and export data automatically. The figure 1 presents the process for evidencing the accounting information with the XBRL. Based on Watson (2005).

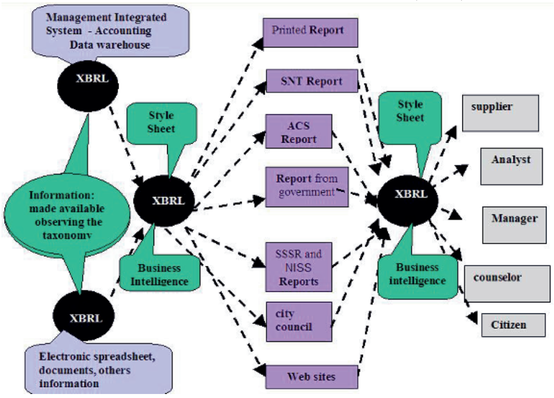

Figure 1 – shows as the use of the XBRL is important in the public area. This language makes possible a direct interaction with the information between the diverse supervisory agencies and users. This prevents waste of time, resources and a better dynamics between supervisors and users regarding the information itself.

Therefore, what must be pointed out is the efficiency and ease in transmitting information by electronic means, which bring to an end incessant reworks to: 1) compile a report for the ACS; 2) recompile the same report as it is required by the SNT; 3) release a series of reports on the Internet, opting for one of these two (ACS or SNT), which results in the same information in different formats and even with different values; 4) forward financial information from SSSR to NISS; 5) send reports to the municipal legislative etc. We emphasize that each phase of the retyping process can invite errors that distort the information. With XBRL, information will be accurate, since it will be released only once, allowing much more time for conferences and analyses to confirm its reliability and to eliminate reworks. Thus, XBRL can reduce the cost of information, accelerate and improve its flow, and all people involved in this information chain will be able to carry out interpretations and comparisons easily, as a result of its flexibility.

Moreover, the article 51 of the FRL determines that the Union promotes, nationally and in each sphere of government, the consolidation of the accounts of public entities related to the previous fiscal period and publishes it, even in electronic means for broad public access, up to June 30. For this purpose, States and Municipalities must send accounts consolidated with SSSR to the Union, up to May 30 and April 30, respectively.

As we can observe, in all these aspects, the accurate, reliable, and agile information is the basis for compliance with legal provisions and for efficiency in the administration. In this new scenario imposed by legislators, information must cross several spheres of government, serving as a basis for new reports and statements that seek to inform the society of its administrators' performance.

5. PLANNED AND EFFICIENT ACTION BASED ON THE SCALABILITY OF INFORMATION.

The enactment of the FRL has culminated in the need to introduce a new form of public administration that leads to efficiency, efficacy, productivity, and that translate into effective and concrete results and benefits for the society. In this sense, it must also observe the principle of economicity, which aims to reduce public costs continually. As a way of motivating public administrators to change their old habits of governing, the Federal Law 10.028 was formulated. It altered the Brazilian Penal Code and defined the crimes against public finances, so that now it is possible to render accountable those administrators who make decisions against legal principles and to punish them.

In this context, it is necessary to develop methods to evaluate administrators' performance and control it more strictly to measure the efficiency of public administration. Thus, one of the fundamentals of public administration is the use of available information about administrators' performance to develop an evaluation process based on the comparison of a public entity with different entities in different places.

Government accounting has had to proceed from a legalist accounting to one turned to organization management. It also had to stop concerning solely with complying with legal rules to start concerning with the generation of information for decision making, aiming at the efficiency, efficacy, and economicity of the public service, and at managerial actions that encourage results that place an entity within the limits and norms of the FRL.

This scenario will definitely require the use of XBRL, which will be essential to produce timely, accurate and cost-effective information for decision making, with the objective of providing for opportune inspection and a national consolidation to be carried out by the SNT. However, understanding how to make information available to administrators should necessarily include adequacy to their ways of making decisions. This is the reason why information must also be scalable, that is, final users should be able to structure their own reports or query screens by means of different searches provided by the datawarehouse (DW), a large non volatile (i.e., that can be stored) collection of data organized by subject, integrated and parameterized by date, to support decision making processes (INMON, 1997). The DW can be manipulated by BI systems in which the administrator finds relationships that are not offered by traditional information management systems and their extremely structured reports.

6. CONCLUSIONS.

Many professionals, mainly in the public sector, need to turn away from releasing the same information in different formats and for different purposes, because it simply means reworks that can be avoided. XBRL serves to ease this inconvenient situation, since its use allows for the release of information on the Internet only once. After that, users would be able to extract information in the format they want, according to their needs. Therefore, in management terms, information becomes more reliable, timely and flexible. Users can, through XBRL, interact with information in any format (Word, Excel, PDF etc.), as well as use BI to perform financial and economic analyses of one or several institutions.

This interaction will bring a considerable reduction of costs to produce and distribute information, because it avoids wasting time and resources in repetitions. It will provide a reduction in the structures of production of information or their better use in management, where XBRL allows for a true "dive" in the database and the possibility of extracting, analyzing, and comparing information. The trend is that XBRL will be supported and accepted by corporations and regulatory agencies in most countries, since at the moment it seems to be one of the best solutions for problems related to the lack of standardization in the information they have to release. However, it is worth considering the exception made by Hannon (2006), that one of our concerns should be the risk for corporations of raising their costs above the benefits obtained from this migration to XBRL. For this reason, the regulatory agencies of the public sector have to conduct new research to formulate taxonomy for an effective use of XBRL in Brazilian public organizations, and also to study different ways of using it in management.

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Information Security in S&M Brazilian Companies

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Abstract - The Managing Committee for Internet Use in Brazil has verified a migration of the target of attacks from the larger companies to individuals and small and medium (S&M) sized companies, largely due to a lack of security in relation to Internet access. Most of the S&M companies have a total lack of a culture of security. They tend to have problems with employees accessing Internet sites with inappropriate content matter, as well as the unauthorized on-job utilization of instant communicators. This means that it has become relatively easier and more efficient to attack residential and small business users directly. True digital inclusion must be based on an education consistent with the safe and adequate use of computational resources, not just their availability: this will contemplate both the existing networked companies and entrepreneurial networks. Governmental organs are taking initiatives to facilitate the acquisition of informational products by smaller companies and the population in general, but educational campaigns are also necessary so that such resources will truly bring about an economic and social evolution. ***This work deals with these concerns and intends to instigate a reflection about the evolvement of society information processing and its related securities aspects (the motivation of why security is important in such a context) as well as on the negative social impact on S&M companies.***

I. INTRODUCTION.

1.2 - An synopsis on Small and Medium Enterprises in Brazil.

The process of globalization of the markets is bringing constant changes in the sectors economic, technological, cultural, social and enterprise. This globalization narrowed the relation between companies and contributed to increase the competition in international level, forcing the Brazilian companies to adapt to a new standards of production and commercialization and moving its relationship with consumers: these became more demanding and conscientious of the organizations roles [1]. Also, the development of the Information Technology changed the market relations (digital networks) and promoted modifications in the productive process, placing the information and entrepreneurial network (*social organizations offering different types of resources to start or improve entrepreneurial projects. Having adequate human resources is a key factor for entrepreneurial achievements. Combined with leadership, the entrepreneurial network is an indispensable kind of social network not only necessary to properly run the business or project, but also to differentiate the business from similar projects social organizations*), as essential in the elaboration of managemental and marketing politics for the companies [2]. In the enterprise environment, this globalization of the economy and international competitiveness is bringing about the reduction of trade barriers and hierarchic levels of the great organizations as well as the acceleration of processes of merger and acquisitions of companies[3]. These factors, allied to the higher requirement level of flexibility and agility and, as well, the necessity of rationalization of process on the part of the company, has transformed the world of the business, as for example: the increase of the productivity with less manpower (collaborative networking), and reduction the chances of job. This has led the individuals to the necessity and desire to undertake new businesses, contributing for the sprouting of Small and Medium Enterprises (S&M Enterprises) in Brazil and, as well, the recent entrepreneurial network. Nowadays, the security questions permeate these S&M companies and the respective emerging networks. This article instigates and develops this question concerned with security.

In Brazil, the S&M companies are identified as more flexible and less bureaucratized, allowing them to faster and more suitable answer to their environment. They are basic elements of the Brazilian economy and the S&M contribute with about 97% in the participation of the number of companies,, about 48% in the production, 60% of the Labor contract and about 25% of the Brazilian Gross Domestic Product [4] .

In general the Brazilian S&M companies work on the basis of the principle of "free enterprise system" (establishing the free competition: making the networking a "special case"). These S&M companies are responsible for the health of the market economy and the guarantee for a just price of the products and services. Therefore, they are directly linked on to the proper social welfare, source of jobs and renewal of the economy: they have social and economic commitment with Brazil.The factor security, either in a network of companies or individual companies (considered in this article), are extremely important.

There is a series factors, such as, economic, financial (tax burden), marketplace, production and managemental that affect the S&M companies and, therefore, these have

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difficulties in reaching and supporting, on your own, an ideal standard in R&D since they do not have an initial capital of investment, making very difficult their technological development and, therefore, the investment in the development of new technologies (securities, networking, integrated system) [5] . This corroborates for these S&M companies to do not fit their productive capacity and installations to the market requirements (they do not have working capital to finance its projects of technological innovations): investment in security deals with the core of this concern and, this article argues about the fragility that undergo the S&M companies. In section V (SMALL BRAZILIAN COMPANIES), we will present a few more details about S&M companies and its security concern.

2.2 - Securities Concern.

Over ten years of observation of more than 1300 **s**mall and **m**edium (***S&M***) sized companies has shown that one of the most important assets in the present economy is information. Independent of the size of the company, the treatment of information may make the difference between survival and failure in business dealings. Analyzing the importance that companies give to information and consequently to actions in pro of the preservation and availability of such information, we have noted a certain lack of correlation between actions and speech. Ten out of ten businessmen or executives readily affirm that they recognize the importance of information, but not all of them have adopted plans for the preservation of the confidentiality, availability and integrity of this information or, implementation of controls for keeping the business continuity, which will guarantee its availability and the maintenance of business activities when there are problems. This untenable situation, represented by knowing about the importance of information while at the same time not attributing enough resources for its preservation is extremely relevant and warrants investigation.

Since antiquity [6], man has tried to control that information which he judges to be important. In ancient China, the written language itself served as a kind of code [*secret code?*] because only the upper classes could learn to read and write. Peoples such as the Egyptians and Romans also left a historical register of their concern with the treatment of certain information, especially that with strategic and commercial value.

We have verified that nowadays many organizations dedicate much of their attention to tangible physical and financial assets, but little to informational assets We [2] have also verified that information has taken on a fundamental importance in conducting business, since this has become ever more dynamic and globalized.

This lack of concern and preparation of many executive officers and businessmen running small companies may result in material losses for these companies, as well as considerable social loss when critical information about clients is disclosed.

II. INFORMATION SECURITY.

The evolution of information technology [3] has wrought changes in social behavior while at the same time increasing ethical problems involving crime, privacy, individuality, jobs, health and work conditions.

According to [7], the concept of computer crime includes the following categories:

1. Unauthorized use, access, modification and destruction of hardware, software, data or network resources.
2. Unauthorized liberation of information
3. Unauthorized software copies
4. Negation of access to one's own hardware software, data or network resources

Constant concern [8] with the security of information exists in all sectors, from the data banks of the credit card companies to those who furnish a variety of services. Breaches of security occur with ever greater frequency, and these involve ever more varied targets and reasons, including hackers pursuing personal gain, users with criminal intentions and even corporations involved in industrial espionage, [9]. Nor are entities which are well prepared in relation to information security totally free of such problems. A Brazilian journal specialized in information security [10] brings the account of an attack on the US Marine Corps, which led to the divulgation of confidential data of some one hundred thousand employees; this information was made available for more than six months on an Internet site, invading privacy and revealing information about the financial life of those involved, since data such as full name and social security number were revealed.

III. VIRTUAL FRAUD.

Cases of virtual fraud [11.a], which include attacks which result in the negation of services, as well as Zumbi networks, viruses, worms, spyware, identity theft, social engineering and invasions, are estimated to have caused damage in the neighborhood of US$ 67.2 billion in 2005 in the United States.

In Brazil, according investigations of the Brazilian Computer Emergency Response Team, [11.b] attempts at virtual fraud registered in 2005 showed a 579% increase over those of the preceding year , with some 68 thousand incidents; although some 40% of these were unsuccessful. Some two thousand companies [11.a] were investigated; 1,300 of these (64%) had registered incidents involving information security in 2005.

IV. POSITION OF INFORMATION SECURITY IN BRAZIL.

In recent research, the Managing Committee for Internet Use in Brazil [12] has verified a migration of the target of attacks from the larger companies to individuals and smaller companies, largely due to a lack of security in relation to

Internet access. Various factors have contributed to this migration, including the fact that there has been an increase in the number of residential users and a corresponding increase in the time spent surfing the web, with the consequent exposure to attack; many of these residential users and small businessmen are not especially concerned with Internet security and limit their protection to a simple antivirus software. Moreover, attacking large corporate servers has become more difficult, especially with investments in protective software such as firewalls and intrusion detection systems.

This means that it has become relatively easier and more efficient to attack residential and small business users directly, whereas attacks aimed at corporate servers represent a smaller percentage.. Such attacks tend to involve the use of technique of social engineering, which involves contact of the user in some way and tricking him/her into revealing sensitive information or in some way exploiting his/her trust. This leads to the sabotage of the user's computer through the introduction of various types of programs, including viruses, Trojan horses and worms. Viruses and Trojan horses are the most common, accounting for 50.34% and 31.13% of the attacks, respectively. Worms are the most recent invention. They are self-repeating programs similar to a virus except that a virus infects a program and needs this host program to propagate itself, whereas a worm is a complete program by itself and does not require another program for propagation and the infection of the boot sector. Even though worms do not reproduce, they can be transmitted by means of e-mail. Once a worm is installed, the machine can be controlled remotely, and the invader can utilize it for such nefarious activities as sending Spam or blocking of sites. These mechanisms often lead to the loss of system configuration (11.2%) and unauthorized attempts at access (10.89%)

Improvement in security [12] depends on a variety of actions, especially the education of users about the threats and forms of protection available. Of the companies investigated in, 19.69% had a training program aimed at information security, and for those with more than 500 employees, this rose to over 40%.

V. SMALL BRAZILIAN COMPANIES.

The National Bank for Economic and Social Development [13] classifies companies according to size on the basis of the Form Letter no 64/02 of October 14, 2002: For the classification of small companies, the category of industry includes micro companies (for 1-10 employees), and small ones (from 20-99). In the area of commerce/services, micro companies involve 0-9 employees, whereas small ones have 10-49.

--Microcompanies: These have gross annual operational earnings of up to US$0.6 million.

--Small companies: These have gross annual operational earnings between US$0.6 million and U$ 4.6 million.

--Medium-sized companies: These have gross annual operational earnings between US$4.6 million and US$30 million.

--Large companies: These have gross annual operational earnings greater than US$30 million [13]

According [14] to the information made available by Brazilian Micro and Small Business Support Service, the number of micro companies in Brazil grew from 2.956,749 in 1996 to 4.605,607 in 2002, an accumulated growth of 55.8%. The total number of people employed by microcompanies also increased from 6,878,964 to 9,967,201, an increase of 44.9%. The total volume of salaries accounted for by these companies grew from 7.3% in 1996 to 10.3% in 2002. During the same period, the number of small companies increased from 181,115 to 274,009, an increase of 51.3%, while the number of people employed in them increased from 4.054,635 to 5.789,875, a growth of 42.8%. The total volume of salaries paid by these small companies increased from 12.8% in 1996 to 15.7% in 2002.

Taken together [14], these two kinds of small companies accounted for 99.2% of the total number of formally-constituted companies in 2002, including 57.2% of the total jobs and 26.0% of the salaries. This information confirms the importance of these small companies in the Brazilian economy.

VI. S&M BRAZILIAN COMPANIES IN RELATION TO INFORMATION SECURITY.

Information security [15] is subjected to differential treatment, depending on the local culture. Specific concerns about the protection of information depend on the kind of activity in which a company is involved, as well as its size and, even more importantly, the individual culture of the CEO or founder.

Information security refers to the preservation and integrity of information about a company in relation to three aspects:

-- Confidentiality - the firms seeks to protect its data and information from disclosure to unauthorized persons, i.e., data and information should be protected according to the need for secrecy and be made available only for those persons for whom it is intended.

-- Integrity - all of the information systems should provide an accurate representation of the physical systems that they represent, i.e., information should be maintained in the form in which it is made available by their owner and should be protected from intentional or accidental alteration

-- Availability - the purpose of the firm's information infrastructure is to make its data and information available to those who are authorized to use it, i.e., protected information should be made available whenever necessary, but only to those individuals for whom it is intended.

Large companies which have accumulated strategic information of enormous aggregate value also tend to have highly specialized personnel who are regularly made aware of the need for information security; these companies have established a culture involving the implantation and maintenance of systems of information security. The level of automation which has been achieved by the market, especially in relation to the organization of information, has made it necessary to establish a specific norm for the management of information security. Such norms originate in the [15] Brazilian norms for Information Security which has the following dominions:

- -- Security policy (*)
- --Organizational security
- --Classification and control of information assets
- --Personnel safety
- --Physical and environmental safety
- --Management of operations and communication
- --Control of access
- --Development and maintenance of systems
- --Management of business continuity
- --Conformity

(*) - Security Policy: *is a plan of action for tackling security issues, or a set of regulations for maintaining a certain level of security. It can span anything from the practices for securing a single computer, to building/ premises security, to securing the existence of an entire nation-state.*

The smaller companies, as seen in the survey reported here, were found to reveal a lack of coherence between true needs and the effective preventive actions taken for information security. For financial reasons and a lack of a wider view of what and how to protect security assets, these companies focus primarily on small investments in limited technological tools which are generally inadequate for the job. These companies have no sense of the risks involved and thus tend to have a false sense of security, giving their users/employees the feeling that they are safe during manipulation of data and information; whereas the tools actually adopted are generally inadequate and can provide only partial coverage in relation to security.

It is a fact that there is no such thing as 100% security for any sector, but inadequate or partial implementations are more problematic in relation to computer security than is the lack of a program for safety in internal access or external access to the Internet, as long as users are aware of this absence. In such a situation involving the lack of coverage, users can be oriented and trained to be careful; this is much better than propagating the unrealistic image that a company is secure. The failure to develop such training/awareness programs increases the likelihood of successful attacks.

Most of the smaller companies have a total lack of a culture of security. They tend to have problems with employees accessing Internet sites with inappropriate content matter, as well as the unauthorized on-job utilization of instant communicators (such as MSN, SKYPE, etc.); they also open e-mails of doubtful precedence.

VII. SOCIAL IMPACT.

The dynamism and rapidity with which communication has developed, especially in relation to the digital transmission of data, contrast with the slow rate of cultural evolution of society, especially of the less privileged. Much has been said about digital inclusion, but true digital inclusion must be based on an education consistent with the safe and adequate use of computational resources, not just their availability. Governmental organs are taking initiatives to facilitate the acquisition of informational products by smaller companies and the population in general, but educational campaigns are also necessary so that such resources will truly bring about an economic and social evolution, rather than introducing undesired collateral effects due to a lack of knowledge.

VIII. CONCLUSION & PERSPECTIVES.

The Managing Committee for Internet Use in Brazil has verified a migration of the target of attacks from the larger companies to individuals and small and medium (S&M) sized companies, largely due to a lack of security in relation to Internet access. Over ten years of observation of more than 1300 small and medium (*S&M*) sized companies has shown that ten out of ten businessmen or executives *readily affirm that they recognize the importance of information, but not all of them have adopted plans for the preservation of the confidentiality, availability and integrity of this information or, implementation of controls for keeping the business continuity, which will guarantee its availability and the maintenance of business activities when there are problems.* Independent of the size of the company, the treatment of information may make the difference between survival and failure in business dealings. Most of the S&M companies have a total lack of a culture of security. They tend to have problems with employees accessing Internet sites with inappropriate content matter, as well as the unauthorized on-job utilization of instant communicators. This means that it has become relatively easier and more efficient to attack residential and small business users directly. True digital inclusion must be based on an education consistent with the safe and adequate use of computational resources, not just their availability. Governmental organs are taking initiatives to facilitate the acquisition of informational products by smaller companies and the population in general, but educational campaigns are also necessary so that such resources will truly bring about an economic and social evolution. As next work a broader view and description for the NBR/ISSO 17799:1 will be carried out based on field prospection (questionaries on S&M companies in the ABC region) and conclusions on the requirements for software requirements procedures aiming information securities concerning are to be found. Finally, as this is a work in progress, for a future work and the broader potential contribution such as how will the study when completed contribute to theory and practice will be researched on.

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Collaborative e-Business and Software Agents

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Abstract- **The paper proposes the functions for two new agents in collaboration e-business, since we know that collaborative processes are growing up and we have a large way to work on researches about e-business. The start point is after we identify collaboration between people, processes and systems, we believe that each important process of the organization during its daily routine has two elements like context and content joined itself. Then, we can understand those processes are important when you look at them as a result for another one and it can resolve a problem. We want to get the ability to collaborate and leverage the knowledge. We have been based the analysis on the intangibles assets joined with the technology in order to build a design to understand the collaborative e-business processes. We present the results of the process and collaboration that already exist between themselves each time when a task has been started, in order to obtain new knowledge. To do that, we are based in new DirCCI model for collaboration of intellectual capital, where the business needs and collaboration are defined and then we propose the concepts to design and understand the main elements of the new system. The result is a new way to understand collaboration based on software agents in order to gain significant and sustainable competitive advantage and knowledge management in the organizations and now we should think in how help people, processes and systems to think about virtual decisions through e-business.**

Key words: Collaborative system, intellectual capital, agents for collaborative e-business.

I. INTRODUCTION

We know that traditional collaboration is good, and with technology we can help organizations to improve their work. We can look the past in this way: "Collaboration is a behavior, and not a new behavior at that!. Humans have used tools to collaborate since the dawn of time (smoke signals, carrier pigeon, telegraph, telephone), so what is different new?" [6]. The first answer is the Internet, infrastructure and tools. But previous researches show that there currently exist some limitations in collaborative tools, because they enable groups to communicate and coordinate but do not target critical processes. In other hand the traditional c-business is designed for actors send and receive information of the systems when they want, we think that is not enough. We believe in a new theory which the system must identify the moment when the process needs collaboration and information without it waits for a specific request. We propose it may happen when the process starts if we can analyze the workflow and previous rules of the business in each situation.

We think that business processes are a part of a major system, this system is called Intellectual Capital (IC). IC identifies human capital and the transformation processes towards others resources like structural capital and relational capital. The research presents a new collaboration system, in order to understand the concept of Intellectual Capital (IC), into a model that joins the approach of information systems and e-business concepts. The initial point is that e-business usually has high information systems in use like CRM, ERP o SCM. Our approach is to identify the collaboration needs inside the processes and others tasks of the daily routine. To identify collaboration needs we propose use software agents. The first agent can be designed to obtain the process needs just in the moment when they are happening; to do that, we need the previous support of workflow process and information systems. Then for the second part, in order to give an action to the system or some advices for a decision, the research propose another agent which must do the analysis of environment and it will obtain the development actions for each process.

The main advantage of the research is the analysis and development of the new DirCCI model (Management and Collaboration in Intellectual Capital, in Spanish is "**Dir**ección y **C**olaboración del **C**apital **I**ntelectual - DirCCI") [2]. The model shows the concept design of the collaboration system under the approach of information systems joined to the elements and the transformation processes of the resources of intellectual capital.

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II. COLLABORATIVE PROCESSES

We present three basics concepts: (a) How to look at e-business in the present and future; (b) the user requirements and the collaboration in organizations.

A. How to look at e-business?

For researchers at Collaborative Strategies, the interactions are augment since, we as humans, have established a context for our relationship and a level of trust, it is much easier to carry on that relationship electronically. Also we know that collaboration is critical when there needs to be an ongoing exchange of complex information between two or more people [6]. In other hand we know that the organization obtains the business intelligence when it can understand the role of each element in the global performance:

1) Integration B2B;

2) Personalized software;

3) Information management;

4) Security;

5) Quality service, etc.

For many years organizations have had trouble to work together, that's why the most viable way for the organizations work together was the acquisition of one for another. Now we have a new opportunity to work in harmony [12]:

1) Delivery through independent business: each one with the own users and systems;

2) Deals with spanning independent businesses: each with its own set of applications and users, each one interoperates with heterogeneous systems without being tied to one specific system technology.

The best examples of integration are in a family business, where the leader of the family supplies the activities and always he (or she) helps to resolve the problems, processes and also gives different ways of collaboration, it is getting very good results. A strong culture is needed to do that and the business itself identifies as a part of the market and specific supply. We know that the value chains B2B is the process of moving goods from customer order through the raw materials, supply, production, and the distribution of products to the customer. The new A2Z approach connects all the links of value chain via partnership, the link would be turned from physical connections to digital ones. You can obtain clear visibility of each stage of business process [1], [11]. The link involves all activities from the requirements to register needs of functional areas, problems and solutions, until the end of the processes.

B. Users Requirements

We show some selected factors from two viewpoints common to many organizations. The factors have seen from both end users and vendors of technology. These factors are critical to the successful adoption of collaborative technology [7]. The selected factors are:

From the end users.

1) Collaboration technology projects need to be tied to specific and important business needs by the actual users of the technology;

2) clear business processes exist are well defined, and are compatible with the technology.

Then from the vendors.

3) Usage is growing, but key decision-makers are not convinced of the effectiveness of the technology;

4) Although tied to a clear business needs and a metric showing success where it is used, people who otherwise would use it are instead claiming they do not have time to learn how.

We have others problems called cultural factors. For the propose of this research we agree with the authors when they say that these factors "are usually hidden, difficult to address, and boil down to not only changing the technology, but also creating changes in the organizational culture simultaneously with the introduction of collaborative technology" [7].

C. Collaboration in organization

"Collaboration is a term that is often misused as a technology and marketing 'buzz word'. And the most of tools do not support either collaboration or coordination but often only different types of communication or only the ability to access specific data" [6]. We need understand the three aspects of collaboration: content, context and process [9]. For us the definition is:

1) We have a problem that needs one or more process to resolve;

2) Theses processes have activities as a context and the information which will be transformed is the content. See Figure 1.

If we think that in the business world the collaborative processes are often highly negotiated, geographically distributed, and highly sensitive, and they are more vulnerable to error in execution, then the results consume more time, effort and resources; now we are ready to think in a new way of collaboration. In otherwise we know that not all things have been identified and the researchers know that there exists a collaboration that they have not found.

So due collaboration was forgotten for a long time, and some papers have appeared, we need a model to understand it.

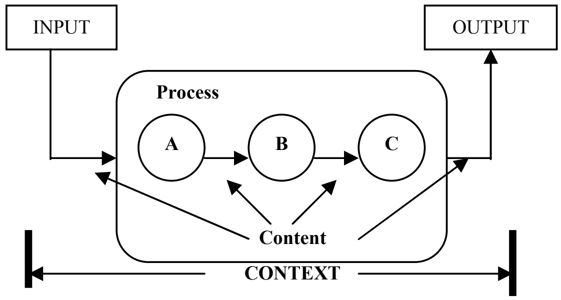

Fig.1. Collaboration aspects. Source: Own

A new system is defined for give collaboration to the processes in order to obtain the performance for intellectual capital system, and then we are ready to developed news e-commerce applications for the news activities. The collaborative system is a new way to use the technology software.

The model has three main parts (Figure 2): the top, the middle and the low part, based on Intellect model [3], [5].

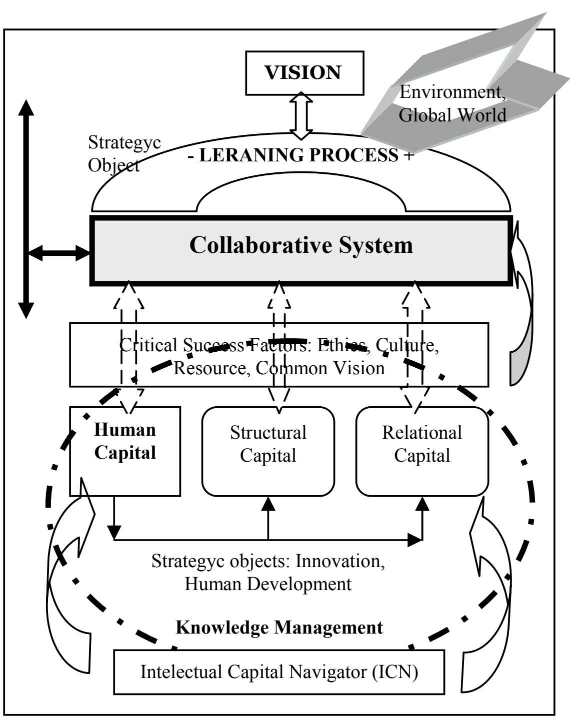

Fig. 2. DirCCI Model. Source: Own

In the top we see:

1) The management as an entity;

2) The intellectual capital as a result of learning;

3) The environment which always gives different changes or rules.

In the middle we look the collaborative system as the main component gives instructions, actions and help the decision.

This new system is based on software agents. One agent is designed to look the environment and internal processes; it will identify when a task start. Then the agent may resolve the critical factor for success in order to deploy the solution to one specific process. A second agent is designed to resolve a process task, its work is evaluates the best way to make each task in order to decide which information is needed from the processes rules. In the low part we can identify the intellectual capital based on transformation processes from human resources to structural capital and relational capital.

It's here when organization creates knowledge and the most important elements are the innovation, the competitive skills and development of people. The transformation processes are studied in another research called intellectual capital navigator [10].

We can say that this transformation is based on rules that we call "workflow".

III. COLLABORATIVE SYSTEM

We have some collaborative tools like FileNet, eRoom, and Collabrix, they were developed to solve a very targeted problem: in the case of eRoom, an easy-to-use virtual team solution, for FileNet, an EIM (Enterprise Instants Messaging) and conference calling services to support application sharing, but now we need to create a value into a virtual space team or across a transaction-based business systems [9]. Also "Effective collaboration represents the most value, top-line gain for organizations today. It can unlock the potential of the collective knowledge and intellectual capital of a given organization, as well as its value network" [6].

The new system is based on information system and agents concepts like transaction-based business systems. We have two agents:

1) The main agent is designed in order to look the environment and identifies when a task is started. Then the agent may resolve the critical factors for success in order to deploy the best solution for each process.

2) The second agent is designed to complete the process, its work is evaluates the best way for each task in order to analyze the information from the previous expertise and process rules (Figure 3).

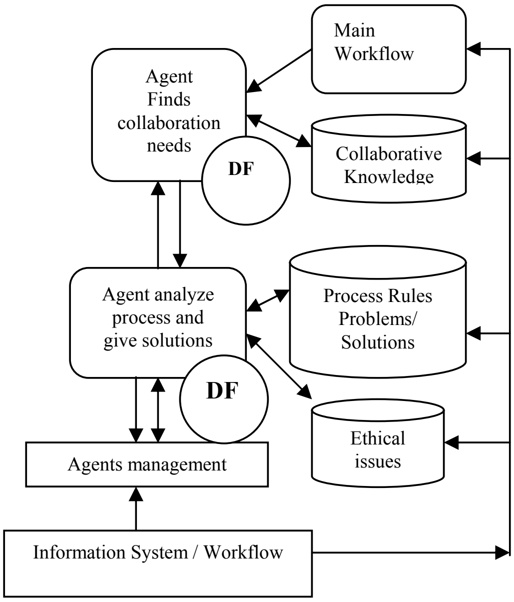

Fig. 3. Collaborative processes. Source: Own

With these agents we can identify an explicit business process around a specific context, and then there is a new process to collaborate looking for coordination and interaction. The new processes are based in repositories where content and context like workflow rules are stored.

With these functions we want to give an answer to traditional problems that have identified such as: "...is usually not a single simple solution that can easily be identified" [7]. After identifying collaboration into the system, we can define the relation between intellectual capital and knowledge management:

1*) Human capital is the source of knowledge;*

2) An action over transformation process gives the performance and new knowledge towards all the system;

3) The new knowledge is ready for be registered and used. The knowledge is shared in organization and in order to obtain a successful goal.

Due to changes are so quickly in daily routine, we do not have enough time to analyze new situations, but the system has old and new experience in order to begin again. The relation is started with human activity, in any place where the needs have been created. Each activity may be registered in previous step, and the processes rules may be designed in workflow system. We have an important result in the transformation process between different parts of intellectual capital, this transformation process produce new knowledge [4].

The agents are defined for:
1) To identify status of the process, look for starting and ending processes points;
2) Analyze previous and similar processes;
3) Analyze problems in previous processes;
4) Analyze new problems with available information;
5) Preparing recommendations and send to executor;
6) Registering the results of the process.
For example, now the functions of the each agent are:
1) Agent which find task and process needs
(a) Search and find when a task starts, inside a specific context;
(b) Identifying who starts task
(c) Obtaining information about process
(d) Obtaining information about main objectives
(e) Match with the rules of workflow and to obtain next sequence;
(f) Delivery to second agent, based on previous experiences and rules;

2) Agent who analyzes process and give the best solution
(h) Just receive the previous results from the first agent and evaluate the alternatives in order to give the best solutions:
(h.1.) attention process,
(h.2.) common problems
(h.3.) complex problems,
(h.4.) recommended solutions;
(i) For example it evaluates if exist new problems for resolve or the action needs more information. When the response is positive, the agent analyzes:
(i.1.) other recommended solution,
(i.2.) formal reports,
(i.3.) changes in the environment,
(i.4.) other main information;
(j) Prepare actions
(k) Delivery actions
(l) Prepare a final report;
(m) Learning process for agent.

The benefits are in the way and performance when each process is executed. Some of the particular results are in Table I.

TABLE I: OPTIONS FOR EACH PROCESS WITHOUT COLLABORATION

Receptor	Executor
Initial activity, may delivery	Must be solve
If need collaboration, but not to ask for	May or not to need collaboration
Resolved	Resolved
Possible mistake	Possible mistake

The result is a new model for collaborative e-business keeps the main elements when the organization needs important processes like SCM, CRM an ERP, but now we have a new approach. We introduce a collaborative system which it has an intermediate function before makes the decision. This new system has two entire points, first the requirements of human capital and the rules of the processes. After do that, the results are new transformations over the structural capital and relational capital (Figure 4).

We know that a possible mistake is always a bad situation, because experience shows that more than 50% of possible mistakes ending in problems, and then people must start again with new resources and new time, and the worse situation is loses the credibility and loyalty from the users and the customs.

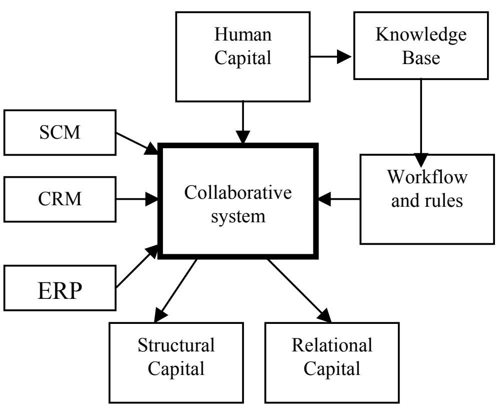

Fig. 4. Collaborative concepts for e-business. Source: Own.

Now with the collaboration system we can to resolve any situation, common or complex, without lose time and improvement performance. We are analyzing all different situations, those will be improving with the new system, but previous results are in Table II.

TABLE II: PERFORMANCE AROUND COLLABORATIVE SYSTEM

	Common problems	Complex problems
Alone	Must be solve	Must be solve
Alone	May or not to ask for help	Must ask for help
Alone	Resolve without help	Resolve without help
Alone	It can or not to has errors	With errors
Collaborative system	Resolve with help, without errors	Resolve with help, without errors

This example of performance will have a better measurement with the development of e-business activities, like supply chain management (SCM), customer relationship management (CRM) or enterprise resource planning (ERP).

IV. EXPECTING RESULTS

We are having answers to some requirements called collaborative problems, like:

1) Tracking down content;

2) It Help people in critical information;

3) Ensuring that task are being completed properly.

That's why we need a robust application based on modules that allow organizations to manage critical business interaction. Now, we need that the members of their value chain can interact with information and processes of their day to day business but with rapid speed and very high efficiency, then we are looking for the trend in order to obtain previous results like:

1) Real time process and synchronous work between actors of the value chain;

2) Create a learning curve for rapid productivity;

3)A holistic approach to challenges of collaboration, between people, processes and systems. We think this is the point to create intelligence in collaborative e-business.

In other hand in looking for the value of collaboration within an organization, we have to look at artifacts, like time and money, which are better with a new collaboration process. We propose to translate some human behavior to systems, and we think we have better results across the organization. It is possible when we are able to produce rich information for decision making and capture best practices for future situations like the Collaborative Project Portfolio Management [8], but now these functions will have the ability for start to work in automatics ways.

V. CONCLUSIONS

We need to identify and to share information; also we need to capitalize the value of intellectual capital gained over the history of the client's relationship. We have recognized problems of collaborations in some activities inside the processes, like spending an inordinate amount of time looking for information, or documents, or contents or people expertise. But now theses activities can be improved with the technology of information systems and software agents. Since researchers worked for more than 20 years, each time is better the understanding of the knowledge management and use of the intangibles concepts like intellectual capital. From the approach of systems engineering we have the opportunity of design technological models in order to improve our processes each time doing new ways to get better results than now in our daily activities.

At the same time we know that knowledge management is present in each activity of people, organizations, processes and systems, and its value is in to obtain itself, to register itself, to analyze itself and finally to give to others users. People need to share knowledge, but in the most of the cases they aren't prepared for do that, because they feel that their knowledge is the source to keep their job and their reason to live. This concept will be changed in order to understand that the value of people is in the new skills for learning and development new ways to get a better performance.

In other hand we are working in a new system, first to identify collaboration needs, and second to see new ways for give collaboration based in software agents and information systems.

The system creates value while people put their experiences and knowledge about different problems and solutions, without fears, always having the main goal on the organization. This reflection presents new conditions for the success, like ethic, culture, transcendental motivation, that will be possible interchange with e-business processes. The new concepts will be development in order to improve the competitive advantage in the organization.

Finally, while we can think in new ways to create something, we are working in a major space for creation and innovation in each organization, because the reason for not do it is the complex and urgent activity of the daily routine. Now people can work in creation and R+D solutions.

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A Compression Algorithm Based on the Polarity of the Speech Signal

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ABSTRACT

In this study, a compression algorithm is presented for speech signals. With this compression algorithm, signals are compressed around the positive and the negative peak values of the speech signal. This algorithm compresses a speech signal by removing reproducible portions of the signal while retaining reference signal portions that are enough to reproduce the original speech signal with a very small loss in quality. Obtained compression results are evaluated visually and using the signal to noise ratio (SNR) values. SNR value is calculated to be over 29dB for a 30% compression rate. Because the method is data-dependent, SNR values can vary at the same compression rate for the different signals. The zero crossings, pitch period, and the peak values of the original speech signals are preserved. Additionally, the proposed algorithm is very efficient and there is reduced processing overhead.

1. INTRODUCTION

Signals such as speech and image data are stored and processed in computers as files or collections of bits. Signal compression is concerned with reducing the number of bits that is required to describe a signal to a given accuracy. The compression of speech signals has many practical applications such as digital cellular technology. Compression allows more users to share the system than otherwise possible. In addition, for a given memory size, compression allows longer messages to be stored such as in digital answering machines [1].

The proposed compression method is based on a sinusoidal representation of the speech signal. Sinusoidal speech compression or coding methods are studied especially for obtaining low-bit rates [3], [5], [6]. Conventional systems and methods for reducing speech data storage are typically based on frequency domain processing. One conventional approach to speech compression relies on a sort of time domain processing, based on periods of silence, voiced sound, and unvoiced sound. Another approach to speech compression uses pitch waveform decimation to reduce data [4].

Zero crossings and pitch periods are very important parameters for speech recognition and speaker recognition. With this study, it is aimed to preserve the peak values of the speech signal, zero crossings, and pitch periods of the signal. The coding model is given with Figure 1. There are two steps in the model.

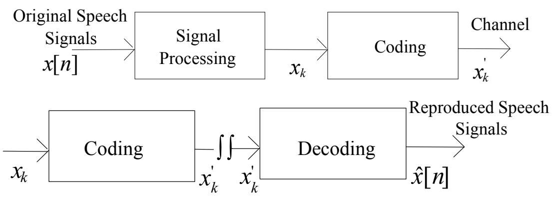

Figure 1: Coding/Decoding Model.

The first step is signal processing. If the speech sample equals to 1, it is changed to the nearest positive integer that is 2. This is done because 1 is a special code data in this compression algorithm. This step introduces data loss that is very small and harmless for the speech parameters such as pitch period, and zero crossing. After the signal-processing step, the SNR was found to be over 65dB for the speech signals that are used in this study.

The second step is the coding step. Some of the speech samples around the peaks are eliminated.

2. THEORY OF THE ALGORITHM

At the beginning of the algorithm, speech signal data is divided into vectors of different lengths. These vectors are formed based on signal polarities. Each vector is then coded independently. This is shown with Fig.2. These vectors can be defined as;

$$\mathrm{X} = \begin{bmatrix} x_0 & x_1 & x_2 & \dots & x_{N-1} \end{bmatrix} \tag{1}$$

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 127–130.

$$x_0 = [x_{0,0} \quad x_{0,1} \quad x_{0,2} \quad \ldots \quad x_{0,n}] \tag{2}$$

$$x_1 = [x_{1,m+1} \quad x_{1,m+2} \quad x_{1,m+3} \quad \ldots \quad x_{1,n}] \tag{3}$$

$$x_2 = [x_{2,n+1} \quad x_{2,n+2} \quad x_{2,n+3} \quad \ldots \quad x_{2,p}] \tag{4}$$

$$x_3 = [x_{3,p+1} \quad x_{3,p+2} \quad x_{3,p+3} \quad \ldots\ldots \quad] \tag{5}$$

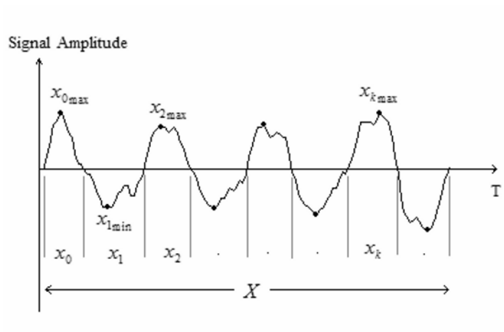

Figure 2: Preparation of the vectors in the algorithm.

x_k signal is defined as ;

$$x_k = [x_{k,h+1} \quad x_{k,h+2} \quad \ldots \quad x_{k,i_{\max}} \quad \ldots \quad x_{k,j}] \tag{6}$$

$k = 0,1,2,\ldots$

If x_k *th* vector has a positive polarity, x_{k+1} th and x_{k-1} th vectors have a negative polarity. Each x_k vector can be a different length. $x_{k,i_{\max}}$ is the peak value for the vector. In the following equations, a is the difference between index of the first data and the maximum peak of the x_k vector. b is the difference between index of the last data and the maximum peak of the x_k vector. R is the number of the samples that will be removed from the original data. R-value is chosen at least 3. If the speech signals have mostly voiced speech signals, the R-value can be chosen more than 3.

$$a = i - h + 1 \tag{7}$$

$$b = j - i \tag{8}$$

$$a > R \Rightarrow \begin{cases} x_{k,i-1} = 1 \\ x_{k,i-2} = 1 \\ \ldots\ldots\ldots\ldots \\ x_{k,i-R} = 1 \end{cases} \qquad b > R \Rightarrow \begin{cases} x_{k,i+1} = 1 \\ x_{k,i+2} = 1 \\ \ldots\ldots\ldots\ldots \\ x_{k,i+R} = 1 \end{cases}$$

(9)

$$x_k = [x_{k,h+1} \quad x_{k,h+2} \quad \ldots \quad x_{k,i-R-1} \quad 1 \quad 1 \quad \ldots \quad 1 \quad x_{k,i_{\max}} \quad 1 \quad \ldots \quad 1 \quad 1 \quad x_{k,i+R+1} \quad \ldots \quad x_{k,j}] \tag{10}$$

$$x'_k = [x_{k,h+1} \quad x_{k,h+2} \quad \ldots \quad x_{k,i-R-1} \quad 1 \quad x_{k,(i_{\max}-(R-1))} \quad 1 \quad x_{k,(i+R+1-2(R-1))} \quad \ldots \quad x_{k,(j-2(R-1))}] \tag{11}$$

If $a > R$ and $b > R$, coded speech signals, x'_k are represented with the Eq.11.

If $a > R$ and $b < R$, coded speech signal is represented with Eq.12.

$$x'_k = [x_{k,h+1} \quad x_{k,h+2} \quad \ldots \quad 1 \quad x_{k,i_{\max}-(R-1)} \quad \ldots \quad x_{k,j-(R-1)}] \tag{12}$$

If $a < R$ and $b > R$, coded speech signal is represented with Eq.13.

$$x'_k = [x_{k,h+1} \quad x_{k,h+2} \quad \ldots \quad x_{k,i_{\max}} \quad 1 \quad \ldots \quad x_{k,j-(R-1)}] \tag{13}$$

Each vector size may be reduced up to $2(R-1)$ data at every iteration. The algorithm can be repeated until $a < R$ and $b < R$.

An algorithm for a new data-dependent compression algorithm:

A repetitive algorithm can be written for this study. With this algorithm, all possible data are deleted for the maximum compression rate as follows:

1. If the speech sample equals to 1, change it to the nearest positive integer which is 2.

2. Divide speech signals into vectors, x_k based on signal polarity.
3. Select a R value.
4. Find the maximum data, $\max(x_k) = x_{k\max}$ in each vector.
5. Calculate a and b for each vector.
6. If $a > R$ or $b > R$ for a vector, delete R samples which are not equal to 1, around $\max(x_k)$. If $a < R$ and $b < R$ for all vectors, go to step 9.
7. Put 1 in the place where samples that were deleted.
8. Go to step 4.
9. Stop.

This algorithm allows compressing the speech signal with the maximum compression rate. The maximum compression rate depends on the signal. If the number of the zero crossings is small in the signal that means the size of the vector is bigger, the compression rate will be much higher.

3. DECOMPRESSION USING THE CUBIC-HERMITE INTERPOLANT

In the decompression process, the cubic hermite interpolant method is used to calculate the data, which are deleted in the compression step to compress the signal. Using standard algorithms it is often necessary to sacrifice interpolation of the data in order to preserve monotonicity, or conversely, to sacrifice monotonicity in order to preserve interpolation. Using the cubic hermite interpolant method, the data are sufficiently accurate to warrant interpolation, rather than a least squares or other approximation method. With this interpolation method, the necessary and sufficient conditions are derived for a cubic to be monotone in an interval. These conditions are then used to develop an algorithm that constructs a monotone piecewise cubic interpolant to monotone data. The curve produced contains no extraneous bumps or wiggles [2].

On each subinterval, $x(l) <= x <= x(l+1)$, $P(x)$ is the cubic hermite interpolant to the given values and certain slopes at the two endpoints. Therefore, $P(x)$ interpolates y , $P(x(j)) = y(j)$ and the first derivative, $P'(x)$ is continuous, but $P''(x)$ is probably not continuous; there may be jumps at the $x(j)$. The slopes at the $x(j)$ are chosen in such a way that $P(x)$ is shape preserving and respects monotonicity.

Obtained decompression results are evaluated using SNR values. SNR of the reproduced speech signals is given with Eq.14. $x[n]$ is the original speech signal, $\hat{x}[n]$ is the reproduced speech signals.

$$SNR(dB) = 10\log_{10}\frac{\sum_{n=0}^{data\_size-1} x[n]^2}{\sum_{n=0}^{data\_size-1} (error[n])^2} \tag{14}$$

$$error[n] = (x[n] - \hat{x}[n]) \tag{15}$$

4. COMPRESSION RESULTS

In this section, results of the new compression algorithm are presented. We checked the results of two different original speech samples. One of them is in English ("Card games are fun to play") and was sampled 8000 Hz at 128 kbps. The other one is in Turkish ("dokuz") and it was sampled at 9600 Hz at 115.2 kbps. Fig.3 shows the first original speech signal and its decoded speech signal with 20.1% compression rate.

Figure 4 shows the second original speech signal and its decoded speech signals with 30.1% and 41.1% compression rates. R-value is chosen 3 for all signals. Reproduced speech signals for the two compression rates are given with Fig.4.b and Fig.4.c. With the proposed coding technique, the waveform of the speech signal is not changed. All the peaks of the signal are preserved.

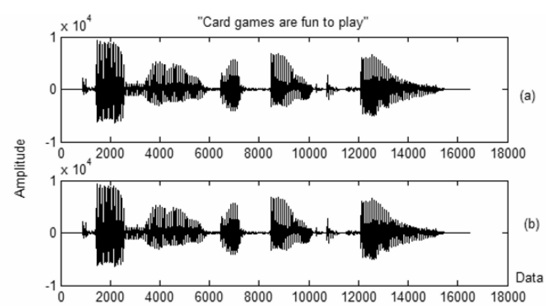

Figure 3: (a) Original speech signals (b) reproduced speech signals, 20.1% compression rate, SNR=14.5 for R=3.

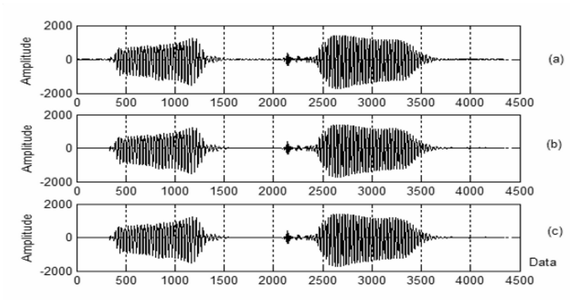

Figure 4: (a) Original speech signal, “dokuz” in Turkish, (b) Reproduced speech signal for 30.1% compression rate, 80.5 kbps, SNR=29.4, (c) Reproduced speech signal for 41.1% compression rate, 67.9kbps, SNR=27.8 for R=3.

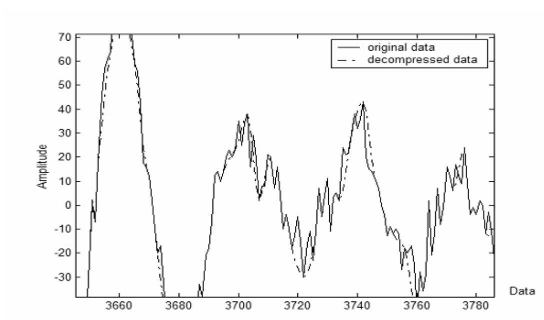

Figure 5: Comparison between original signal and the reconstructed signal with 41.1% compression rate.

Comparison between the original signal and the signal reproduced using the cubic hermite interpolant method is shown with Fig.5. With this method, reproduced signal catches the original signal quite well.

5. CONCLUSIONS

A compression method for speech signals has been proposed. The algorithm of this method is very easy to apply to signals. The algorithm is very efficient and there is reduced processing overhead. Since this method is data-dependent, the compression rate is limited with the structure of the speech signals. Voiced sounds have more compression rates with this method. Some signals that have low frequency content may allow much higher compression rates. The cubic hermite interpolant method has been used to reconstruct the data.

Based on current speech compression standards, one can think the 30% or 40% compression rates presented with this method is low. However, since this method has a high SNR, it can be used to supplement other compression techniques to achieve higher compression rates.

Neural networks are currently used for speech processing. It may be of interest to investigate how neural networks may be used to reconstruct the speech data using the proposed method.

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Building Teaching Timetables Using Random Variables: Algorithms and Techniques

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***Abstract* - As noted by a number of researchers, teaching timetable construction is a multi-parameter and complex task. Various techniques have been proposed in order to tackle timetable constraints. This study focuses on the School Timetable Builder (STB) application, which enables the construction of teaching timetables for secondary education. The STB application works with random variables and outputs results considerably fast, evaluating all the constraints and conditions that should be taken into account. Moreover, the algorithms and the techniques used by STB application are discussed.**

I. The STB Application

The creation of a teaching timetable is an extremely complex problem [1]; [2]; [3]; [4]; [5]. Various approaches, examples and techniques which will solve timetabling problems have been proposed over the years [6]; [7]; [8]; [9]; [5]. Relevant research is focused on reducing the time required for timetable creation [10].

In this study, special attention is paid to timetable construction problems in Greek secondary education schools. The problem in these schools, as a result of our educational system, usually is delimited to fixing teachers in specified classes, under certain conditions. All teaching hours are defined and follow specified constraints and limitations, according to the way they are interconnected with the classes. This is exactly the context of the school teaching timetabling problem [11].

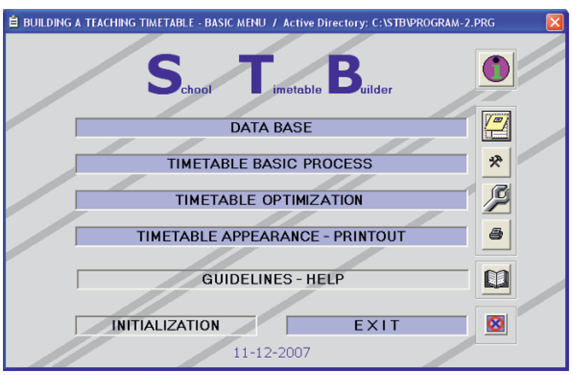

Fig. 1. Screenshot of the main menu of the STB application.

The School Timetable Builder (STB) application, as has been analyzed by Panagiotakopoulos & Kameas (2006) [12], faces the problem reliably, within reasonable time, provides different versions of the same teaching timetable, according to pre-defined limitations. STB is constructed using the high level programming language Microsoft Visual Basic 6.0, runs under Microsoft Windows environment and consists of four modules. Each module can be called from a central menu and can be executed using a shell.

Although the processing module plays a significant role, the data base module is also important, since it registers all relevant information and defines the relations between the various variables.

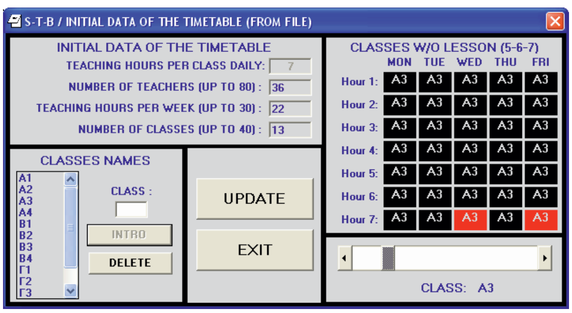

Fig. 2. Screenshot of an STB application instance from data base module. In this, are appearing the initial data of the timetable.

In school timetabling, the teaching hours offered by every teacher are interdependent and connected with specific constraints [13].

The data base module of STB application, allows the appearance and the adjustment of all relative parameters, e.g. variables, limitations and constraints for every teacher, related to his/her teaching hours.

The same module can also manipulate and cross-check the registered data in order to inform the user whether the limitations and constraints allow the construction of a teaching timetable version or not. For example, the application may not afford to build a version of a teaching timetable because of a teacher's absence at a specific day/hour for a class, due to a commitment from the user of a part or all the teachers.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 131–136.

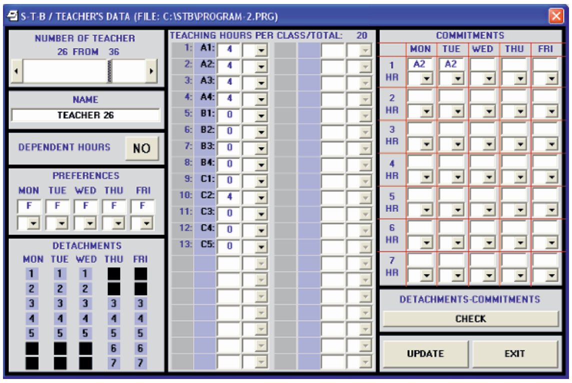

Fig. 3. Screenshot of an instance from the data base module. This contains all the data and all the constraints, concerning to a specific teacher.

The processing module works with random variables, and manages all data and its association. Also manages the available teachers registered in the data base.

Users can utilize the processing task through two alternatives. These are: processing with counters view or processing with a teaching timetable view. Under the first choice (Fig. 3) the application runs faster, compared to the alternative.

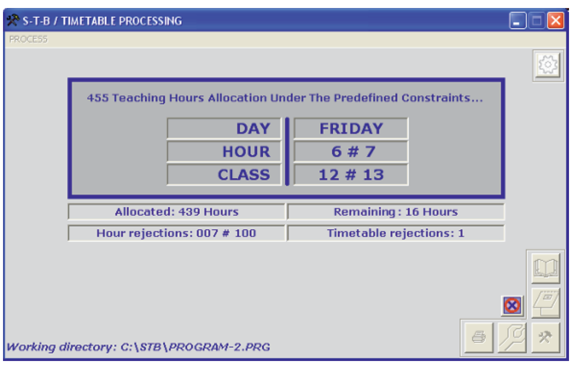

Fig. 4. Screenshot of an instance from the data processing module for the construction of a version of a timetable, using counters view.

If constraints allow the creation of a timetable version, STB application searches for an available teacher by auditing the specific candidate on the basis of his/her constraints and accepts or rejects this random choice. The process described so far utilizes continuous random sampling techniques.

Specific counters are used for controlling processing during the creation of the teaching timetable. The main objective is to limit the expending time on the repeated processes. When values exceed the pre-determined thresholds, the current hour or even the entire program is disassembled. Used hours return to the teaching schedule and the application begins once again a new effort to build a new version either from the point before the current day or completely from the beginning.

II. STB Application and The Typical Permissions of a Teaching Timetable

For the needs of this study let us presume that:

- T_i denotes the array of the teachers (i = 1 up to T_{max}, when T_{max} denotes the number of teachers).
- C_j denotes the array of the classes (j = 1 up to C_{max}, when C_{max} denotes the number of classes).

The total sum of the lessons offered by all the teachers must be equal to the number of the lessons that the application has to associate with all the classes.

- D_m denotes the array of the working days (m = 1 up to D_{max}, when D_{max} = 5).
- $H_{n,j}$ denotes the array of the teaching hours per day for each class (n = 1 up to H_{max}, when H_{max} = 7 (maximum teaching hours per day) and j = 1 up to C_{max}).

In general, the basic constraints and limitations of a teaching timetable for a typical secondary education school in Greece fall within four categories:

(ct1) T teachers teach simultaneously students coming from C classes (division of original classes and recreation of new classes from groups of students coming from various classes). In other words, T specified teachers are associated with C specified classes. May be $T_{max} = C_{max}$ or $T_{max} > C_{max}$ or $T_{max} < C_{max}$.

(ct2) T teachers are associated with specified classes, in specified hours of a day (and in a specified day of the week) with a correspondence 1 - 1 (for example: $T_1 \rightarrow C_3$, $T_5 \rightarrow C_7$, $T_2 \rightarrow C_4$, etc.).

(ct3) T teachers do not teach at specified hours of a day or specified days of the week.

(ct4) Among all the constraints of a typical teaching timetable is the one related to the distribution of the teaching hours per class of a teacher in a week. For example let it be considered that the teacher T_1 has to teach for 25 hours equally distributed in a week in 5 classes (5 hours * C_1, 5 hours * C_2, 5 hours * C_3, 5 hours * C_4, and 5 hours * C_5). An important and critical constraint is the dissemination of his hours, so as to teach one hour in the same class per day.

All the above mentioned constraints are shaped in the data base module and STB deals with them during data processing. The underlying rationale behind this design is to search for the optimal treatment. The application ignores the number of classrooms, because such a problem is not common. Usually, the number of necessary classrooms is ensured a priori by the school management.

III. The Processing Module

For the random sampling the function RND is used, which in Visual Basic 6.0, according to Microsoft, starts from the command RANDOMIZE. The function RND can return random numbers of single precision, uniformly distributed in the [0,1] range [14].

The application begins with the admission that each class, hierarchically, beginning from the first till the last (except in the case of something different) should be connected with a valid teacher. So, supposing that the basic data of the program includes classes C_1, C_2, C_3, C_4, C_5, … C_{15}, the application begins from Monday's first teaching hour and tries to connect hierarchically, each one of the classes at least with one teacher (will be found, that is to say, for the particular hour at least, 15 teachers). When the first teaching hour is covered, the application moves to the second teaching hour of Monday, the third, etc. When Monday's schedule is completed, the application continues with Tuesday etc, until all the days of the week are covered.

Teaching hours and teachers, which are related to constraint of type ct1 and ct2, are imported automatically when processing a timetable version. Thus, these classes are committed, as they are already connected with teachers and they are not available for further search. At the same time, the used teaching hours are removed from the teaching hours of the "selected" teachers. After this:

With continuous random samplings the application searches for a candidate teacher, who in his teaching schedule has available the specific class. This search, as it will be analyzed below, occurs using three different ways, which seems to be effective.

If that search is successful, the remaining constraints of type ct3 and ct4 are checked. The last two constraints, check whether the selected teacher will be accepted or rejected. If the teacher is accepted, the teaching hours of this teacher are decreased.

The non valid random samplings are countered by specific counters, and their values are indicative of the beginning of a new effort, as previously mentioned.

IV. Facing the Permissions

In the previous section, we discussed how we tackled constraints in cases ct1 and ct2. Constraints of type ct3 were handled with integer programming techniques. More analytically, during registering, in the data base module, the application had created the array YesNoTeach(D_m, $H_{n,j}$, T_i). The elements of this array are equal to 0 whenever teacher T_i is available for teaching at the hour $H_{n,j}$ / day D_m and in different case, are equal to 1. Thus, during data processing, the appropriate element of the array YesNoTeach is checked, which is determined by the variables of running day, hour and the teacher. If the value of this element is equal to 1, the current choice is cancelled and the application moves to the next sampling. If the value is 0, the control passes so as to check the constraints of type ct4. The constraints of type ct4 are checked in the following way:

In the data base module, during registering the teaching hours of each teacher and the interconnections between them, two integer arrays are created: the array Distribution(T_i, C_j), and GivenHours(T_i, C_j, D_m).

The array Distribution(T_i, C_j) is created on the basis of the teaching hours of each teacher per class. The array GivenHours(T_i, C_j, D_m) is created during the timetable construction and their elements contain the number of teaching hours of a teacher that are connected with a class in a day. The constraint of type ct4, works on the basis of the next reasonable structure:

```
PassFlag=0
If Int((Distribution(Ti, Cj) + 2) / Dmax + 0.5) <=_
GivenHours(Ti, Cj, Dm) Then
  PassFlag = 1
End If
```

With the type Int((Distribution(T_i, C_j) + 2) / D_{max} + 0.5), the teaching hours of the teacher T_i in the class C_j, are "broken" and distributed according to the number of the teaching days.

The comparison of Int((Distribution(T_i, C_j) + 2) / D_{max} + 0.5) with the element of the array GivenHours(T_i, C_j, D_m) leaves unaffected or alters the value of the integer flag PassFlag. This alteration from 0 to 1 signifies that the randomly found teacher T_i is not valid, since there is an accumulation of their teaching hours in the class C_j.

Of course, in the case that something like this is desirable, a specific constraint of type ct2 has to be set in the data base module of the application.

V. Random Sampling for a Valid Teacher

For the random search of a teacher who is valid for a current class, it appears empirically that the effectiveness of the STB application is increasing using three different ways (three levels). However, before we analyze these three ways of search, lets hypothesize that for one hour/day and for the class C, the application is searching for a teacher T. Let we suppose also that the available teaching hours of the teacher T are analyzed with the form that appears in the middle column of table 1:

TABLE 1
THE FORM OF THE AVAILABLE TEACHING HOURS OF THE TEACHERS (WEEKLY)

Teacher	Teaching Hours / Classes								Total
T_1	C_1	C_1	C_1	C_3	C_3	C_3	C_6	…	22
T_2	C_4	C_4	C_4	C_4	C_5	C_5	C_5	…	24
T_3	C_3	C_3	C_3	C_5	C_5	C_5	C_6	…	26
T_4	C_1	C_1	C_2	C_2	C_3	C_3	C_4	…	28
T_5	C_5	C_5	C_6	C_6	C_6	C_6	C_6	…	22
T_6	C_6	C_6	C_7	C_7	C_7	C_8	C_8	…	24
…	…	…	…	…	…	…	…	…	…

The efforts for the schedule completion for the current class (C), develop hierarchically in three levels, as follows:
(1) At this level, a random sampling between 1 and T_{max} begins to locate a teacher. Afterwards, the validity of this search is checked. The random sampling begins with seed number, that each time is potentially differentiated since it is supported from the value of the counter CL4 (its use is analyzed below). The used code has as follows:

```
If (CL4 mod 2) = 0 Then
  Randomize Timer
Else
```

```
 Randomize Val(Mid$(Time$, 7, 2))
End If
T = Int(Tmax * Rnd + 1)
```

If the candidate teacher is not available for the current class (hour/day) or the teacher has no available teaching hours, the random sampling is repeated. If a temporary valid teacher is found, a random sampling in his teaching hours is accomplished also, in search of a teaching hour, which is suitable for the current class.

If the class that is found in the teaching hours of candidate T is the same with the class C, this may be accepted, since it is not in contrast with the constraints. If this class is finally accepted, the application searches for a new teacher for the next class. If the class is not accepted, a new random search begins for a valid teacher and a random search of his available teaching hours, also. Rejection of the class that found is possible when:

- If the random search in the teaching hours of the teacher T yielded a different class from C.
- If the random search in the teaching hours of T yielded the class C (that is to say, C is available for use in T teaching hours), but T is already busy in another class on the current hour.
- If the teacher T had already taught in the class C, and it is not allowed to teach again in the same class for another teaching hour.

In the case where the sampling with the previous form is not efficient, the control passes to the next (second) level.

(2) At this level, a random sampling begins once again between 1 and T_{max} for the search of a teacher T. If T is not available for the current hour/day or he has not available teaching hours, the sampling is repeated. If the teacher is temporarily valid, a sequential search for a class, within his teaching hours begins, in order to be found the class C.

If class C is not found in the teaching hours of T, the application begins an attempt to locate an available teacher (a new teacher), using once again a random sampling procedure. Then, if a new teacher is available, the application searches for class C using a sequential search design, within a teacher's available teaching hours.

If class C is found within T's available teaching hours, since no conflict arises between teacher T and his/her constraints, the candidate is accepted. Rejection of the selected teacher is possible when:

- The random search in the teaching hours of T yielded the class C, but teacher T is already busy in another class during the current hour.
- The teacher T had already taught before in the class C, and is not permitted to teach again in the same class, on the same day, for another hour.

In the case where the sampling with the second form is again not efficient, the control passes to the next (third) level.

(3) At this level, a random sampling is repeated between 1 and T_{max} , in order to locate an available teacher T_1. If T_1 is not available for the current hour/day or there are no available hours within his teaching hours, the sampling is repeated.

If the candidate teacher is temporarily valid, a sequential search for a class begins, within his teaching hours, in order for the class C to be found. If class C is not found within the teaching hours of T_1, a new attempt starts with a new random sampling for a valid teacher, in order to find out the class C using sequential search, within his available teaching hours. In this level, after repeated random sampling, there is a high possibility to find a teacher T_1 with the class C within his available teaching hours, but there is also a high possibility for the following:

- To be unavailable (the candidate teaches in another class).
- To have already taught an hour in the same day in class C, and it is not permitted to teach again in the same class.

Then, the application:

- Searches for a teacher T_2 from a past day/hour, who could teach at the current hour in the class C, but without contravention of the constraints.
- Checks whether teacher T_1 could be interchanged with T_2, but without contravention of the constraints.

At the same time, if it is not feasible to match the class C with a teacher, the random sampling is repeated from the beginning.

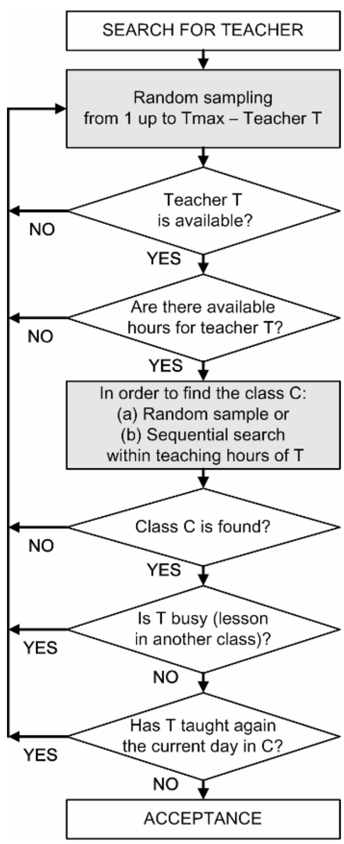

Fig. 5. Logic diagram for the search of a valid teacher (levels 1 and 2).

VI. APPLICATION'S SELF-ADJUSTMENT

Five counters are available for controlling data processing functions. Details on how these counters operate are described in a previous study [12]. Three of these counters (CL1, CL2, CL3) are used for the flow's redirection from a level of teacher search to the other. The value of all three, if a valid teacher is found in any level, is altered to zero and the search begins again from the start.

The value of the fourth counter (CL4) is every time equal to the sum of values of the previous three counters. When its value exceeds a specified threshold, the creation of the timetable of the current day is cancelled and the application restores all teaching hours to the candidates' schedule. Then, the flow of the program is redirected once again to the beginning of the level three, in order to construct a new timetable version. Thus, the consumed time for the timetable construction is controlled, since the application avoids searching for a candidate teacher for a long time.

The fifth counter (CL5) holds the daily rejections and when its value exceeds a specified threshold, the application rejects the whole schedule that is created until this time and the construction starts from the beginning.

The values of these counters can be controlled and can be adjusted (even in real time) by the user or be adjusted automatically by the application. The most optimal adjustments that are done automatically by the application have resulted from a huge number of repeated empiric measurements. Data and constraints of two types of timetables, which appear in table 2, were used for their calculation.

TABLE 2
GENERAL CHARACTERISTICS OF THE TWO TYPES OF TEACHING TIMETABLES USED FOR APPLICATION TESTING

	P1	P2
Teachers (T_{max})	36	36
Classes (C_{max})	13	13
Working days (weekly) (D_{max})	5	5
Teaching hours for each class (daily) (H_{max})	7	7
Total available hours (weekly)	1225	1225
Total working hours (weekly)	455	455
Total dependent working hours	0	35
Total reserved hours	0	92
Total detachment hours	84	84

With value of counter CL4 = CL1 + CL2 + CL3 + 1 and value of CL5 = 100, we asked from the application to create 25 different versions of teaching timetables for each case using specified data and constraints. The values of the counters CL1, CL2, CL3 are adjusted to cover all the values consecutively in the closed interval [T_{max} * Int(T_{max}/4), T_{max} * Int(T_{max} + T_{max}/2)], using a step (3 * T_{max}) between them. This range is empirically found, as a result of constructing timetables in minimal time.

Thus, a total of 102,400 teaching timetables for each specified type of timetable were completed and several variables were calculated. One such example was the time consumed for each group of timetables.

For the optimization of the counters' values (consumed time) five completely identical personal computers (CPU Intel P4/2,8GHz, 2GB RAM, HDD WD 80GB / 8MB cache), with Microsoft Windows XP Professional Edition were used.

Results, concerning the time required for the completion of a timetable, suggest that the application could create improved timetable versions, in terms of time-effectiveness, when the counters' values CL1, CL2 and CL3 are fluctuated closed to the points:

- for CL1: T_{max} * (T_{max}/3)
- for CL2: T_{max} * (2 * T_{max}/3)
- for CL3: T_{max} * (5 * T_{max}/6)

With the values above, which depend on the number of the teachers (that is from the interval range used for the sampling), the application can be self-adjusted, so that the creation of a teaching timetable requires minimal time.

VI. CONCLUSIONS

The timetable construction for education (schools, universities), is an extremely complex problem, whose manual solution requires much effort. So, various approaches have been followed so far for the solution of the problem, to produce acceptable solutions that satisfy all the problem constraints.

The operation of School Timetable Builder (STB) application was analysed in this study. STB operates effectively, using random variables and requires a reasonable time duration for the creation of a functional teaching timetable. As it has been analyzed, the time duration for a typical teaching timetable, that includes some specified constraints and limitations, is varying between 0 seconds to several minutes [12]. STB application uses three hierarchical different methods in order to "appoint" a valid teacher to a certain class. In all these three methods, the Rnd function is used, in conjunction with the Randomize command, so that an integer number from 1 up to the total number of teachers is produced every time.

Issues concerning the types of constraints and the ways of confrontation during the construction of a teaching timetable for a secondary education school were also discussed. Another point of analysis was the search procedures for a valid teacher for a specified class during data processing. The search is performed with the use of specified counters, which enforce the redirection of the flow, from one to the other among three hierarchical different levels. These counters enable the application to work optimally, decreasing the time that is required for the creation of a timetable version. The application can auto-adjust the counters' thresholds in order to be more effective, taking into consideration the number of teachers. The optimal values of the thresholds were certified with extended tests, so the time needed for a functional timetable version is minimal.

The previous approaches, as it appeared, produce satisfactory results. At the end, it should be noted that the STB operation constitutes a different approach towards the solution of timetabling problems.

STB is currently cross-examined with similar applications. The comparison focuses mainly on checking its effectiveness. Ongoing research focuses also on using different programming languages (e.g. C++) for implementing the STB application. The main goal is to reduce the time needed for the timetable completion.

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Effects of Computer Skill on Mouse Move and Drag & Drop Performance

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Abstract - **The present study attempts to investigate difficulties that appear during mouse use control. It reports a series of experimental mouse tasks, using software constructed for this research. Our main objective is to investigate children and adults' performance on various drag & drop tasks. Differences in performance between high and low-skill computer users on low-level mouse operations tasks are also examined. Results revealed within-subject variation in performance on mouse operation tasks and performance differences among groups.**

I. Introduction

The integration of computers and other automated technologies into modern work settings has been rapidly increased during the last decade [1]. Curriculum designers across Europe follow that trend, introducing computer applications in the form of educational software into classrooms [2]; [3]. Due to this technological revolution an increasing number of people are confronted to the use of computers [1]; [4], or are involved in using some form of computer technology [5]; [6]. Thus, users are no longer restricted to computer specialists, but to a vast range of professionals [7].

Computers are an increasingly used learning tool within all levels of formal education. Studies report a transition from traditional teaching methods, such as lectures, into interactive teaching approaches [8]. These changes have a positive influence on student motivation. Computer-assisted instruction offers plenty of gains such as efficiency, portability, consistency and effectiveness [9]. The graphical user interface of the educational software used is not always static but rather dynamic and involves movements or changes [10]. The implementation of educational software and the use of computers introduced into schools seem to challenge not only the role of teachers, but students' learning strategies as well. Teachers have incorporated computers into their teaching practice although there are still some teachers who are reluctant to use computers, probably because they sometimes experience control difficulties [3] or exhibit stress symptoms [11].

A key notion to computer's effectiveness is usability. Despite the continuously expanding use of computers, we as yet know little of how children and adults respond to this novel situation, for example, how children interact with educational software, or how well children can handle input devices, or how human factors are associated with mouse use [12]; [13]. Another important question to be addressed is whether performance differences could be observed between children and teachers and consequently affect the control of educational software. Educational software, used in schools, is constructed for running under a graphical interface, which is the standard mode of human – computer interaction [14]. Despite that fact, educational software interfaces are not studied in detail [12]. Thus, practical issues concerning computer use need to be studied thoroughly. The use of a mouse, for example, is a topic that requires further investigation [13]. This necessity becomes imperative since *"a mouse is the input device children principally use to control a computer in school"* [15]. It is also the case that occasional computer users are sometimes under the impression that that they could easily use a mouse. But this certainty turns out to be fallacious when confronted with a specific task [16]. Simple hand movements, such as grasping the mouse or accurately pointing to a target become considerable obstacles for many people. Things get worse when tasks become more complex. Clicking, dragging or dropping a target, despite being frequent tasks while using a computer, are considered to be very difficult [17]. The necessity to explore users' efficiency to use a computer derives mainly from two reasons. The first is that previous relevant computer experience leads to improved performance [18]. The second stems from the fact that both students and teachers' ability to use a computer varies. Since "ease-of-use" is the key to successful application development [19], software designers should limit operational difficulties [18].

II. Methodology

One experimenter collected the data in order to control variations in testing. Experiments were conducted in an ordinary small office. Subjects sat on an ergonomic, height adjustable office chair, approximately 80 cm away from the screen, so that the visible screen surface was within their optical field. It should be noted that the light had no effect, such as reflections, on the computer screen.

A. Software and Hardware

For the needs of this research, specific software was constructed, using the object based, high level programming

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language Microsoft Visual Basic 6.0, under Microsoft Windows XP Professional Edition.

The desktop PC (IBM compatible) that was used during the testing process had a microprocessor P4 / 2.8 GHz / 512 RAM, a CRT display adapter 17''. The screen resolution was 800x600 pixels with a refresh rate of 85 Hz.

A typical wired optical two-button + wheel with USB connection, Microsoft Intellimouse type mouse was used for controlling the software with Microsoft Windows default settings. A typical dark blue mouse pad was also used.

The software involved a simple user interface. Icons or features that may mislead subjects or distract them were not used.

B. Experimental Procedure

A simple task was used for assessing "computer skill". A white frame and a progressive bar appeared on the screen. After 10 sec the progressive bar disappeared and two "open" square shapes appeared at the top of the screen. Subjects were asked to place (drag & drop) successively two round red-colour stimuli, with a diameter of 32 pixels, into the "open" shapes. After the completion of the task the elapsed time was recorded. A median split was employed for allocating subjects into one of the two computer skill categories.

The main experimental procedure did not exceed 10 min for all tasks. The experimenter used exactly the same verbal instructions to guide subjects through the experiments. Additional coloured cards were used to explain what exactly the subject was about to see on the screen and what actions he/she was supposed to take for successfully completing the tasks. The sequence of the five main tasks was rotated across sessions.

C. Sample

Overall, 131 subjects participated in this study. Individuals with special needs, either learning or kinetic, were excluded from the sample. All subjects had normal or corrected to normal vision and all of them were right-handed. Individuals, for all four groups, were allocated to one of the two computer-skill categories (Table 1).

TABLE 1
SUBJECTS ALLOCATED TO GROUP AND COMPUTER SKILL CATEGORIES

Group	High computer skill	Low computer skill
Primary school pupils	14	23
Secondary school pupils	16	15
University arts students	16	15
Primary school teachers	17	15
Total	63	68

D. Pilot test

Overall 18 subjects participated in the pilot test and all were excluded from the main experimental procedure. Data from the pilot test led to revisions both on software and calibration functions.

E. Description of experiments

Each subject completed five main experiments. Tests were presented in rotation for each subject.

As mentioned before, apart from the software and the hardware, coloured cards were also used in this research, which represented the interface of each experiment. The tests were the following:

Experiment 1 – Mouse move and click to create shapes with straight-edged sections and shapes with curved lines.

For the first test, a white frame and a progressive bar appeared. After 10 sec the progressive bar disappeared. Then, as shown in Figure 1 (Left), two different shapes appeared, consisting of straight-edged sections and on the right of each there was a set of 10 dots. From that moment the experiment was timed.

The experimenter had already prepared the subject verbally by showing the appropriate coloured card. After left clicking the mouse and moving it across the dots, the subject could rebuild the shapes. After completing the task the subject had to click on the "stop" button at the bottom of the screen. Thus, the time elapsed was recorded. When subjects managed to cover at least 7 of the 10 dots given, then the attempt was marked as "correct". Otherwise it was marked as "incorrect".

Fig. 1. Left: The interface for the first trial of the 1st test – Right: The interface for the second trial of the 1st test.

Soon after, a white frame and a progressive bar appeared again. After 10 sec the progressive bar disappeared. Figure 1 (Right) shows the software interface with two different curved-line shapes. On the right of each shape there was a set of 10 dots. From that moment the experiment was timed.

After left clicking the mouse and moving it across the dots, subjects could rebuild the shapes. At completion, subjects had to click on the "stop" button at the bottom of the screen. Thus, the time elapsed was recorded.

If subjects managed to cover at least 7 of the 10 dots given, then the attempt was marked as "correct". Otherwise it was marked as "incorrect".

Experiment 2 – Drag and drop with successive constant stimuli

For the second test, a white frame and a progressive bar appeared. After 10 sec the progressive bar disappeared. Figure 2 shows the software interface. From that moment the experiment was timed. The experimenter had already prepared the subject verbally by showing the appropriate coloured card. Subjects had to move the red round shape into the "open" square shape. In the case of failure, they could try to move it again as many times as required to succeed. After the first successful attempt another four attempts followed

(Figure 2). After the successful completion of all five attempts the elapsed time was recorded.

The red round shape had a diameter of 32 pixels, while the open square shape was 52x42 pixels. Tractive forces at some pixels around the open shape helped subjects place the red circle exactly into the right position.

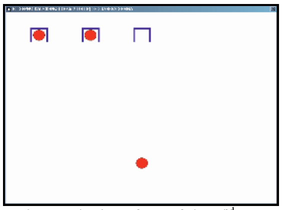

Fig. 2. The interface of the 2nd test – in the middle of an attempt.

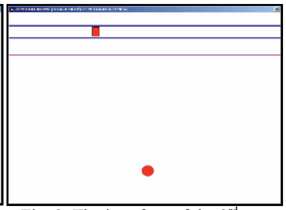

Fig. 3. The interface of the 3rd test.

Experiment 3 – Drag and drop on a moving (with constant velocity) stimulus.

For the third test, a white frame and a progressive bar appeared. After 10 sec the progressive bar disappeared. From that moment the experiment was timed.

A red square shape, sized 29x21 pixels, started to move, at a pre-determined speed, from the left to the right side of the screen and vice versa. The stimulus stopped at pre-determined positions. Simultaneously a red round shape of 32 pixels diameter appeared in the centre of the screen.

Figure 3 shows the software interface. Subjects had to move the round shape and place it under the square stimulus. One could attempt more than once in case of failure. Tractive forces at some pixels around the shape helped subjects place the red circle into the right position. This procedure was repeated five times and subjects had to complete five successful attempts. At the end of the task the elapsed time was recorded.

Experiment 4 – Drag and drop for a simple picture synthesis.

For the fourth test, a white frame and a progressive bar appeared. After 10 sec the progressive bar disappeared. From that moment the experiment was timed. Figure 4 shows the software interface.

The experimenter had already prepared the subject verbally by showing the appropriate coloured card. Subjects had to move the five available segments and place them correctly for reconstructing a picture.

The picture, in the bottom left corner of the screen, had dimensions of 300x206 pixels. Each of the five segments had dimensions of 60x206 pixels.

The experiment was timed right after the appearance of the picture on the screen. After completing the task, the elapsed time was recorded and an exit button appeared in the bottom right corner of the screen.

Each segment could be relocated if it was placed incorrectly. Tractive forces at some pixels around the shape helped subjects put the segments to the right place.

Experiment 5 – Drag and drop for a complex picture synthesis.

For the fifth test, a white frame and a progressive bar appeared. After 10 sec the progressive bar disappeared. From that moment the experiment was timed. Figure 5 shows the interface.

The experimenter had already prepared the subject verbally by showing the appropriate coloured card. Subjects had to move the picture's five segments and place them in the right position on the left of the screen. The picture, in the bottom left corner of the screen had dimensions of 339x246 pixels.

The experiment was timed right after the appearance of the picture on the screen. After completing the task, the elapsed time was recorded and an exit button appeared in the bottom right corner of the screen.

Each segment could be removed even if it was placed incorrectly. Tractive forces at some pixels around the shape helped subjects place segments into the right position.

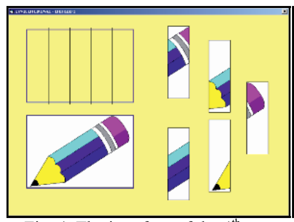

Fig. 4. The interface of the 4th test.

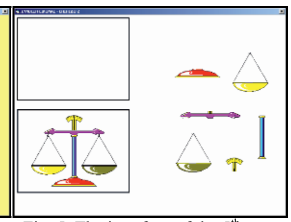

Fig. 5. The interface of the 5th test.

III. RESULTS

A. Experiment 1

The following tables show the mean accuracy scores and the mean duration times for the first experiment for all groups allocated to the two computer skill categories.

TABLE 2
MEAN DURATION TIME AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

Mean duration time (straight lines)	High skill		Low skill	
	M	SD	M	SD
Primary	32.44^{a}	8.34	35.96^{a}	9.74
Secondary	18.82^{b}	5.91	22.50^{b}	6.90
University	24.56^{b}	11.03	29.55^{a}	8.99
Teachers	20.27^{b}	5.74	23.92^{b}	6.11

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison (the Tukey HSD (*Honestly Significant Difference*) test is used to find significant differences between any pair of means in a balanced design).

TABLE 3
MEAN ACCURACY TIME AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

Mean accuracy (straight lines)	High skill		Low skill	
	M	SD	M	SD
Primary	85.71^{a}	23.44	80.43^{a}	36.12^{a}
Secondary	71.88^{a}	36.37	70.0^{a}	31.62
University	93.75^{a}	17.08	100.0^{b}	.00
Teachers	79.41^{a}	35.61	60.0^{a}	33.81

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison.

TABLE 4
MEAN ACCURACY AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

Mean accuracy (curved lines)	High skill		Low skill	
	M	SD	M	SD
Primary	85.71[a]	23.44	82.61[a]	28.64
Secondary	56.25[b]	44.25	63.33[a]	44.19
University	100.0[a]	.00	93.33[a]	17.59
Teachers	88.24[a]	28.11	66.67[a]	36.19

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison.

TABLE 5
MEAN DURATION TIME AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

Mean duration time (curved lines)	High skill		Low skill	
	M	SD	M	SD
Primary	21.98[a]	6.54	25.55[a]	6.46
Secondary	13.23[b]	3.92	16.88[b]	4.99
University	16.78[b]	4.87	22.29[a]	7.39
Teachers	16.60[b]	5.47	23.72[c]	7.33

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison.

Analysis of drawing shapes with straight-edged sections failed to reveal significant differences either on accuracy or on duration time scores between skilled and less skilled computer users across groups. The same results were also observed on accuracy scores for drawing shapes with curved lines. Analysis on duration time scores for university students and primary school teachers revealed significant differences between skilled and less skilled users. For university students significant difference was found on mean duration time between the two groups, $t(27)= -2.40$; $p<0.05$. The mean for high skill users was $\bar{x}=16.78$, SD=4.87, and for low skill users, $\bar{x}=22.29$, SD= 7.39. Significant difference was also found on mean duration time between the two computer skill categories in the primary school teachers' group. Specifically, $t(30)= -3.13$; $p<0.05$. The mean for high skill was $\bar{x}=16.60$, SD=5.47, and for low skill, $\bar{x}=23.72$, SD= 7.33.

A second analysis was carried out to compare performance between groups. For drawing shapes with straight-edged sections, a 2x4 ANOVA was calculated for accuracy scores. Computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) were the between-groups factors and accuracy scores was the dependent measure. There was a significant main effect of group, $F_{(3,123)}=5.59$; $p<0.005$. Despite the fact that accuracy scores did not vary significantly, the observed group effect can be attributed to variations found in the low computer skill category. Multiple comparisons (Tukey's HSD) revealed significant differences (for subjects assigned to the low skill category) between teachers and university students, and between secondary school children and university students (Table 2). There were no significant interactions.

A second 2x4 ANOVA was calculated for mean duration time scores with computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) as the between-groups factors and duration time scores as the dependent measure. There was a significant main effect of group, $F_{(3,122)}=18.58$; $p<0.001$. Tukey's HSD test showed significant differences between primary school children (low computer skill) and all the other groups. Significant differences were also revealed between primary school pupils (high computer skill category) and secondary school students (high and low computer skill category), secondary school students (high computer skill category) and university students (low computer skill category). Finally, significant differences were also found (Table 3) between teachers (high computer skill category) and university students (low computer skill category). There was also a significant main effect of computer skill, $F_{(1,123)}= 7.54$; $p<0.01$. Analysis revealed no significant interactions.

The same design was used for analyzing accuracy scores. A 2x4 ANOVA was calculated with computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) as the between-groups factors and accuracy scores as the dependent measure. There was a significant main effect of group, $F_{(3,123)}=7.71$; $p<0.001$. Multiple comparisons (Table 4) revealed significant differences between secondary school children assigned to the high skill category and university students (both for high and low skill category). Significant differences were also found between secondary school children assigned to the low skill category and university students (for high skill category). There were no significant interactions.

A 2x4 ANOVA was calculated for mean duration time scores, with computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) as the between-groups factors and duration time scores as the dependent measure. There was a significant main effect of group, $F_{(3,121)}=11.72$; $p<0.001$. Tukey's HSD test showed significant differences (Table 5) between primary school children (low computer skill) and secondary school students (high and low computer skill), university students (high computer skill) and teachers (low computer skill. Significant differences were also revealed between primary school pupils (high computer skill category) and secondary school students (high and low computer skill category) and teachers (high computer skill), secondary school students (high computer skill category) and university students (low computer skill category). Finally, significant differences were found between teachers (high computer skill category) and university students (low computer skill category). There was also a significant main effect of computer skill, $F_{(1,121)}= 21.73$; $p<0.01$. Analysis revealed no significant interactions.

B. Experiment 2

A 2x4 ANOVA was calculated with computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) as the between-groups factors and duration time scores as the dependent measure. There was a significant main effect of group, $F_{(3,123)}=22.26$; $p<0.001$. Tukey's HSD test showed significant differences (Table 6) between teachers and all the other groups for the "low computer skill" category. Significant differences were also revealed between primary school pupils and secondary school students and between secondary school children and teachers for the "high computer skill" category. There was a significant main effect of computer skill, $F_{(1,123)}=58.39$; $p<0.001$. Analysis also revealed a significant group X computer skill interaction, $F_{(3,123)}=8.22$; $p<0.001$.

TABLE 6
MEAN DURATION TIME AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

	High skill		Low skill	
	M	SD	M	SD
Primary	11.16^{a}	3.40	14.30^{a}	5.13
Secondary	7.08^{b}	1.72	10.67^{a}	4.12
University	9.06^{a}	1.70	11.81^{a}	2.80
Teachers	10.76^{a}	3.04	21.49^{b}	5.43

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison.

C. Experiment 3

A 2x4 ANOVA was calculated with computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) as the between-groups factors and duration time scores as the dependent measure. There was a significant main effect of group, $F_{(3,123)}=7.68$; $p<0.001$. Tukey's HSD test showed significant differences (Table 7) between primary school children and secondary school students, and between primary school children and university students for the "high computer skill" category. Significant differences were revealed between teachers and secondary school children, and between teachers and university students in the "low computer skill" category. It seems that primary school children and teachers form a distinct subset, which is differentiated significantly from the other two groups. Secondary school children and university students form a second subset. There was a significant main effect of computer skill, $F_{(1,123)}=23.07$; $p<0.001$. Analysis also revealed a significant group X computer skill interaction, $F_{(3,123)}=2.91$; $p<0.05$.

TABLE 7
MEAN DURATION TIME AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

	High skill		Low skill	
	M	SD	M	SD
Primary	9.94^{a}	5.81	10.81	3.43
Secondary	5.63^{b}	2.35	8.95^{a}	3.60
University	5.56^{b}	2.43	8.11^{a}	3.01
Teachers	7.06	3.19	13.43^{b}	5.95

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison.

D. Experiment 4

A 2x4 ANOVA was calculated with computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) as the between-groups factors and duration time scores as the dependent measure. There was a significant main effect of group, $F_{(3,123)}=10.80$; $p<0.001$. Tukey's HSD test showed significant differences (Table 8) between primary school children and university students for the "high computer skill" category. Significant differences were also revealed between primary school children and university students and between university students and teachers for the "low computer skill" category. There was a significant main effect of computer skill, $F_{(1,123)}=22.98$; $p<0.001$. There were no significant interactions.

TABLE 8
MEAN DURATION TIME AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

	High skill		Low skill	
	M	SD	M	SD
Primary	16.16^{a}	7.60	22.79^{a}	12.26
Secondary	12.36	7.80	16.55	6.84
University	8.99^{b}	2.46	12.23^{b}	3.16
Teachers	13.81	3.86	24.00^{a}	3.87

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison.

E. Experiment 5

A 2x4 ANOVA was calculated with computer skill (high and low) and group (primary school pupils, secondary school pupils, university students and teachers) as the between-groups factors and duration time scores as the dependent measure. There was a significant main effect of group, $F_{(3,123)}=13.38$; $p<0.001$. Tukey's HSD test showed significant differences (Table 9) between primary school children and all the other groups for the "high computer skill" category. Significant differences were also revealed between primary school pupils and university students, between secondary school children and university students. Finally, significant differences were revealed between university students and teachers. It seems that primary school children and teachers form a distinct subset, which is differentiated significantly from the other two groups. Secondary school children and university students form a second subset. There also a significant main effect of computer skill, $F_{(1,123)}=18.80$; $p<0.001$. Analysis also revealed a significant group X computer skill interaction, $F_{(3,123)}=3.92$; $p<0.05$.

TABLE 9
MEAN DURATION TIME AMONG DIFFERENT GROUPS AND COMPUTER SKILL CATEGORIES

	High skill		Low skill	
	M	SD	M	SD
Primary	26.14^{a}	8.05	27.03^{a}	7.56
Secondary	16.60^{b}	9.07	20.51^{a}	7.03
University	15.23^{b}	4.08	19.93^{b}	6.66
Teachers	19.21^{b}	5.42	31.83^{a}	8.67

a, b, c: means in the same row that do not share the same superscripts differ at $p<0.05$ in the Tukey's HSD multiple comparison.

IV. DISCUSSION

Our data suggests that different competence levels can be found in a normal classroom. This diversity poses difficulties when students or teachers have to use a computer.

We examined the performance of different groups on a number of drag and drop mouse operation tasks. Reaction and duration times were used as dependent variables. Subjects were allocated into two computer-skill categories (e.g. high and low skill). Five simple and complex experimental tasks evaluated performance on mouse drag and drop operations.

For test 1 significant variation was observed within the low computer skill category. Teachers achieved the lowest accuracy scores among participants. This outcome seems logical, since younger subject are more familiar with computer technology such as video games [20]. Duration times on the other hand, vary both across group and computer skill categories. Data supports the view that secondary school children and university students form a separate competence group, differing significantly from primary school children and teachers in relation to duration time scores. They seem to be more familiar with such mouse operation movements.

High accuracy scores were also observed for creating shapes with curved lines. A surprising outcome was the low score obtained by secondary school children. A closer look at the results led us to speculate that they had probably underestimated the difficulty level of the task. This is related to the duration times achieved by secondary school children. Homogeneous subsets were also observed in drawing curved lines. As mentioned before, primary school children and teachers showed significant performance difference to secondary school children and university students, in terms of speed.

Experiments 2 and 3 tested participants' performance on dragging tasks. The second experimental task assessed speed in dragging an object and placing it in a pre-determined position, while in the third experiment objects were placed in random positions. For both experiments skilled users were quicker at completing the task. Secondary school children and university students' dragging performance excelled that achieved by primary school children and teachers.

Drag and drop operation was tested in the fourth and fifth tests. Results were similar to those observed in the previous tasks. Computer skill was found to be the main element of performance variation among groups. But, we cannot underestimate the effect of group.

Often, educational software used in a typical classroom poses difficulties to users, who use a mouse as the main data entry and computer control device. It is not yet known how less skilled computer users interact with educational software, since research in this field has little to present. It is though a common belief among educators that novice or less skilled computer users consume a large portion of the available time in their attempt to use educational software. Thus time could be better exploited for acquiring new knowledge or skills.

V. Conclusions

Educational software used in the classroom offers teachers the ability to disengage from the standard teaching practice and engage in an interactive learning process. Control of this type of software demands simple mouse operations, such as mouse move, clicks and drag & drop. A number of usability factors may pose certain difficulties to some users. Difficulties in using a mouse for controlling a graphical interface are inhibitory factors for less skilled computer users. Within a normal classroom setting we often observe differences in mouse use performance between competent and less competent users.

Our results suggest that less skilled computer users often require more time to master educational software than to actually focus on the learning process. So, educational software designers should thoroughly investigate parameters linked to the ease-of-use and mouse constraints in mouse move and drag & drop tasks.

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Morphological Enhancement and Triangular Matching for Fingerprint Recognition

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***Abstract.* Among the principal problems for realizing a robust Automated Fingerprint Identification System (AFIS) there are the images quality and matching algorithms. In this paper a fingerprint enhancement algorithm based on morphological filter and a triangular matching are introduced. The enhancement phase is based on tree steps: directional decomposition, morphological filter and composition. For the matching phase a global transformation to overcame the effects of rotation, displacement and deformation between acquired and stored fingerprint is performed using the number of similar triangular, having fingerprint minutiae as vertexes.**
The performance of the proposed approach has been evaluated on two set of images: the first one is DB3 database from Fingerprint Verification Competition (FVC) and the second one is self collected using an optical scanner.

I. INTRODUCTION

An Automated Fingerprint Identification System (AFIS) can work in two different ways [6] [7]:

- Verification mode: the on-line biological signature is compared with the registered signature. With more details, a username is used to select the database item and perform a 1 -> 1 match.
- Identification mode: the on-line biological signature is compared with the registered signatures. With more details 1 -> n matches must be performed in order to select the candidate with the highest matching score.

Fingerprint are composed by a unique pattern of locally parallel ridge and valley with well defined orientation. To compare fingerprints in an Automatic Fingerprint Identification System (AFIS), local ridge structure characteristics, called minutiae, are usually used. The minutiae used in an AFIS are ridge ending and ridge bifurcation.

Generally, the AFIS performance depends by image quality. The ridge-valley structures are rather well visible in a poor quality fingerprint image because a wrong fingerprint impression produces imperfections, genuine features alteration and many false features introduction.

In addition, an incorrect image acquisition due to a wrong finger displacement produces partial fingerprint acquisition, roto-translation displacements and non linear deformation.

In this paper, a directional morphological filter, in image enhancement process for improving the clarity of ridge/valley structures, is applied. In addiction, an enhanced matching algorithm with respect to [14], has been used to overcame the effects of rotation, displacement and deformation between acquired and stored fingerprint. The matching score is related to the number of similar triangular, having fingerprint minutia as vertexes, in the acquired and the stored images.

In literature different approaches for fingerprint image enhancement were proposed [19][20]. Usually dedicated enhancement algorithms are time consuming applications.
Most of fingerprint image enhancement approaches are based on Gabor filter [17][18]. The Gabor approach is a frequency-selective and orientation-selective filter tuned by ridge directions and ridge frequencies in fingerprint. It works both on spatial and frequency domains.

Hong et al. [5] adaptively improves the clarity of ridge and valley structures in fingerprint images by a bank of Gabor filters.

In reference [8] a Gabor filter is applied on input images using 8 predefined directions and 4 predefined frequencies without the above computation.

In reference [11] the image is decomposed into 8 directional components using a directional filter bank to obtain directional energy distribution for each block.

In reference [12] the method in [5] is modified by discarding the inaccurate prior sinusoidal plane wave assumption.

In reference [13] a different approach respect Gabor filter has been proposed. A directional-frequency Fourier domain filter to reduce low and high frequency noise has been used.

Maio at al. in [9] have proposed an approach for extracting fingerprint minutiae directly from grey scale images, without binarization phase and ridge thinning phase.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 143–147.

In references [1] [2] Conti et al. present a match operator based on Tanimoto distance to overcame the roto-translation displacement achieving a better recognition rate.

The objective of the proposed fingerprint enhancement algorithm based on morphological filter is to overcome some limits of the standard fingerprint enhancement software, such as time consuming, ridge frequency calculation and empirical parameter selection, exploiting only local ridge orientation.

Experimental results are comparable to the TGF [5] with respect the average number of real extracted minutia, while execution time is decreased (more than 3 times faster than TGF [5]). The effectiveness of the proposed approach has been evaluated on two set of images: the first one is DB3 database supplied by Fingerprint Verification Competition (FVC2002) [16] and the second one is self collected database composed by 492 grey scale fingerprint images of 300 X 260 pixels, supplied by the Hamster Secugen optical sensor [15].

This paper is organized as follows: in section II an overview about morphological filters is presented, in section III and IV the proposed system, features extraction phase and the matching phase, is described, in section V the experimental results are proposed and finally in the section VI the conclusions are reported.

II. MORPHOLOGICAL FILTERS OVERVIEW

Greyscale morphological image processing is a non-linear shape-oriented technique that uses a small 3-D template known as Structuring Element (SE). The SE is positioned at all possible locations in the image's surface and it is compared with the corresponding neighbourhoods pixels in grey levels [4]. Greyscale erosion is the process of placing the SE under the grey level of the image and pushing it so far as it will go without any part of it rising above the landscape. In other terms, denoting an image by $\boldsymbol{f}$ and the SE by $\boldsymbol{s}$, the greyscale erosion $\boldsymbol{f} \ominus \boldsymbol{s}$ at a pixel $\boldsymbol{(x,y)}$ is

$$(f \ominus s)(x, y) = \min_{j,k} [\, f(x+j, y+k) - s(j,k) \,]. \tag{1}$$

where $\boldsymbol{j}$ and $\boldsymbol{k}$ index the pixels of $\boldsymbol{s}$.

Grayscale dilation is defined in a dual manner, thus greyscale dilation is

$$(f \oplus s)(x, y) = \max_{j,k} [\, f(x-j, y-k) + s(j,k) \,]. \tag{2}$$

where $\boldsymbol{j}$, $\boldsymbol{k}$ vary as before.

Two important compound operations are opening and closing. The opening of an image by a SE s is denoted $f \circ s$ and is defined as an erosion followed by a dilation.

$$f \circ s = (\, f \ominus s \,) \oplus s. \tag{3}$$

The closing of an image by a SE s is denoted $f \bullet s$ and is defined as a dilation followed by an erosion.

$$f \bullet s = (\, f \oplus \hat{s} \,) \ominus \hat{s}. \tag{4}$$

where $\hat{\boldsymbol{s}}$ is a rotated version of the structuring element $\boldsymbol{s}$.

III. FEATURES EXTRACTION PHASE

In Fig. 1 the full process proposed in [1][2] is illustrated with the main phases about image preprocessing and minutiae extraction process. These phases are: image acquisition and normalization, segmentation, directional image extraction, image enhancement process (ridge structure reconstructing), thinning, post-processing and minutiae extraction.

The used fingerprint images in this work have a non-uniformity brightness distribution due to different contact pressure between user's finger and sensor. Consequently an adaptive normalization algorithm [5] is adopted. This algorithm reduces the dynamic range of the grey scale between ridges and valleys.

Successively through image segmentation the corrupted and not recoverable regions are erased [6], while the good and its recoverable regions are returned.

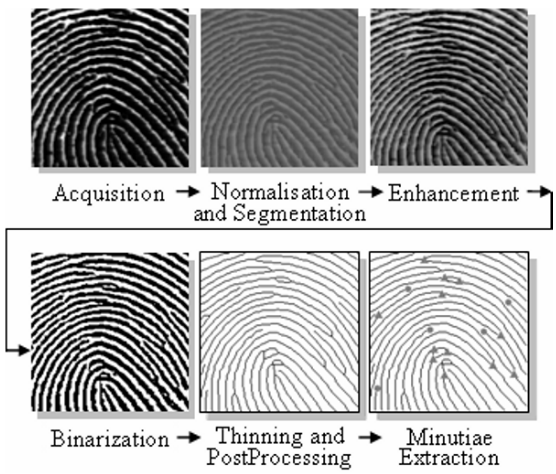

Fig. 1. Minutiae extraction process

Then the Local orientation image is calculated. It is an N*M block matrix (Local Direction Matrix: LDM) where the value in *(x, y)* is the local ridge direction in the *(x, y)* block of fingerprint image. In our approach the pre-arranged fingerprint direction number is $D=16$ and flank block is $r=8$ pixels. Discontinue ridges and not clear structures in fingerprint flow give some error in orientation image, but far from singular point the ridge local orientation varies slowly thus an orientation smoothing is applied to uniform the direction in a neighbourhood [5]. Then the proposed filter is applied on grayscale image and differently by the authors in [9] a fingerprint binarization is carried out.

Minutiae from skeleton image are extracted and a post-processing algorithm, to reduce extraneous false minutiae structures, is applied [1][2].

A. *The Proposed Enhancement Approach*

In this paper the authors use a morphological-based enhancement [10]. The main steps of the proposed fingerprint image enhancement algorithm are:

- Directional Decomposition
- Morphological Filter
- Composition

In Directional Decomposition Step (DDS), the Local Direction Matrix (LDM) is required (see fig. 2).

In this step, segmented image is decomposed in D Regions Of Interest (ROIi i=0, 1…D). Where D is the number of different local ridge orientations. The ROIs are non-overlapping subsets of the segmented image having uniform local ridges orientation.

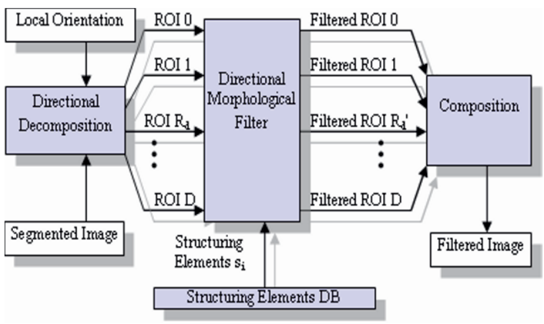

Fig. 2. Block diagram of the proposed fingerprint enhancement process.

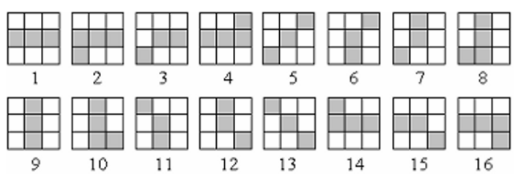

Fig. 3. SE Database with the own local orientation. Grey color denotes pixel locations with high grey level, white color denotes low grey level locations

The principal purpose of DDS is to gather, for each local direction *d*, only the regions that have the direction *d*. Successively, for each local direction *d* the relative regions of interest (ROI Rd) are enhanced with a morphological greyscale closing operation. Morphological closing is defined as a dilation followed by an erosion. The greyscale dilation removes small bright structure without enlarging dark feature, while greyscale erosion shrinks bright features of an image and enlarges dark features.

The directional morphological enhancement process acts on the region following their local orientation, closing holes and preserving the ridge from fragmentation and discontinuities.

A morphological direction-oriented structuring elements database has been created in off-line experimental phases (Fig. 3). For each ROI R_d with a local orientation d_i, a suitable structuring element s_i has been created and it represents the basis of a morphological greyscale closing operation. Our Structuring Element Database is composed by *D* different 3-D structuring elements, (see Fig. 3).

In the filtering algorithm for each given R_d, characterized by a local ridges orientation *d*, the proper structuring element is obtained from the Database (ES_d). Thus a greyscale morphological closing is carried out to act on R_d following their local orientation, closing holes and preserving it from fragmentation and discontinuities occurring along the ridge. In this way a regions $R_{dClosed}$ is obtained. In order to obtain the final greyscale filtered image, (see Fig. 2). Every possibility of composition overlapping or superposition between obtained ROI_s is avoided by registering translation to refer $R_{dClosed}$ to its position in original image spatial coordinates

In figure 4a) a corrupted image is reported. The figure 4c) shows the obtained experimental result after morphological filter application. The algorithm improves the clarity of ridge and valley structures and reconnects broken ridges.

Fig. 4. Demonstration of proposed technique: bullets are endpoints and triangles are bifurcations: (a) Original image (c) enhancement algorithm result (b) thinned original image with 52 extracted minutiae (d) thinned enhanced image with 30 extracted minutiae

IV. MATCHING PHASE

In the matching phase, the considered minutiae parameters are spatial position and ridge direction inside fingerprint image. The proposed matching is based on the similarity measure between triangular shapes present in a pair of fingerprints. The triangles have the candidate minutiae as vertexes, (see fig. 5).

The matching score is valued on a particular similarity measure proposed in [14].

In this paper the considered algorithm, to calculate the similarity measure, differs respect the original approach for the used parameters and the application scheme [3]. The matching score depends by maximum number of similar triangles between the stored fingerprint and a linear transformation of the on-line acquired fingerprint. The obtained minutiae set is compared with stored template, building, for each minutia of the first set, a list of corresponding potential minutiae using the Euclidean distance as metric. All considered triangular shapes pairs must have an Euclidean distance less of a given threshold. Each built triangle with minutiae of the first set is compared with all relative obtained triangles with corresponding potential minutiae, (see fig. 5).

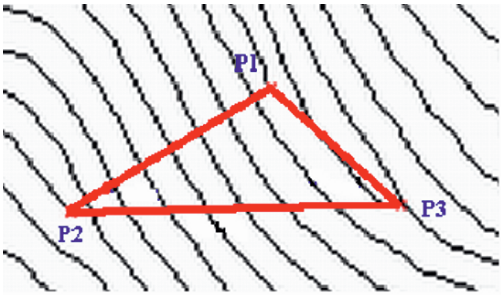

Fig. 5. A triangle having the candidate minutiae (P1,P2,P3) as vertexes.

In our approach respect to the criterion proposed in [14], only small dimensions triangles have been considered to be relatively tolerant to the problem of non-linear deformation. Let SM_A and SM_B be the sets of the extracted minutiae from stored and acquired fingerprint with displacement and rotation transformations, respectively. The main steps of triangular extraction phase follows:

1. for every minutia in SM_A the relative minutiae in SM_B, within a given Euclidean distance Th_{dist}, are found. If there is at least one corresponding then the minutia of SM_A is stored in a new set named SM'_A and its (one or many) corresponding minutiae matched are stored in SM'_B.
2. if the number of stored minutiae from SM'_A not exceeds a threshold Th_{cand}, than considered fingerprint images are not comparable; else go to step 3.
3. for every set of three minutiae in SM'_A, the obtained triangle is compared with the ones formed from every potential corresponding minutia in SM'_B. Each flank of every triangle must have a dimension less than a threshold Th_{flank}. The similarity criterions are in [14].
4. the number of similar triangles is returned.

Where Th_{dist}, Th_{cand} and Th_{flank} are thresholds fixed with experimental trials and based on fingerprints knowledge.

V. EXPERIMENTAL RESULTS

In order to evaluate the effectiveness of the proposed approach, two experimental trials were conducted on two different databases. In first trial the enhancement algorithm is not applied whereas in second experiment it is applied. In both cases minutiae features are extracted. Two datasets are used, the first one is the DB3 database supplied by the FVC2002 collected by using a capacitive sensor [16]; and the second one is a self created database with grey scale fingerprint images, captured by a Hamster Secugen optical sensor [15]. The first dataset is composed by 800 images of 300X300 pixel collected from 100 people. The second dataset is composed by 492 medium quality images of 300X260 pixel collected from 96 people.

To estimate the effectiveness of the proposed enhancement algorithm the matching score distributions was calculated on the two datasets by means matching algorithm proposed by the authors in [14]. The algorithm has been tested using the following matching parameters for both dataset: $Th_{dist}= 3$, $Th_{cand}= 10$ and $Th_{flank} = 30$.

In figure 6 and figure 7 the Receiver Operating Characteristic (ROC) curves, plotting the Genuine Accept Rate (GAR) against the False Accept Rate (FAR) at various matching threshold, are shown. We observe that with the filter application FAR decreases and GAR increases.

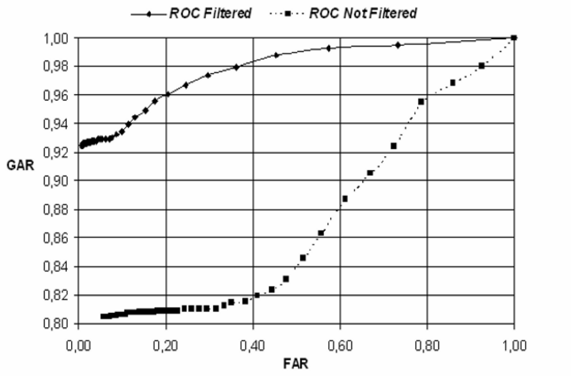

Fig. 6. ROC related to the FVC DB

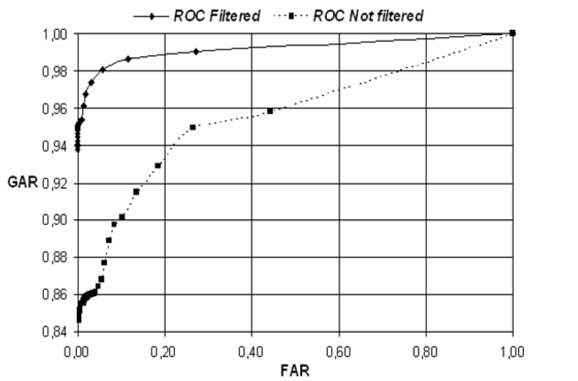

Fig. 7. ROC related to the Secugen optical sensor DB

In figure 4a) corrupted image with the results of our proposed algorithm is reported. The algorithm improves the clarity of ridge and valley structures and reconnects broken ridges. Consequently, the number of false minutiae decreases improving matching results. As example, the minutiae reduction for the images of figure 4a is about 42%. In addition in figure 4b a noisy area with a high number of false minutiae is highlighted. The same area after proposed filter application shows a reduced number of false minutiae (fig. 4d).

The Goodness Index (GI) [8] was calculated for the same images applying morphological and Gabor approaches. GI is defined as:

$$GI =[\Sigma_i q_i (p_i - m_i - s_i)] / [\Sigma_i q_i v_i]. \quad (5)$$

where p_i, m_i, s_i and v_i are respectively, the number of paired minutiae, the number of missed minutiae, the number of spurious minutiae and the number of minutiae visually located in i-th block.

The quality factor q_i is equal to 1, in useable blocks, and to 0 in discarded blocks. The ideal case is GI =1. The GI on original image (fig. 4a) is 0.03. The GI on enhanced image by means morphological filter (fig. 4c) is 0.43, while it is 0.45 on enhanced image by means TGF [5] .

On two sets of good and bad images extracted from the two datasets, we have calculated average number of real minutia after application of the morphological filter and the Gabor filter. Table I shows that our filter performances are comparable with the Gabor filter [5] performance taking into account the average numbers of real minutia.

TABLE I
AVERAGE REAL MINUTIA NUMBER IN THE TWO DATASET

	good images [12]	bad images [12]	good images [13]	bad images [13]	time (s)
morfological filter	28	29	34	36	3
gabor filter [4]	32	33	34	34	9

Using a Pentium IV 2.4GHz with JDK 1.4 in Win XP OS, enhancement algorithm execution time for each fingerprint is about 3 seconds. Java implementation of the Gabor filter proposed by Hong at al. in [5] requires about 9 seconds on the same images.

VI. CONCLUSIONS

In this paper a new fingerprint enhancement algorithm based on morphological filter and a triangular matching have been proposed.

The proposed filter exploits only the local ridge orientation, while Gabor filters calculate ridge frequency that is computationally expensive. Experimental results are comparable to the TGF [5] with respect the average number of real extracted minutia, while execution time is decreased (more than 3 times faster than implementation of the Gabor filter proposed by Hong at al. in [5]).

To evaluate the proposed approach in the context of an AFIS the improvement in False Accept Rate (FAR) and Genuine Accept Rate (GAR) was plotted by means ROC curves (fig.7-8). The Equal Error Rate EER is about 7% and with a FAR of 0.1% the GAR is 93% for the FVC DB [16]. The EER is about 2.5% and with a FAR of 0.1% the GAR is 95.5% for the Secugen optical sensor DB [15].

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About Sign Function and Some Extensions

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Abstract - Starting from the well.known definition (1) of the Sign function, a different compact definition has the advantage that leads also to compact definitions for other useful functions as, for example, the Dirac impuls and the unity step function. Extensions using single periodical functions leads to compact relations for some test periodical signals, the Ladder function and its derivative, the periodical Dirac impulse and then, to a 'quantized' operator. An extension for double (and/or multiple) periodical functions is also presented. It may find out applications in image and/or field theory.

I. INTRODUCTION

Sign function may be mathematically defined as :

$$Sgn[t] = \begin{cases} 1.\,for\; t > 0; \\ 0.\,for\; t \equiv 0; \\ -1.\,for\; t < 0; \end{cases} \tag{1}$$

Sign function may be also considered as the projection on the OX axis (vector $\vec{X}$) of a semi-infinite ray vector $\vec{V}$ with the origin in the origin of the system of coordinates, which spin around this point, normalized by the vector $\vec{X}$. As the vector $\vec{V}$ is infinite, its projection on the OX axis will be always equal to the equivalent vector $\vec{X}$ except for the case where these vectors are orthogonal. In this last case, the projection of the vector $\vec{V}$ will be zero. Another way to define the Sign function is to observe that the square root operator has the property to avoid the sign of the results. To get only the sign, we can then normalize a given function by its square root. For the real variable 't', [1], [2], [10], we get:

$$Sgn[t] = \frac{t}{\sqrt[2]{t^2}}; \tag{2}$$

This function has a discontinuity for t=0 an then, its derivative is not defined in this point. To avoid this discontinuity, we may define the Sign as a distribution:

$$Sgn[t] = \lim_{T \to 0} [Tanh[\frac{t}{T}]]; \tag{3}$$

The graphical representations of sign function for the relations (2) and (3) before the limit, with T=1, are shown in the figure 1.

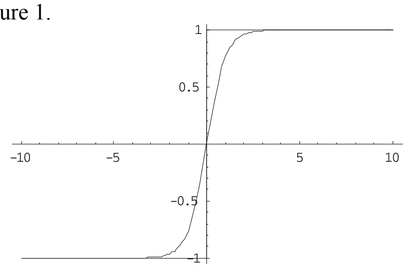

Fig. 1. Sign function and its equivalent as distribution for T=1.

We can observe that in this case $\sqrt[2]{t^2} = |t|$ and then:

$$\begin{cases} Sgn[t] = \frac{t}{\sqrt[2]{t^2}} = \frac{t}{|t|}; \\ \int Sgn[t].dt = \sqrt[2]{t^2} = |t|; \\ |t|.Sgn[t] = |t|.\frac{t}{|t|} = t; \end{cases} \tag{4}$$

The derivative of the Sign function (considered in a sense of a distribution) may be associated with the Dirac impulse function. From geometrical considerations we have for the unity step function $U[t]$ the representation:

$$U[t] = \frac{1}{2}.(1 + \frac{t}{\sqrt[2]{t^2}}); \tag{5}$$

and then it results for Dirac impulse function $\delta[t]$:

$$\delta[t] = \frac{d(U[t])}{dt} = \frac{d(Sgn[t])}{2.dt}; \tag{6}$$

II. EXTENSIONS.

A first extension may be made using single periodical uniform trigonometric functions. As the variable 't' is odd, we must use odd trigonometric uniform functions. Then, it results:

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$$\begin{cases} Sgn[t_p] = \dfrac{Sin[t]}{\sqrt[2]{Sin^2[t]}} = \dfrac{t_p}{|t_p|}; \\ \int Sgn_p[t].dt = ArcCos[Cos[t]] = |t_p|; \\ |t_p|.Sgn_p[t] = ArcCos[Cos[t]].\dfrac{Sin[t]}{\sqrt[2]{Sin^2[t]}} = t_p; \end{cases} \tag{7}$$

relations formally similar with (4). It can be observed that the derivative of $Sgn[t_p]$ (in the sense of distributions) leads not to periodical Dirac function but the derivative of t_p did. These functions are shown in the figure 2.

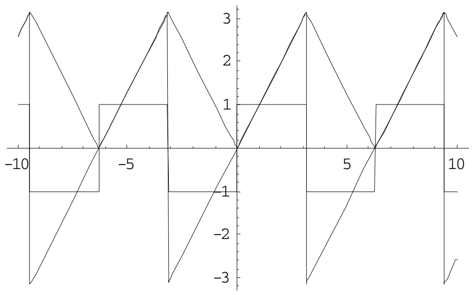

Fig. 2. Rectangular periodical function $Sgn[t_p]$, its integral $|t_p|$ and their product, the seesaw function.

It is interesting to remark that in this case the inverse function of $ArcCos[Cos[t]]$ loose the rank of its branch. This is seen in the figure 3.

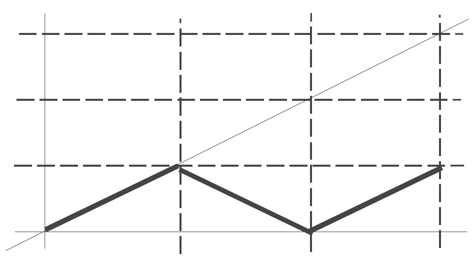

Fig. 3. Function $ArcCos[Cos[t]]$ and its argument.

An 'uniform step Ladder' function $L[t]$ may be built taking into account the relations (4) and (7) or simply, integrating the periodical Dirac function. Then:

$$\begin{cases} L[t] = t - t_{p1}; \\ \dfrac{d(L[t])}{dt} = \delta_1[t] = \sum_{k=-\infty}^{\infty} \delta(t-k); \\ LI[t] = \int L[t].dt; \end{cases} \tag{8}$$

with

$$t_{p1} = \frac{\dfrac{Sin[2.\pi.(t-0.5)]}{2\sqrt{Sin^2[2.\pi.(t-0.5)]}}.ArcCos[Cos[2.\pi.(t-0.5)]]}{2.\pi} - 0.5;$$

The function $L[t]$ is represented in figure 4 where we had added its 'interpolator LI[t]', the line y=t.

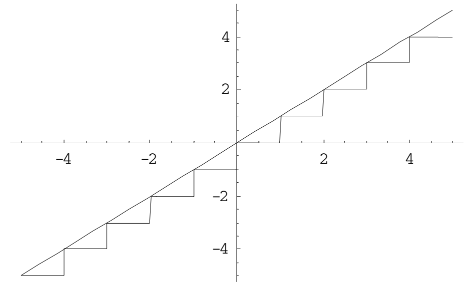

Fig. 4. The functions $L[t]$ and $LI[t]$.

The function $L[t]$ may be regarded as a 'quantification operator' for which only certain amplitude values (quanta) are possible. For $F[t]$ and $G[t]$ some real function, two situations are possible: $F[L[t]]$ where the quanta intervals remains constant but the amplitude of quanta may vary and $L[G[t]]$ where the amplitude of quanta remains constant but the quanta intervals may vary.

The function $LI[L[t+1]]$ is shown in the figure 5. We get the same results as for the numerical sum of an arithmetic progression represented by the numerical equivalent of the function $L[t]$. The advantage of this procedure consists in the possibility to compute the sum by integrals. As an other example is a ladder function where the amplitude of its steps grows as a factorial. We may use as 'interpolator' a modified Gamma function of Euler.

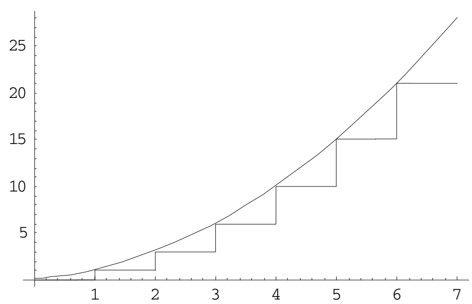

Fig. 5. $L[t]$ and $LI[L[t+1]]$ functions.

We have for $Gamma_m[t]$ and this new function $NF[t]$:

$$\begin{cases} Gamma_m[t] = U[t] - U[t-1] + U[t-1].Gamma[t+1]; \\ NF[t] = e^{Log[L[t]].(L[t]-1)}; \\ NF[t] = [t] - U[t-1] + U[t-1].Gamma[L[t]+1]; \end{cases} \tag{9}$$

Figure 6 shows these functions .

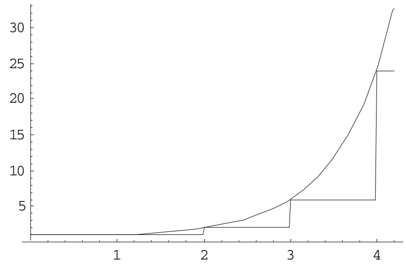

Fig. 6. $Gamma_m[t]$ and $NF[t]$ functions

An example for $L[G[t]]$ is shown in the figure 7 for $G[t] = 5.Sin[t]$. This case is useful for signals quantification.

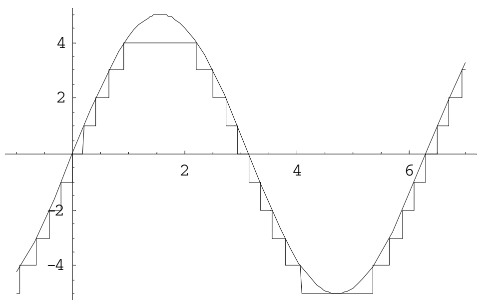

Fig. 7. $L[5.Sin[t]]$ and $5.Sin[t]$ functions.

III. EXTENSIONS IN COMPLEX

Still yet we had consider a single spinning angle. In space, we have to consider 2 (or more) spinning angles. A simple way to separate algebraically 2 spinning angles is to consider complex numbers: one real angle and one 'imaginary'. Then, a function having 2 periods leads to a complex argument. A well-known example is the Laplace transform kernel $e^{s.t} = e^{(\sigma + i.\omega).t}$ based on uniform analytic functions. An extension to 2 (or more) period meromorphic functions leads to elliptic functions.

For the function $u = t + i.\tau \Rightarrow i = \sqrt[2]{-1}$ we get:

$$\begin{cases} Sgn[u_c] = \dfrac{u_c}{\sqrt[2]{u_c^2}}; \\ \int Sgn[u_c].dt = \sqrt[2]{u_c^{\,2}}; \\ \int Sgn[u_c].dt .Sgn[u_c] = u_c; \end{cases} \tag{10}$$

relations very similar to (4). The real part of the function $Sgn[u_c]$ is shown in the figure 8.

We can remark that this function has a discreet period and that in the sense of distributions, its derivative leads to a special periodical 'line' impulse Dirac functions. Its imaginary part is equal to zero and its absolute value to one.

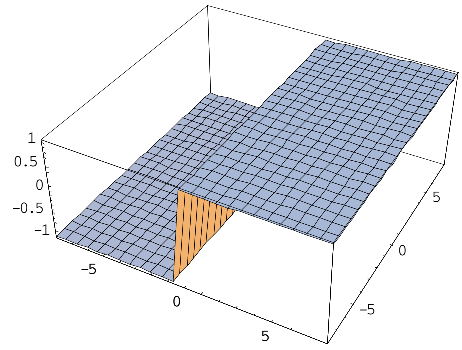

Fig. 8. Real part of $Sgn[u_c]$ function.

The uniform single-periodical trigonometric function $Sin[t]$ is a special case of the elliptical 2-periods meromorphic Jacobi function $Sn[u,m] \to u = x + i.y$ for the elliptical module $m \equiv 0 \Rightarrow 0 \le m < 1$. The Jacobi functions $Sn[u,m], Cn[u,m], Dn[u,m]$ have the properties:

$$\begin{cases} SN^2[u,m] + CN^2[u,m] = 1; \\ DN^2[u,m] + m.SN^2[u,m] = 1; \end{cases} \tag{11}$$

Then, using the same tips as for the relations (7), we get the relations:

$$\begin{cases} Sgn_{2p}[u] = \dfrac{Sn[u,m]}{\sqrt[2]{Sn[u,m]}}; \\ \int Sgn_{2p}[u].dt = \sqrt[2]{u_{2p}^2}; \\ \int Sgn_{2p}[u].dt .Sgn_{2p}[u] = u_{2p}; \end{cases} \tag{12}$$

with $\dfrac{EllipticF\left[ArcSin\{Cn[u,m]\}, \dfrac{m^2}{m-1}\right]}{\sqrt[2]{1-m}} = \sqrt[2]{u_{2p}^2}$. Relations (12) are formally very similar to (7) .These functions are shown in the figures 9, 10 and 11 as real part.

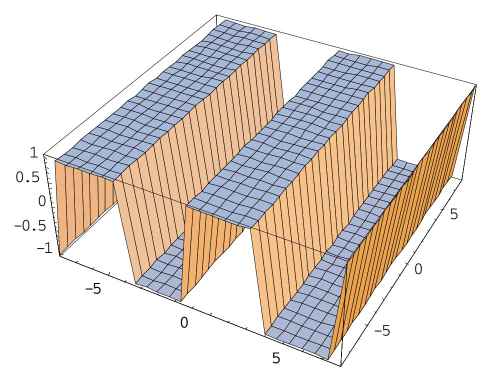

Fig. 9. Real part of $Sgn_{2p}[u,0.6]$ function.

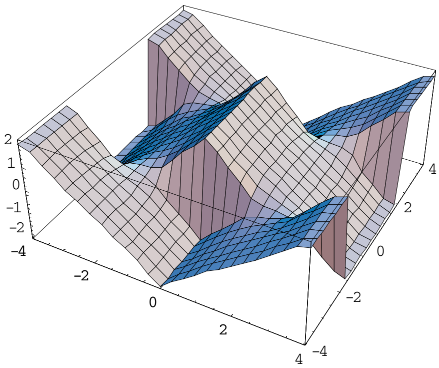

Fig. 10. Real part of $\int Sgn_{2p}[u\ ,0.6].du$ function.

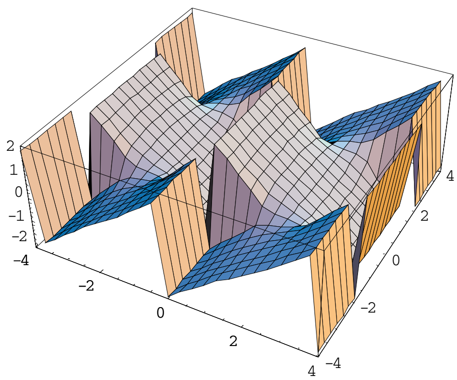

Fig. 11. Real part of $Sgn_{2p}[u\ ,0.6].\int Sgn_{2p}[u\ ,0.6].du$ function.

By analogy with the relation (7) we can define an equivalent Dirac function as:

$$\Delta[u] = \frac{d(u_{2p}[u])}{u}; \qquad (13)$$

which has not 'lines poles' but discreet as for elliptical functions. This function can be shown in the figure 12.

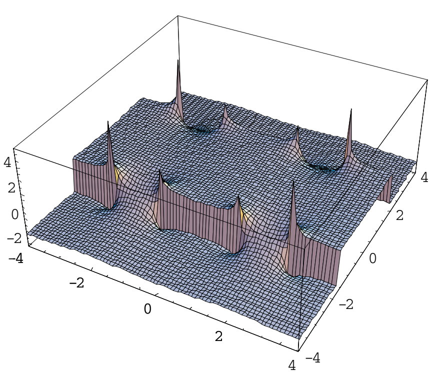

Fig. 12. Real part of $\Delta[u\ ,0.6]$ function.

By analogy with the relation (8) we can define a 3D equivalent Ladder function $L_3[u\ ]$:

$$L_3[u] = u - u_{2p} = u - Sgn_{2p}[u].\int Sgn_{2p}[u].du; \qquad (14)$$

This function is shown in the figures 13 and 14.

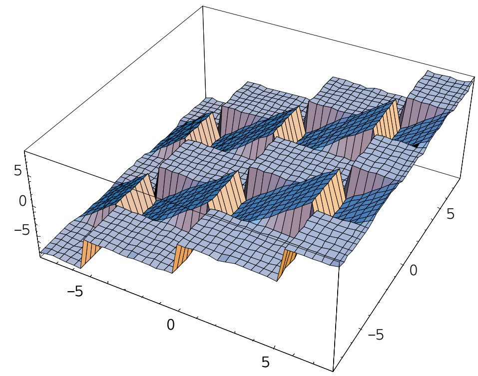

Fig. 13. Real part of $L_3[u\ ,0.6]$ function.

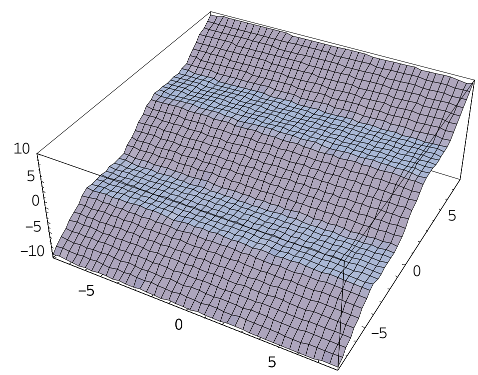

Fig. 14. Imaginary part of $L_3[u\ ,0.6]$ function.

If we consider the relations (3), (9) and (13), it can be observed that we can generalize the definition of a sign function as:

$$\begin{cases} \int Sgn[t].dt \Rightarrow \int \frac{dx}{\sqrt[2]{x^2}}; \\ \int Sgn_p[t].dt \Rightarrow \int \frac{dx}{\sqrt[2]{1-x^2}}; \\ \int Sgn[u].du \Rightarrow \int \frac{dz}{\sqrt[2]{z^2}}; \\ \int Sgn_{2p}[u].du \Rightarrow \int \frac{\alpha.dz}{\sqrt[2]{(1-z^2).(1+k.z^2)}}; \\ \int Sgn_{mp}[u].du \Rightarrow \int \frac{P(z).dz}{\sqrt[2]{(R(z))}}; \end{cases} \qquad (15)$$

where x is real, z complex and the last relation imply ultra elliptical functions with many periods (m complex periods).

Here, $P_3[z]$ is a polynomial in x and $R_3[z]$ another polynomial in z which degree is higher or equal to five.

Remarks:

1/ Except for the discontinuities, the Sign function is a normalized binary function. It may be regarded as the results of the projection of some 'spinning' angles.

2/. Special test functions (unity step, Dirac impulse, etc) may be easily defined started from the Sign function.

3/. Intelligent extension for simple periodical functions leads to the sampling theorem.

4/. The Ladder function makes discreet the amplitude of a function called 'interpolator'. It represents then a manner to 'quantize' a continuous function.

5/. The composed function $\sqrt[2]{u^2}$ loose the rank of the branch of the square root function which, in this case, is equivalent to the 'sign'. For signals as represented in the figure 2, the branch is selected due to energy conservation. The composed function $Log[e^u]$ loose also the rank of the branch of $Log[.]$.

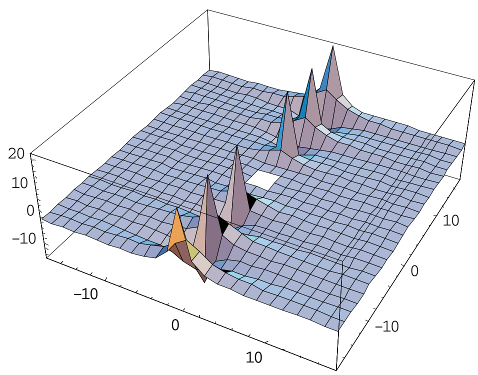

Fig. 15. Real part of $\dfrac{u}{Log[e^u]}$ function.

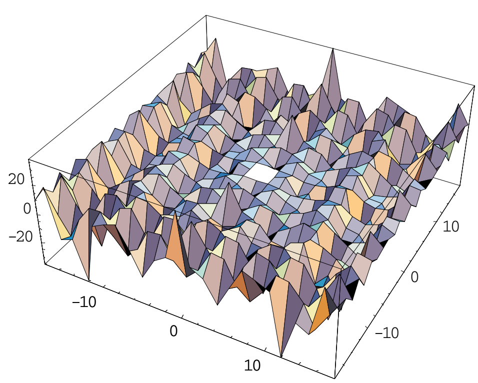

Fig. 16. Real part of $\dfrac{u}{InversSn\ [Sn[u,9.6],0.6]}$ function.

The normalized $\dfrac{u}{Log[e^u]}$ function is represented in the figure 15. Of course, others functions like these can be envisaged as, for example, Jacobi functions. Figure 16 shows this case. It can be remarked that in this last case some poles appears also on the real axis.

6/. A vector extension of Sign function leads to $Sign[\vec{r}] = \dfrac{\vec{r}}{\sqrt[2]{\vec{r}\bullet\vec{r}}} = \dfrac{\vec{r}}{\sqrt[2]{|\vec{r}|^2}}$ and its integral to $\int Sign[\vec{r}].d\vec{r} = \sqrt[2]{\vec{r}\bullet\vec{r}} = \sqrt[2]{|\vec{r}|^2} = S(\vec{r})$ which correspond to the arc of the hodograph of the vector $\vec{r}$. Here we had denoted by ' ● ' the dot product.

7/. The sign function may be interpreted as a 'choice function': for example: 'yes' correspond to 1, 'no' to 0 and 'may be' represent the discontinuity at zero point. The periodicities introduced by trigonometric functions leads to a 'multiple choices with the memory of the precedent choice' and then to a Ladder function. The introduction of elliptical Jacobi functions leads to a 'double multiple choices with the memory of the precedent choice'.

IV. CONCLUSIONS

A compact relation for the Sign function as in (2) leads also to compact relations for some very important signals in signal processing theory as, for example, the Dirac impuls and step unity function [1], [2], [10]. The Sign function may be regarded as a normalized binary function. The discontinuity for its argument equal to zero may be interpreted as a third possibilty ('may-be'). Extension for trigonometric periodical functions leads to compact relations for some periodical 'test signals' as rectangular, triangular, seesaw, periodical Dirac impulse function, Ladder function, etc. [3], [8], [9], [10]. The big advantage is the fact that these relations may be used to process signals more easily than with the old representations and leads mainly to compact representations too. The quantized signals are a good example. Extension of the compact Sign function representation allows also to compute some sum by integrals. Extension in complex leads to relations with a form very similar to those for the trigonometric functions. The advantage of the use of elliptical functions which generally represents electromagnetic fields in the electrical motor air-gap or the behavior of mechanical gyroscopes, is that the resulting extensions presents only discreet poles. These new tools may be used in 2D and 3D representation as, for example in [14] and/or field theory.

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Ambiguity and Contradiction From a Morpho-Syntactic Prototype Perspective

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***Abstract*— In this paper, the contradiction and ambiguity problems in Natural language Processing are briefly introduced. We also present the morpho-syntactic WIH (Web Intelligent Handler) prototype and the overall approach it takes to process any Spanish text. Finally, we analyze how it processes Spanish sentences with contradictions or ambiguities using its own perspective, despite deeper linguistic considerations.**

Index Term — Text-Mining, Automatic summarization, morphosyntactic analysis.

I Introduction

Communication through written texts has a certain coherence [1] and structure[2] according to the type of text. When text is coherent, there is a unique sense in it (named intra-textual coherence). But there is a conceptual coherence that exceeds the text. It is inherent of the culture and historical situation. This paper does not study any kind of coherence. It just analyzes parts of texts. In the following sections, the contradiction and ambiguity linguistic concepts are introduced along with some of the state of the art in that area. Afterwards, WIH (Web Intelligent Handler) prototype is presented and tested in several test cases with contradiction and ambiguity.

II Contradiction

Contradiction has been studied since Aristotle [3]. In general, it is considered as a concept:

-Logical or analytic: as a reasoning distortion. One example is the Aristotle no contradiction principle.

-Dialectic or synthetic: in a philosophic context, as an expression with its own essence. Unamuno Kiekergaard and Hegel had done part of this work.

From the computational perspective, only the first alternative can be considered. Work done with contradiction could be for Natural Language Processing frameworks [4], Distributed processing, Knowledge Based Systems [5] [6] [7] [8], etc. Some working prototypes are ASKER [9], CLASSIC [10], ECD [11], and XLE [4].

III Ambiguity

Written text is mostly disambiguated by the reader knowledge of its context and its main topic [12] [13]. But ambiguity is intrinsic to natural Languages morphology, syntax and semantics [2]. They can be classified as:

Lexical ambiguity: it is generated by the polysemy of a word.

Syntactic ambiguity: due the combination of the syntactic structure with certain nesting and embedding clauses.

Semantic ambiguity: originated by the logical structure of a sentence (not related to subordinated sentences as in the previous case).

Anaphora: originated by textual components without semantics. But they are syntactically right.

Some of the work in this area is related to lexical [14] [15], syntactic [16], morphologic [17], phonemic [18] [19], semantic [20] [21] levels or certain combination of them [22] [23]. In other prototypes as CLICK-TALP this task is performed manually [24] [25].

IV WIH Prototype

The Web Intelligent Handler (WIH) is a partially working prototype to process Spanish based web content in order to generate a meta-web for browsing and querying support. It was first introduced in [26]. It has a three-layered design (see Fig. 1):

-Internal Structure: gets data and metadata from the WWW and processes it to derive a set of Homogenized Basic Elements (HBE). These elements constitute a representation in an internal language.

-Virtual Structure: processes the actual stream of HBE and makes a structure named E_{ci} for the name in Spanish Estructura de Composición Interna, Internal Composition Structure in English. An E_{ci} is an oriented graph representing a statement in the original text. Sets of E_{ci} are then processed to make an E_{ce} (for the name in Spanish Estructura de Composición Externa, External Composition Structure in English, a supra-structure composed by a set of E_{ci} structures, all of them related to the same text).

-Visual Structure: it works with a Virtual Network composed by the set of E_{ci} and E_{ce} structures. It can be considered as an interface between the Virtual Structure and any user.

The WIH prototype has implemented a first version of the Internal Structure and Virtual Structure. In the Virtual Structure, there is a set of components to perform: the composition of E_{ci} and E_{ce} structures (Composition Engine, CE), insertion into the Virtual Net (Assimilation Engine, AE) through a set of functions (named effect functions, f_e) regulated by a set of dynamic parameters (metric functions, fm). All the activity is controlled by a feedback system (composed by a general controller System Controller, a Manager for f_e named Metrics Manager, and a manager for fm named Metrics Engine).

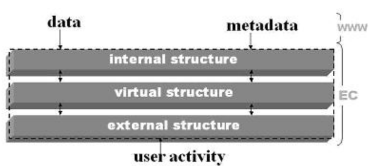

Fig. 1.Three layered structure of the EC. Data and metadata are extracted from the WWW.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 154–158.

The system performs all the activity through a set of f_e that can be changed dynamically. Some of such effect functions define the way a set of HBE is transformed to an E_{ci}, and how a set of E_{ci} is converted into an E_{ce}. For this activity, a metric p_o was defined as a set table to categorize each HBE and therefore determine how to process it. In general terms E_{ci} weight is in $p_o^{Eci} \in [-1, +1]$, and the range for E_{ce} is $p_o^{Ece} \in [-1, +1]$. Values near 0.0 denote closeness to the main topic. The system uses a combination of weight, descriptor values and sequence in the sentence to mark some HBEs as *indicator words*.

A The Opposite HBE

From the WIH perspective, the contradiction is not directly processed. It remains to the reader the interpretation and solving of them. But when certain Spanish negative prefixes are detected the related HBE is detected as an *opposite HBE*. This kind of detection is used to bias the value of p_o^{Eci} related, and therefore to reflect some subjacent handling of words in the sentence and text. In these cases, the weight p_o is the multiplied by –1. This value is combined with the rest of the HBE weights within the same E_{ci} (namely p_o^{Eci}), resulting in a p_o^{Ece} weight. Table 1 shows a sample of prefixes and words considered this way. The main characteristics of this approach are:

-The opposite pattern list is mutable and perfectible. The original p_o value shares the same property.

-The patterns are related to the f_e active set. A change in the set could make a change in the actual list.

-The valuation is just a rough qualification of the original text words.

B Ambiguity in WIH

Several algorithms can be used to solve the ambiguity problem, most of then use Natural Language Processing (NLP) which is a hard task due the language expressions are related to writer's culture, education, geographical situation, etc. [8]. The WIH prototype [26] has many differences with NLP:

-In NLP it is required to process the entire document. WIH is not intended to process semantics but to extract certain features and words to represent it approximately.

-NLP works at a semantic level. WIH performs a simple morpho-syntactic processing. It does just an analysis in the neighbor of each word.

-NLP considers five main levels of information: lexical, morphologic, syntactic, semantic and pragmatic. WIH just makes use of some aspects from the first three levels.

-In NLP each level has to be robust enough to support upper abstraction levels. In WIH this is not necessarily true due to the simplicity of the process.

TABLE 1
SPANISH WORD WEIGHTS.

Opposite
No X
Sin X
desX
inhX
antiX
disX

The approach followed in the prototype does not intend to solve ambiguities but to detect and use them as a weighing bias of the word and sentence (through p_i^{Ece} and p_o^{Ece} respectively). Each time one of such words is detected, a special non-zero value is used for the corresponding HBE to express a vagueness degree of the related statement. Table 2 shows part of the Spanish words and prefixes detected. It is important to say that the list is mutable and related to the actual set of f_e as in the previous section's list.

V CASE STUDIES

The original test cases were extracted from web sites in Spanish but they were translated into English here. As the HBE describes many morpho-syntactic aspects of the related words, it has a complex structure. For simplicity they were replaced by the Spanish word from the site followed by its p_o value within parenthesis.

It is important to note that the system sometimes can decide to eliminate some words and not to represent them into a HBE. All the fields and data presented here are extracted automatically by the system.

A Contradiction

Three cases of Spanish contradictions were processed. For each one, the sentence is first presented in Spanish and translated in parenthesis. HBE for E_{ci} and E_{ce} are finally described with the originating words. All the web pages processed have the orchids as main topic.

1) Test case 1:

TEXT: "Las orquídeas vegetativamente son muy diferentes pero iguales."(Orchids are vegetatively very different but similar).

URL: orquidea.blogia.com temas – que-es-una-orquidea-.php

E_{ci}WORDS: orquídeas (0.0) + vegetativamente (0.0) + son (0.0) + muy (0.7) + diferentes (0.0) + pero (0.0) + iguales (0.0)

E_{ce}WORDS: orquid (0.0) + son (0.0)

As can be seen, the semantic contradiction is not taken into account and remains hidden in the original text. The system detects some important information (in fact a definition) and promotes two HBEs to E_{ce} level. At this time, the user that is looking for information at an E_{ce} level is notified there is a definition, but is not provided with the definition itself (the information content value at the E_{ce} is 0.0).

2) Test case 2:

TEXT: "Las estructuras reproductivas están fusionadas separadas.". (Reproductive structures are merged separately).

URL: orquidea.blogia.com temas – que-es-una-orquidea-.php

E_{ci}WORDS: estructuras (0.0) + reproductivas (0.0) + estan (0.0) + fusionadas (0.0) + separadas (0.0)

E_{ce}WORDS: none

Table 2. Ambiguity patterns

Ambiguity
muy X
tan X
algo X
poco/a(s) X
mucho/a(s) X
bastante(s) X
escaso/a (s) X

The contradiction is not taken into account again. Even the system decides there is no important information in the sentence and does not promote any HBE.

3) Test case 3:

TEXT: "Tienen y no tienen dos tipos básicos de crecimiento simpodial.". (They have and don't have two main kinds of sympodial growth).
URL: orquidea.blogia.com temas – que-es-una-orquidea-.php
E_{ci}WORDS: Tienen (0.0) + no (-1.0) + tienen (0.0) + dos (0.0) + tipos (0.0) + básicos (0.0) + crecimiento (0.0) + simpodial (0.0)
E_{ce}WORDS: none

The contradiction is not taken into account again. The system does not find important information in the sentence and does not promote any HBE.

B *Ambiguity*

In this section three test cases for each kind of ambiguity were selected. The information provided is organized as in the previous section, but here indicator words of the E_{ci} are denoted with an asterisk to make explicit the ambiguity. An SAQ (Spanish Ambiguity Question) is included to make apparent the ambiguity in the Spanish language.

1) Lexical ambiguity: Test case 1

TEXT: "Ha despertado las más inimaginables pasiones en los hombres.".(It has inspired the most unimaginable passion in men/mankind).
AMBIGUITY QUESTION: whose passion is? Men/mankind?
URL: es.wikipedia.org wiki Orchidaceae
E_{ci}WORDS: ha (0.0) + despertado (0.0) + más (0.0) + inimaginables (-1.0) + pasiones* (0.0) + hombres(0.0)
E_{ce}WORDS: pasiones

The prototype finds the HBE related to the word pasiones to be the one that express the essence of the sentence and promotes it to E_{ce}. It is therefore denoting that the phrase is speaking something regarding orchid passion with lack of interest about whose passion it is. Note that the special combination of the mopho-syntactic characteristics with the closeness to p_o(inimaginables)^{1}0 results in the word pasiones to generate an indicator HBE.

2) Lexical ambiguity: Test case 2

TEXT: "se conocen plantas recolectadas a mediados del siglo pasado que todavía están creciendo y floreciendo saludables en muchas colecciones". (Last century collected plants are known to be still growing and flowering healthily in several collections).
SAQ: collection as a group or as part of collections stock?
URL: es.wikipedia.org wiki Orchidaceae
E_{ci}WORDS: conocen (0.0) + plantas (0.8) + recolectadas (0.0) + mediados (0.0) + del (0.0) + siglo* (0.0) + pasado (0.0) + todavía (0.0) + están (0.0) + creciendo(0.0) + floreciendo (0.0) + saludables (0.0)
E_{ce}WORDS: siglo

The prototype finds the HBE related to the word siglo to be the one that expresses the essence of the sentence and promotes it to E_{ce}. It expresses the sentence is describing something about century and orchids. Note that this indicator is found even with all p_o values being zero.

3) Lexical ambiguity: Test case 3

TEXT: "Las orquídeas son realmente las flores de lo superlativo ". (Orchids are really flowers of superlative).
SAQ: superlative as something big or cute?
URL: es.wikipedia.org wiki Orchidaceae
E_{ci}WORDS: orquídeas(0.0) + son (0.0) + realmente (0.0) + flores (0.0) + lo (0.0) + superlativo (0.0)
E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one (the original text describe main considerations about taking care of orchids and general description).

4) Syntactic ambiguity: Test case 1

TEXT: "¿Sabías que la orquídea Brassavola Digbyana es flor nacional de Honduras?" (Did you know Brassavola Digbyana orchid is the Honduras national flower).
SAQ: is it from Honduras or is the flower representing Honduras?
URL: 63.173.68.43 sites 7dias content.cfm id 276 PageName Insolit
E_{ci}WORDS: sabías (0.0) + orquídea(0.0) +Brassavola (0.0) + Digbyana (0.0) + es (0.0) + flor (0.0) + nacional (0.0) + Honduras (0.0)
E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one. But it is important to say the original text described mainly the Brassavola´s maintenance requirements.

5) Syntactic ambiguity: Test case 2

TEXT: "La parte de la flor que produce el polen." (The part of the flower that produces pollen).
SAQ: is it speaking of part of a plant or about a plant?
URL: orquidea.blogia.com temas -que-es-una-orquidea-.php
E_{ci}WORDS: parte(0.0) + flor (0.0) + produce (0.0) + polen (0.0)
E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one without studying the ambiguity.

6) Syntactic ambiguity: Test case 3

TEXT: "Las orquídeas son realmente las flores de lo superlativo". (Orchids are really flowers of superlative).
SAQ: superlative as something big or cute?
URL: orquidea.blogia.com
E_{ci}WORDS: orquídeas(0.0) + son (0.0) + realmente (0.0) + flores (0.0) + lo (0.0) + superlativo (0.0)
E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one. It is important to say this case was extracted from a very small dictionary at the end of the page.

7) Semantic ambiguity: Test case 1

TEXT: "Se corta unos 3 cms por arriba y por abajo de la yema." (Perform a cut 3 cm upper and bellow the leaf bud).
SAQ: 3 cm bellow and 3 cm upper? Or 3 cm upper and just bellow the bud?
URL: orquidea.blogia.com
E_{ci}WORDS: se(0.0) + corta (0.0) + unos (0.0) + cms (0.0) + arriba (0.0) + abajo (0.0) + yema (0.0)
E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one.

8) Semantic ambiguity: Test case 2

TEXT: "Según he leído en algunas webs, se puede acelerar agregando al agua algunas hormonas, pero yo nunca las he usado, así que no opino al respecto.". (According to some webs, it can be accelerated by adding some hormones, but I've never used them, so I don't give any opinion).

SAQ: did not use hormones at all or did not use them for this?

URL: orquidea.blogia.com

E_{ci}WORDS: según(0.0) + he leído (0.0) + algunas (0.3) + webs* (0.0) + puede (0.0) + acelerar (0.0) + agregando (0.0) + agua (0.0) + algunas (0.3) + hormonas (0.0) + pero (0.0) + yo (0.0) + nunca (0.0) + he (0.0) + usado (0.0) + así (0.0) + no (-1.0) + opino* (0.0)

E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. Despite all the statement is not considered as part of the main topic, the system finds that there is a secondary topic regarding an opinion and other webs.

9) Semantic ambiguity: Test case 3

TEXT: "A cada yema le he quitado la especie de piel que la cubre en forma de triangulo para dejar al descubierto la yema con unas pinzas." (I have peeled off each leaf bud to leave uncovered them with tweezers).

SAQ:is the peel triangular? Has it been taken off with this cutting shape ?

URL: orquidea.blogia.com

E_{ci}WORDS: cada (0.0) + yema (0.0) + he (0.0) + quitado (0.0) + especie* (0.0) + piel* (0.0) + forma* (0.0) + triángulo (0.0) + dejar (0.0) + descubierto (0.0) + yema (0.0) + unas (0.0) + pinzas (0.0) + piel (0.0) + cubre (0.0)

E_{ce}WORDS: especie, piel, forma

The prototype finds the words to express there is a mention of this topic but it does not find interesting to describe the technique details. The ambiguity is preserved within the E_{ci} to be solved by the reader.

10) Anaphora: Test case 1

TEXT: "Unas serán fértiles, otras estériles.". (Some of them will be fertile, others will be sterile).

SAQ: who?

URL: orquidea.blogia.com

E_{ci}WORDS: unas(0.0) + serán (0.0) + fértiles (0.0) + otras (0.0) + estériles (0.0)

E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one.

11) Anaphora: Test case 2

TEXT: "Debajo de esta hay una pequeña "ramita"." (Bellow it there is a small branch).

SAQ: below what?

URL: orquidea.blogia.com

E_{ci}WORDS: ramita(0.0)

E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one. In the original text it is a part of a more complete description on how to handle leaf buds.

12) Anaphora: Test case 3

TEXT: "Con ello conseguiremos que les llegue más luz.". (With that we allow them to get more sunlight).

SAQ: who?

URL: orquidea.blogia.com

E_{ci}WORDS: con(0.0) + conseguiremos (0.0) + llegue (0.0) + más (0.0) + luz (0.0)

E_{ce}WORDS: none

The prototype doesn't find an indicator HBE. The weight values are also all 0.0. As a consequence all the statement is considered as an irrelevant one.

VI Conclusions

Some real cases handling contradictions and several ambiguities were presented. They were processed by WIH and handled as in any other case without being disturbed by the lack of enough context information. On the other hand, proposals like Classic [10] need to go further, with a detailed purpose clause analysis, and try to encode linguistic knowledge about actions as a way to provide some context to interpret new statements through a formal analysis, making the whole process too complex.

In spite of the fact WIH neither disambiguates nor eliminates contradiction, it is able to extract main words and represent the main sentence content into an E_{ce} level. In contrast, in systems like Asker [9] the contradictions and ambiguities are detected, collected and reorganized into structures that need to be constantly updated and used to overcome contradictions while performing text processing.

Other proposals such as Click-talp [24], perform a morpho-syntactic level annotation, while WIH does all its work skipping that step.

From all this, it can be said that p_o could be a simple tool to provide support to some NLP.

VII Future work

There is a lot of work that remains yet: to tune and complete the list of p_o weights and to test them against real cases in order to validate results. Although this paper is a summary of the results on 100 web pages it is important also to perform a test with a higher number of pages. It will be very interesting to perform an extension of WIH to other languages as well.

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A Metric for Automatic Word Categorization

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***Abstract*—This paper presents a metric to be used by the working prototype WIH (Web Intelligent Handler). This metric (referred here as p_o) is designed to reflect main topic words and discriminate certain text profiles through word weightings. The actual version is designed only for Spanish web texts. Statistical analyses show that it is possible to differentiate text profiles upon p_o behavior. A poll is presented also, showing that it is a good main words discriminator. This paper is posted here as a new algorithm useful for Spanish text processing.**

Index Term — Text-Mining, Automatic summarization, morphosyntactic analysis.

I Introduction

Document handling is a part of index and retrieval process in the Web. It is a hard task due the complexity involved. Many approaches have been developed: noun phrases processing [1], morphemes processing [2], morphosyntactic analysis [3], etc. Sometimes the words searched in the documents are *expanded* with related words to improve recall. There are several alternatives to accomplish this work: expand upon morphology and syntax relations [4], morphosyntactic normalization of texts [5], conceptual and phonologic frames for word processing [6], etc.

Additional approaches have been developed for automatic paraphrasing ([7], [5], etc.), summarization of texts ([8], [9], [10], [11], etc.) and profiling of language usage of documents in the web ([12], [13], [14], [15], etc.). The contribution of this paper is twofold: to propose a metric to allow a kind of qualification of written texts and to provide an automatic word weighting for summarization activity.

Several mentioned algorithms use Natural Language Processing (NLP) which is a hard task due to the language expressions related to the writer's culture, education, geographical situation, etc. [16]. This paper presents a very simple proposal to process Web Spanish texts as part of the WIH (Web Intelligent Handler) prototype [17]. Its approach has many differences with NLP:

-In NLP it is required to process the entire document. WIH is not intended to process semantics but to extract certain features and words to represent it approximately.

-NLP works at a semantics level. WIH performs a simple morphosyntactic processing. It does just an analysis in the neighbor of each word.

-NLP considers five main levels of information [18]: lexical, morphologic, syntactic, semantic and pragmatic. WIH just makes use of some aspects from the first three levels.

-In NLP each level has to be robust enough to support upper abstraction levels. In WIH this is not necessarily true due to the simplicity of the process.

The remainder of this paper is organized as follows: section II presents the WIH prototype, which is the system that actually makes use of the proposed metric p_o; section III describes the metrics constraints; section IV defines p_o, section V shows statistical foundations; section VI, conclusions; and finally section VII states the main work remaining to be done.

II WIH Description

The Web Intelligent Handler (WIH) is a partially working prototype, first introduced in [17]. It has a three layered design (see Fig. 1):

-*Internal Structure:* gets data and metadata from the WWW and processes it to derive a set of Homogenized Basic Elements (HBE). These elements constitute a representation in an internal language.

-*Virtual Structure:* processes the actual stream of HBE and makes a structure named E_{ci} for the name in Spanish *Estructura de Composición Interna*, Internal Composition Structure in English. An E_{ci} is an oriented graph representing a statement in the original text. Sets of E_{ci} are then processed to make an E_{ce} (for the name in Spanish *Estructura de Composición Externa*, External Composition Structure in English, a supra-structure composed by a set of E_{ci} structures all of them related to the same text).

-*External Structure:* it works with a Virtual Network composed by the set of E_{ci} and E_{ce} structures. It can be considered as an interface between the Virtual Structure and any user.

The WIH prototype has implemented a first version of the Internal Structure and Virtual Structure. In the Virtual Structure there is a set of components to perform: the composition of E_{ci} and E_{ce} structures (Composition Engine, CE), insertion into the Virtual Net (Assimilation Engine, AE) through a set of functions (named effect functions, f_e) regulated by a set of dynamic parameters (metric functions, f_m). All the activity is controlled by a feedback system (composed by a general controller System Controller, a Manager for f_e named Metrics Manager, and a manager for f_m named Metrics Engine). The system performs all the activity through a set of f_e that can be changed dynamically. Some of such effect functions define the way a set of HBE is transformed to an E_{ci}, and how a set of E_{ci} is converted into an E_{ce}. For this activity, a metric p_o was defined to categorize each HBE and therefore determine how to process it.

III Metric Constraints and Working Hypotheses

As the metric p_o is used by the f_e to perform CE and AE activities, it has some constraints:

a-domain values must not exceed [-2.0; +2.0].

b-most of the values are 0.0.

c-must depict simple morphosyntactic considerations.

d-a 0.0 value must reflect an HBE with no influence in the global process.

e-an absolute value far from 0.0 must reflect an HBE with more influence than values nearby 0.0.

f- HBE's metric value must be able to be projected in some way to the E_{ci} level and E_{ci} value to an E_{ce} level.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 159–163.

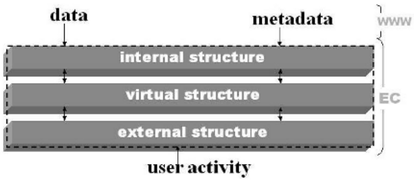

Fig. 1. Three layered structure of the EC. Data and metadata are extracted from the WWW.

g-must provide a qualification for HBE selection.
h-must provide a fuzzy manipulation of original texts.

There is a set of working hypotheses also:

a-There are many writer profiles.
b-Some words in a text are more representative of the main topic than others.
c-Sentences can have different quality.
d-Texts can have different styles.

IV The Metric Description

Set tables defining a numeric weighting of HBE whose original words match with certain prefix, define the quantification for each HBE. Table 1 presents a short list of some weightings and the Spanish word related to the HBE. The same procedure is followed for a list of special words (i.e. Some articles, conjunctions, modifiers, etc.).

The weightings range in [-2, +2]. When sentences are processed, this p_o values for HBE are combined with equation 1 to get a p_o^{Eci} for the sentence.

$$p_o^{Eci} = p_0^{HBE} + \sum_{i=1}^{n}\left(\frac{p_i^{HBE}}{2^{n-i+1}}\right), \text{ n =number of HBE}_i \text{ in this Eci} \qquad (1)$$

The metric is further used to evaluate the resulting E_{ce} with a p_o^{Ece} value (the more optimistic[1] p_o^{Eci} value is used).

V Statistical Analysis

Following the ***h*** constraint, the main steps to derive a fuzzy HBE model [19] are: frequency analysis, pattern and relationship extraction, model description and validation. The model description was performed in the previous section. The rest of the steps are detailed below.

A Frequency Analysis

Two samples were used:

1) Testing hypothesis a: To perform this testing, the profiles proposed are: forum, web index, document, and blog. A sample with 200 individuals was selected (50 of each one). The dot plot in Fig. 2 shows some potential outliers. There is not any confirmed outlier according to the information processed.

2) Testing hypothesis d: To perform this testing, three document styles are proposed: Literary, Technical and Messages. A sample with size 150 was selected (50 of each one). Texts were downloaded from sites specifically dedicated to each of these topics. The dot plot in Fig. 3 shows some potential outliers. There is not any confirmed outlier according to the information processed.

[1] The CS could change the optimistic concept.

TABLE 1
SPANISH WORD WEIGHTING.

muy	0.7
tan	1
pocos	0.2
mucha	0.75
muchos	0.8
bastantes	0.7
escaso	0.15
excesivamente	1.5
abundantemente	1.3
abundante	1.1
demasiada	1.7
exagerados	1.9
no	-1
sin	-1
desX	-1
inhX	-1
antiX	-1
disX	-1

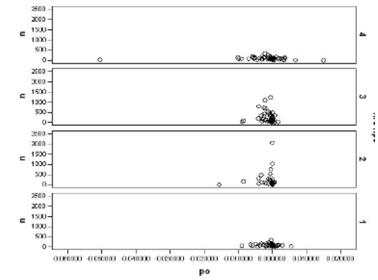

Fig. 2. Profiles dot plot. Set 1: forum. Set 2: web index. Set 3: documents. Set 4: blog. The p_o value corresponds to average of eq. (1).

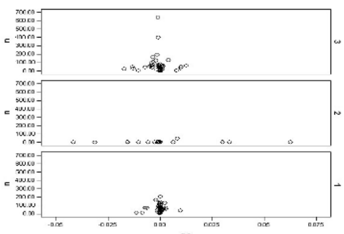

Fig. 3. Styles dot plot. Set 1: literary. Set 2: technical. Set 3: messages. The p_o value corresponds to average of eq. (1).

3) Testing hypothesis c: To perform this testing, the p_o mean value for each statement was calculated (let it be p_o^m). Afterwards, the GREATER_ZERO, LESS_ZERO and EQUAL_ZERO variables were defined as the number of p_o^m greater, less and equal 0.0 respectively. This value is intended to represent quality according to the profile/style (hypotheses c). The distribution was tested as a Binomial (eq. 2).

$$F(x_i) = p(X \le x_i) = \binom{n}{0}p^0q^n + \binom{n}{1}p^1q^{n-1} + \ldots + \binom{n}{k}p^kq^{n-k} \qquad (2)$$

Table 2 shows results: a Chi-Squared value greater than 0.05 indicates a Binomial distribution in all the subsets. The parameters are ***n***=50 cases and a specific ***p*** value found out for each subset in the table.

Therefore, each subset has a specific Binomial distribution of p_o^m. Parameter p indicates predominant EQUAL_ZERO for messages, decreasing for technical and literary.

GREATER_ZERO is higher for literary, decreasing for technical and message. All this can be thought as a valuation scale: literary, technical, and message; where literary seems the opposite of message and technical is in between. But a similar analysis for profiles can't give a definite scale for: blog, doc, forum and web index.

TABLE 2
BINOMIAL TEST FOR STYLES (A) AND PROFILES (B).

	subset	GREATER ZERO	LESS ZERO	EQUAL ZERO
		p (Chi²)	p (Chi²)	p (Chi²)
Style	literary	0.00720 (0.9434)	0.01000 (0.6795)	0.00280 (0.9998)
	message	0.00320 (0.9996)	0.00640 (0.9718)	0.01040 (0.6196)
	technical	0.00600 (0.9809)	0.01000 (0.6795)	0.00400 (0.9984)
Profile	doc	0.00200 (>0.9999)	0.01520 (0.0577)	0.00280 (0.9998)
	forum	0.00400 (0.9984)	0.00960 (0.7347)	0.00640 (0.9718)
	webindex	0.00280 (0.9998)	0.00640 (0.9718)	0.01080 (0.5563)
	blog	0.00720 (0.9434)	0.01200 (0.3608)	0.00080 (>0.9999)

4) Testing hypothesis b: To perform this testing, the maximum and minimum p_o^m values from each document were selected for processing. The related sentences were checked against main topic and secondary topic of the original document. Table 3 shows this analysis performed for one subset from styles (messages) and one from profiles (documents). These subsets were selected because the average number of sentences is the fewest for messages and the highest for documents.

It can be seen that the main topic is represented in most cases. Sometimes the secondary topic is represented. Considering the main topic as being represented by the max. and/or min. p_o, the main topic in the first sentence was studied also. Table 4 shows the percentage of cases where the max and min values appear in the first sentence.

B Pattern and Relationship Extraction

To find out the behavior of p_o, the Shapiro Wilks test was performed for profiles and styles. In all cases the estimator is $p<<0.05$, therefore it can be said the populations have not a normal distribution (see Table 5).

TABLE 3
MAIN TOPIC FOR STYLES (A) AND PROFILES (B).

	%messages	%documents
Min. p_o		
-main topic	92	77
-secondary topic	8	15
-other phase	0	8
Max. p_o		
-main topic	83	38
-secondary topic	17	54
-other phase	0	8

TABLE 4
PERCENTAGE FOR STYLES (A) AND PROFILES (B).

	%messages	%documents
Max. p_o	29	0
Min. p_o	33	0
other	38	26

Pearson correlation between n (number of non-zero p_o values) and p_o denotes an increasing correlation according to the specific profile and style (see Table 6). The technical and blog subsets have a significant correlation (.94 and .97 respectively).

TABLE 5
SHAPIRO-WILKS TEST FOR STYLES (A) AND PROFILES (B).

a)

class	Var.	n	Media	SD.	W*	p (1 tail)
literary	po	50	-3.5E-04	2.9E-03	0.73	<0.0001
	n	50	61.74	41.38	0.89	0.0005
message	po	50	1.1E-03	0.02	0.66	<0.0001
	n	50	4.14	5.94	0.34	<0.0001
technical	po	50	-1.0E-03	0.01	0.86	<0.0001
	n	50	65.46	105.24	0.56	<0.0001

b)

class	Var.	n	Media	SD.	W*	p (1 tail)
blog	po	50	-2.0E-03	0.01	0.64	<0.0001
	n	50	97.74	66.09	0.92	<0.0108
doc	po	50	-1.2E-03	2.0E-03	0.77	<0.0001
	n	50	246.08	273.52	0.78	<0.0001
forum	po	50	-6.8E-04	2.3E-03	0.85	<0.0001
	n	50	80.00	63.03	0.86	<0.0001
webindex	po	50	-9.1E-04	2.6E-03	0.47	<0.0001
	n	50	167.54	336.36	0.51	<0.0001

As the distribution for subsets in both samples is not normal but a different correlation was detected for each one, a Krustal-Wallis test for p_o variability was performed. Therefore it can be stated if the subsets are statistically from different populations (that is, if Literary, Technical and Message belong to a main sample Style, and if Forum, Doc, Web index and Blog belong to a main sample Profile, as proposed here). As can be seen from Table 7, Chi-squared value p is 0.056 and 0.602 for styles and profiles. Both values are greater than the cut value 0.05. As a consequence, it cannot be said they are different populations.

The p_o median study for Styles and Profiles are depicted in Table 8. The Styles gives $p=0.071 > 0.05$, so it cannot be stated the median values are different. But for Profiles $p=0.00 < 0.05$ instead.

In a similar way, Krustal-Wallis test and median studies have been performed for n (number of non-zero p_o values). Table 9 shows a Sig. Value less than 0.05. As a consequence it can be said the n value is a good subset discriminator.

TABLE 6
CORRELATION TEST FOR STYLES (A) AND PROFILES (B).

a)

style	correlation
literary	.43
message	.69
technical	.94

b)

profile	correlation
doc	.44
foro	.49
webindex	.84
blog	.97

TABLE 7
KRUSTAL-WALLIS TEST FOR P_O WITH STYLES AND PROFILES.

Style	SubSet	parameter	value
	literary message technical	Chi-squared	1.014
		Degrees Freedom	2
		Sig. (p)	0.602
Profile	doc forum webindex blog	Chi-squared	7.555
		Degrees Freedom	3
		Sig. (p)	0.056

TABLE 8
MEDIAN FOR P_O WITH STYLES AND PROFILES.

Style	subset	parameter	value
	literary message technical	Chi-Squared	5.303
		df	2
		Sig.	0.071
Profile	doc forum webindex blog	Chi-Squared	18.720
		df	3
		Sig.	0.00

TABLE 9
KRUSTAL-WALLIS AND MEDIAN FOR N WITH STYLES AND PROFILES.

set	parameter	Median analysis	Variability analysis
Style	Chi-Squared	74.880	90.848
	df	2	2
	Sig.	0.000	0.000
Profile	Chi-Squared	15.607	11.680
	df	3	3
	Sig.	0.001	0.009

C Validation

A web page in Spanish was downloaded and text processed. A set of E_{ci} was derived. The p_o value for each HBE was calculated also. In order to perform the validation of p_o, HBE´s original words in Spanish were used. The resulting sets of words were classified according to the associated p_o value:

-The set of words whose $|p_o| < 0.110$ was selected and composed into two sentences representing the main topic. Let them be the Type I words.

-The set of words whose $|p_o| > 0.856$ was selected and composed into five sentences representing secondary topics. Let them be the Type II words.

A poll with 35 volunteers aged between 22 and 56 years old. All of them spend less than 10 hours a week in the Internet. They were asked to read the original text and to extract 2 to 10 most representative words from it. Fig. 4 shows a chart with the frequency of types of words extracted by volunteers. From the chart, it can be said that type I words are preferred mostly. There are a number of invalid words, because they weren't in the text (denoted by label *out of text*). A set of words extracted from text, describing heterogeneous details were represented and classified as *other in text*. *T*ogether, *type* I and II represent a bigger portion of the graph than *other in text* words.

As a part of the poll, volunteers also wrote 2 to 4 short sentences describing the topics in the text. These were compared with the set of 7 sentences (main and secondary topics) derived previously. Fig. 5 shows that derived sentences represent a high percentage of the topics answered.

Some additional characteristics of the metric results are:

-the p_o values have not a bias other than the opposite and ambiguity HBEs mentioned in the metric description section.

-the lexical categories of the HBEs detected as Type I and II from the text can be mainly classified as noun, verb and other (Fig. 6).

Fig. 4. Types of words 1: $|p_o| < 0.110$, 2: $|p_o| > 0.856$, other from text other not in text.

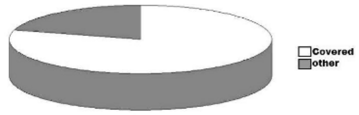

Fig. 5. Matching between topics declared and the 7 sentences derived.

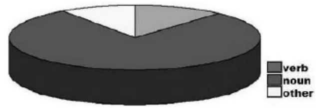

Fig. 6. Lexical category of words in the processed text.

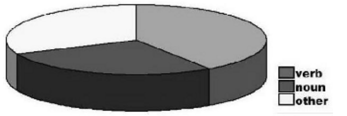

Fig. 7. Average p_o for each lexical category.

-the average p_o value is almost the same for each lexical category depicted previously (Fig. 7).

VI CONCLUSIONS

A p_o metric was presented and used here. From the results follows that it is not statistically possible to differentiate the *Text Styles* proposed here. But there is a clear distinction for the following *Writing Profiles*: doc, forum, webindex and blog. Therefore, this metric is invariant to document size and mentioned Text Styles but it is useful to detect certain writing profiles.

The p_o value was evaluated in a poll to assess its ability to detect main topics and relevant words. Statistics show that there is a pretty good relation between them and p_o values close to zero.

VII FUTURE WORK

It remains to do a better tuning of the weighting for different HBEs for texts in Spanish. It also has to be applied in other languages as well.

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Non-Technological Aspects on Web Searching Success

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***Abstract*— This paper studies the influence of social, cultural and emotional background of typical Web users into the web searching process. Several variables, describing such aspects, are represented and statistically analyzed with well known clustering and classifying algorithms such, as COBWEB, J48, Bayes classification, and Correspondence analysis. Results indicate that the efficiency of the complete process of Information Retrieval will not be fully understood without considering subjectivity and personality facts.**

I Introduction

A long research area studies Web information retrieval. Several approaches involve improvement in visualization[1], performance, management, processing and use of information (mainly evaluated by its precision and recall metrics). Much attention has been directed at the design and usability of communication technologies. Both W3C Consortium and GVU (Graphic, Visualization & Usability Center) [2]are to provide related documentation, collected data, publications, etc.

Earliest systems ignored subjectivity and used fixed algorithms. The first kind of adaptive systems intended to perform some consideration of the user behavior [4] [3]through parameters adjusted by users directly or indirectly by relevance feedback. Pickard proposed in [4] that the strategy for considering subjectivity is a three step process: first, share some of the common sense of the user; second, observe and model user's actions and third, learn from these interactions.

For image retrieval, Jaimes [5] explained that "the effectiveness of retrieval depends not only on the metadata description, but also on how the user performs the query, his expectations, and other factors". In this field, the user activity is taken to combine with other descriptions and automatically derive the meaning of certain image. It also could be used to detect user's level of interest. Pickard in [6] proposed a similar approach, combining inference and learning to improve content image retrieval. She also proposes to apply a kind of relevance feedback to image content retrieval, combined with a nonlinear learning algorithm with dynamic bias. But it requires a number of queries to detect the kind of problem and switch the activity bias.

For Web design considerations, the satisfaction is referred to as part of the ISO-9241-11, to define web accessibility [7]. User subjective perception was also elaborated as a comparison between latency perception and time delay [8], and related with navigation behavior. Web applications are designed consequently in such a way to improve internet QoS (Quality of Service) perception. In [7] several virtual communities[a] are evaluated and the study concludes that existing guidelines are mainly technological oriented and the design of websites that support virtual communities should be based on a multidisciplinary corpus of guidelines. At this moment, only cognitive psychology and ergonomics are incorporated into design knowledge.

The e-commerce industry [7] define the customer experience, that includes usability factors such as ease–of–use in finding a product or service on the first visit; understandable and efficient navigation; good feedback as to the success of a transaction; and clear cues to the options. A highly usable site lets customers feel in control and affords them appropriate flexibility, while providing interaction that appears simple. There are a set of rules and guidelines to perform expert's evaluation of such items.

From the accessibility point of view, the subjective assessment has been criticized [9]. The main problem is that it has no proved effectiveness. Therefore simpler methods like conformance testing (standard review) are claimed to be problem free and also preferable even when it is not fully understood their coverage of the userside problem. But cultural incidence in the accessibility has been proved to be significant [10][b]. It remains to this paper to analyze other subjective factors.

This paper studies the user point of view of the whole retrieval process, mood and personality aspects from the user's opinion. It is something important to be considered as it settles the degree of preference for selecting and remain using a certain browser. To analyze it, this work uses a public database from GVU [11] but for Web searches evaluation. It is statistically shown here that the user insecurity and personality play an important role in the user results examination. The main objective and subjective behavioral aspects of internet users in Web interaction are defined disregarding the flexibility and optimization provided by the internet tools being used.

The rest of this paper is organized as follows: section 2, describes the database. Section 3 describes the Data Mining methods used to analyze data. Section 4 presents the results of applying such methods and finally in section 5 are some conclusions.

II Database

Although the selected GVU Data Set is not a new one, it was selected because it has an important number of instances, the proven expertise in pollings and because it is published in internet. Besides, the essential subjectivity user behavior, as a

[a] A particular category of web sites: those that are designed to support groups and communities.

[b] The results of an on-line experiment that exposed American and Chinese users to sites created by both Chinese and American de-signers indicate that users perform information-seeking tasks faster when using web content created by designers from their own cultures.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 164–169.

human being, is not expected to change so quickly even with the proven internet technology evolution.

A *Data Source*

GVU is an entity devoted to study specific aspects of internet and internet users. As mentioned previously, the database used is a public database from the GVU's site, identified as 'April 1997 GVU 7 Flow Data Set[c], previously used by Hoffman y Novak [11]. The data was collected by an online form. Many volunteers filled the form freely. The form, composed by 77 items, was published in internet from April 10th to May 10th of 1997. More than a million records were gathered.

The original dataset was filtered by GVU to take only the experienced and frequent user volunteers. The result is a dataset containing 4232 records.

B *Data Description*

Each record represents an internet user and depicts three main aspects of him or her:

-Web usage: it is reflected in the first form section. The related answers are represented by V1 and V2 attributes.

-Attitudes and opinions about using the web: it is in the second section. At this time it has three kinds of questions:

1.Agree or disagree with certain opinion or sensation (V3 to V46).

2.Degree of certain characteristic or quality, choosing between two opposite extremes (V47 to V58).

3.Evaluation between two non-opposite characteristics or qualities (V59 to V74).

-Background data: it is in the third section, with three questions (V75 to V77).

The processed database instances have 78 attributes (77 from the poll plus a unique identifier field).

C *Attribute Description*

Poll attributes are numeric, generally ranging from 1 to 9. The remaining field (named who) is a unique user identification string. Table 1 depicts each one.

TABLE 1

SUMMARY OF ATTRIBUTES.

Atrib.	Question
Section I	
V1	How much time would you estimate that you personally use the Web?
V2	When did you start using the Web?
Section II	
V3	I am very skilled at using the Web
V4	I feel unimaginative when I use the Web
V5	I feel worried when I use the Web
V6	It is hard for me to find information on the Web
V7	Using the Web makes me think
V8	I feel uninventive when I use the Web
V9	The Web provides many possible things for me to do
V10	I feel "in flow" when I use the Web
V11	When I use the Web there is very little waiting time between my actions and the computer's response
V12	I feel stimulated when I use the Web
V13	I can concentrate fully when I use the Web
V14	I like to click on a link just because it looks interesting
V15	Using the Web challenges me to perform to the best of my ability
V16	I forget about my immediate surroundings when I use the Web
V17	I feel spontaneous when I use the Web
V18	I feel I am more in the "computer world" than in the "real world" around me when I use the Web
V19	The Web provides many opportunities for action
V20	I enjoy browsing the Web to see what is out there
V21	I feel apathetic when I use the Web
V22	I think about other things when I use the Web
V23	I can be easily distracted when I use the Web
V24	I know how to find what I want with a search
V25	I feel relaxed when I use the Web
V26	I feel flexible when I use the Web
V27	Time seems to go by quickly when I use the Web
V28	I feel like I am in a "virtual reality" when I use the Web
V29	I find that using the Web stretches my capabilities to the limits
V30	I feel playful when I use the Web
V31	Interacting with the Web is slow and tedious
V32	Using the Web challenges me
V33	Using the Web provides a good test of my skills
V34	Downloading software is easy for me to do on the Web
V35	I feel creative when I use the Web
V36	When I use the Web I am totally assorted in what I am doing
V37	The number of different ways that one can interact with the Web today is limited
V38	I consider myself knowledgeable about good searching techniques on the Web
V39	I feel bored when I use the Web
V40	I know less about using the Web than most users
V41	I feel unoriginal when I use the Web
V42	I like to experiment when I use the Web
V43	I feel in control when I use the Web
V44	I find the Web easy to use
V45	I feel anxious when I use the Web
V46	Navigating the Web with today's Web browsers is:
V47	When you use the Web you usually:
V48	Interacting with the Web is:
V49-V58	Aspects on How do you feel, in general, about the Web
V59-V74	Aspects on How do you feel, in general, when you use the Web
Section III	
V75	Gender
V76	Age
V77	What is the highest level of education you have completed?

Some missing values were detected. Because such missing values represent a refusal to answer the question, it was interpreted as a different answer and therefore filled with a new value (zero).

In the following, V6 is taken as the classification variable. To do so, it is processed as follows:

-Data values ranging from 1 to 4: represents class NO

-Data values from 5 to 9: represents class YES

-Missing values: are ignored (they represent 10 instances that are discarded).

III METHODS APPLIED

The database (composed by three main sections) is analyzed with a dependency study in order to reduce the number of attributes. Variable V6 (corresponding to the question: Is it hard for me to find information on the Web?) was selected as a class variable. Afterwards, the inference power of each section variables is tested with an induction tree (J48) and verified with COBWEB. From the best classification subset, four inference rules are derived. An alternate subset of variables used with Bayes classification provides a complementary set of inference rules. A final analysis between variables and their meaning is provided using also correspondence analysis.

A *Decision Trees*

In order to get the natural ordering of the observed data without previous knowledge, the first analysis performed here is classification, using WEKA software (Waikato Environment for Knowledge Analysis[d])[13].

[c] Copyright 1994-1998 Georgia Tech Research Corporation. All rights Reserved. Source: GVU's WWW User Survey www.gvu.gatech.edu/user_surveys.

[d] Weka is a collection of machine learning algorithms for data mining tasks. The algorithms can either be applied directly to a dataset or called from your

The J48 algorithm is an implementation of the well-known C4.5 [14]. Some of its characteristics are: It works with binary split on numeric attributes, performs a weighted bifurcation when missing values, performs post pruning, evaluates the node error as a Bernoulli process in order to evaluate future pruning activities, works with a confidence interval for the estimated error and performs a pruning of the resulting set of tree derived rules. It works with nominal and numerical data fields, deals with missing values and does not need to reduce the training set to evaluate the node error.

Result evaluation: To measure the predictor success, Kappa statistics is used[15]: let c_1, c_2,...c_n be the row margin frequency of the confusion matrix with elements named f_{ij}. Let also correspond each f_{ij} to attributes a_1, a_2, ...a_n; N the number of instances and e_{ij}, the elements of expected value matrix (the one that represents a random predictor), then κ is defined as in (1).

$$\kappa = \frac{N - \sum_{i=1}^{i=n} e_{ii}}{\sum_{i=1}^{i=n} f_{ii} - \sum_{i=1}^{i=n} e_{ii}} .100\% \qquad \textbf{(1)}$$

The maximum value of Kappa statistic is 100% and the minimum is 0%, and evaluates the agreement between predicted and observed categorizations of a dataset, while correcting for agreement that occurs by chance [14].

B Bayesian Classification

Bayesian learning [15] assumes that there is a probability distribution followed by the dataset, and that better decisions are derived from a mix-up of probabilities and experience. The Naïve Bayes Classifier, which assumes that any instance has a number of attributes, tries to learn a target function capable to classify a new instance.

Bayesian nets take the advantage of a good graph representation when the node distribution is simple. They also overcome the typical node replication problem of induction trees.

This work uses an extension of Naïve Bayes Classifier named BayesNet in Weka Software [17]. It uses Bayes for nominal attributes with no missing values. The following searching algorithms are applied to the exploratory phase:

-K2: hill climbing. At each node evaluates adding new arcs to previously processed nodes, followed by a fixed ordering of the variables. It is a quite a fast algorithm and, by construction, doesn't produce cyclic graphs.

-TabuSearch: performs a hill-climbing search until it hits a local optimum. The next step is selected as the least worse. Last n steps are held in a list to be eliminated in further steps.

-HillClimber: it uses the traditional hill climbing, adding, removing and reverting arcs as needs. The search is not restricted to the variable order.

-TAN: Tree Augmented Naïve Bayes. Starts with the class node as father node. It then evaluates whether to add a new father to each node. The tree is formed by calculating the maximum weight-spanning tree using Chow and Liu algorithm [18].

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own Java code. Weka contains tools for data pre-processing, classification, regression, clustering, association rules, and visualization. It is also well-suited for developing new machine learning schemes. Weka is open source software issued under the GNU General Public License.

Result evaluation: To measure the predictor success, Kappa statistics is used. Correctly Classified Instances and Incorrectly Classified Instances are used as well.

C COBWEB Clustering

This is a simple method of incremental conceptual clustering [15]. Its parameters handle the precision of the clustering (acuity parameter) and preserve from overfitting (cutoff parameter). Unlike k-means, it does not partition instances into disjoint clusters. It was developed in late 1980s for nomimal attributes. Perform a hierarchical grouping of instances and use a measure of cluster quality (category utility). It evaluates each new instance and inserts it in a tree representing the actual set of clusters. The root node represents the entire collection and the leaves each instance. Operations like merging and splitting of the tree nodes allow a convenient restructure anytime it is considered needed. The main advantage of this algorithm is that does not need to pre-set a certain number of clusters a priori. A similar method is Classit, but for numeric attributes.

The Category utility [15] measures the overall quality of a partition of instances into clusters. It is a kind of quadratic loss function defined on conditional probabilities defined as in (2).

$$CU(C_1, C_2, \ldots, C_k) = \frac{\sum_l \Pr[C_l] \sum_i \sum_j \left(\Pr[a_i = v_{ij} \mid C_l]^2 - \Pr[a_i = v_{ij}]^2 \right)}{k} \qquad \textbf{(2)}$$

Where C_1, C_2, …,C_k, are the k clusters. The outer summation is over these clusters and the inner one sums over the attributes. Attribute a_i is the i^{th} attribute and takes on values v_{i1}, v_{i2},--,v_{ij}, …

Result evaluation: the cutoff value and the number of leaves will be used to set the desired result.

D Correspondence Analysis

This is used to reflect variable associations graphically [19]. It translates relations to dot positions using a translation table: closer row dots reflect similar profiles or conditional distributions. The proximity to column dots denotes a co occurrence that doesn't respond to the independent model. This analysis provides the dot coordinate and its inertia (the measure of the information from each dimension).

Result evaluation: the p value will be used, corresponding to the area under observed value Chi-squared curve. The significance value is set to α= 0.05.

IV Results

A Variable Selection

Due to the limited processing capacity and the high number of attributes, the attributes are reduced using dependency study. Variables V3 through V74 represent answers to some redundant questions. These questions provide certain cross validation. Taking that in consideration, variables were processed as follows:

1.-The attributes for each subset described in the Data Description section are represented in a dispersion matrix (SPLOT) to examine candidate interdependencies.

2.-Correspondence analysis is performed for each subset to refine dependences from the previous step.

3.-The categories correspondences between the most probable dependencies were studied.

4.-Variables V1, V2, V75 and V76 are related to user description and do not represent user behaviors. They will be referred to as profile variables. They will not be considered in the following analysis.

B Clustering Analysis

1) Selecting variables with J48: To perform a comparison among reduced subsets, the 4222 instances are initially classified using all the variables. As described previously (see Data Description section) the subsets are:

i-Web usage (first form section).

ii-Attitudes and opinions about using the web (second section)

iii-Background data (third form section)

From the results it is important to say that:

-classification with first section is much better than the other two and than the average from all variables. 3331 instances (78.9%) are correctly classified by this subset.

-filtering rules taking only the ones with more than 20 instances, an interesting subset remains (see Table 2). They can be summarized in the following rules:

r1)$V24 > 6 \Rightarrow$ No

r2)$V24 < 5 \Rightarrow$ Yes

r3)$V24 = 5 \wedge V4 = 5 \Rightarrow$ No

r4)$V24 = 6 \wedge [(V9 = 8 \wedge V27 = 8) \vee (V9 = 7 \wedge V42 = 6) \vee (V9 = 6)] \Rightarrow$ Yes

These rules are denoting that users say they find information because they know how to search for it (r2), and the opposite happens when they don't know how to search for it (r1). But the main finding is reflected in r3 and r4, with the values 5 and 6 for V24:

-The ones who don't know how to search and are worried, they don't find what they are searching.

-The ones who don't know how to search and feel with ability to explore and experiment, they find what they are searching.

TABLE 2

RULES WITH SUPPORT OVER 20.

Rule	Classification	Support
$V24 = 9$	No	1211.0
$V24 = 8$	No	1254.0
$V24 = 7$	No	826.0
$V24 = 6 \wedge V9 = 9$	No	72.0
$V24 = 6 \wedge V9 = 8 \wedge V27 = 8$	No	28.0
$V24 = 6 \wedge V9 = 7 \wedge V42 = 6$	No	25.0
$V24 = 6 \wedge V9 = 6$	Yes	52.0
$V24 = 5 \wedge V4 = 5$	No	22.0
$V24 = 4$	Yes	150.0
$V24 = 3$	Yes	143.0
$V24 = 2$	Yes	87.0

*2) Hierarchical Clustering with COBWEB:*The following analysis intends to make an approximate confirmation of the previous subsection findings: if the selected variables are good for classification of instances from people that consider hard "to find information on the Web" (classification variable V6), they should be good for clustering with the same criteria.

Therefore, to analyze the classification power of these variables, the data is clustered just considering the variables handled in rules r1 through r4 (see Table 2) COBWEB is applied to V4, V9, V24, V27, and V42. Due the sensitivity of this technique to sampling sequence, several orderings were considered.

The processing is performed 36 times with WEKA software, tuning the parameters each time. From the output is interesting to observe:

-The biggest tree with the optimal sample sequence is 4075 clusters. It has the highest over fitting, and has cutoff values less than 0.0028209479.

-When cutoff is 0.18, the tree has three clusters (see Fig. 1). Table 3 shows the instance assignment detail. It can be seen that assignment corresponds to 30% ± 12%.

-Distribution of clusters according to classes is significant. Clusters 0 and 1 are deeply related to class "Yes". Group 2 and 3 correspond to class "No". There are no mixtures. The group or cluster 0 corresponds to initial node, with instances not classified by the rest of the nodes. Therefore, there are 4 clusters instead of 3. It could mean that these attributes are good to perform an acceptable description of both classes.

-When cutoff value is 0.167, the tree increases (see Fig. 2), with instance assignments depicted in Table 4. It can be seen a homogenized distribution among leaves.

-Cluster distribution preserves the trend observed previously: clusters 0 and 1 represent class Yes, while the rest of the clusters represent class "No".

C Alternate Classification Analysis

The previous classification and clustering results indicate that the first section has a good clustering subset of variables (related with the set of rules r1 through r4). In this section, a new classification analysis is performed to find out if there are other variables in other sections that are also good to infer data classification. First, the kind of Bayes classification is studied according to the Kappa value. Afterward, the search approach is studied deeply to get the best Kappa value.

1) Bayes Classification: As an initial classification study, the dataset was processed with Naïve Bayes. All the variables and each of the three subsets were processed with Naïve Bayes classifier (see Table 5). From the table arises a difference from J48: classification is better for Background data (third subset).

To confirm this result, the same procedure was repeated with BayesNet with K2 searching approach (see Table 6). The table shows that the Kappa value is better for BayesNet. Therefore the remaining analysis is performed with BayesNet.

As the best classification result is again for the third subset, it will be considered for analyzing deeply the behavior with other search approaches available for BayesNet.

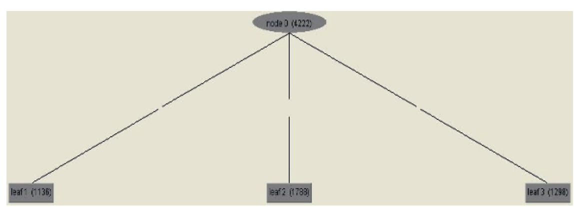

Fig. 1. COBWEB: tree with cutoff 0.18.

TABLE 3

INSTANCE ASSIGNATION TO GROUPS.

Group	Number of instances	Percentage
1	1136	27
2	1788	42
3	1298	30

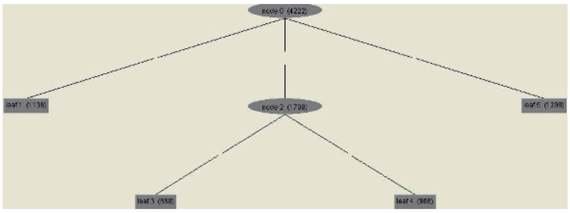

Fig. 2. COBWEB: tree with cutoff 0.18.

TABLE 4
INSTANCE ASSIGNATION TO CLUSTERS.

Cluster	Number instances	Percentage
1	1136	27
2	1788	42
3	880	21
4	908	21
5	1298	31

TABLE 5
NAÏVE BAYES CLASSIFICATION.

Variables	A priori Prob. for "No"	A priori Prob. for "Yes"	classified OK	Kappa
All (41)	.72	.28	68.33%	0.304
Group i(27)	.72	.28	70.13%	0.326
Group ii(6)	.72	.28	70.04%	0.182
Group iii(10)	.72	.28	71.55%	0.253

TABLE 6
BAYESNET CLASSIFICATION USING K2.

Variables	A priori Prob. for "No"	A priori Prob. for "Yes"	classified OK	Kappa
all (41)	.72	.28	68.23%	0.303
Group i(27)	.72	.28	70.01%	0.325
Group ii(6)	.72	.28	69.85%	0.189
Group iii(10)	.72	.28	71.55%	0.254

2) Bayes Classification search approach: To get the best search approach, the third subset is classified again with BayesNet several times with alternate search approaches. Table 7 details the classification comparison. From all the alternates, the best Kappa value is obtained with TAN as long as the best correctly classification percentage. Fig. 3 shows the net graph obtained with BayesNet with TAN.

It is interesting to observe in the graph a number of conditions on the variable probabilities and to interpret them in terms of the questions represented by the related attributes. Some are:

-The difficulties to find information (V6, the class variable) are related to the lack of browsing knowledge (V67).

-The difficult to get information because of missing navigation knowledge (V67) takes incidence in the control feeling (V64) and the knowledge need (V73).

-The happiness when using the Web takes incidence in navigation stimulation (V65), socialization (V69) and challenging personal capabilities to their limits (V60).

It is possible to extract a common subtree from all the search alternatives approaches available for BayesNet except for K2 (see Fig. 4). This sub-tree shows the relation between the information search success and the questions represented by those variables.

TABLE 7
SEARCHING ALTERNATIVES FOR BAYESNET.

Algorithm	Classified OK	Kappa
BayesNet/K2	71.55%	0.254
BayesNet/TabuSearch	73.14%	0.212
BayesNet/HillClimber	73.07%	0.245
BayesNet/TAN	76.05%	0.293

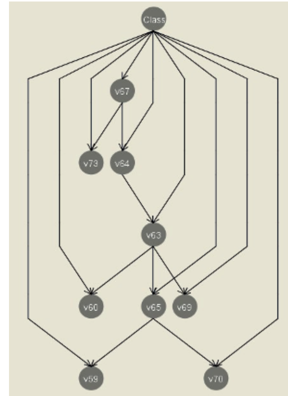

Fig. 3. TAN search with BayesNet.

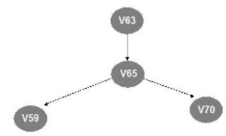

Fig. 4. BayesNet common subtree.

V CONCLUSIONS

The GVU poll provides user profiles with special focus in frequent web users. It is related to the study of questions on personal feelings and opinion about success or failing in the information retrieval activity. The analysis is performed on three groups of data variables:

-Group 1:Degree of agreement with certain statements.

-Group 2:Selection between two opposite characteristics.

-Group 3:Selection between two alternatives which are not necessarily opposite, but related.

Two different techniques were applied: induction tree (J48) and Bayesian net classification. The J48 classification denotes the Group 1 to be the best set of variables to infer the satisfaction in user interaction. A set of rules can be derived and used to perform such an inference with reasonable degree of precision. It was verified with COBWEB clustering.

Naïve Bayes and BayesNet classifiers were used to determine the classification variables for satisfied/non-satisfied users. Both of them define Group 3 as the best classifier subset. Furthermore, the best BayesNet search approach is TAN.

From the J48 relevant set of rules it can be said that to find what user is looking for, it is important:

-To know how to search (the lack of browsing knowledge is related to the possibility to find information)

-To be confident (not worried)

-To feel ability to explore and experiment

-Browsing happiness

From all the BayesNet nets there is a common subgraph which represents a set of questions necessary but not sufficient to describe profile of people with difficulties to find information in the Web:

-Do you feel happy when using the Web?

-Do you feel stimulated when using the Web?

-Do you feel excited when using the Web?

It is interesting to observe also the simple correspondence analysis with the perspective of the variables preferred by the classifications. Some new answers can be obtained doing that:

-V4, V8 and V41 are alternatives for the same topic. It could be a way to verify consistency in the answers of the poll. Other similarities were detected as well.

-V63 and V71 relate browsing happiness with the frustration feelings. It is interesting to observe that people unhappy do not necessarily feel frustrated.

-V68 and V48 relate interface knowledge with simplicity. People who don't know the right things to do, do not define that the Web is whether or not counterintuitive. But people who do know the right things to do, define the Web as something intuitive.

-V72 and V62 describe hostile attitudes due to the navigation stress. People that feel irritated feel hostile also.

As final conclusion from the above results presentation, it could be stated that there are in fact subjective conditionings using the Web, and they are transient to interface quality and technical effectiveness. The remaining question now is how to and where to search for solutions for this inconvenience. It seems to start as social and cultural problems and takes deep incidence in the man-machine interaction, degrading the interface utility and making people search for information and interaction in alternate sources that are more costly in time and money.

VI Future work

It is interesting to extent this analysis with other databases and evaluate the changes in the human factors considered here. It will be very useful to perform an analysis of the influence of each recent web technology in the user subjectivity as well.

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A Novel Technique to Avoid Similarity Criterion Calculations in a Multi-Processor Environment

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***Abstract* - In a multi-target scenario more than one observation may fall within the prediction ellipse of a filter and prediction ellipses of different filters may interact. Several approaches may be used for this situation, one of that is called the track splitting algorithm. In this algorithm, if *'n'* observations occur inside a prediction ellipse, then the filter branches or splits into *'n'* tracking filters. This process may result in more than one filter, tracking the same target. Therefore, similarity criterion is used to prune similar filters tracking the same target. In this study a novel approach is used to avoid this criterion to speed-up the computation, additionally it also substantially reduce system storage requirements.**

***Index Terms* - Target Tracking, Similarity Criterion, Multi-processor, Track Splitting Filter.**

I. INTRODUCTION

Tracking of a single target, in the ideal situation where one noisy observation is obtained at each radar scan can be done using standard Kalman filter technique. In the multi-target case, an unknown number of observations are received at each radar scan and assuming no false observations, each observation has to be associated with an existing or new target tracking filter. When the targets are well apart from each other then forming an observation prediction ellipse around a track to associate the correct observation with that track is a standard technique [1]. When targets are near to each other, then more than one observation may fall within the prediction ellipse of a filter. Track Splitting Filter algorithm can be used for the data association in such cases [2] [3] [4] [5]. In this algorithm if *'n'* observations occur inside a prediction ellipse then the filter branches or splits into '*n'* tracking filters. This situation which results in an increased number of filters require not only more computational power but also more storage capability. Therefore, multi-processor implementation is one alternative to have more computation power and system storage capability. However, some mechanism for restricting the excessive tracks that originated from track splitting is required since eventually this process results in more than one filter tracking the same target. One such mechanism is the similarity criterion which uses a distance threshold to prune similar filters tracking the same target [6].

The multi-processor architecture for such applications requires that the incoming data from sensors to be collected at a central place for further processing. Also, after final processing results are to be displayed and any further action must be commanded through this central place. This means the possible architecture for such applications is Master-Slave that is the master act as a central place and all other processors in the system works as slaves. In other words the architecture is a loosely coupled set of processors having their own independent memory. The master processor is not only connected with sensor but also with a display console/panel.

At each radar scan the master processor can equally distribute the incoming observations and existing tracks to all the slave processors and they can function independently. However, it should be noted that at each radar scan during distribution, collection of observations and tracks by the master processor the other (slave) processors remain idle. Also, when the master processor receives updated filter tracks from the other processors in the network, global similarity criterion has to be applied on all the tracks collectively, which can lead to a bottle neck (detailed later in the implementation). Our novel technique implemented here avoids such comparison and ultimately no bottle neck is possible.

II. TARGET MODEL

The motion of a target being tracked is assumed to be approximately linear and modeled by the equations;

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 170–175.

$$\underline{x}_{n+1} = \Phi \underline{x}_n + \Gamma \underline{w}_n \qquad (1)$$

$$\underline{z}_{n+1} = H \underline{x}_{n+1} + \underline{v}_{n+1} \qquad (2)$$

Where the state vector

$$\underline{x}_{n+1}^T = (x \quad \dot{x} \quad y \quad \dot{y})_{n+1} \qquad (3)$$

is a four-dimensional vector, $\underline{w}_n$ the two-dimensional disturbance vector, $\underline{z}_{n+1}$ the two dimensional observation vector and $\underline{v}_{n+1}$ is the two-dimensional observation error vector. Also Φ is the assumed (4x4) state transition matrix, Γ (4x2) is the excitation matrix and H (2x4) is the observation matrix and they are defined below. Where Δt is the sampling interval and corresponds to the time interval (scan interval) assumed constant, at which radar observation data is received.

$$\Phi = \begin{bmatrix} 1 & \Delta t & 0 & 0 \\ 0 & 1 & 0 & 0 \\ 0 & 0 & 1 & \Delta t \\ 0 & 0 & 0 & 1 \end{bmatrix}, \Gamma = \begin{bmatrix} \Delta t^2/2 & 0 \\ \Delta t & 0 \\ 0 & \Delta t^2/2 \\ 0 & \Delta t \end{bmatrix}, H = \begin{bmatrix} 1 & 0 & 0 & 0 \\ 0 & 0 & 1 & 0 \end{bmatrix}$$

The system noise sequence $\underline{w}_n$ is a two dimensional Gaussian white sequence for which

$$E(\underline{w}_n) = 0 \qquad (4)$$

where E is the expectation operator. The covariance of $\underline{w}_n$ is

$$E(\underline{w}_n \quad \underline{w}_m^T) = Q_n \delta_{nm} \qquad (5)$$

where Q_n is a positive semi-definite (2x2) diagonal matrix and δ_{mn} is the Kronecker delta defined as

$$\delta_{nm} = \begin{Bmatrix} 0 & n \neq m \\ 1 & n = m \end{Bmatrix}$$

The observation noise sequence v_n is a two-dimensional zero mean Gaussian white sequence with a covariance of

$$E(v_n \quad \underline{v}_m^T) = R_n \delta_{nm} \qquad (6)$$

where R_n is a positive semi-definite symmetric (2x2) matrix given by

$$R_n = \begin{bmatrix} \sigma_x^2 & \sigma_{xy} \\ \sigma_{xy} & \sigma_y^2 \end{bmatrix} \qquad (7)$$

σ_x^2 and σ_y^2 are the variances in the errors of the x, y position observations, and σ_{xy} is the covariance between the x and y observation errors. It is assumed that the observation noise sequence and the system noise sequence are independent of each other, that is

$$E(v_n \quad w_m^T) = 0 \qquad (8)$$

The initial state $\underline{x}_0$ is also assumed independent of the $\underline{w}_n$ and v_n sequences that is

$$E(\underline{x}_0 \quad w_n^T) = 0 \qquad (9)$$

$$E(\underline{x}_0 \quad v_n^T) = 0 \qquad (10)$$

$\underline{x}_0$ is a four dimensional random vector with mean $E(\underline{x}_0) = \hat{\underline{x}}_{0/0}$ and a (4x4) positive semi-definite covariance matrix defined by

$$p_0 = E\left[(\underline{x}_0 - \bar{\underline{x}}_0)(\underline{x}_0 - \bar{\underline{x}}_0)^T\right] \qquad (11)$$

where $\bar{\underline{x}}_0$ is the mean of the initial state $\underline{x}_0$. The Kalman filter is an optimal filter as it minimizes the mean squared error between the estimated state and the true (actual) state provided the target dynamics are correctly modeled. The standard Kalman filter equations for estimating the position and velocity of the target motion described by equations 1 & 2 are;

$$\hat{\underline{x}}_{n+1/n} = \Phi \hat{\underline{x}}_n \qquad (12)$$

$$\hat{\underline{x}}_{n+1} = \hat{\underline{x}}_{n+1/n} + K_{n+1} \underline{v}_{n+1} \qquad (13)$$

$$K_{n+1} = P_{n+1/n} H^T B_{n+1}^{-1} \qquad (14)$$

$$P_{n+1/n} = \Phi P_n \Phi^T + \Gamma Q_n^F \Gamma^T \qquad (15)$$

$$B_{n+1} = R_{n+1} + H P_{n+1/n} H^T \qquad (16)$$

$$P_{n+1} = (I - K_{n+1} H) P_{n+1/n} \qquad (17)$$

$$\underline{v}_{n+1} = \underline{z}_{n+1} - H \hat{\underline{x}}_{n+1/n} \qquad (18)$$

Where $\hat{x}_{n+1/n}$, $\hat{x}_{n+1}$, K_{n+1}, $P_{n+1/n}$, B_{n+1} and P_{n+1} are the predicted state, estimated state, the Kalman gain matrix, the prediction covariance matrix, the covariance matrix of innovation, and the covariance matrix of estimation respectively. The covariance of the observation noise Q_n^F assumed by the filter is normally taken equal to Q_n. In a practical situation, however, the value of Q_n is not known so the choice of Q_n^F should be such that the filter can adequately track any possible motion of the target. To start the computation an initial value is chosen for P_0. Even if this is a diagonal matrix, then clearly from the above equations the covariance matrices B_{n+1}, P_{n+1}, $P_{n+1/n}$ for a given n do not remain diagonal when R_n is not diagonal [7][8][9][10][11].

III. SIMILARITY CRITERION

The similarity criterion is used to account for those instances where filters are producing similar estimates and may in fact be tracking the same target. Figure 1 shows a typical scenario for such a case. Tracks T1 and T2 are updated with observations M1 and M2 as these observations fall inside their respective gates. Now hypothetically assume that M1 is the true observation for T1 and M2 is the true observation for track T2 (shown by same color). Therefore, T1M2 and T2M1 pairings are false associations. It also suggests that tracks T1M1 and T2M1 are following the same observation M1 and obviously one is true and the other one is false. Similar is the case for tracks T1M2 and T2M2. The similarity criterion measures the nearness of the two tracks and if this is below a certain threshold one track can be eliminated. Two tracks i and j are deemed to be similar if;

$$(\hat{x}_i - \hat{x}_j)^T P^{-1} (\hat{x}_i - \hat{x}_j) \le D_{th}$$

where $\hat{x}_i$ and $\hat{x}_j$ are the two filter estimates. The weighting matrix P is chosen to be the sum of the covariance matrices of the two filters with block diagonal elements set to zero and D_{th} is the chosen threshold. In a multi-processor architecture once the individual computation in each processor is finished all the updated filter tracks are sent to the master controller/processor to locate if any target filter is tracking the same target by applying the similarity test described above. This test which is performed inside the master controller is called global similarity test/criterion. The individual processors also perform the similarity test during their individual computation which is called local similarity test. The global similarity calculations could be the biggest source of bottle neck in a multi-processing architecture if the distribution of observation/tracks is not handled intelligently. In our implementation we have avoided the global similarity calculations through our intelligent method of distribution for observation/tracks.

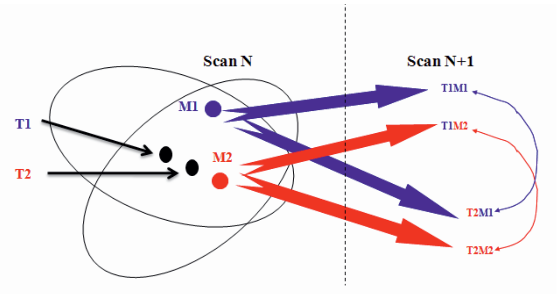

Figure 1: Similar Tracks

IV. IMPLEMENTATION

The processing architecture model is shown in figure 2 for our implementation. Simulated target data is fed to the master controller TMS320C6416 which is responsible for distribution of tracks and observations to other processors in the network and is working as master controller/processor. The main feature as pointed out earlier is our technique of distribution (observation/tracks) and it is made in such a way that no global similarity criterion calculations are required inside the master controller instead only local similarity calculations are performed by individual processors. The technique is explained through an example later in the text. All the processors including the Master controller have common routines of Filtering, Data Association and Local Similarity. Master controller has two additional routines one for distribution of observations which is like distributing cards (equally to all players) to available processors and all the tracks are sent to each processor. Once the computation in each slave processor is finished master receives the tracks which are stored for further processing in the next scan.

V. RESULTS & CONCLUSION

The central objective in multi-processor system is to distribute the work load in such a way that each processor ideally start/finish processing at the same instance to achieve maximum processing speed. Ideally, the speed of processing should linearly increase with each additional processor, however, in practice this is difficult to obtain. Amdahl's law states that a fraction ε of a sequential algorithm cannot be process concurrently and the speed up is given as;

$$speedup = \frac{N}{1+(N-1)\varepsilon}$$

where N is number of processors in the system. In a multi-processor target tracking system this fraction ε becomes very significant especially for global similarity calculations.

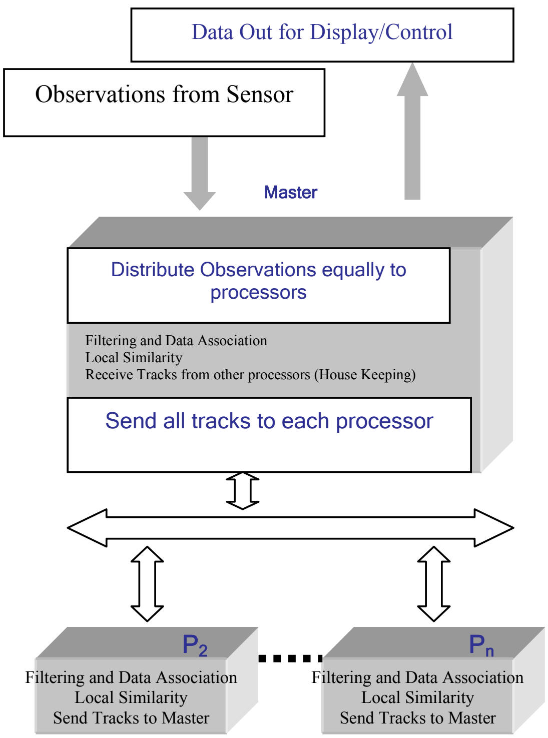

Figure 2: System Configuration

Now imagine a target tracking system with five processors (Master, P1, P2, P3 & P4) and five targets (T1, T2, T3, T4 & T5), therefore, each processor can handle one target each as shown in figure 3. Each processor is given one target and all the observations (M1, M2, M3, M4 & M5) as one does not know which target belongs to which observation. Now when targets are distinct meaning there ellipses are not interacting each target will accept one observation each and no global similarity calculations will be required in the master controller so no bottle neck. The system speed up will be nearly optimal (ignoring the house keeping and other communication overheads). However, imagine at some point in time the targets gets near to each other (cross), this means each target may accept other targets observations. Considering for example targets T2 (inside processor P2) and T3 (inside processor P3) accepts observations M2 and M3 this means both are following the same targets M2 and M3. However, the problem is they are located on two different processors and to establish that they are indeed following the same target either T2M2, T2M3 are to be sent to processor P3 or vice versa T3M2 and T3M3 are to be sent to P2. It is also possible that another target may also have accepted M2 and M3, therefore, rather spending time (over heads) in locating similar tracks all tracks are typically collected in the master processor and global similarity calculations are performed, such an implementation is given in reference [6]. However, results presented in there show exponential increase in computation time with such distribution of targets and observations technique during ambiguity region (ellipses interaction).

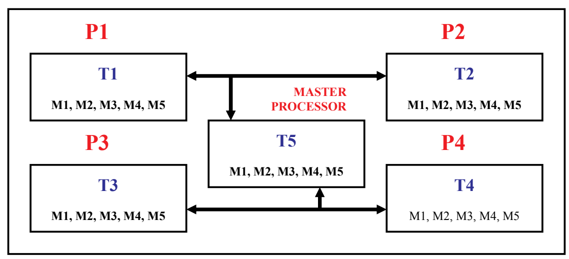

Figure 3: System with 5 Processors and 5 Targets

In our implementation we have changed this methodology of distribution of tracks and observations. Instead of distributing targets we distribute observations to the available processors as shown in figure 4, each processor has only one observation and all the tracks (T1, T2, T3, T4 & T5).

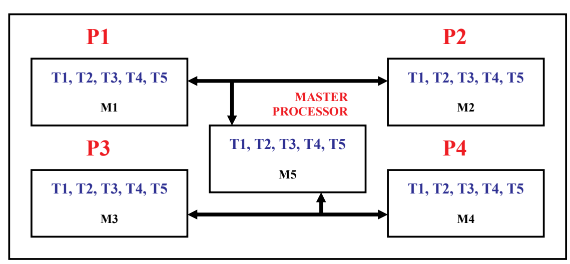

Figure 4: Global Similarity Avoidance Strategy

Now imagine the same situation of ellipse interaction; targets T2 and T3 accept observation M2 in processor P2 and M3 in processor P3, only local similarity calculations in the respective processor (p2 & P3) are required. This way not only global similarity calculations are avoided but also the overhead of sending all the targets to the master processor is saved. Also if our method of distribution is not used then the master processor have to store individual tracks from each slave processor for global similarity. Therefore, with our technique the storage requirement is also tremendously reduced.

Although many scenarios were considered and executed with our methodology however, only one sample scenario is shown in figure 5 to avoid repetition, the emphasis is on technique rather than describing scenarios. In this scenario there are 6 targets and they start at various positions and all of them cross after 30 seconds (same velocity for all). The tracking sensor is also moving in this case towards the targets along y-axis with a high speed (0.3 Km/sec). Figure 6 shows the processing time achieved with our method in comparison to the technique used in reference [6] during the ambiguity region. It can be seen that our technique is far superior during the ambiguity region scan 26 – 34, the reason is global similarity calculations are not performed. However, one could see that outside the ambiguity region the processing time for our algorithm is slightly higher which is due to the local similarity calculations performed individually by each processor. Another important point we observed in our study is that it is impossible to achieve a balance processing load even though equal number of targets are distributed among the available processors. Because tracks in individual processors may accept varying number of observations resulting individual processor to finish processing at different time instances causing the average processing time to rise.

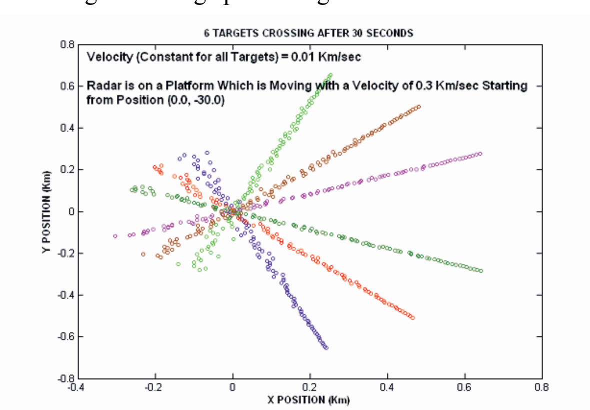

Figure 5: 6 Targets Crossing after 30 Seconds

To further illustrate the novelty of our implementation we compared the average speed-up obtained from many different scenarios with the strategy discussed in reference [6]. Figure 7 shows the results of such speed up comparison, it can see that our technique gives a better speed up compared with earlier technique. In the future we would like to study and observe speed-up by using a grid of 16, 32, 64 and 128 processors.

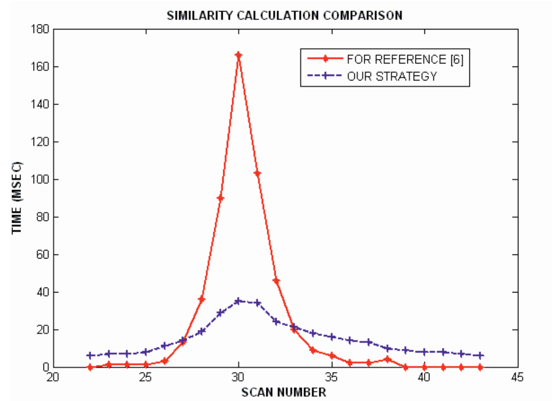

Figure 6: Peak Processing Time (Scan 22 – 43)

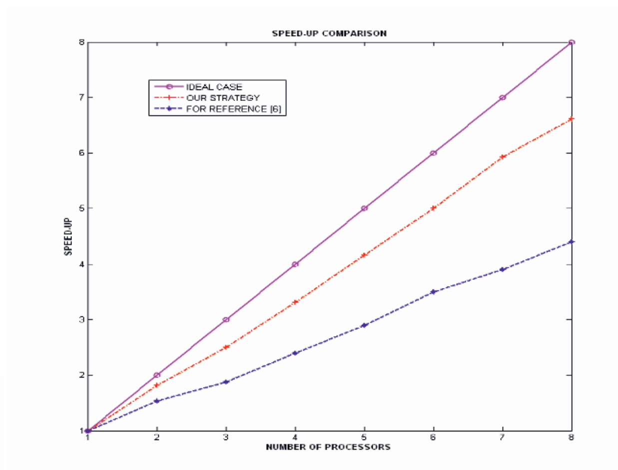

Figure 7: Speed-up Comparison

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Modelibra Software Family

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Abstract **-This paper provides a brief overview of Modelibra, the open source software family that is used to develop dynamic web applications based on domain models. The software family consists of a graphical design tool, a domain model framework, a web component framework, a collection of CSS declarations, and XML, database and Java code generators. Modelibra facilitates the definition and the use of domain models in Java. It uses Wicket for application views of a domain model. Wicket is a web framework that provides web components to construct, in an object oriented way, web concepts, such as web pages and page sections. Modelibra interprets the application model and creates default web pages based on the model. A default application may help developers validate and consequently refine the domain model. In addition, Modelibra has a collection of generic web components that may be easily reused in professional web applications to display or update entities.**

I. MODELIBRA

In computer terms, a domain model is a model of specific domain classes that describe the core data and their behaviour [1]. The heart of any software is a domain model. When a model is well designed and when it can be easily represented and managed in an object oriented language, a developer may focus on views of the software and they are what users care about the most.

Modelibra has been designed to help developers in representing and using application domain models in a restricted way. The main restriction of Modelibra, and at the same time its main feature, is that all data must be present in main memory. This and other restrictions of Modelibra minimize the number of decisions that a domain designer must make. This makes Modelibra easy to learn. Modelibra is an Open Source Software (OSS) [2]. It is hosted at JavaForge [3]. Developers of OSS may find Modelibra useful for developing their software around a domain model, for providing an easy installation for their users and for developing a web application to introduce their software to the public audience.

By default, Modelibra uses XML files to save model objects. However, Modelibra allows the use of both relational and object databases. The upgrade of an application from XML data files to a database does not require a single line of programming code to be changed. Modelibra also provides the data migration from XML files to a database. Although the focus of Modelibra is a software application with relatively small amount of data, it provides some advanced features such as transactions and undo. A domain may have several models. One of them may be a reference model where common data, common to all models within a domain, are kept. A domain model may also inherit some of its definitions from another model in the same domain. In Modelibra, a part of the base model may be exported to another model, which can be taken for an off-line work, then returned back to synchronize changes with the base model.

The Modelibra software family consists of a graphical design tool, a domain model framework, a web component framework, a collection of CSS declarations, and code generators for XML configurations, database schemas and Java classes. Modelibra facilitates the definition and the use of domain models in Java. Modelibra interprets the application model and makes it alive as a default web application, which may help developers validate and consequently refine the domain model.

The software closest to Modelibra is Eclipse Modeling Framework (EMF) [4]. The objective of EMF is to provide code generation facility for building tools and other applications based on a structured data model. It has additional components for queries, transactions, model integrity validation and service data objects. It is quite a complex software that is not that easy to learn.

The next section will start with a simple domain model, which will be used to show how a web component is built from the model.

II. DOMAIN MODEL

A domain model is a representation of user concepts, concept properties and relationships between concepts. The easiest way to present a domain model is through a graphical representation. Fig. 1 represents a domain model of a simple web application called Web Links. It features web links that are of interest to certain members.

In our case, the domain model's concepts are: Url, Question, Category, Member, Interest and Comment. Url describes a web link. Urls are categorized. Categories are organized in a tree of subcategories. Question is a frequently asked question about the use of the web application. Questions are optionally categorized. Members express their interests in categories of web links. Comments can be made about anything related to the web application.

A concept is described by its properties and neighbors, called together the concept's attributes. The Url concept has only one neighbor, the Category concept. However, the Category concept has four neighbors: Url, Question, Interest and

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Category concepts. A relationship between two concepts is represented by two neighbor directions, displayed together as a line. A neighbor direction is a concept special (neighbor) property, with a name and a range of cardinalities. A neighbor is either a child or a parent. A child neighbor has the max cardinality of N (or a number greater than 1). A parent neighbor has the max cardinality of 1. If a parent neighbor has the min cardinality of 0, the parent is optional.

A concept is represented as a list of entities. The retrieval of entities starts with the entry concepts of the domain model

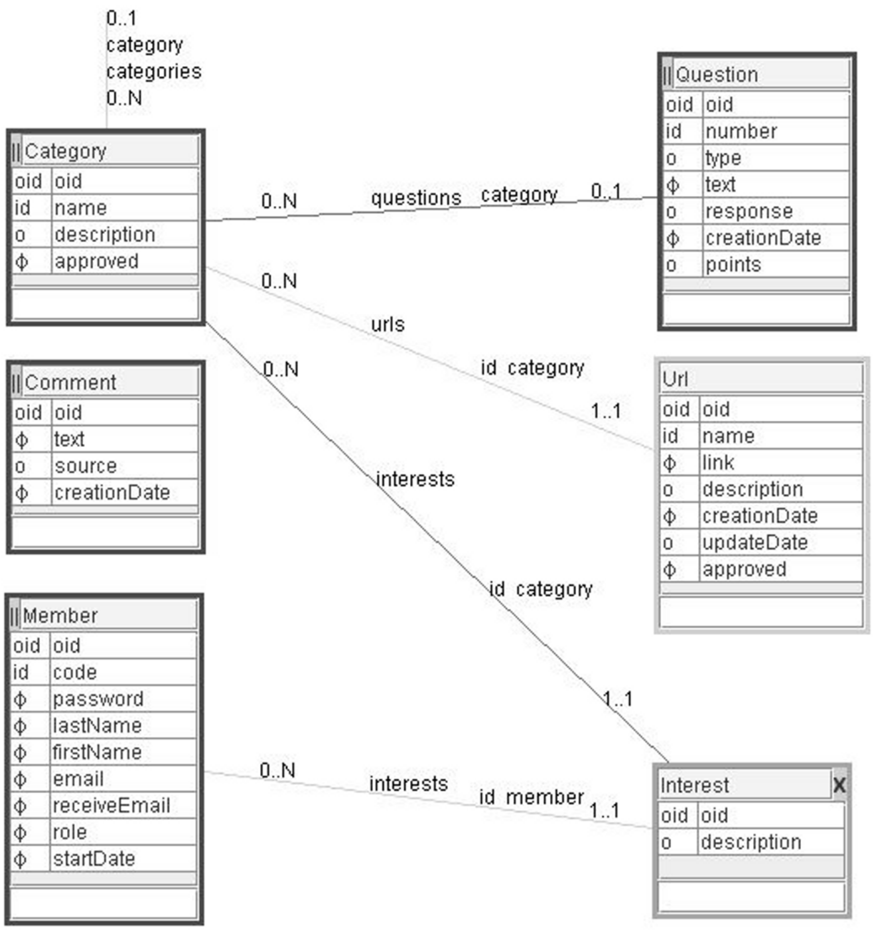

Fig. 1. Web Link domain model

In the Web Link domain model, the entry concepts are Category, Question, Member and Comment. They all have a darker border and the || symbol in the upper left corner of the concept. Once an entity of the entry concept is retrieved in the list of entities, the retrieval of neighbor entities may start. A child neighbor is represented as a list of entities. A parent neighbor is represented as a single entity. The Url concept is not an entry concept. Hence, entities of the Url concept may be reached only through its parent Category concept. The Interest concept has two parents. Thus, interests may be retrieved either from the Member concept or the Category concept. A concept that has more than one parent is called an intersection concept and has the **X** sign in the upper right corner of the concept.

Every concept has a predefined property called oid. The oid property is mandatory. It is used as an artificial identifier and is completely managed by Modelibra. Its value is unique universally. In addition, a concept may have at most one user oriented identifier (id) that consists of the concept's properties and/or neighbors. A simple id has only one property. In an entry concept, all entities must have a unique value for the concept id. However, in a non-entry child concept, the id is often unique only within the child parent.

A graphical design tool called ModelibraModeler was used to create the Web Link domain model. The tool generates a relational database schema and an XML configuration of the domain model. Modelibra reads the XML configuration and uses it quite often in interpreting the model to support different actions on the model. The following is a short excerpt of the domain configuration for the Web Link model.

```
<domain oid="1101453345237">
   <code>DmEduc</code>
   <type>Specific</type>
   <models>
      <model oid="1101453347786">
         <code>WebLink</code>
         <author>Dzenan Ridjanovic</author>
         <concepts>
            <concept oid="1101453347834">
               <code>Category</code>
               <entitiesCode>
                  Categories
               </entitiesCode>
               <entry>true</entry>
               <properties>
                  <property oid="1101453347889">
                     <code>name</code>
                     <propertyClass>
                        java.lang.String
                     </propertyClass>
                     <required>true</required>
                     <unique>true</unique>
                  </property>
                  …
               </properties>
               <neighbors>
                  <neighbor oid="1101453348237">
                     <code>urls</code>
                     <destinationConcept>
                        Url
                     </destinationConcept>
                     <type>child</type>
                     <min>0</min>
                     <max>N</max>
                  </neighbor>
               </neighbors>
            </concept>
            ...
         </concepts>
      </model>
   </models>
</domain>
```

By default, each entry point into the model is saved in its XML data file. The following is a part of categories data.

```
<categories>
  <category oid="1193171129084">
    <name>Software</name>
    <approved>true</approved>
    <urls>
      <url oid="1194575051451">
        <name>
          Open Source Software in Java(tm)
        </name>
        <link>http://java-source.net/</link>
        <creationDate>2007-11-08</creationDate>
        <approved>true</approved>
      </url>
    </urls>
  <categories>
    <category oid="1193171173850">
      <name>Framework</name>
      <approved>true</approved>
      <urls>
        <url oid="1193171279804">
          <name>Wicket</name>
   ...
```

III. ENTITIES

A concept is represented in Modelibra as two Java classes, one to represent an entity and the other to represent a list of entities, both Plain Old Java Objects (POJOs) [5]. For example, the Category concept has two classes: `Category` and `Categories`. The `Category` class extends the `GenCategory` class. The `Categories` class extends the `GenCategories` class. The generic classes are generated by Modelibra from the XML configuration. The specific classes, also generated the first time by Modelibra, are almost empty. A developer may add a specific code to the specific classes, which will not be lost when the model changes and when the generic classes are regenerated.

```
public class Category extends GenCategory
public class Categories extends GenCategories
```

The abstract `GenCategory` class extends the `Entity` class passing the `Category` class as a generic type parameter. Similarly, the abstract `GenCategories` class extends the `Entities` class.

```
public abstract class GenCategory extends
   Entity<Category>
public abstract class GenCategories extends
   Entities<Category>
```

The abstract `Entity` class implements the `IEntity` interface. The abstract `Entities` class implements the `IEntities` interface.

```
public abstract class Entity<T extends IEntity>
   extends Observable implements IEntity<T>
public abstract class Entities<T extends IEntity>
   extends Observable implements IEntities<T>
```

Modelibra has an Eclipse project skeleton, called ModelibraWicketSkeleton, where the XML configuration is generated from ModelibraModeler. The skeleton has predefined directories for a Wicket web application with a collection of CSS declarations for a web page layout. In addition to Java classes, different files, such as XML data files and property files with display text, are also generated to make a complete web application.

IV. WEB COMPONENTS

A web application is a collection of web pages. A dynamic web application has some pages that are generated dynamically based on some data. When those data change, the content of the corresponding web pages changes as well.

A web framework, called Wicket [6], is used to compose a default web application from the domain model. Wicket is a web application framework for creating dynamic web pages by using web components for web application concepts such as web pages and page sections. It uses only two technologies: Java and HTML. Wicket pages can be designed by a visual HTML editor. Dynamic content processing and form handling is all handled in Java code.

Modelibra makes a domain model alive as a web application, so that model data may be displayed as web pages and updated through forms. The model is metamorphosed into a web application with the help of the XML configuration by a web framework of Modelibra that is called ModelibraWicket. With some small changes in the XML configuration, the web application may be somewhat customized. It is important to realize that this web application is not a version that one would like to install as a web site. Its main purpose is to validate a domain model by designers and future users of the web application and consequently refine the domain model. In addition, ModelibraWicket has a collection of generic web components that may be easily reused to customize the web application.

In order to construct a web component, which represents a section of a web page, two component arguments must be prepared. In order to prepare those two arguments the domain model must be reached from the web application.

```
DmEducApp dmEducApp = (DmEducApp) getApplication();
DmEduc dmEduc = dmEducApp.getDmEduc();
WebLink webLink = dmEduc.getWebLink();
```

The `WebLink` model is obtained from the `DmEduc` domain, which is found in the `DmEducApp` that extends the `DomainApp` class from ModelibraWicket. The `DomainApp` class extends the `WebApplication` class from Wicket. Once the domain model is reached, in three lines of code, the model of the web component is determined by the `ViewModel` class of ModelibraWicket.

```
ViewModel commentsModel = new ViewModel(webLink);
Comments comments = webLink.getComments();
commentsModel.setEntities(comments);
```

The view model of the web component is based on the `WebLink` model. Its entities are `Comments` that are entry point into the model.

The view of the web component carries a Wicket id that connects the Java web component with its HTML code.

```
View commentsView = new View();
commentsView.setWicketId("commentTable");
```

Finally, the web component is constructed and added to its page container.

```
add(new EntityDisplayTablePanel(commentsModel,
      commentsView));
```

The web component comes from ModelibraWicket. Its name reflects its purpose. Starting with the end of the name, the component is a **panel** that will be placed in a section of a web page. The dynamic data will be presented as a **table** of entities. The entities will only be **display**ed without possibility to update them with this web component. The web component scope is only one **entity** concept. In short, the web component will display a table of comments, originally in color but transformed in gray in this paper. Only the `text` property is shown, because it is the only `essential` property of the `Comment` concept, defined as such in the XML configuration.

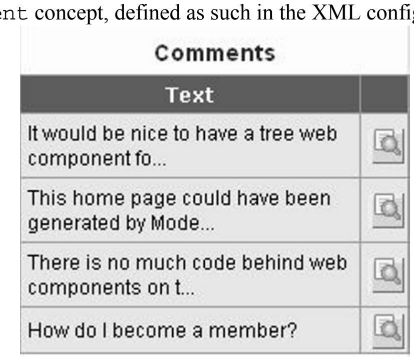

Fig. 2. Comments Web Component

The corresponding HTML code for the web component is placed in an HTML page that contains the component's dynamic data. The HTML element where the data will be displayed is determined by the opening and closing `div` tags. The opening tag has the `wicket:id` attribute whose value must be identical to the `id` value of the web component.

```
<div wicket:id="commentTable">
To be replaced dynamically by the table of comments.
</div>
```

Another generic web component from ModelibraWicket whose scope is only one concept is called `EntityPropertyDisplayListPanel`. The component is a **panel** that will be placed in a section of a web page. The dynamic data will be presented as a **list** of entities. The entities will be **display**ed without possibility to update them with this web component. Only one **property** of the concept will be shown. The component's scope is one **entity** concept.

```
ViewModel questionsModel = new ViewModel(webLink);
Questions questions = webLink.getQuestions();
questionsModel.setEntities(questions);
questionsModel.setPropertyCode("text");

View questionsView = new View();
questionsView.setWicketId("questionTextList");
questionsView.setTitle("Questions");
```

The view model consists of questions and the view title is determined by the `Questions` key in a property file of the web application.

```
Questions=Questions
```

The web component is constructed from the prepared arguments and added to the same web page but to a different section of the page.

```
add(new EntityPropertyDisplayListPanel(
        questionsModel, questionsView));
```

The given Wicket id is found in the corresponding HTML element.

```
<div wicket:id="questionTextList">
To be replaced dynamically by the list of questions.
</div>
```

The Java and HTML code of the web component, together with CSS declarations of the web application, gives the following content and look.

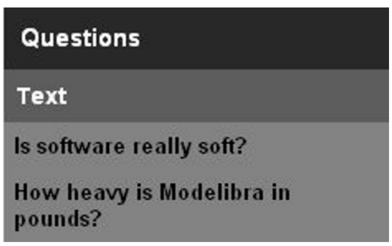

Fig. 3. Questions Web Component

The last web component shown in this paper covers two concepts and one relationship. The concepts are `Category` and `Url`. The first concept is an entry into the model, while the second concept is not. As a consequence, urls can be retrieved only through their parent category. Only approved categories will be ordered by name, which will be the only property shown. For each approved category the `urls` neighbor will be reached and only the `link` property of the neighbor concept will be displayed. In order to indicate the child neighbor and the child property two user properties are introduced.

```
ViewModel categoryUrlsModel = new
      ViewModel(webLink);
Categories categories = webLink.getCategories();
Categories orderedApprovedCategories =
        categories.getApprovedCategories()
        .getCategoriesOrderedByName();
categoryUrlsModel.setEntities(
        orderedApprovedCategories);
categoryUrlsModel.setPropertyCode("name");
categoryUrlsModel.getUserProperties()
        .addUserProperty("childNeighbor", "urls");
categoryUrlsModel.getUserProperties()
        .addUserProperty("childProperty", "link");
```

```
View categoryUrlsView = new View();
categoryUrlsView.setWicketId(
        "categoryNameUrlLinkList");
categoryUrlsView.setTitle("Category.WebLinks");
```

The `Category.WebLinks` key is used to determine a title of the web component.

```
Category.WebLinks=Category Web Links
```

The key with its value is located in a property file of the web application. In an international version of the web application there are different files with the same keys but different values. The following is the title of the web component in French.

```
Category.WebLinks=Liens Web des catégories
```

The web component has a rather long name, but the name indicates that component is a **panel** in which a **list** will be **display**ed. Only one **property** from both the **child** and **parent** concepts will be shown.

```
add(new ParentChildPropertyDisplayListPanel(
        categoryUrlsModel, categoryUrlsView));
```

The HTML code, with the help of Wicket id, determines a place within a web page where the component will be displayed.

```
<div wicket:id="categoryNameUrlLinkList">
To be replaced dynamically by the list of categories
each with its web links.
</div>
```

The web component displays a list of approved root categories, each with its web links. Even this web component, whose scope is larger than a single concept, is easy to prepare. Both Java and HTML contain only a few lines of code.

Fig. 4. Category Web Links Web Component

V. Conclusion

Modelibra is the OSS family that is used to develop dynamic web applications based on domain models. The software family consists of the ModelibraModeler graphical design tool, the Modelibra domain model framework which carries the same name as the software family to indicate its core status, the ModelibraWicket web component framework, CSS declarations in the ModelibraWicketSkeleton project, and various code generators in ModelibraModeler, Modelibra and ModelibraWicket. Modelibra facilitates the definition and the use of domain models in Java. With Modelibra, a programmer does not need to learn a complex data framework in order to create, maintain and save domain models. ModelibraWicket uses Wicket for application views of domain models. Modelibra interprets the domain model and ModelibraWicket creates the default web application, which can be customized by the XML configuration and by adding a specific code. The web application helps developers validate and consequently refine the domain model. The generic code can be regenerated without loosing the added specific code.

ModelibraWicket has a collection of generic web components that may be easily reused, within a few lines of code, in specific web pages in order to convert the default web application into a web application that responds well to user needs. If there is no generic web component in ModelibraWicket for a particular need, a specific web component may be developed by reusing Wicket components.

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SOA Design Pattern for Distributing the Object Model in n-tier Applications

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***Abstract*—This paper discusses the proposed architecture for distributing the object model (set of user defined classes) across n-tiers when building SOA, Service Oriented Architecture, based applications using WinFX set of technologies that as of now are part of the .NET Framework 3.0 [1]. It presents a qualitative commentary on the architectural issues pertinent to the way a distributed application using WinFX is structured.**

Keywords- Design Pattern for Services based architectre; Architecture guidelines for Service distribution and mapping; .Net 3.0; WinFX

I. Introduction

The foreseeable next evolution in software development is going to be the adoption of SOA, Service Oriented Architecture, by the industry [2], [14]. The path to the adoption of SOA is the next logical conclusion after transitory era of XML based Web Services [15] and revolves around the same underlying concepts of interoperatability, modularity, isolation and independency, yet it expands upon the existing model based on XML and SOAP by adhering to W3C's new WS-* standards to offer what Web Service lagged in terms of security, transactions, reliability and performance besides solving the host of issues concerning a service specification in 'soft' terms [14] and the infamous middleware problem [16]. Microsoft is the current leader in providing the developers a unified programming model for achieving all of these tenets on a single platform [17] with complete support for developing an end-to-end solution that supports SOA, bridging the tool gap [14], in the shape of its new framework for communication, the WCF [3], Windows Communication Foundation, that forms part of the WinFX (now .Net 3.0) [4], Windows Framework Extensions, set of technologies.

In this paper I aim to discuss the proposed design pattern for distributing the object model across n-tiers when using WinFX set of technologies. It outlines the experiences gained and lessons learnt from a year of application development using the pre-release WinFX SDK from Microsoft. Design Patterns have become popular tool of choice for expressing and promoting the guidelines for reusable software architecture/components, experiences and best practices among the Programmer community since the famous text by GoF [11] and the numerous follow ups such as [12], [13] and others.

II. A WinFX Premier

Before discussing the pattern for distributing the object model itself I would be outlining the technology specifics from the WinFX (now .Net 3.0) that pertain to the discussion to follow. WinFX, comprises a host of revolutionary technologies that would change the scenario of application development.

On client side the technology introduced is WPF [5], Windows Presentation Foundation that would aid in the development of rich client applications offering a unifying platform for 2D, 3D drawings, animations and images be it using Vector Graphics, text or animation and even multimedia. It would be compiled and would put both developers and designers on-board [18] by allowing files to be written in mark-up or programming language such as C# or VB.Net and even a combination of the two, allowing for separation of concerns and freeing both to learn a host of presentation and design technologies/softwares. One feature that would be of particular interest to us is Data Binding that aids in gluing the GUI, Graphical User Interface, to object model (set of user-defined classes) so that the changes to UI (user interface) are reflected in object model and vice versa. This helps a great deal in standardizing and minimizing the effort to update the two views (object model and UI components) of the same model (information stored in a data store, for example database). Furthermore we can Databind not only between UI Components and object model but also between two UI Components and even between UI Components and XML data.

On the communication side, WCF, Windows Communication Foundation, is the main workhorse providing support for writing SOA based applications supporting WS-* set of standards [3] using built-in features simply by adding attributes and/or adding tool generated XML configuration descriptors allowing for developers to choose from a variety of built-in channel stacks depending on the features of the platform that the application wish to benefit from such as Reliable Transport, Security, Reliable Sessions, Interoperatability and Message Patterns for writing services that can even be hosted in .Net CLR, Common Language Runtime, apart from conventional support for hosting in IIS, Internet Information Server.

Then there is more to WinFX in the form of set of technologies such as WF [6], Windows Workflow Foundation

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that aims to aid in extending the concept of workflows to day to day application development allowing for definition of flow of application using GUI, for example UML Activity Diagrams. Then there are enhancements to the C# language itself that would change the facet of application development offering unprecedented level of leverage to developers in areas such as Database access in the shape of LINQ [7], the Language integrated Query that would allow the developers to query any data store, be it a Database, Object Collection, Data Sets or XML data, uniformly using a single language (that would be a part of the C#/VB.Net) that would be compiled, allowing for compile time checks, and translated into the specifics by the framework, for example say SQL, for querying database, allowing the developers to query using an abstract declarative format rather than specific monolithic construct. DLINQ [8], .NET Language-Integrated Query for Relational Data, is the name given to LINQ to SQL converter that's now a part of the new version of C# language.

Having built the foundation now lets delve deeper into exploring the alternatives for writing applications when building distributed applications supporting SOA using these WinFX set of technologies.

III. SCENARIOS FOR DISTRIBUTED APPLICATION USING

A typical WinFX distributed application would consist of the following tiers: a Client tier (View/ViewModel), an Application tier (Services/DLINQ) and a Database tier (or any other data store such as XML data, so called Model).

A typical scenario consists of a WPF based or a traditional HTML/JavaScript client (possibly on a client machine) interacting with either one of the following:

1) A WCF based service (possibly on a server machine) that talks to database (possibly on a database server) at the backend using DLINQ.

2) Alternatively the client can directly talk to the backend data store, again either using Databinding or DLINQ

3) A mixture of the above two (Databinding to DLINQ classes).

It would be not of place to mention here that John Gossman, the architect for Microsoft Expressions Designer, had proposed a similar design pattern that he calls Model/View/ViewModel [9]. In this pattern he talks about directly binding the UI Components (View) to the Model (data store) or alternatively binding the View to a ViewModel (user defined classes that contain business logic) that in turn binds to the Model (data store). This model, that he claims is nurtured from his experience with developing Expressions Designer using WPF [10], in my opinion, fits nicely for the WPF-only applications where the Model/ViewModel would also be residing on the client machine, apart from the View (UI) itself. This pattern corresponds closely to the second and third options, as mentioned above, of binding the client (the View in WPF) directly to the data store, excepting DLINQ. The pattern, however, manifests its weaknesses when applied to a distributed tiered application, when he mentions that the View would bind to ViewModel that comes from backend Services (this option corresponds closely to first alternative for binding View, WPF client, to (WCF based) service, again excepting DLINQ). Here, he goes no further to describe the other side of the story such as what happens on the server machine viz. the object model and technologies to be used (as mentioned above in the first scenario). I have tried to address these issues based upon my experience from the end-to-end application development and consequently would limit myself to exploring the first alternative.

Now let's discuss the alternatives for distributing the Object Model (the set of classes implementing the application services) across different tiers. I would be comparing the alternatives from the developer perspective, offering a qualitative commentary on the architectural issues pertinent to the way a distributed application using WinFX is structured. In this context, I would be using the following terminologies: View (to refer to the UI i.e. User Interface, that allows interaction with the client, this is normally what gets displayed in browser), View Model (to refer to the set of classes acting as intermediary between the interaction UI in the View and higher tier components), Message Model (refers to the set of classes implementing the services, these typically implement the business logic), Entity Model (refers to the set of classes providing object to relational mapping between the underlying data store and the higher level services, these may be implemented using built-in services of DLINQ facilities beginning .Net Framework 3.0 onwards), Data Store (to refer to any structure for holding of user generated data, it can be classes acting as containers or a full fledged DBMS offering permanent storage), Services (to refer to the set of exposed functionality satisfying user's business needs realized through interaction of objects and/or possibly other services).

A. Fully Mapped ViewModel and MessageModel

In this approach we bind the View (client using WPF) to a ViewModel (using application specific classes) that both execute on the client machine (though they can be further fragmented). The ViewModel talks to a MessageModel that interacts with the EntityModel both executing on the server side (with possible another layer of fragmentation between the two). MessageModel refers to the set of classes that expose the EntityModel, representing the object-to-relational mapping for the data base (or any other data store) entities (possibly using DLINQ set of classes), in the form of Services. This model is as depicted in the Fig. 1 below:

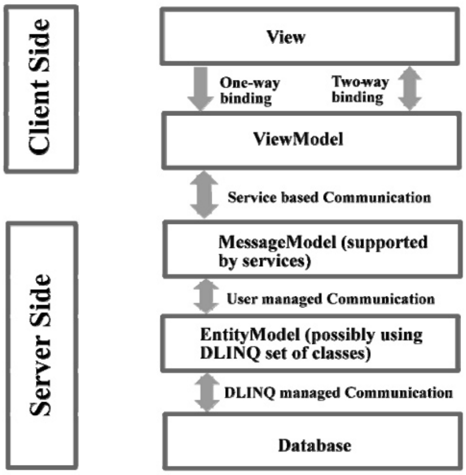

Figure 1. Fully Mapped ViewModel & MessageModel.

The disadvantage of this approach is that it requires changing the server side MessageModel to support different ViewModel while offering the advantage of simplicity of implementation due to one-to-one correspondence/direct talking to the exposed services from the MessageModel. A further con can be the bloated ViewModel because of the need to accommodate different Views in the same ViewModel. Also, on the server side, it means that a single MessageModel need to cater to all the Clients, which again results in bloating the MessageModel. This also renders full exposure of the MessageModel to the outside world, with restrictions implemented in ViewModel, this while again adding to the size of the ViewModel, can lead to rogue clients plying on the application's (unauthorized) services. This model can be improved in the following modification to the above pattern.

B. Using different ViewModels that map to the same MessageModel

We can modify the above ObjectModel so that we have different ViewModels depending on the type of user, say, User1ViewModel, User2ViewModel and so on, that talk to the same MessageModel on the server side. This tackles the brittleness in the above ObjectModel by fragmenting the change in the ViewModel that abstracts the same MessageModel. This model is as depicted in the Fig. 2 below:

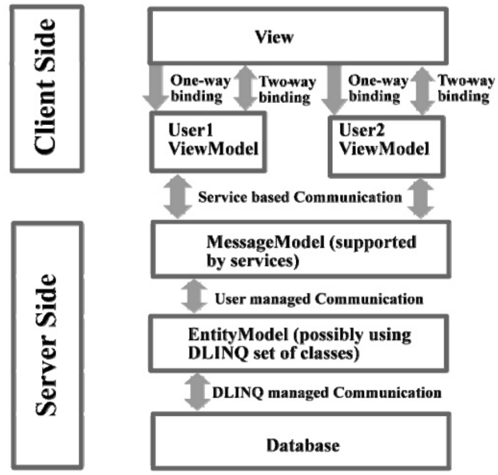

Figure 2. Different ViewModel's for same MessageModel.

This ObjectModel overcomes the coupling problem from the first distribution scenario as we keep changing the ViewModel's corresponding to the user/type of user for the core set of service as exposed by the same MessageModel. This lets us modify the clients in isolation, a typical scenario could be pay as you go SOA based application that can be updated to consume as much services the client has paid for. Obviously, exposing the services bare allows us to isolate the impact of change at the Client Side at the peril of opening over services to possibly rogue clients (written by developers or plagiarized UserViewModel). This possible carry over problem from the previous pattern can be addressed by assigning a unique identification to client (for example registration key/token) that can only be used per machine, per session. We explore a final alternative using the following modification.

C. Using different ViewModels that map to the different MessageModels

A further modification to the above model is to expose a single MessageModel using different MessageModels that have a one to one correspondence with user's ViewModels. This model is as depicted in the Fig. 3 below:

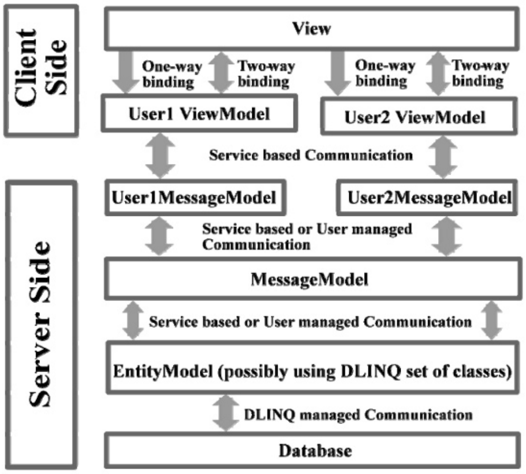

Figure 3. Tying ViewModels to MessageModels.

This ObjectModel has the advantage that it allows the developer to customize the MessageModel as well as the ViewModel depending on the user. This means that per client changes are easy to configure (because they are needed to be done only at the Server Side), while still allowing for the flexibility of individuality from the previous distribution pattern, due to the retention of individual UserModel. This also allows for limiting the public offering by restricting only the services through MessageModel's corresponding to some client (corresponding ViewModel). This also means that we are free to change the underlying MessageModel while keeping the exposed UserMessageModel the same to allow for modifications reflecting architectural restructuring of services, plug and play (in context of SOA) or performance enhancements such as creating a separate per user UserMessageModel for important clients while offering same UserMessageModel to less important ones and so on.

A final modification to the above architecture might be to factor out the centralized MessageModel and have the user specific MessageModels talk directly to the EntityModel. This approach might be used side by side with the aforementioned approach allowing some MessageModels to talk to the EntityModel and requiring others to go through the centralized MessageModel depending on the application requirements. This allows for performance enhancements for it results in decreasing level of indirection at the cost of increasing the coupling of the exposed service's (UserMessageModel's) to the underlying Entities, so that changing one necessitates change in the other which may require making change to several UserMessageModel's.

IV. CONCLUSION

I have outlined some of the possible alternatives for distributing the object model when developing SOA based applications using WinFX set of technologies offering a qualitative commentary on the architectural issues pertinent to the way the distributed application using WinFX may be structured. These alternatives are just one possible viewpoint and I am sure more would be introduced in the coming days as the API nears finalization stage. I look forward to constructive criticism from the developer community regarding the design patterns outlined here.

ACKNOWLEDGMENT

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CLS and CLS Close: The Scalable Method for Mining the Semi Structured Data Set

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Abstract— **Semistructured pattern can be formally modeled as Graph Pattern. The most important problem to be solved in mining large semi structured dataset is the scalability of the method. With the successful development of efficient and scalable algorithms for mining frequent itemsets and sequences, it is natural to extend the scope of study to a more general pattern mining problem: mining frequent semistructured patterns or graph patterns. In this paper, we extend the methodology of pattern-growth and develop a novel algorithm called CLS (Canonical Labeling System), which discovers frequent connected subgraphs efficiently using either depth-first search or breadth-first search strategy.**

A novel canonical labeling system and search order are devised to support efficient pattern growth. CLS has advantages of simplicity and efficiency over other methods since it combines pattern growing and pattern checking into one procedure. Based on CLS, we develop CLS Close to mine closed frequent graphs, which not only eliminates redundant patterns but also substantially increases the efficiency of mining, especially in the presence of large graph patterns.

Keywords — **frequent pattern, closed pattern, graph mining, CLS code, canonical label**

I. INTRODUCTION

Semistructured data appears when the source does not impose a rigid structure on the data, such as the web, or when data is combined from several heterogeneous sources. Unlike unstructured raw data (like image and sound), semistructured data does have some structure, but unlike structured data (such as relational or object-oriented databases), semistructured data has no absolute schema or class fixed in advance. For example, in the movie XML database [2], some movies have more actors than others; some fields (e.g., *Award*) are missing for some movies; some actors have birthday recorded and some do not; etc. As a result, the structure of objects is irregular and a query over the structure is as important as a query over the data. This structural irregularity, however, does not imply that there is no structural similarity among semistructured objects. On the contrary, it is common for semistructured objects describing the same type of information to have similar structures. In mathematical terms, we called semi structured data set as graph data set

As a general data structure, labeled graph can model complicated structures. A labeled graph has labels associated with its vertices and edges. We denote the vertex set of a graph g by $V(g)$ and the edge set by $E(g)$. A label function, l, maps a vertex or an edge to a label. A graph g is a subgraph of another graph g' if there exists a subgraph isomorphism from g to g'.

The remaining of the paper is organized as follows. In Section 2, the preliminary concept of subgraph isomorphism, Canonical Labeling System (CLS) Coding & Lexicographic Ordering is introduced in Section 3. Section 4 formulates cls extension & completeness. We conclude our finding into CLS Algorithm that discussed in Section 5. Closed Frequent Graph Mining will be discussed in Section 6 and conclude in Section 7.

II. PRELIMINARY CONCEPT

Definition 1 (***Subgraph Isomorphism***) A subgraph isomorphism is an injective function $f : V(G) \rightarrow V(G')$, such that (1)$\forall u \in V(G)$, $l(u) = l'(f(u))$, and (2)$\forall(u, v) \in E(G)$, $(f(u), f(v)) \in E(G')$ and $l(u, v) = l'(f(u), f(v))$, where l and l' are the label function of G and G' respectively. Let g be the graph with the vertex set and the edge set mapped by the subgraph isomorphism f in G'. Graph g is an embedding of G in G'.

If G is a subgraph of G', then G' is a supergraph of G, denoted by $G \subseteq G'$. G' is the proper supergraph of G if $G \subset G'$.

Definition 2 (***Frequent Graph***) Given a labeled graph dataset, $D = \{G_1, G_2, \ldots, G_n\}$, support($g$) (or frequency($g$)) is the percentage (or number) of graphs in D where g is a subgraph. A frequent graph is a graph whose support is no less than a minimum support threshold, *min_support*.

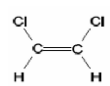

Figure 1. Example of Graph

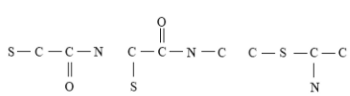
Figure 2: A Sample Graph Dataset

Figure 1 shows a sample chemical structure dataset. Figure 2 depicts two of frequent subgraphs in this dataset if the minimum support is set at 66.6%. A graph g can be extended by adding a new edge e. Let $g \diamond_x e$ be the new graph. Edge e may or may not introduce a new vertex to g. If e introduces a new vertex, we denote the new graph by $g \diamond_{xf} e$, otherwise, $g \diamond_{xb} e$. Algorithm 1 (IntuitionGraph)

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 186–191.

illustrates a naive frequent graph mining algorithm. For each discovered graph g, it performs extensions recursively until all the frequent graphs with g embedded are discovered. The recursion stops once a graph is not frequent any more.

Algorithm 1 ***IntuitionGraph***(g, D, *min_support*, S)
Input: A frequent graph g, a graph dataset D, and *min_support*.
Output: The frequent graph set S.
1: if $g \in S$ then return;
2: else insert g to S;
3: scan D once, find all edges e such that g can be extended to $g \diamond_x e$;
4: for each frequent $g \diamond_x e$ do
5: Call IntuitionGraph($g \diamond_x e$, D, *min_support*, S);
6: return;

IntuitionGraph is simple, but not efficient. The key issue is the inefficiency of extending a graph. The same graph can be discovered many times. For example, there may exist n different ($n-1$)-edge graphs which can be extended to the same n-edge graph. The repeated discovery of the same graph is computationally inefficient. Our method have the following salient properties: (1) it reduces the generation of duplicate graphs; (2) it need not search previous discovered frequent graphs for duplicate detection; and (3) it does not extend any duplicate graph but still guarantees the complete set of frequent graphs.

III. Canonical Labeling System (CLS) Coding & Lexicographic Ordering

In this section we develop main techniques to represent and extend graphs efficiently, which includes mapping a graph to a CLS code (CLS Coding), building a lexicographic ordering among these codes, and mining CLS codes based on this lexicographic order.

Depth-limited search is one way of traversing graph. Initially, a starting vertex is randomly chosen and the vertices in a graph are marked so that we can tell which vertices are visited.

The visited vertex set is expanded repeatedly until a full CLS tree is built. Given a graph G with a CLS tree T, the *forward edge* (*tree edge* [3]) set contains all the edges in the CLS tree, denoted by E_T^f , and the *backward edge* (*back edge* [3]) set contains all the edges which are not in the CLS tree, denoted by E_T^b .

IntuitionGraph extends a frequent graph in every possible position, which may generate a large number of duplicate graphs. In our method, we introduce a more sophisticated extension method.

The new method restricts the extension as follows: Given a graph G and a CLS tree T in G, a new edge e can be added between the CLS vertex and other vertices on the CLS path (backward extension); or it can introduce a new vertex and connect to vertices on the CLS path (*forward extension*). Since both kinds of extensions take place on the CLS path, we call them *CLS extension*, denoted by $G\diamond_r e$ (for brevity, T is omitted here).

A. CLS Coding

We transform each subscripted graph to an edge sequence so that we can build an order among these sequences. The goal is to select the subscripting which generates the minimum sequence as its base subscripting. There are two kinds of orders in this transformation process: (1) edge order, which maps edges in a subscripted graph into a sequence; and (2) sequence order, which builds an order among edge sequences, i.e., graphs. We introduce edge order in this subsection and sequence order in the next subsection.

Formally we define a linear order, $\prec_T$, in N^2 such that $e_1 \prec_T e_2$ holds if and only if one of the following statements is true (assume $e_1 = (i_1, j_1)$, $e_2 = (i_2, j_2)$):

(i) $e_1, e_2 \in E_T^f$, and $j_1 < j_2$ or $i_1 > i_2 \wedge j_1 = j_2$.

(ii) $e_1, e_2 \in E_T^b$, and $i_1 < i_2$ or $i_1 = i_2 = j_1 < j_2$.

(iii) $e_1 \in E_T^b$, $e_2 \in E_T^f$ and $i_1 < j_2$.

(iv) $e_1 \in E_T^f$, $e_2 \in E_T^b$ and $j_1 \le i_2$

Definition 3 (*CLS Code*) Given a subscripted graph G_T , an edge sequence (e_i) can be constructed based on relation $\prec_T$, such that $e_i \prec_T e_{i+1}$, where $i = 0, \ldots, |E| - 1$. The edge sequence (e_i) is a CLS code, written as *code*(G, T).

B. CLS Lexicographic Order

The formal definition of CLS code order is given as follows:

Definition 4 (***CLS Lexicographic Order***) Let Z be the set of CLS codes of all graphs.

Two CLS codes α and β have the relation $\alpha \le \beta$ (**CLS Lexicographic Order** in Z) if and only if one of the following conditions is true. Let $\alpha = \text{code}(G_\alpha, T_\alpha) = (a_0, a_1, \ldots, a_m)$ and $\beta = \text{code}(G_\beta, T_\beta) = (b_0, b_1, \ldots, b_n)$.

(i) $\exists t$, $0 \le t \le \min(m, n)$, $a_k = b_k$ for all k s.t. $k < t$, and at $<_U b_t$

(ii) $a_k = b_k$ for all k s.t. $0 \le k \le m$ and $m \le n$.

Definition 5 (***Minimum CLS Code***) Let Z(G) be the set of all CLS codes for a given graph G. Minimum CLS Code of G, written as **CLS**(G), is a CLS code in $Z(G)$, such that for each $\gamma \in Z(G)$, $\text{CLS}(G) \le \gamma$.

Theorem 1 Given two graphs G and G$'$, G is isomorphic to G′ if and only if **cls**(G) =**cls**(G$'$).

Proof. If G is isomorphic to G′, then there is an isomorphic function $f: V(G) \rightarrow V(G')$.

Given a CLS subscripting of G, by assigning the subscript of v for each $v \in V(G)$ to $f(v)$, a CLS subscripting is thus built in G'. The CLS code produced by these two subscriptings of G and G′ must be the same, otherwise, f is not an isomorphic function between G and G′. Therefore, $Z(G) \subseteq Z(G')$. Similarly, $Z(G) \supseteq Z(G')$. Hence, $Z(G) = Z(G')$ and **cls**(G) = **cls**(G').■

IV. CLS EXTENSION & COMPLETENESS

So far, we have defined CLS code, minimum CLS code, and base subscripting. In this paper, minimum CLS code is treated as canonical label. For every frequent graph, we only conduct the CLS extension on its base subscripting and ignore other subscriptings.

Below, the CLS extension of G specifically means the CLS extension on the base subscripting of G.

Definition 6 (***k CLS Extension Set***) Let C_α^k be a set of CLS codes generated from a CLS code α through k times of CLS extensions. That is, $C_\alpha^k = \{\beta | \exists b_1,...,b_k, \beta = \alpha\diamond_r b_1 \diamond_r,...,\diamond_r b_k, \forall i, 0 \le i \le k, b_i \in U, and\ \alpha, \beta \in Z\}$.

C_α^k is called the k right most extension set of α.

Let O_γ be the set of all CLS codes which are less than a CLS code γ (γ ∈ Z), $O_\gamma = \{\eta | \eta < \gamma, \eta \in Z\}$, η where η and γ are not necessary from the same graph..

Lemma 1 (***CLS Code Extension***) Let α be the minimum CLS code of a graph G and β be a non-minimum CLS code of G. For any CLS code δ generated from β by one CLS extension, i.e., $\delta \in C_\beta^1$,

(i) δ is not a minimum CLS code,
(ii) **cls**(δ) cannot be extended from β, and
(iii) **cls**(δ) is either less than α or can be extended from α, i.e., cls(δ) ∈ $O\alpha \cup C_\alpha^1$ and cls(δ) < β

Proof. First, statement (i) can be derived from statement (ii). Assume to the contrary that δ is a minimum CLS code. It is contradicting to statement (ii). Secondly, if statement (iii) is true, statement (ii) must be true. The reason is as follows. Since α and β have the same size and $\alpha < \beta$, $O_\alpha \subseteq O_\beta$ and $C_\alpha^1 \subseteq O_\beta$. If cls($\delta$) ∈ $O\alpha \cup C_\alpha^1$, then cls(δ) < β, which means that cls(δ) cannot be extended from β.

Now we prove that statement (iii) is true. Let the edge set of G be $\{e_0, e_1, \ldots e_n\}$, the edge sequence of α be $e_{i0}, e_{i1}, \ldots, e_{in}$, where $0 \le i_m \le n$. CLS code δ is extended from β by adding a new edge, e, i.e., $\delta = \beta \diamond_r e$. Let G_δ be the graph represented by δ.

G_δ is a new graph built from G_α with a new edge e. There are two situations: (1) e introduces a new vertex; or (2) e connects two existing vertices in G. Consider the first situation. We construct an alternative CLS code of G_δ based on α as follows. Let v_x be the vertex of e in G. If v_x is on the CLS path of α, then $(e_{i0}, \ldots, e_{in}, e)$ forms an alternative CLS code for G_δ. Otherwise, there must exist a forward edge (v_{w1}, v_x) in α such that $w_1 < x$ by CLS subscripting. Since v_x is not on the CLS path, there exist forward edges (v_{w2}, v_{w3}) in α s.t. $w_2 < w_1$ and $w_1 < w_3$. Let e_{im} be the smallest edge among them according to the linear order ($\prec_T$). By inserting the new edge e right before e_{im} in the edge sequence, $e_{i0}, e_{i1}, \ldots, e_{im}, \ldots, e_{in}$, the new sequence forms a CLS code of G_δ.

This code is less than α. Therefore, there is an alternative CLS code existing for G_δ, which should be in one of the following two formats: (1) $(e_{i0}, \ldots, e_{in}, e)$, which belongs C_α^1; (2) $(e_{i0}, \ldots, e_{im-1}, e, e_{im}, \ldots)$ and the code formed by this sequence is less than α. Similarly, the same conclusion holds for the second situation. In summary, an alternative CLS code δ′ of G_δ exists such that δ′ ∈ $O\alpha \cup C_\alpha^1$. Since ***cls***(δ) ≤ δ′, ***cls***(δ) ∈ $O\alpha \cup C_\alpha^1$ and ***cls***(δ) < β.■

Theorem 2 (Completeness) Performing only the CLS extensions on the minimum CLS codes guarantees the completeness of mining results.

Theorem 2 is equivalent to the following statement: *Any CLS code extended from non-minimum CLS codes is not minimum*. Thus, it is not necessary to extend non minimum CLS codes at all. Formally, given two CLS codes, α and β, if α = ***cls***(β) and α ≠ β, then for any CLS code, δ, extended from β, i.e., $\delta \in \bigcup_{k=1}^{\infty} C_\beta^k$, ***cls***(δ)<$\beta$.

Proof. Assume that the following proposition is true,

$\forall p \in \bigcup_{k=1}^{\infty} C_\beta^k$, if ***cls***($p$) <$\beta$, then $\forall q \in C_p^1$, ***cls***(q) <β (Proposition 1)

By Lemma 1, $\forall p \in C_\beta^1$, we have cls(p) < β since β ≠ ***cls***(β) (initial step). Using the above proposition, by induction we have $\forall p \in \bigcup_{k=1}^{\infty} C_\beta^k$, ***cls***($p$) <$\beta$. That means any k CLS extension from a non-minimum CLS code must not be a minimum CLS code. Furthermore, its minimum CLS code is less than β. Now we prove Proposition 1.

For any $p \in \bigcup_{k=1}^{\infty} C_\beta^k$ p if ***cls***(p) <β, then for any $q \in C_p^1$, by Lemma 1, ***cls***(q) ∈ $O_{cls(} \cup C_{dfs(p)}$. Since ***cls***(p) <β and the length ***cls***(p) is greater than that of β, according to CLS lexicographic ordering, $\forall \delta \in C_{dfs(p)}^1$, $\delta < \beta$. Therefore, ***cls***(q) < β. ■

The above theorem says that we need not extend non-minimum CLS codes, but minimum CLS codes. It is interesting to figure out how many non-minimum CLS codes may be generated from minimum CLS codes. The following lemma bounds the number of non- minimum CLS codes that a given graph can generate from minimum CLS codes. These non-minimum CLS codes are the duplicate graphs we have to detect and prune.

Lemma 2 (CLS Duplicate Code Cardinality) Let Z(G) be the set of CLS codes that a graph G can generate from minimum CLS codes. If $|E(G)| > 2$, then $|Z(G)| \leq |E(G)| \times |V(G)|$. That is, the number of G's codes which are extended from other minimum codes is bounded by the product of the number of its edges and vertices.

Proof. Please refer to [8]

This bound is not tight because the *m*-th edge cannot be added to an arbitrary position mentioned above in order to keep the topology of G. For many graphs, $|Z(G)|$ is less than |E(G)|. Since we prune all non-minimum CLS codes and their descendants, we only access those duplicate graphs which are extensions of other minimum CLS codes. Therefore, for each frequent graph, the number of duplicate graphs accesses is limited. Considering we are mining frequent graphs, we have the following straightforward anti-monotonicity property.

Lemma 3 (Anti-monotonicity of Frequent Patterns) If a graph G is frequent, then any subgraph of G is frequent. If G is not frequent, then any supergraph of G is not frequent. It is equal to say $\forall \beta \in \bigcup_{k=1}^{\infty} C_{\alpha}^{k}$, if a CLS code α is infrequent, β is infrequent too.

Proof. Please refer to [8]

These lemmas and theorems set the foundation of our mining algorithm. The pruning of non-minimum CLS codes in the tree ensures that the search is complete while the anti-monotonicity property can be used to prune a large portion of the search space.

V. CLS ALGORITHM

In this section, we present the implementation details of our algorithm, CLS. Two major pruning techniques will be introduced, followed by the analysis of CLS.

Algorithm 2 ***MainLoop*** (D, *min_support*, S).
Input: A graph dataset D, and *min_support*.
Output: The frequent graph set S.
1: remove infrequent vertices and edges in D;
2: $S^1 \leftarrow$ all frequent 1-edge graphs in D;
3: sort S^1 in the increasing CLS lexicographic order;
4: $S \leftarrow S^1$;
5: **for each** edge $e \in S^1$ do
6: initialize s with e, set $D_s = \{g \mid g \in D$ and $e \in E(g)\}$; (only graph id is recorded)
7: ***CLS***(s, D_s, *min_support*, S);
8: $D \leftarrow D \setminus e$;
9: **if** $|D| <$ *min_support*;
10: **break**;

Algorithm 3 ***CLS*** (s, D, *min_support*, S)
Input: A CLS code s, a graph dataset D, and *min_support*.
Output: The frequent graph set S.
1: **if** s ≠ ***dfs***(s), then
2: **return**;
3: insert s into S;
4: set C to $\varnothing$;
5: scan D once, find all edges e such that s can be *right-most* extended to $s \diamond_r e$;
insert $s \diamond_r e$ into C and count its frequency;
6: sort C in DFS lexicographic order;
7: for each frequent $s \diamond_r e$ in C do
8: Call **CLS**($s \diamond_r e$, *D, min_support*, S);
9: return;

A. Analysis

Since subgraph isomorphism problem is an NP-complete problem, the worst case runtime of CLS is exponential. If measured by the number of subgraph and/or graph isomorphism tests conducted by CLS, the time complexity is bounded by $O(kFS + rF)$, where k is the maximum number of embeddings existing between a frequent subgraph and a graph in the dataset, F is the number of frequent subgraphs, S is the dataset size, and r is the maximum value of $Z(g)$, where g is a frequent subgraph pattern. The major advantages of CLS are listed as follows.

• **No complicated candidate generation and downward closure test**. In CLS, a frequent (k+1)-edge graph is grown from a k-edge frequent graph directly. It does not perform complicated candidate generation and downward closure test which usually are conducted in an Apriori-like algorithm.

• **Memory efficient**. CLS can perform either depth-first search and breadth-first search, while an Apriori-based approach usually has to conduct breadth-first search. BFS suffers from much higher memory usage than DFS. While the graph dataset is too huge and has to be disk resident, BFS can reduce the disk scans over the dataset. CLS can work in either way.

VI. CLOSED FREQUENT GRAPH MINING

According to the Apriori property, all the subgraphs of a frequent graph must be frequent. A large graph pattern may generate an exponential number of frequent subgraphs. For example, among 423 confirmed active chemical compounds in an AIDS antiviral screen dataset, there are nearly 1,000,000 frequent graph patterns whose support is at least 5%. This renders the further analysis on frequent graphs nearly impossible. For the AIDS antiviral dataset mentioned above, among the one million frequent graphs, only about 2,000 are closed frequent graphs. If further analysis, such as classification or clustering, is performed on closed frequent graphs instead of frequent graphs, it will achieve similar accuracy with less redundancy and higher efficiency.

Since the maximal pattern set is a subset of the closed pattern set, usually it is more compact than the closed pattern set. However, it cannot reconstruct the whole set of frequent patterns while the closed frequent pattern set can.

Definition 7 (Closed Frequent Graph) A frequent graph G is closed if and only if there is no G' such that $G \subset G'$ and *support*(G) = *support*(G').

A. Equivalent Occurrence

In frequent graph mining, whenever CLS Algorithm finds a non-minimum CLS code, CLS stops searching its descendants in the lexicographic search tree. We want to identify other conditions under which some search branches can be pruned for closed frequent graph mining.

Given two graphs g and G, where g is a subgraph of G, the number of embeddings (the number of subgraph isomorphisms) of g in G is written as $\varphi(g,G)$.

Definition 8 (Occurrence) Given a graph g and a graph dataset $D = \{G_1, G_2, \ldots, G_n\}$, the occurrence of g in D is the number of embeddings of g in D, i.e., $\sum_{i=1}^{n} \varphi(g,G)$, written as $\mathcal{I}(g,D)$.

Let g' be the graph formed by a graph g with a new edge and ρ be a subgraph isomorphic function between g and g'. Assume both graphs g and g' are subgraphs of a graph G. It is possible to transform an embedding of g in G to an embedding of g' in G. Let f be a subgraph isomorphism of g in G, and f' be a subgraph isomorphism of g′ in G. If $f(v) = f'(\rho(v))$ for each v in $V(g)$, we call f an extendable subgraph isomorphism and f′ an extended subgraph isomorphism of f. Intuitively, if a subgraph isomorphism f was already built between g and G, we can extend it to obtain a subgraph isomorphism between g′ and G. The number of extendable isomorphisms is written as $\phi(g, g', G)$.

Definition 9 (Extended Occurrence) Let g′ be the graph formed by a graph g with a new edge. Given a graph database D = $\{G_1, G_2, \ldots, G_n\}$, the extended occurrence of g′ in D with respect to g is the number of extendable subgraph isomorphisms of g in D, i.e., $\sum_{i=1}^{n} \phi(g, g'/G_i)$, written as $\mathcal{L}(g, g', D)$.

The extended occurrence is the number of embeddings of g that can be transformed to the embeddings of g' in a graph dataset.

Definition 10 (Equivalent Occurrence) Let g' be the graph formed by a graph g with a new edge. Given a graph database D, g and g' have equivalent occurrence if $\mathcal{I}(g,D) = \mathcal{L}(g, g', D)$.

Let e be the new edge added in g such that $g' = g \diamond_x e$. Given a graph G, if $g \subset G$ and $g' \subseteq G$, the inequality $\varphi(g,G) > \phi(g, g', G)$ always holds. Hence, we have $\mathcal{I}(g,D) > \mathcal{L}(g, g', D)$. When $\mathcal{I}(g,D)$ is equal to $\mathcal{L}(g, g', D)$, it means wherever g occurs in G, g' also occurs in the same place. Let h be a supergraph of g. If h does not have the edge e, then h will not be closed since *support*(h) = *support*($h \diamond_x e$) ($h \diamond_x e$ is an abbreviation of $h \diamond_{xf} e$ and $h \diamond_{xb} e$. Unless specifically noted, $h \diamond_x e$ is constructed by adding e into an embedding of g in h such that g′ becomes a subgraph of $h \diamond_x e$). Therefore we only need to extend g' instead of g. Whenever we "extend" a new frequent graph g, we perform two actions: find all the embeddings of g in the graph dataset and then count the embeddings of its supergraphs that have one more edge. If one of its supergraphs is frequent and has not been discovered, the extension will be repeatedly performed on this supergraph. We name this search strategy **Early Termination** since we stop searching the supergraphs of g except g'.

B. Failure of Early Termination

Unfortunately, Early Termination does not work in one case. The following example shows the situation where Early Termination fails.

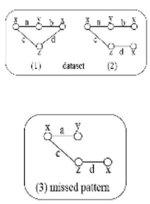

Figure 3: Failure of Early Termination

Failure Case Suppose there is a dataset with two graphs shown in Figures 3(1) and 1(2) and we want to find closed frequent graphs whose minimum support is 2. Let g be $x \underline{\;a\;} y$ and g' be $x \underline{\;a\;} y \underline{\;b\;} x$. For brevity, $x \underline{\;a\;} y$ represents a graph with one edge, which has "a" as its edge label and "x" and "y" as its vertex labels. As one can see, edge $y \underline{\;b\;} x$ is always associated with edge $y \underline{\;b\;} x$ and $\mathcal{I}(g,D)$ is equal to $\mathcal{L}(g, g', D)$ for this dataset. Can we apply early termination and stop extending g? No, we cannot. Otherwise, the pattern shown in Figure 3(3) will be lost. This graph cannot be extended from $x \underline{\;a\;} y \underline{\;b\;} x$. If we attach edge $y \underline{\;a\;} x$ to the vertex labeled "y" in Figure 3(3), the newly formed graph will not be frequent any more. This case shows a failure situation of Early Termination and the vertex with edges $y \underline{\;b\;} x$ and $z \underline{\;d\;} x$ forms the failure point.

The following theorem sets the condition where Early Termination can be applied.

Theorem 3 Let g' be the graph formed by a graph g with a new edge e. If g′ has equivalent occurrence with g in a graph dataset D and for each graph h, s.t., $g \subseteq h$ and $g' \not\subset h$, one of the following conditions holds,

(i) $\exists h \diamond_{xf} e$, s.t., $g' \subset h \diamond_{xf} e$ and $I(h,D) = L(h, h \diamond_{xf} e, D)$,

(ii) $\exists h \diamond_{xb} e$, s.t., $g' \subset h \diamond_{xb} e$ and $I(h,D) = L(h, h \diamond_{xb} e, D)$,

then it is unnecessary to extend g except g'.

Proof. Since h contains g, but not g', edge e can be added into an embedding of g in h to construct an embedding of g' in $h \diamond_{xf} e$ or $h \diamond_{xb} e$. If $I(h,D) = L(h, h \diamond_{xf} e, D)$ or $I(h,D) = L(h, h \diamond_{xb} e, D)$, then one of the following statements must be true: (1) $\forall G \in D$, if $h \subset G$, then $h \diamond_{xf} e \subseteq G$; or (2) $\forall G \in D$, if $h \subset G$, then $h \diamond_{xb} e \subseteq G$. Therefore, either *support*(h) = *support*($h \diamond_{xf} e$) or *support*(h) = support($h \diamond_{xb} e$). In each case, h is not closed. That is, for any graph that is a supergraph of g, but not of g', it must not be closed. Thus it is unnecessary to extend g except g'.■

C. Close Frequent Graph (CLS Close)

In previous sections, we discussed two techniques, right-most extension for frequent graph mining and early termination for closed graph mining. Now we put them together and formulate *CLS Close* for closed frequent graph mining.

Besides using Early Termination to remove some non-closed frequent graphs, we also need to get rid of those remaining non-closed frequent graphs. One approach is to compare the newly discovered graph with the previously discovered graph patterns.

Lemma 4 Given graphs g and h, if $g \subset h$ and support(g) = support(h), then there exists a graph g' and an edge e, $g' = g \diamond_x e$ and $g' \subseteq h$, such that support(g) = support(g').

Proof. If $g \subset h$, we can always construct a graph g' with an edge e such that $g' = g \diamond_x e$ and $g' \subseteq h$. We have *support*(g') $\geq$ *support*(h) and *support*(g′) $\leq$ *support*(g). If *support*(g) = *support*(h), then *support*(g) = *support*(g').■

Lemma 4 shows that if we want to determine whether a graph is closed, we only need to check the support of its supergraphs that have one more edge. If there exists a graph g′ such that $g' = g \diamond_x e$ and *support*(g) = *support*(g′), then g is not closed.

Algorithm 4 ***CLS Close*** (*s, p, min_support, D, S*)

Input: A CLS code s, its parent p, a graph dataset D, and *min_support*.

Output: The closed frequent graph set S.

1: **if** s $\neq$ **CLS**(s), **then**

2: **return**;

3: **if** there exists a graph g' such that $g' = g_p \diamond_x e$, $g' < gs$, $I(g_p,D) = L(g_p, g',D)$, and g' does not have the crossing point on e **then**

4: **return**;

5: check the crossing situation in g_s;

6: insert s to S if it is closed;

12: for each frequent graph $s \diamond_r e$ do

13: Call CLS Close($s \diamond_r e$, s, D, min support, S);

14: return;

VII. CONCLUSIONS

In this paper, we have investigated several issues related to efficient and scalable mining of frequent graph patterns in large graph databases and addressed the possible inefficiency in previously developed candidate-generation-and-test algorithms. A new lexicographic ordering system is formulated in CLS, and several optimization techniques are proposed to further leverage the performance. CLS is promising since it can be applied (with some minor extensions) to mining nearly all kinds of frequent substructures, including sequences, trees, and lattices. It provides a general framework for scalable mining of complicated patterns.

We built *CLS Close* based on CLS to mine closed frequent graph patterns, a critical problem in graph pattern mining because mining all patterns is inherently inefficient and redundant. Several new concepts are introduced in *CLS Close* including equivalent occurrence, early termination, and its failure detection.

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Goal Functions from Minimax to Maximin in Multicriteria Choice and Optimization

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Abstract - A technique is proposed for constructing a continuous range of goal functions from minimax to maximin using the generalized form of a weighted power mean (WPM). Each item of multicriteria choice or alternative variant in an optimization problem is characterized by a vector of performance parameters and it is assumed that each performance parameter is constrained by its target value. An "imperfect maximin-minimax" principle of multiobjective optimality is suggested as well as the technique for its implementation. The technique is based on expert evaluation of each performance parameter's degree of freedom expressed as the worst compensable deviation from its target value. The data obtained in an expert evaluation is sufficient for calculating the order and weights of a WPM on the basis of which the goal function is to be built.

I. Introduction

We assume that every item in a multi-criteria choice (solution in an optimization problem) is characterized by *a performance vector* $Y = (Y_1, ..., Y_n)$, where Y_i is the i-th *performance parameter* $(i \in 1:n)$. We assume that Y is constrained by a vector $Y_T = (Y_{1.T}, ..., Y_{n.T})$ of *target values* of performance parameters. In design, e.g., it is planned values of parameters, given in the requirements specification of the object of design, that can be treated as target values.

We suppose, without loss of generality, that every parameter Y_i is a non-negative real number, characterized by *preferable change* or *orientation*: either increase for parameters which characterize positive (useful, desirable) factors, or decrease for parameters which characterize negative (harmful, undesirable) ones. Parameters with increasing orientation will be called *maximum-oriented parameters*, or *max-parameters*, whereas parameters with decreasing orientation will be called *minimum-oriented parameters*, or *min-parameters*. We will use notations $I_{\max}$ and $I_{\min}$ for the subscript numbers of parameters characterizing max- and min-oriented parameters in Y, respectively.

Our purpose is to construct a goal function (GF), the numerical value of which would serve as an *efficiency (integral performanve) index* of any item (solution) in question. Goal functions have also their specific orientation, so we can speak of *maximum-oriented goal functions (max-GF) and minimum-oriented goal functions (min-GF).*

To ensure that all components of Y have the same orientation as the goal function, we introduce an orientation-unifying transformation, which transforms vector $Y = (Y_1, \ldots, Y_n)$ into vector $\overline{Y} = (\overline{Y}_1, \ldots, \overline{Y}_n)$:

$$\overline{Y}_i = \begin{cases} Y_i, \text{ if } i-\text{th parameter and the goal function} \\ \quad \text{have the same orientation;} \\ 1/Y_i \text{ otherwise.} \end{cases}$$

It is expedient to normalize performance parameters using target values $Y_{1.T}, ..., Y_{n.T}$ as normalizing coefficients. Then the normalized value of Y_i will look the following way: $y_i = \overline{Y}_i / \overline{Y}_{i.T}$.

We introduce *a normalization function* $y = N(Y)$ which transforms performance vector $Y = (Y_1, \ldots, Y_n)$ into *a normalized performance vector* $y = (y_1, \ldots, y_n)$ in such a way, that all its components could have the same orientation. It is worth mentioning that $y_{i.T} = N(Y_{i.T}) = 1$, i.e. the use of this normalization function makes normalized target values equal unity. This feature is useful from the practical point of view.

The main problem of goal functions in multi-criteria choice and optimization consists of

a) formulating the principle of optimality which should determine the properties of an optimal solution and answer the question in what sense an optimal solution surpasses all other alternatives in the given problem situation;

b) choosing the form of the goal function and its coefficients for appropriate implementation of the accepted principle of optimality.

Unlike single criterion optimization, in multi-criteria optimization one can use diverse principles of optimality (trade-off schemes) that would lead to selecting different "optimal" solutions (see, e.g., [1]). The majority of approaches focus on trade-off analysis im kleinen (in the small), i.e. on determining the goal function weights based on pairwise comparison of reference performance vectors $Y = (Y_1, \mathrm{K}, Y_n)$, made by experts. In other methods, such as the goal programming approach, they minimize one objective

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while constraining the remaining objectives to be less than given target values.

An approach proposed in this paper differs in the respect, that it is based on mutual-parameter-compensability analysis im grossen (in the large), namely, on expert evaluation of the worst compensable deviation of each parameter from its target value.

II. Convolution and Goal Functions

Consider the following formulation of a multi-criteria (vector) optimization problem (confining ourselves without loss of generality to minimization)

$$\left.\begin{aligned} & g(y)\to \min,\\ & y=N(Y),\ Y=F(X),\\ & y\le y_T,\ \ y_T=N(Y_T),\ X\in\Psi. \end{aligned}\right\} \qquad (1)$$

In this formulation, $Y=F(X)$ is a functional dependence of the performance vector Y on the vector $X=(X_1,\dots,X_m)$ of control variables (control parameters): $Y_i=F_i(X)$ $(i\in 1:n)$, where vector X is also constrained within some region Ψ. The nature of Ψ usually involves physical laws and structural constraints and, in any case, is different from the nature of constraints imposed on performance vector Y.

The composition $G(X)=g(N(F(X)))$ is usually called *a goal* or *objective function* and $g(y)$ will be called *a convolution function (CF)*. The value of $g(N(Y_T))$ will be called *a target value of the efficiency index (target efficiency)*.

Choosing the orientation of the CF $g(y)$ and, consequently, goal function $G(X)$, should be done to make it match the orientation of the majority of performance parameters. In practice, a limited number of forms are used as $g(y)$: linear, multiplicative, maximin and minimax.

The general approach to constructing maximum-oriented convolution functions can be based on using a utility function $u_i(y_i)$ that expresses the utility of a numerical value of each performance parameter y_i $(i\in 1:n)$. The assumption of additive character of utility allows one to express the integral efficiency of an item, characterized by vector y, the following way: $g(y)=\sum_{i=1}^{i=n}u_i(y_i)$. Similarly, constructing minimum-oriented convolution functions can be based on using a concept of a disutility function $d_i(y_i)$: $g(y)=\sum_{i=1}^{i=n}d_i(y_i)$.

We will use *the weighted power mean (WPM)* [2,3] as a CF. This mean is defined as follows [4]

$$g(y)=M_r(w,y)=\left(\sum_{i=1}^{i=n}w_i y_i^r\right)^{1/r}, \qquad (2)$$

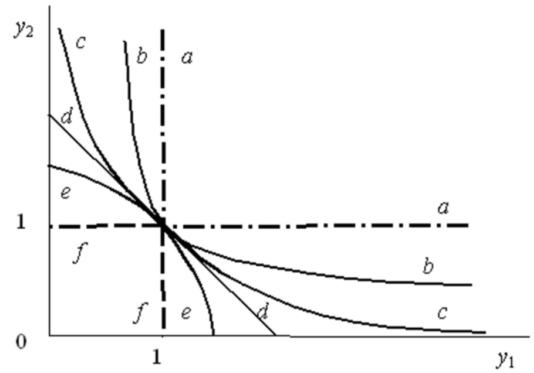

Fig. 1. The isoefficiency curves for several values of WPM convexity parameter: *a)* $r\to-\infty$, *b)* $r<0$, *c)* $r=0$, *d)* $r=1$, *e)* $r>1$ and *f)* $r\to+\infty$

where $y_i\ge 0$ is a normalized value of the i-th performance parameter; $w_i\ge 0$ are the weights such that $\sum_{i=1}^{i=n}w_i=1$; $r\in(-\infty,+\infty)$ is the order of the mean, which we will call *a convexity parameter*.

When r varies from $+\infty$ to $-\infty$, $M_r(w,y)$ provides a continuous range of means from $\max(y_1,\dots,y_n)$ to $\min(y_1,\dots,y_n)$, including a linear and a multiplicative form.

The isoefficiency curves for $g(y)=M_r(w,y)=1$ and various values of r are given in Fig. 1 (for purposes of graphical representation we consider the case of $n=2$):

a) $r\to-\infty$, *b)* $r<0$, *c)* $r=0$, *d)* $r=1$, *e)* $r>1$, and *f)* $r\to+\infty$.

At $r=1$, the WPM gives the most widely used mean with linear dependence on performance parameters: $g(y)=\sum_{i=1}^{i=n}w_i y_i$. The main drawback of the linear mean as a multiobjective criterion is that the decrease in any performance parameter even to zero can be compensated by a finite increase in other performance parameters. This property usually means that, in the case of contradictory performance parameters, a linear form cannot adequately represent the decision maker's preference relation over a wide range of parameter values.

The limit of (2) at $r\to-\infty$ is $\min\{y_1,\dots,y_n\}$ [4], which can be used to formulate a maximin principle: $g(y)=\min\{y_1,\dots,y_n\}=\ \to\max$. It was proposed, e.g., by the philosopher Rawls [5], as a justice criterion for social systems. According to this principle, the system should be designed to maximize the position of those who will be worst off in it. In technical systems, the maximin principle can also be used, e.g., as a criterion to eliminate traffic bottle-necks in computer networks.

It can easily be seen that maximin is an uncompromising criterion because, in the above example about social systems, even a slightest decrease in the advantages of the least fortunate leads to an overall decrease of justice which cannot be compensated even by an infinite increase in the advantages of the most fortunate.

The limit of (2) at $r \to +\infty$ is $\max\{y_1,\dots,y_n\}$ [4]. It can be used as a minimax principle $g(y) = \max\{y_1,\dots,y_n\} \to \min$, which is just as uncompromising as the maximin principle. It can be used as a criterion for minimizing the maximum possible deviation from the target values of performance parameters, e.g., to estimate an integral effect of several factors of environmental pollution.

III. A Principle of Multiobjective Optimality

In management and design, adequacy of multi-criteria evaluation of decisions depends on the principle of optimality built into the convolution function. To formulate the principle of optimality we need to give some definitions using the notion of a performance parameter target value, introduced above.

Definition 1. A set of all possible values of max-parameter Y_i, meeting the constraint $Y_i < Y_{i.T}$ (a set of values of min-parameter Y_i, meeting the constraint $Y_i > Y_{i.T}$), will be called *a pre-target region* of Y_i. A set of all possible values of max-parameter Y_i, meeting the constraint $Y_i > Y_{i.T}$ (a set of values of min-parameter Y_i, meeting the constraint $Y_i < Y_{i.T}$), will be called *a post-target region* of Y_i.

Definition 2. If $g(N(Y + \Delta Y)) = g(N(Y_T))$ and $\Delta Y = (\Delta Y_1,\dots,\Delta Y_n)$, where all deviations ΔY_j of parameters Y_j $(j = 1,\dots,i-1,i+1,\dots,n)$ are in their post-target regions and ΔY_i is the deviation of Y_i into its pre-target region, we will say that ΔY_i is *adequately compensated at the expense* of deviations ΔY_j $(j = 1,\dots,i-1,i+1,\dots,n)$.

Definition 3. The greatest deviation $\Delta Y_{i.\mathrm{comp}}$ of parameter Y_i from its target value into its pre-target region, which can be adequately compensated at the expense of other parameters, will be called *a worst compensable deviation* of Y_i.

Definition 4. The value of $Y_{i.\mathrm{comp}} = Y_i + \Delta Y_{i.\mathrm{comp}}$, where $\Delta Y_{i.\mathrm{comp}}$ is a worst compensable deviation of Y_i, will be called *a worst compensable level* (WCL) of Y_i.

The following formulation of the principle of multiobjective optimality, consisting of three statements, is proposed:

1. The orientation of a max-parameter is increase and the orientation of a min-parameter is decrease.
2. An increment of utility of every succeeding unit added to a max-parameter in its post-target region is less than an increment of utility of any preceding unit added to the parameter in question. Similarly, a decrement of disutility of every succeeding unit subtracted from a min-parameter in its post-target region is less than a decrement of disutility of any preceding unit subtracted from it.
3. The possibility of adequate compensation of any parameter in its pre-target region must satisfy, first of all, the decision maker's preference idea, expressed as the worst compensable level of the parameter in question.

The first statement of the principle of optimality expresses the natural preference to the growth of positive factors and to reduction of negative ones.

The second statement for max-parameters is an analogy of Gossen's First Law in economic science [6]: the continuance, increase or repetition of the same kind of consumption yields a continuously decreasing satisfaction (pleasure) up to a point of satiety. Sometimes it is also formulated as the hypothesis of diminishing marginal utility that states that as the quantity of a good consumed by an individual increases, the marginal utility of the good will eventually decrease.

The third statement imposes constraints on possible mutual compensation of parameters.

In the next section of this paper, we will focus on the capability of the WPM to realize all the three statements.

IV. WPM as a Multiobjective Function

The partial derivative of $M_r(w,y)$ with respect to each performance parameter y_i

$$\frac{\partial M_r(w,y)}{\partial y_i} = \frac{w_i y_i^{r-1}}{M_r^{r-1}(w,y)}$$

is always positive, what corresponds to the first statement of the principle of optimality.

The sign of the second partial derivative

$$\frac{\partial^2 M_r(w,y)}{\partial y_i^2} = \frac{w_i y_i^{r-2}(r-1)}{M_r^{r-1}} \times \left(1 - \frac{w_i y_i^{r-1}}{M_r^r(w,y)}\right)$$

depends only on the sign of the factor $r-1$.

When $r \ge 1$, we have $\partial M^2(w,y)/\partial y_i^2 \ge 0$ and, in accordance with the second statement of the principle of optimality, the WPM for this region of r can be used as a min-CF $g(y) = \left(\sum_{i=1}^{i=n} d(y_i)\right)^{1/r} \to \min$. In this case, $d(y_i) = w_i y_i^r$ is the disutility of the i-th performance parameter and $g(y)$ can be treated as a monotonic transformation of integral disutility $\sum d(y_i)$ of vector y.

Similarly, when $r \le 1$, we have $\partial M^2(w,y)/\partial y_i^2 \le 0$. and therefore, the WPM for these values of r, can be used as a max-CF.

When $0 \le r \le 1$, $g(y)$ is a monotonic transformation of the integral utility $\sum u(y_i)$, representing the efficiency of vector y: $g(y) = \left(\sum_{i=1}^{i=n} u(y_i)\right)^{1/r} \to \max$.

When $r \le 0$, $g(y)$ is a monotonic transformation of the inverse of the integral disutility $\sum d(y_i)$ of vector y: $g(y) = \left(\sum_{i=1}^{i=n} d(y_i)\right)^{1/r} \to \max$.

To show the dependence of the WPM on the value of one performance parameter, the values of all the other parameters being fixed at the target level, we introduce the following function: $\varphi(Y_i) = \left[1 - w_i\left(1 - N^r(Y_i)\right)\right]^{1/r}$.

Fig. 2 shows the curves of $\varphi(Y_i)$ for a max-CF and $r < 0$, whereas Fig. 3 shows similar curves for a min-CF and $r > 1$. The curves for max-parameters, when $Y_i \to \max$, demonstrate the saturation effect in their respective post-target regions. The curves for min-parameters demonstrate the same effect, when $Y_i \to 0$. The saturation effect for max-parameters corresponds to Gossen's first law, whereas for min-parameters, the effect in question can be treated as an analogy of Gossen's first law for min-parameters.

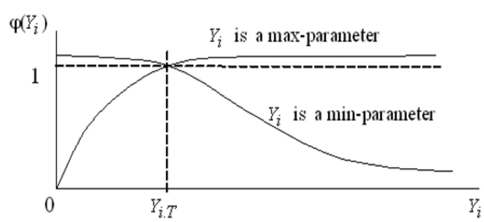

Fig. 2. The dependence of the max-CF for $r < 0$ on the value of one performance parameter Y_i, the values of all the other parameters being fixed at the target level

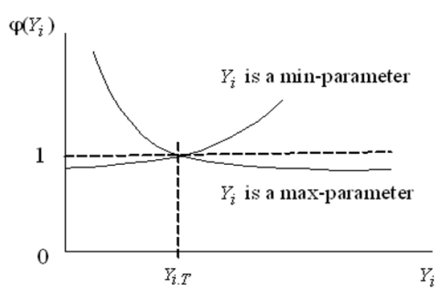

Fig. 3. The dependence of the min-CF for $r > 1$ on the value of one performance parameter Y_i, the values of all the other parameters being fixed at the target level

TABLE 1
THE UPPER AND LOWER BOUNDARIES OF y_i

Actual range of r	Boundary condition	Boundary of y_i
$r < 0$	$\bigwedge_{j \ne i}(y_j \to \infty)$	Lower $c / w_i^{1/r}$
	$c^r - \sum_{j \ne i} w_j y_j^r = 0$	Upper ∞
$r = 0$	$\bigvee_{j \ne i}(y_j \to \infty)$	Lower 0
	$\bigvee_{j \ne i}(y_j = 0)$	Upper ∞
$r > 0$	$c^r - \sum_{j \ne i} w_j y_j^r = 0$	Lower 0
	$\bigwedge_{j \ne i}(y_j = 0)$	Upper $c / w_i^{1/r}$

In the pre-target regions, these curves show the impossibility to compensate neither a zero value of max-parameters, nor an infinite increase in min-parameters.

To analyze the possibility of adequate compensation of performance parameters in the pre-target region, consider the following formula that expresses the value of a performance parameter y_i for the fixed value of $M_r(w, y) = c$:

$$y_i = \begin{cases} w_i^{-1/r}\left(c^r - \sum_{j \ne i} w_j y_j^r\right)^{1/r}, & r \ne 0; \\ \sqrt[r]{\dfrac{c}{\prod_{j \ne i} y_j^{w_j}}}, & r = 0. \end{cases} \tag{3}$$

From (3), one can derive the upper and lower boundaries of performance parameter y_i, as well as formulae, in terms of other parameters y_j $(j = 1, \ldots, i-1, i+1, \ldots, n)$, expressing the conditions for obtaining the upper and lower boundaries in question (see Table 1).

The presented formulae show that for max-CF, i.e. for $r \le 1$, the possibility of mutual parameter compensation for $r \in (-\infty, 0]$ and $r \in (0, 1]$ differs. When $r < 0$, parameter y_i cannot be less than $c / w_i^{1/r}$, the rest of performance parameters y_j $(j \ne i)$ having to take an infinite value to compensate a decrease in y_i down to $c / w_i^{1/r}$. At $r = 0$, the zero value of y_i can only be compensated with an infinite increase of any other performance parameter.

Finally, when $1 > r > 0$, the form $M_r(w, y)$ displays the same characteristics as at $r = 1$, i.e. zero values of $n - 1$

performance parameters can be compensated with a finite value $y_i = w_i^{-1/r} c^r$ of one parameter.

Denote $y_{i.\text{comp}}$ the normalized WCL of parameter y_i. At $c = 1$ when $r \le 1$, it is the lower boundary, and when $r \ge 1$, it is the upper boundary, that represent respective worst-compensable levels.

Therefore we can write from Table 1 the following formula for $y_{i.\text{comp}}$:

$$y_{i.\text{comp}} = \begin{cases} 1/w_i^{1/r} & \text{for } r < 0; \\ 0 & \text{for } 1 \ge r \ge 0; \\ 1/w_i^{1/r} & \text{for } r \ge 1. \end{cases} \quad (4)$$

Finishing the theoretical part of this paper, two remarks should be made.

First of all, we consider only the case when $r < 0$ or $r > 1$, and we do not consider the case of $g(y) = M_r(w, y) \to \max$, $1 \ge r \ge 0$, where a decrease in any performance parameter y_j to zero can be compensated by a finite increase in all the other parameters and evaluation of weights of the CF can be based on the traditional trade-off analysis im kleinen, e.g., ranking or pairwise comparison technique.

Secondly, we have assumed that the WCL of any parameter Y_i can be compensated at the expense of infinite compensatory values $\tilde{Y}_j$ of all other max-parameters and zero compensatory values of all other min-parameters. Suppose we have adopted the following boundary compensatory values of normalized performance parameters: $\tilde{y}_i$ and their worst compensable values $y_{i.comp}$ $(i \in 1:n)$. Then we can find the parameters of the WPM by solving the equation

$$\left.\begin{aligned} \sum_{j=1,\ldots,i-1,i+1,\ldots,n} w_j \tilde{y}_j^r + w_i y_{i.comp}^r = 1 \ (i \in 1:n), \\ \sum_{j \in 1:n} w_j = 1 \end{aligned}\right\}$$

with respect to w_j $(j \in 1:n)$ and r. If one restricted the compensatory values of max-parameters to some finite, greater than their target, levels, and for min-parameters – to some non-zero, smaller than their target, values, the parameters of the resulting parameters of the WCL would not change appreciably.

V. EXPERT ESTIMATION OF WPM PARAMETERS

As has already been stated, an approach proposed in this paper differs in the respect, that parameters of the WPM are estimated by using a target value $Y_{i.T}$ and *a worst compensable deviation* from the target value $\Delta Y_{i.\text{comp}}$.

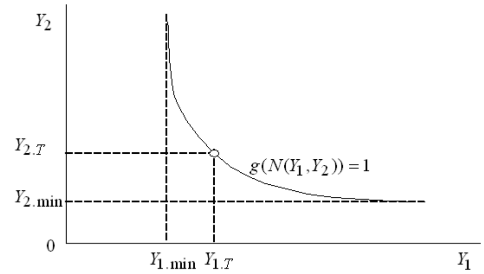

Fig. 4. The isoefficiency curve for $r < 0$, $Y_1 \to \max$, $Y_2 \to \max$, as well as the target and worst compensable levels of performance parameters Y_1 и Y_2

Keeping in mind that $\Delta Y_{i.\text{comp}} < 0$ for $i \in I_{\max}$ and $\Delta Y_{i.\text{comp}} > 0$ for $i \in I_{\min}$, we will use the following notation

$Y_{i.\min} = Y_{i.T} + \Delta Y_{i.\text{comp}}$ and $Y_{i.\max} = Y_{i.T}$ for $i \in I_{\max}$,

as well as

$Y_{i.\max} = Y_{i.T} + \Delta Y_{i.\text{comp}}$ and $Y_{i.\min} = Y_{i.T}$ for $i \in I_{\min}$.

Keeping in mind that $\Delta Y_{i.\text{comp}} < 0$ for $i \in I_{\max}$ and $\Delta Y_{i.\text{comp}} > 0$ for $i \in I_{\min}$, we will use the following notation

$Y_{i.\min} = Y_{i.T} + \Delta Y_{i.\text{comp}}$ and $Y_{i.\max} = Y_{i.T}$ for $i \in I_{\max}$,

as well as

$Y_{i.\max} = Y_{i.T} + \Delta Y_{i.\text{comp}}$ and $Y_{i.\min} = Y_{i.T}$ for $i \in I_{\min}$.

Thus, for the purpose of analysis im grossen, it is possible to represent an interval between the upper and lower boundaries of any performance parameter independent of its orientation as $[Y_{i.\min}, Y_{i.\max}]$.

As an example, Fig. 4 shows the isoefficiency curve $g(N(Y_1, Y_2)) = 1$, as well as target and worst compensable levels of Y_1 and Y_2, for $r < 0$, when Y_1, Y_2 and convolution $g(N(Y_1, Y_2))$ are max-oriented. One can see that the decrease in Y_1 down to $Y_{1.\min}$ can only be compensated by an infinite increase in Y_2. Similarly, the decrease in Y_2 down to $Y_{2.\min}$ can only be compensated by an infinite increase in Y_1.

Experts are asked to evaluate, e.g., the “optimistic” target value $Y_{i.T}$, and “the worst permissible deviation” $\Delta Y_{i.\text{comp}}$ of parameter Y_i from its target value, or “the optimistic ($Y_{i.\max}$ for max-parameters, $Y_{i.\min}$ for min-parameters) and pessimistic ($Y_{i.\min}$ for max-parameters, $Y_{i.\max}$ for min-parameters) estimates of the required value of Y_i“, or “the boundaries of the interval $[Y_{i.\min}, Y_{i.\max}]$, where a satisfactory value of parameter Y_i should be”.

In any case, we obtain the WCL $Y_{i.\text{comp}}$ and, using the normalization function, we find $y_{i.\text{comp}} = N(Y_{i.\text{comp}})$

$(i \in 1:n)$. In the case of a max-CF, i.e. $r < 0$, the WCL of parameter y_i has the value

$$y_{i.\mathrm{comp}} = y_{i.\min} = y_{i.T} + \Delta y_{i.\mathrm{comp}} < 1,$$

and, in the case of a min-CF, i.e. $r > 0$, it has the value

$$y_{i.\mathrm{comp}} = y_{i.\max} = y_{i.T} + \Delta y_{i.\mathrm{comp}} > 1.$$

It is sufficient to have $y_{i.\mathrm{comp}}$ $(i \in 1:n)$ for calculating the weights and convexity parameter of the WPM. Using (4), we obtain the following formula for calculating the weights

$$w_i = 1 / y^r_{i.\mathrm{comp}} \quad (i \in 1:n). \tag{5}$$

Using the normalization condition $\sum w_i = 1$, one can write the equation

$$\sum_{i=1}^{i=n} 1 / y^r_{i.\mathrm{comp}} = 1. \tag{6}$$

Solving (6) with respect to r and putting the received value of r in (5) we calculate the weights of performance parameters.

VI. Example

The performance vector of a vehicle has three components: highway speed Y_1 (kilometers per hour), load capacity Y_2 (kilograms), and fuel consumption Y_3 (litres per 100 kilometers). The target and worst compensable levels are given by experts:

$Y_{1.T} = Y_{1.\max} = 100$, $Y_{1.\mathrm{comp}} = Y_{1.\min} = 90$,

$Y_{2.T} = Y_{2.\max} = 800$, $Y_{2.\mathrm{comp}} = Y_{2.\min} = 750$,

$Y_{3.T} = Y_{3.\min} = 7$, $Y_{3.\mathrm{comp}} = Y_{3.\max} = 8$.

In this example, $y_{1.\min} = 90/100$, $y_{2.\min} = 750/800$, $y_{3.\min} = 7/8$. Equation (6), in this example, is

$$1 / y^r_{1.\min} + 1 / y^r_{2.\min} + 1 / y^r_{3.\min} = 1.$$

Solving this equation, we find $r = -11.381$. Putting the received value of r in (5), we find $w_1 = 0.301$, $w_2 = 0.480$, and $w_3 = 0.219$.

It can be seen, that the smaller the WCL of max-parameter $y_{i.\mathrm{comp}} = y_{i.\min}$, the larger is its weight. The resulting CF is

$$g(N(Y_1, Y_2, Y_3)) = (w_1 (Y_1 / Y_{1.\min})^r + w_2 (Y_2 / Y_{2.\min})^r + w_3 (Y_{3.\max} / Y_3)^r)^{1/r} \to \max.$$

There are four samples of vector (Y_1, Y_2, Y_3), as well as respective values of the goal function, in Table 2.

Table 2.
Examples of Vector (Y_1, Y_2, Y_3) and the Corresponding Values of the CF

No	Y_1	Y_2	Y_3	$g(N(Y_1, Y_2, Y_3))$
1	100	800	7	1.000
3	90	900	8.5	0.866
4	110	700	8	0.890
5	120	740	7	0.9488

Conclusion

Constructing goal functions on the basis of the WPM $M_r(w, y)$ and the proposed principle of multiobjective optimality, called "imperfect maximin-minimax", enables to adequately represent the decision maker's preference over a wide range of performance parameters options.

An experts' estimation technique for evaluating the weights and convexity parameter of the WPM, based on the evaluation of worst compensable levels of performance parameters, has been proposed.

An important feature of the form $M_r(w, y)$ is its homogeneity, i.e. $M_r(w, ky) = k M_r(w, y)$ where k is a constant. It is also useful in practice that the CF is equal to unity if each performance parameter has its target value: $g(y_T) = M_r(w, y_T) = 1$.

It can easily be proved that, for $r > 1$, the inequality $g(y) = M_r(w, y) \le 1$ is true if and only if $\forall i \in 1:n \;\; y_i \le 1$. It means that if one agrees to slacken performance parameter constraints $y \le y_T$ and substitute y_{comp} for y_T in (1), the necessary and sufficient condition for $y \le y_{\mathrm{comp}}$ to be true is $g(y) \le 1$. Consequently, the inequality $y \le y_{\mathrm{comp}}$ will not be necessary and the optimization problem (1) can be rewritten as follows

$$\left.\begin{array}{l} g(y) \to \min, \\ y = N(Y),\ Y = F(X),\ X \in \Psi. \end{array}\right\}$$

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Modular Neural Network Learning Using Fuzzy Temporal Database

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Abstract

Despite the availability of several well-known neural network learning algorithms, we have taken the initiative to propose a new mechanism for initial learning and training of a neural network. Our methodology uses fuzzy temporal database as a storehouse of information that would be used to feed the network for learning and perfecting itself.

Keywords

Temporal Database, Modular Neural Network (MNN), Fuzzy Temporal Database (FTD), Fuzzy Temporal Database Learning (FTDL), initial learning.

1. INTRODUCTION

The need for storing imprecise information necessitates the usage of fuzzy database systems. Being able to trace changes in fuzzy sets over periods of time and the ability to access the state of a fuzzy set at a certain point of time requires the concept of temporal database systems. This extension of temporal databases to fuzzy temporal databases requires a database that can store variables belonging to fuzzy sets and their membership values along with a record of all changes in set variables or changes in membership values.

Shadab et al [10] has given Fuzzy temporal database design.In our paper we have used the same design. A simple and general structure for storing temporal fuzzy sets is been used. An example design of the temporal database consists of two tables, which can be shown diagrammatically as:

Member_ID	Set_Name	Variable

(a)

Member_ID	Membership_Degree	Beginning_Time	End_Time

(b)

Figure 1: Fuzzy Temporal Database Design, Example Tables (a) Member_Variable (b) Member_History

For the sake of fulfilling maximum number of constraints of designing a database, every table in a fuzzy temporal database would basically be divided into two sub – tables or normalized, in order to remove redundancy of data. Similarly, the tables shown in figure 1 has also been divided into two such tables: *Member_Variable* and *Member_History*.

Member_Variable table basically identifies the constituent variables of each set, with the provision that each set has multiple elements and each element may belong to multiple sets. Incase the variables are of complex type; they should preferably be stored in a separate table and referenced here through a foreign key. The combination of *Set_Name* and *Variable* should be unique. Each such tuple should have a distinct *Member_ID* - making it a primary key - so that it can be used as a foreign key in another relation. This *Member_ID* identifies the pair (A, x) where A is a set and x, a member variable of A.

Member_History table specifies the dynamic behavior of member variables of a set with respect to time. It must be noted that the

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history specified here is not specific to either a particular variable or set but to a combination of these. The combination of *Member_ID* and *Beginning_Time* can be considered as the primary key; an alternative approach is to use the combination of *Member_ID* and *End_Time* as the primary key. The column *Member_ID* references the table *Member_Value*.

Membership degree in a fuzzy temporal database is stored as a real number restricted according to the specified range. Storing it as a number enables us to have a much larger range of values and the use of considerably less space vis a vis linguistic variables. Also the generation of linguistic variables is context dependent and can be conveniently performed as demanded by the situation. Also the conversion from a number to a linguistic variable is generally irreversible as the precise value 0.48 cannot be generated from "medium".

The paper is divided into 7 sections. In section 2 we have given an overview of the FTDL (Fuzzy Temporal Database Learning) algorithm. In section 3 , we have the FTDL Architecture. Section 4 deals with the Conceptual FTDL Model. Section 5 and 6 deals with Features and Hitches of FTDL and application of the algorithm respectively. And finally we conclude in section 7.

2. OVERVIEW OF THE FTDL ALGORITHM

Learning in the context of neural network is defined as the process by which the free parameters of neural networks are adapted through a process of stimulation by the environment in which the network is embedded. The type of learning is determined by the manner in which the parameter changes take place.

The basic objectives that a learning algorithm should try to accomplish are:

- Generalization
- Speed of algorithm, and
- Probability of successful convergence.
- Weight should be adjusted on presentation of each sample.
- Learning should be able to capture complex non-linear mapping between input and output pattern pairs, as well as between adjacent pairs in a temporal sequence of patterns.

3. THE FTDL ARCHITECTURE

In this section we introduce the new introductory model that is used for the proposed FTDL algorithm. Before that, the described example of a fuzzy temporal database in the figure 1 is integrated into one for ease of use and its diagrammatic representation. Various explored factors for the design is also discussed in this section. Integration of FTD and modular neural network architecture to be used is also discussed.

Event	Membership_Degree	Beginning_Time	End_Time

Figure 2: Compact FTD Design for FTD Learning

The above figure shows the compact design of a fuzzy temporal database. This has been designed to ease the explanation of the use of FTD in the modular network architecture. In fact, the *Event* field of the above-described architecture is an embedded database table. When this table is disintegrated, it forms two tables as in figure 1, the primary key of the first table is an attribute of the second table and acts as a foreign key. The combination of this foreign key along with the *Beginning_Time* attribute is the primary key of the second disintegrated table.

The value in the *Event* field specifies the changes within the environment where the database is used. The *Beginning_Time* and the *End_Time* attributes together define the time period for which the event was stable at a particular value i.e. it did not experienced any signs of fluctuations or any value changes in the embedded database.

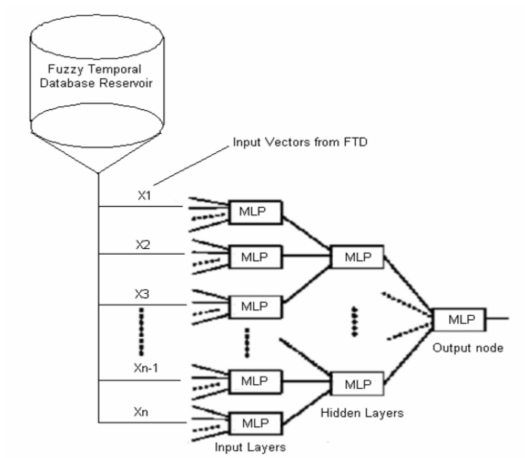

Figure 3: Preliminary Architecture for FTD Learning

The above diagram is the basic structure resulting from the integrated fuzzy temporal database (FTD) and the modular neural network (MNN). This diagrammatic representation is designed for the fuzzy temporal database learning (FTDL) algorithm.

As each module in the modular neural network is entitled to perform their respective task, thus the input to every module would be a different FTD, related to the task to be performed by the module of the network. This is one of the drawbacks to the proposed approach i.e. for every module a different FTD should exist or in other worlds the number of modules of a particular MNN is restricted by the available FTDs of the system in question.

In the figure 3, every MLP can be considered as a module of the entire system but at a larger scale the entire system depicted in the same figure can be considered as a module. This is totally based on the size and complexity of the system in question and the environment within which it is to be implemented.

The input vectors ($X_1 \ldots\ldots\ldots X_n$), in the above diagram is a collection of n FTDs, which acts as input to the modules of the modular neural network. This states that every n is a FTD and it is used as an input vector to the input module of the MNN and result of the processing of these input modules is fed to the subsequent modules in the hidden layers, which perform their intended task. The number of modules in the next layer is always less than the previous layer because a set of output from previous layer is fed to a single module in the next layer. This phenomenon of module integration carries on till the final output from the output module.

4. CONCEPTUAL FTDL MODEL

As stated earlier, this learning algorithm is still in its initial stages of development and the concept too is at a very immature stage. In this section we have described and discussed the theoretical aspect of the proposed model

Briefly, the FTDL algorithm falls under the broad category of supervised learning algorithms in neural networks. The mechanism is thought of under the influence of supervised form of learning and thus would imitate the basic characteristics of this mode of learning.

Very simplest form of fuzzy temporal database learning (FTDL) algorithm can be represented in the mathematically as:

$$\Delta w(n) = f(\eta) \left[\left[(X_{mn})\ \zeta\ (Y_{(m-1)(n-1)})\ \zeta\ (\mu_m(x)) \right] \right]$$

Figure 4: Mathematical formula for FTD Learning

FTDL learning, thus is a function of combination of current input parameters, output from the previous stage and fuzzy membership value of the current tuple from the input FTD with respect to the rate of learning of the modular neural network, within the given environment.

The variables of the FTDL formula prescribed above and their meaning with respect to the proposed algorithm are:

$\mathbf{\Delta w(n)}$ = change in weight of node n.

n = rate of learning of the FTDL algorithm.

$\mathbf{X_{mn}}$ = m^{th} input tuple to the node n form the FTD n corresponding to the n^{th} modular node.

$\mathbf{Y_{(m-1)(n-1)}}$ = Output of the input $X_{(m-1)(n-1)}$

$\mathbf{\mu_m(x)}$ = fuzzy membership value of the tuple m from the input FTD.

ζ = any operation between any two variables required by the environment or the system.

This is the very basic and generic mathematical structure of the fuzzy temporal database learning algorithm. The relation and the dependencies amongst the variables are still to be decided.

Initialization of a neural network is a very important aspect of a NN system. In FTDL approach of learning, initialization of the modular neural network can be easily done by using the FTD tuple of the FTD to be used as input to the module in question, with the minimum membership degree value (only a possible approach). Weight dynamics and weight space problem is also sorted and eased. We can assign initial weights to the system with the help of a function with the membership value as its required parameter. Once initialized, the MNN keeps on learning using the defined formula and the designed rules.

One of the possible relationship between the change in weight ($\Delta w(n)$) and the membership value ($\mu_m(x)$) in the given formula, is that they would be inversely proportional to each other. This is because the greater the membership value of the tuple, the lesser the difference would be between the actual output and the desired output from the modular neural network.

Another aspect explored with this design is that, the order in which we give in the input to the network should be in the ascending order with respect to the membership value of the input FTD. This is because, starting from the tuple having the lowest membership value and going to the maximum ones would result in step by step learning and finally reaching near or to the desired output. Indirectly this states that, the FTD should be sorted before it is fed to the module of the MNN. This sorting should be done such that the passed by events are arranged in ascending order and the event in progress (i.e. the event having End_Time value as "now") should not be considered as input tuple. This criteria also holds for the passed events, which did not had a membership value due to any unspecified reasons.

5. FEATURES AND HITCHES OF FTDL ALGORITHM

Generally a neural network is initialized using random number values. One of the most important features of this network is that no random number guessing is required for initializing and activating the modular network. We can initialize the network by looking at the membership value of the tuples. With this methodology, weight initialization becomes an easier task. For this particular aspect, all membership values are considered and a weight space is formulated, from the formulated weight space, using a function defined according to the environment, the initial weights can be taken.

When any MNN is fully trained with the FTDL algorithm, then with very slight modifications, we can use this very MNN to get a fuzzy temporal database by inputting only temporal databases. This is an important feature of this algorithm and can be viewed as an application of the FTDL algorithm.

One of the major drawbacks of the FTDL algorithm is that its applicability is limited by the availability of the fuzzy temporal databases. In other words, any modular neural network wishing to use FTDL should posses a bank of fuzzy temporal databases related to the system for which the network is intended to be. When any MNN to be designed, wants to use FTDL, it should first look for the FTDs and the number of modules in the entire MNN would also be restricted by the number of FTDs available for the system.

6. APPLICATIONS OF FTDL ALGORITHM

Although this work is still in its initial phases, but certain basic applications of this algorithms are thought of and are described in this section.

One of the most important areas where this algorithm would be of great use is in the predicting the output of an event. Looking at the features of this algorithm, when the network is fully trained then it can be useful in getting the membership values of a tuple whose $\mu_m(x)$ is not known for the FTD, which was used earlier as an input to a module of the MNN. Extending this feature, we can utilize the trained MNN for converting simple temporal databases to fuzzy temporal bases.

Any applications involving fuzzy temporal databases, when they are incorporated with neural networks would find this learning technique useful and easier as compared to other neural network learning algorithms. Several other applications of this mode of learning to the modular neural networks are possible and would exist but due to time limitations they cannot be explored and discussed in this work.

7. CONCLUSION

The proposed method for initial learning of the modular neural network is only theoretical. No experimental verifications have been made because the algorithm is still to be implemented and simulated. Future work in this domain will be focussed on its possible applications, areas of its applicability are also been considered as future endeavors. Searching for the domain of problems is a task to be accomplished.

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TSD: A Proposed Algorithm for Solving Tags Ambiguity

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Abstract- **Tags used to describe web resources bookmarked in social bookmarking services are considered ambiguous without the existence of a proper context. In this paper, we describe an algorithm that uses the semantic relationships in an ontology to resolve the ambiguity in people's tags.**

I. Introduction

Collaborative tagging, popularly know as folksonomies; can be thought of as index keywords that describe what a document is about.

Since people started using social bookmarking services, such as del.icio.us1 in late 2003, many resources have been bookmarked and tagged collaboratively. Using these services, people usually tag a web resource with words they feel best describes what the subject of a resource is about.

Tags are considered one of the crucial assets of social bookmarking services; however, these tags are subject to a well-known problem in the natural language processing field, which is: Word Sense Disambiguation (WSD), where words have multiple meaning in different context (i.e. polysemy).

To solve this problem in the domain of social bookmarking services, we have proposed a novel algorithm to disambiguate the senses of a tag in a given domain.

II. Significance of the Study

Tag sense disambiguation is an important problem in participatory services. It can be considered a variant of Word Sense Disambiguation and the general area of semantic control and knowledge organization discipline.

Tags used in contemporary web services such as del.icio.us are highly ambiguous. Tags ambiguity originated from the semantic problems related to tagging, such as: synonym control, singular and plural forms of words and hidden composite phrases in tags.

Trying to solve all these problems with a single system might be possible in the long run. However, the only problem this paper is trying to propose a solution to is, how to disambiguate a tag sense by exploiting its placement in the stream of tags assigned to a bookmarked web resource with the aid of domain specific ontologies.

III. Tags Sense Disambiguation Algorithm

Tags Sense Disambiguation (TSD) is a derivative idea that comes from Word Sense Disambiguation (WSD) technique, a well-known problem in the natural language processing field. The goal of a WSD algorithm is "… to associate the most appropriate meaning or senses to a word w in document d, by exploiting its window of context (or more simply context) C, that is a set of words that precede and follow w. The senses are selected from a predefined set of possibilities, usually known as sense inventory [such as WordNet]"[1]. In other words, the working idea behind WSD is to enumerate the set of all possible meanings of a word in a given context, and then determine based on some techniques, which sense is the most appropriate for a given context.

Mihalcea and Moldovan [2] have classified WSD techniques into five broad categories:

- Dictionary based WSD,
- Supervised WSD,
- Unsupervised WSD,
- Machine learning WSD, and
- Hybrid methods that combine several techniques with each other.

For TSD algorithm, the dictionary based WSD has been adopted with some modifications, i.e. make use of ontologies in place of dictionaries to resolve the ambiguous tags.

The proposed TSD algorithm is based on constructing a matrix of the semantic relationships between the concepts' instances for a given domain ontology, and then recording the co-occurrence of these instances in the tags list.

[1] http://del.icio.us

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IV. A Working Example

To test the practicality of our proposed algorithm, we need to construct a very simple ontology that consist of some concepts along with their instances and connect them with relationships. Our working example will be from the domain of Cascading Style Sheets (CSS).

Suppose the concept 'property' in the CSS ontology has an instance called 'list', see Figure 1. This instance if put in another context might mean 'a collection of things'. These multiple senses of the 'list' tag are misleading; therefore we need a mechanism to know what the tag actually means.

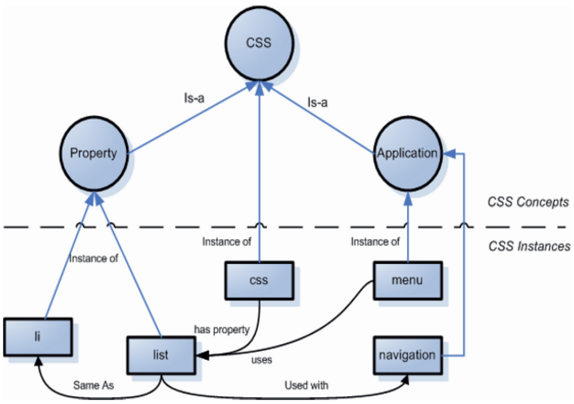

Fig. 1: A schema that depicts the semantic relationships between the 'list' instance and its neighboring instances in the CSS ontology.

To demonstrate the step by step functionality of the TSD algorithm the following real example taken from the del.icio.us bookmarking service is given:

"Listamatic: Rollover horizontal list"[2] web resource was bookmarked by four people in del.icio.us as follows, Figure 2.

To check whether the ambiguous tag, in this case 'list', appeared in the list of tags assigned to the web resources as an instance of the CSS ontology, we first need to construct the semantic matrix for the 'list' instance and populate it with the number of times an adjacent instance has occurred. Therefore, the algorithm starts by traversing the list of all tags in the posts (as one long list), and recording the number of times two instances have co-occurred in the tags list.

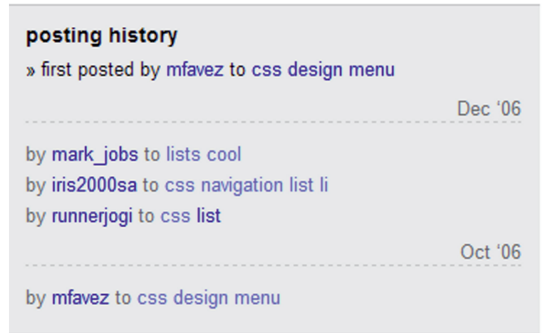

Fig. 2: Screenshot showing the appearance of the 'list' tag in "Listamatic: Rollover horizontal list" web resource.

Table 1 shows the populated semantic matrix built for the 'list' tag from the "Listamatic: Rollover horizontal list" example. The table axes (column and row headings) are constructed based on the semantic relationships between the 'list' instance and its neighbors in the ontology; i.e. the 'list' instance is semantically associated with 'CSS' as 'a property' relationship, with 'menu' as 'a component of' relationship, with 'navigation' as 'used in' relationship and finally with 'li' as 'same as' relationship. The populated numbers in the table represent how many times the tag 'list' co-occurred with a column instance in the given example.

TABLE 1

The Semantic Matrix for the 'list' instance, the row headings represents the ambiguous tags while the columns headings represent the neighbor instances in the ontology.

Concepts	CSS	Navigation	Menu	li
list	3	1	0	1

After finishing recording all the tags co-occurrences, thus building the semantic matrix, a tag ambiguity index is calculated based on the following formula:

$$Tag\_Index(Tag_d) = \frac{\sum_{i=1} co\_occur(Tag_d, Tag_i)}{All\_Tag_d}$$

Where Tag_d refers to the ambiguous tag, Tag_i refers to the co-occurred instance in the semantic matrix and All_Tagd corresponds to the total number of occurrence for Tag_d in the tags list.

So, based on the previous example, we can compute the Tag_index as follows: Tag_Index('list') = 5/4 = 1.25. The value 1.25 is greater than our pre-determined threshold of 0.5; this threshold represents a hypothetical value that we have setup to qualify a tag as being from the ontology instances. The value of the threshold was based on a

[2] http://css.maxdesign.com.au/listamatic/horizontal02.htm[accessed 10/1/07]

candidate value observed after repetitive experimental trails. Therefore, if the Tag-Index for a given tag was not greater than 0.5, the ambiguous tag will not be considered as one of the ontology instances.

Another example is given next to show how the algorithm behaves when the tag 'list' is used for a different purpose. The "CSS - Contents and compatibility"3 web resource was bookmarked by 477 people in del.icio.us. Figure 3 shows all the appearance of the 'list' tag compiled in one shot.


by electricspace to css reference browser browsers list support compatible selector style

by auriumsoft to css reference browser compatibility list

by ad0ac to css

by alexcrown to css compability list

by viniciuscamara to web browser compatibility

by jrichard to browsers business chart code coding compatibility computer css design development documentation dom firefox hack hacks help howto html ie internet javascript list programming reference resources standards style tips tools tutorial tutorials web webdesign work xhtml browser

by franquin to css

by yhassy to css reference browsers

by sambarino to css browser compatibility guide list selectors

Fig. 3: Screenshot showing the appearance of the 'list' tag in "CSS - Contents and compatibility" web resource.

The tag 'list' has appeared in that bookmarked web resource five times; however, the co-occurrence of the tag 'list' with other semantic instances was zero. This yields a Tag-Index value of zero, which indicates that the people who have bookmarked this resource were using the tag 'list' to mean 'a collection of things' and not the property 'list'.

Finally, it is worth mentioning that the idea of this algorithm has originated from our observations of the repetitive patterns spotted while analyzing the list of tags in the del.icio.us bookmarking service. This pattern was then discovered to be useful for eliminating ambiguous tags. The algorithm also exploits the spontaneous act of tagging practiced by del.icio.us users, this act can be recognized in terms of tacit semantic relationships between tags.

V. Related Work

We have encountered very little academic research handling the issue of tags ambiguity/semantics. This might be attributed to the recent appearance of these kinds of Web 2.0 applications.

However, Zhang et al. [3] was among the first researchers tackling this problem. They used a probabilistic generative model called the asymmetric SMM model for data co-occurrence to infer the semantics of the folksonomies and to resolve the ambiguity of folksonomy tags. Their model has succeeded in resolving tags ambiguity and in grouping synonym tags together. However, to be able to use their proposed solution, a very large-scale data set needs to be used in order to train the model and infer tags' senses accordingly. This approach is somewhat problematic given a small focused domain such as ours, thus, we thought that using ontological relationships co-occurrences to resolve tags ambiguity per web resource for a focused domain is less time-consuming and less resource-hungry as opposed to the SMM model.

Another similar investigation was carried out by Kipp and Campbell [4]. Were they have used Multi-Dimensional Scaling (MDS) for co-word clustering to visualize the relationships between tags. While their approach helped picture the relationships between tags for a given URL, yet, their approach is still not practical for resolving tags ambiguity because it does not assign weights to tags based on their ambiguity level in a given context.

VI. Limitations of the Proposed Algorithm and Future Work

This work focused on investigating the potential of using ontological associations to resolve the ambiguity of tags. The derived approach we have pursued should be subject to further evaluation to improve the validity of the proposed algorithm.

Possible improvements include:

- Investigate the difference and similarities between using WSD and TSD, i.e. dictionaries vs. ontologies, and check whether TSD is more advantageous than WSD.
- Incorporate the calculations of semantic distances [5] between ontology instances to help in improving the ambiguity resolving.
- The accuracy of other typical dictionary based WSD methods was between 60-70% [6]; this value can be used to benchmark TSD in future algorithm development and evaluations.

Finally, the TSD algorithm can be potentially useful in application where the ambiguity of tags needs to be resolved, for instance the use of tags in the process of semantic annotation e.g. [7] or in folksonomy based recommender systems [8].

VII. Conclusion

In this paper we have demonstrated the implementation of Tags Sense Disambiguation algorithm and assessed its

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[3] http://www.quirksmode.org/css/contents.html[accessed 10/1/07]

functionality using two simple examples. We claim that the TSD algorithm will solve the problem of multiple meanings of tags used when bookmarking web resources in the del.icio.us social bookmarking service. The TSD utilizes the power of ontological associations to resolve the semantics of tags.

Notice that we have not yet run extensive tests on the accuracy of the algorithm using large scale scenarios; however, our preliminary test shows that the TSD algorithm is so promising.

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Proposal and Field Experiment of Road Facility Management Support System by RFID and GIS

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***Abstract* - Recently, the administrative and maintenance expenses of superannuated public infrastructure and facilities became unfeasible in Japan. Then, Necessity arose for premeditated and efficient maintenance of public infrastructures like road facilities. However, because controlling the maintenance of road facilities such as bridges and road lights depends on human resources and businesses, therefore the introduction of excessively advanced information systems was avoided. This research proposes a road facility management support system architecture that integrates Radio Frequency IDentification (RFID) and Geographic Information Systems (GIS) and evaluates the effectiveness of such a support system. The design goal of the proposed support system is to enable the efficient inspection of road facilities; it comprises (1) road facilities enabled with RFID tags, (2) mobile phones with built-in RFID readers, (3) ITAG (Integration server of e-Tag Application and GIS) to integrate the RFID functionality with GIS servers, and (4) a database to store road facility inspection data and site photographs. A field experiment prototype system was developed based on the proposed system architecture. The field experimentation of this prototype system was conducted with the Morioka City Office Road Administration Division as the experimental subject. A survey questionnaire administered after the completion of the field experiment showed that the system was effective in enabling the unified administration of inspection history. In conclusion, this paper proposes a new type of system architecture that integrates RFID and GIS for GIS-based applications that will be required for the advanced spatial information societies of the future.**

I. Introduction

In Japan, Government construction agencies are focusing their asset management resources on the planned management of public infrastructure. In performing asset management for public infrastructure, gathering and storing inspection data is essential to properly maintain infrastructure such as road facilities and bridges. If information systems can be introduced into road facilities management and the inspection and the repair histories of facilities be recorded in a database, it would be possible to realize the unified management of the state and deterioration data of these facilities. Further, this would also allow us to record the basic data describing these facilities efficiently.

If a system supporting the road facilities management business is developed, it would be crucial to acquire location information because the target facilities are widely scattered in any given region. Some systems that support road pavement and road facilities management by using location information have already been proposed. One such typical system is the groupware GLI-BBS that combines GPS mobile phones with an ASP-type map service [1]. An implementation of this approach on a site where this system has been deployed revealed that the efficiency of intelligence sharing among the administration—the road administrator and the company to which the administration of the road was assigned—had been improved [2].

The study done by Abe, A., et al. [2], clarified the limit of the GLI-BBS system that used the GPS mobile phone. The error margin is attributed to the acquisition of the location information by using GPS mobile phones; this is because the measurement result changes for each measurement. If a rough position is acquired, the repair of the road pavement can be undertaken with a fairly reasonable error margin. However, road facilities management requires highly accurate location information. This is because it is necessary to manage the several road facilities in an urban

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area individually. The positional accuracy necessary for individual management cannot be provided by a GPS mobile phone.

In order to solve this problem, it is necessary to provide each road facility with a direct RFID. This approach can provide the high-accuracy location information that is required by road facilities management business. A system supporting urban road facilities management can be developed by displaying this information on a basic GIS map.

In this study, therefore, we focus on making mobile phones with built-in RFID readers and GIS cooperate. We propose a system that can identify individual operation and maintenance facilities by matching their RFID numbers to the road facilities attribute data based on a basic GIS map. This system is based on the basic road facilities management support system experiment that the author's group conducted earlier [3].

Ⅱ. Issues of Road Facility Management

A. Present situation

We define the road facilities comprising road structures such as road bridges, underground pavements, and tunnels and road attachments such as road lights, curve mirrors, and induction lights. Road facilities management business is defined as the inspection of road structures and the issuing of repair instructions for road attachments.

Road facilities management business is executed by road administrators such as local governments. To execute an appropriate maintenance management program, the road administrator performs inspection. The road structure is examined on a daily basis by visual inspection. The road administrator confirms the damaged site, and the road attachment is repaired by the consignment company that

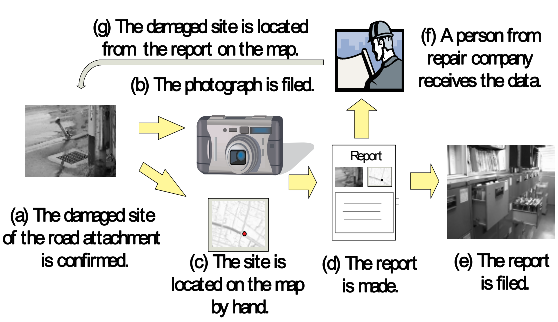

Fig. 1. The conventional process for issuing repair instructions.

receives the instruction. Figure 1 shows the procedure for issuing repair instructions in the road attachment.

B. Issues

We identified two issues by analyzing the interview of the road administrator. The first can be stated as "The retrieval of necessary information for the repair instruction requires a considerable amount of time because the management ledger of road facilities is paper-based. (issue 1)." The second states that "The road administrator cannot perform the unified management of the inspection history because keep the inspection data are stored at multiple locations. Therefore, planning an efficient long-term management is difficult. (issue 2)."

Ⅲ. Proposal of Road Facilities Management Support System

We propose an information system as a support for the road facility management that aims to solve the two issues described in section Ⅱ.*B*. The approach to issue 1 is to prepare the inspection history by combining RFID and GIS and use it for formulating a long-term management plan.

The approach to "issue 2" is from a site to the real-time registration of the inspection data and the damage situation in the database by RFID and GIS referring to the road facilities ledger.

A. Concept of the Proposed System

Figure 2 shows the system concept chart of the proposed road facilities management support system. An RFID is assigned to each target road facility. Mobile phones with built-in RFID readers read the RFID numbers. In this system, RFID and GIS establish this identification number as a key and cooperate. Thus, obtaining the necessary attribute information on-site by employing this mechanism and referring to a road ledger becomes possible.

The inspection data and site photographs are transferred from the site to the GIS database via the Internet. The road administrator at the office confirms the site information registered in the database by using a administrator PC and shares this information. The inspection history and repair instructions information are recorded in the database; this makes the unified management of information possible.

B. Design Policy of System Configuration Elements

B.1 Road Facilities Enabled with RFID tags

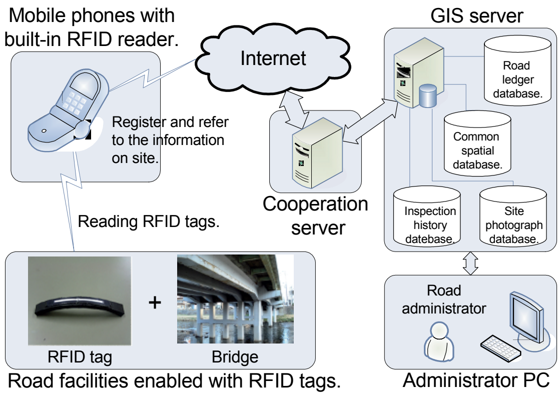

Fig. 2. System concept.

Each target facility is individually identified by an RFID. When a two-dimensional code is used instead of an RFID, it is likely to be deteriorated by wind, rain, or other external factors. Moreover, there exists a possibility that it could be counterfeited by someone with malicious intent.

B.2 Mobile Phones with Built-in RFID readers

Mobile phones with built-in RFID readers are used for reading the RFID tags. There are two reasons for this: First, road facilities personnel always carry mobile phones during road patrolling activities for reporting purposes. Therefore, it is not desirable to add further to their equipment load by making them carry devices such as PDAs. Second, because there exists a possibility that the proposed system may undergo expansion in the future, a prototype of mobile phones with built-in RFID readers has already been developed [4].

B.3 Cooperation Server

A cooperation server integrates RFID and GIS; further, it provides server functions to manage the attestation to generate RFID numbers. The usage of mobile phones with built-in RFID readers makes it possible to retrieve the attribute data of road facilities and match it to the corresponding RFID numbers by accessing the database stored in a GIS server.

B.4 GIS Server

The GIS system used by the road administrator in this study is assumed to be an integrated GIS system, which is advocated by the Ministry of Internal Affairs and Communications (MIC) in Japan. A basic map of the integrated GIS system comprises common spatial data constructed based on the guidelines provided by the MIC. The usage of such an integrated GIS makes it possible to display maps with a largely reduced scale, and the high-accuracy map data required by the road facilities management business can be obtained.

B.5 Administrator PC

The road administrator accesses the integrated GIS via the computer network in the administration office using a administrator PC. They can thus inspect the inspection data, etc. Efficient, exclusive-use PC need not be newly manufactured for the introduction of this system.

IV. FIELD EXPERIMENT

A prototype system was constructed based on the proposed system concept, and the field experiment was conducted using actual road facilities management. The experimental site comprised a part of the road facilities managed by the Morioka City Office Road Administration Division. The duration of the experiment was four months—December 2006 to March 2007.

The construction of the integrated GIS system was started in 2005, and it was operational in October 2006 in Morioka City. It would be difficult to mount the entire information system at the proposed level in such a real operational environment. Therefore, the proposed system and the integrated GIS in Morioka City that had been developed by the authors' groups thus far were combined; thus, an experimental system for proof of use was constructed.

A. Composition of the Experimental Prototype System

Figure 3 shows the system architecture for field experiment. The RFID reader used in this experiment is a prototype mobile phone equipped with a passive-type RFID reader (hereafter referred to as "RFID mobile phone") [4]. The attribute data for the experimental road facilities was stored in a PostgreSQL database (experimental road ledger database). The inspection data and site photographs were recorded in an electronic bulletin board and were accumulated as history.

The integrated GIS uses a system that is actually operational at the business sites in Morioka City. The connection from the RFID mobile phone to various

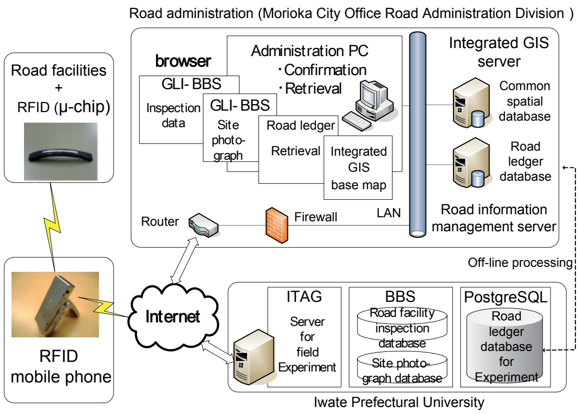

Fig. 3. Composition of the Experimental Prototype System.

databases is realized through the ITAG to integrate the RFID with the GIS server. The person in charge of Morioka City Road Administration Division can confirm the inspection data by using the administrator PC in the office. This experimental system demonstrates the cooperation between the RFID and integrated GIS by using a PC-based browser for management.

B. Field Experiment on Business Site

B.1 Road Facilities enabled with RFID tags

Table I lists the names of the road facilities and the corresponding number of installed RFID tags. There are six types of target road facilities. They were further classified into two groups of three each—road structures and road attachments. A total of 30 locations were considered as sample sites from among these facilities.

The type of RFID tags used was the ultra-micro RFID (μ-chip), which is supported by the passive-type RFID mobile phone. The method of sticking RFID tags was

TABLE I ROAD FACILITIES WITH INSTALLED RFID TAGS.

Management object	Facility name	Number of installed RFID tags
Road structures	Road bridge	4
	Underground pavement	4
	Snow-melted device	4
Road attachments	Road light	7
	Curve mirror	6
	Induction light	5
Total		30

a method similar to the basic experiment done by the author's group [3]. Moreover, it is important to decide the position in which RFID tag is installed beforehand. For instance, RFID tag installed in the road bridge is set up in the plaque (or nameplate) of the bridge. The road administrator is informing the inspection person of such regulations beforehand.

B.2 Maintenance of attribute data

In this experiment, the attribute data of the road facilities in the 30 locations listed in Table 1 was confirmed and transferred to an experimental road ledger database. Then, the identification numbers of the RFID tags and the attribute data were matched.

B.3 Utilization procedure for the RFID mobile phone

Figure 4 shows the screen image of the RFID mobile phone. With regard to the Japanese word input method by the characters, it takes a long time to input the inspection result through the mobile phone. Then, a user interface was devised for the inspection form so that this prototype system can input the main inspection results by a sign method. The retrieval and registration procedures of the inspection data by the inspectors are described as follows.

1) *Step 1:* Starting RFID readers: The power supply of the RFID reader is turned on, and the RFID application program of the RFID mobile phone is run.
2) *Step 2:* Reading the RFID number: The reader of the RFID mobile phone is brought close to the RFID provided to the road facilities, and the RFID numbers are recorded.
3) *Step 3:* Collation of RFID numbers: The RFID mobile phone is accessed by ITAG through of the Internet and the read RFID numbers are collated with the RFID management data.

(a) Top menu of RFID mobile phone. (b) Input form of inspection result for bridge.

Fig. 4. Screen image of the RFID mobile phone.

4) *Step 4:* Indication of the menu: The top-page menu appears, and appending the database becomes possible. Simultaneously, access to various databases becomes possible, and it is possible to register information (Fig.4 (a)).

5) *Step 5:* Search and perusal of the ledger: The inspector performs a ledger search when required. In addition, perusal of the inspection history is also possible.

6) *Step 6 :* Registration of the inspection results: The inspector inspects the road structure. Subsequently, he inputs the results into the inspection form and registers it in the database (Fig.4 (b)).

B.4 Experimental conditions on field

The experimental field spanned an east-west distance of about 6 km and a north-south distance of about 5 km; further, it coincided with the center of Morioka City. The 30 road facilities with installed RFID tags lay scattered within this area. When the person in charge of the road administration division usually patrolled this region, he carried the RFID mobile phone, and this field experiment was performed by registering the inspection data.

The experimental conditions for the duration of four months—December 2006 to March 2007—are described in this section. First, the experimental conditions for the observation of the road structures are described. The experiment on the road bridge was conducted by patrolling it at the rate of one degree per month, and the inspection frequency was four times per degree. The inspection on the underground pavement was executed by patrolling it at a rate of about one degree per week, and the inspection frequency was 13 times per degree. The inspection frequency of the snow-melted device was four times per degree. As a result, the total frequency of the inspections executed by using the actual experimental system was 82 times per degree.

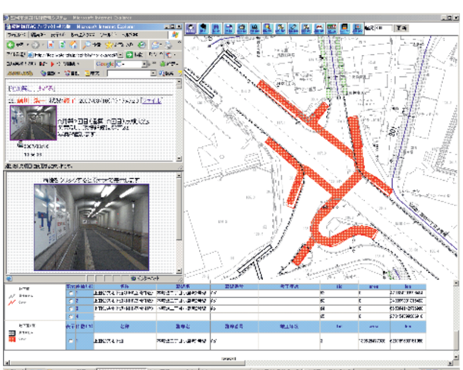

Fig. 5. PC screenshot.

B.5 Experimental conditions in office

The person in charge of management, who registers the inspection data on field, confirms the registered inspection data by using a administrator PC in the office. The road ledger chart layer and the road facilities layer are displayed superimposed on each other to confirm the inspection data against the site photographs registered in the electronic bulletin board; this makes it possible to promptly retrieve the required attribute data. Figure 5 shows an example of the screen.

V. RESULT AND DISCUSSION

A. Time evaluation

The processing times required by the earlier methods in the repair instructions and the proposed system were compared. The purpose of this time measurement is to evaluate the time effect of the proposed system. Working hours to confirm on site damage and report repair instructions to the repair company for the induction light damaged while experimenting was measured. As the result, the time required by the proposed system was one-third that required by the conventional process (the required time was reduced from 32 minutes to 9 minutes).

B. Effectiveness evaluation

To evaluate the effectiveness of the unified management of the inspection history by the proposed system, a questionnaire was prepared for the staff of the Morioka City Road Administration Division. The execution days considered were February 1 and March 20, 2007. The questionnaire was administered to a total of 29 people—19 currently working for the Road Administration Division and 10 who worked for the Road Administration Division in the past. Five people who participated in the experiment as in-charges were also included in this group.

First, the evaluation method explained the function of the proposed system and the conditions of the field experiment through a diagram. Next, each participant was allowed to experience the retrieval of the facilities attribute data and understand the function of the field experiment system

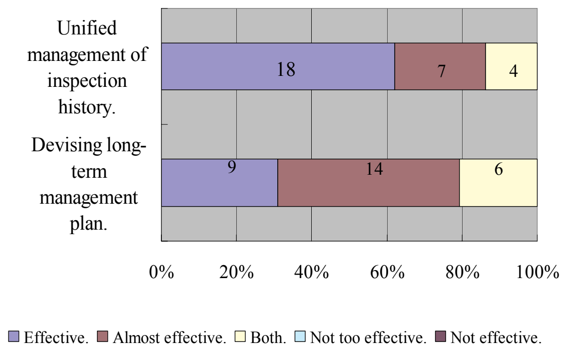

Fig.6. Result of effectiveness evaluation.

sufficiently. Then, they were interviewed and asked to fill an evaluation form on the questionnaire at the end. Concrete evaluation of the following three items: "Unified management of inspection history," and "Devising a long-term management plan," were conducted in five stages.

Figure 6 shows the evaluation results. Almost 86% of the participants voted for the "Unified management of inspection history" option with choices ranging from "effective" to "Almost effective," thereby making it an almost excellent result. However, only 79% of the participants selected the "Devising of a long-term management plan" option.

The results suggest that the constant unified management of inspection history can be realized by introducing the proposed system. However, the results also indicated that the proposed system would have no immediate effect on the devising of long-term maintenance management plans. Thus, there exists a need for an asset management application covering all road facilities management activities; such a system needs to be introduced in conjunction with the proposed system.

VI. Conclusion

We propose a futuristic asset management system for the road management facilities of local governments; the proposed road facilities management support system integrates RFID and GIS. In order to verify the effectiveness of this system, we constructed an experimental system based on the proposed system architecture and operated it in Morioka City, Japan.

The results of the experiment revealed that that the proposed system realizes the constant unified management of inspection history by assigning each road facility an RFID tag and matching its identification number to the facilities attribute data. However, it is necessary to introduce an asset management tool that is effective with regard an aspect in which the proposed system was found to be lacking—devising long-term management plans. Such a system needs to be introduced in conjunction with the present system to realize the efficient maintenance management of road facilities.

The development of an effective road facility tool to realize the unified management of inspection history is a topic for future research. In this study, we examine the effectiveness of the proposed system with regard to the inspection management of road structures such as tunnels, road drain pump facilities through a road bridge, underground pavements, etc.

Acknowledgement

This study was supported by the "Environmental research" project of the Iwate Prefectural University. We gratefully acknowledge the significant support lent by the staff at KDDI Ltd. in providing us with the RFID mobile phone prototype and the required technical support thereof. Furthermore, we wish to thank the staff of the Morioka City Road Administration Division for their help during the investigation and system evaluation.

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Fusion of Remote Sensing Images Using Contourlet Transform

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***Abstract*- Image fusion is a process of producing a single image from a set of input images. Recently, the wavelet transform (WT) has been widely used in image fusion. However, the Contourlet transform give better results because it represents edges better than the wavelets transform. In this paper, fusion algorithms based on the contourlet transform are proposed. These algorithms are tested and compare to an existing similar algorithm using Synthesized and QuickBird images. The experimental results show the superiority of the proposed algorithms over the existing contourlet-based one.**

I. Introduction

Image fusion is a technique of combining complementary information from multiple images, originated from different sources, into a single image. This fusion process improves the quality of the fused image compared to the original input images. Image fusion has emerged as a new and promising research area with the availability of multisensor data in many disparate fields, such as remote sensing, machine vision, robotics, medical images, and military applications, [1,2,3].

In remote sensing imaging systems, such as earth resource satellites; SPOT, IKONOS, and QuickBird, both panchromatic (PAN) images at a higher spatial resolution and multispectral (MS) images at lower spatial resolution are obtained.

The technical limitations of having sensors that acquire multispectral images with high spatial resolution and the need of such images in many remote sensing applications increase the demands of developing an effective image fusion techniques. These limitations and requirements motivated us to study and compare the latest techniques used in this field.

In recent years, wavelet transform (WT) has been widely used in image fusion field. WT is characterized by its ability of capturing the geometry of image edges; however, there are some limitations of the commonly used separable extensions of WT. Wavelets in 2-D are good at isolating the discontinuities at edge points, but will not “see” the smoothness along the contours. In addition, separable wavelets can capture only limited directional information-an important and unique feature of multi-dimensional signals. Though wavelet transform is optimal when it is used in analyzing zero-dimensional or point singularities, it is not optimal when used in analyzing linear/curve singularities in image processing.

Minh N. Do and Martin Vetterli [4,5] recently pioneered a new system of representations named contourlet which is a “true” two dimensional transform that can capture the intrinsic geometrical structures that are key features in visual information. The idea is to construct a discrete domain multiresolution and multi-direction expansion using non-separable filter banks, in much the same way that wavelets were derived from filter banks. This construction results in a flexible multi-resolution, local, and directional image expansion using contour segments, and thus it is named the contourlet transform (also named Pyramidal Directional Filter Bank, PDFB).

The contourlet transform is more appropriate for the analysis signals which have line or hyper-plane singularity than the wavelet transform. Furthermore, the contourlet transform has better approximation precision [4,5,6]. The ability of the contourlet transform for noise reduction is also better than the wavelet transform. In addition, contourlets represent edges better than wavelets. Therefore, the contourlet transform is adopted in this work for fusing images.

The rest of the paper is organized as follows; the second section explains the principle of contourlet transform. In section III the contourlet –based fusion algorithms are detailed. The experiments and the result analysis are presented in section IV. The conclusion follows.

II. Principle of the Contourlet Transform

The wavelet transform is only good at isolating the discontinuities at object edges, but can not detect the smoothness along the edges. Moreover, it can capture limited directional information. The contourlet transform can effectively overcome the disadvantages of wavelet; contourlet transform is a multi-scale and multi-direction framework of discrete image. In this transform, the multiscale analysis and the multi-direction analysis are separated in a serial way. The Laplacian pyramid (LP) [7,8] is first used to capture the point discontinuities, then followed by a directional filter bank(DFB) to link point discontinuities into linear structures. The overall result is an image expansion using basic elements like contour segments. The framework of contourlet transform is shown in Fig. 1.

Contourlet expansion of images consists of basis images oriented at various directions in multiple scales with flexible aspect ratio. In addition to retaining the multiscale and time-frequency localization properties of wavelets, the contourlet transform offer high degree of directionality. Contourlet transform adopts nonseparable basis functions, which makes it capable of capturing the geometrical smoothness of the contour along any possible direction. Compared with traditional image

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expansions, contourlet can capture 2-D geometrical structure in natural images much more efficiently.

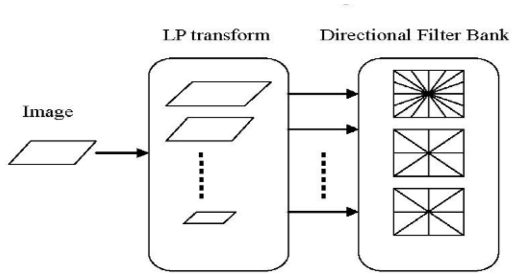

Fig. 1: Framework of the contourlet transform.

III. THE IMAGE FUSION ALGORITHMS

A. *Analysis of the Contourlet Transform Coefficients*

The distribution of the coefficients of contourlet transform is not well-regulated as good as that of wavelet transform. It is related with the parameter ***nlevels*** given in the DFB stage decomposition. ***nlevels*** is a one dimensional vector. It is used to store the parameters of the decomposition level of each level of pyramid for DFB. If the parameter of the decomposition level is 0 for DFB, DFB will use the wavelet to process the subimage of pyramid. If the parameter is l_j. The decomposition levels of DFB is 2^{l_j} , which means that the subimage is divided into 2^{l_j} directions.

Corresponding to the vector parameter ***nlevels***, the coefficients, *Y*, of the contourlet decomposition is a vector, too. The length of *Y* is equal to length (***nlevels***) +1. ***Y*{1}** is the subimage of the low frequency. ***Y*{i}(*i* = 2,... *Len*)** is the direction subimage obtained by DFB decomposition, i denote the *i-th* level pyramid decomposition. The subband images of contourlet decomposition coefficients for Lena image are shown in Fig.2.

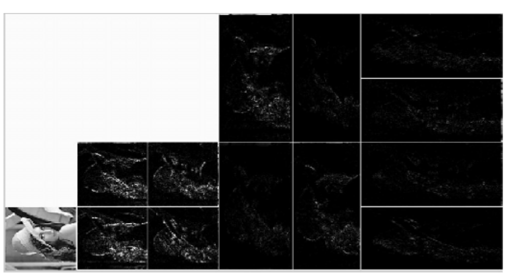

Fig. 2. The subband images of contourlet decomposition coefficients

B. *The Contourlet-based Image Fusion Scheme*

Fusion methods based on contourlet analysis usually combine decomposition coefficients of two or more source images in an appropriate way afterwards, the backward transform is performed on the combined coefficients resulting in the fused image. A general scheme for contourlet-based fusion methods is shown in Figure 3. Where, image1 and image2 denote the input images, CT represents the contourlet transform, and Image F is the final fused image.

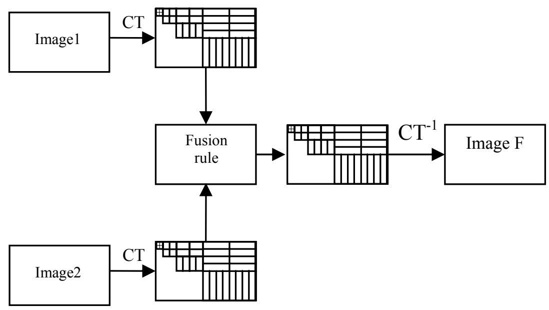

Fig.3: General framework for contourlet-based image fusion techniques.

M. Qiguang et al [9], proposed a contourlet-based image fusion algorithm. The authors demonstrated that the contourlet-based algorithm has better performance compared to a know fusion algorithms based on wavelets and pyramid decompositions. However, the contourlet-based algorithm in [9] calculates the region-energy for every coefficient of each subband which needs more calculations than other simple methods. In this paper three contourlet based fusion algorithms are proposed. In the first algorithm, we replace all the details information of the MS image with that of PAN image through the following operational procedure:

1) The two images taking part in the fusion are registered to each other.

2) Apply histogram matching between PAN image and first band of MS image and get new Pan Image.

3) Decompose the histogram-matched panchromatic image and the first band of MS image into contourlet planes.

4) Replace all the details coefficients of the first band of MS image with that of panchromatic image

5) Perform an inverse contourlet transform, and generate the first band of the fused image.

6) Steps (2-5) are repeated for the other MS image bands.

The flow chart of fig. 4 shows these steps.

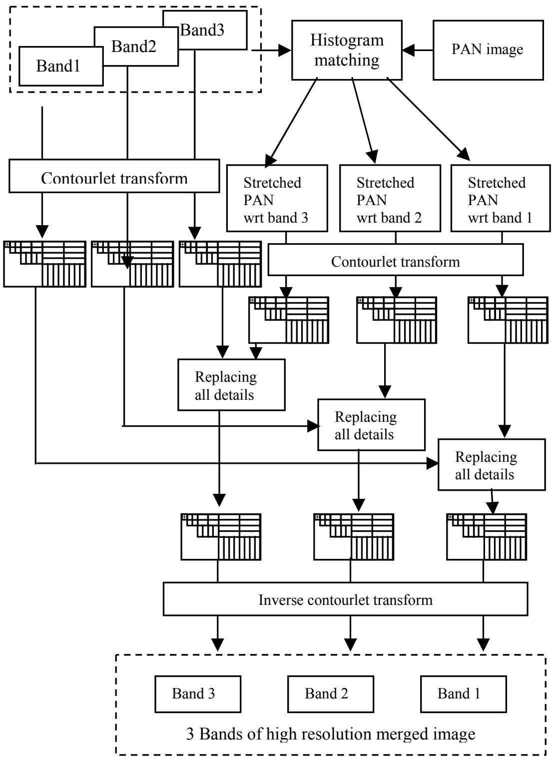

Fig. 4.Flow of the fusion based on contourlet by replace all details

The second and third algorithms differ only in the applied fusion rule. Fig.5 shows the schematic diagram of these two algorithms. In the second algorithm, the contourlet coefficients of the fused image are constructed by selecting the coefficient that has maximum energy. Whereas in the third algorithm the contourlet coefficients of the fused image are constructed by averaging out the coefficient that result from the contourlet transform of both the PAN and MS images which is demonstrated in the following operational procedure:

1) The two images taking part in the fusion are registered to each other.

2) Transform the multispectral image (MS) into the IHS (Intensity-Hue-Saturation) components (Forward IHS transform).

3) Apply histogram matching between panchromatic image and Intensity component (I), and get new panchromatic image (New Pan).

4) Decompose the histogram-matched panchromatic image and Intensity component (I) to contourlet planes.

5) Apply the fusion rule to construct the contourlet coefficients of the fused image.

6) Perform an inverse wavelet transform, and generate a new intensity.

7) Transform the new intensity, together with the hue, saturation components back to get the fused multispectral image. .

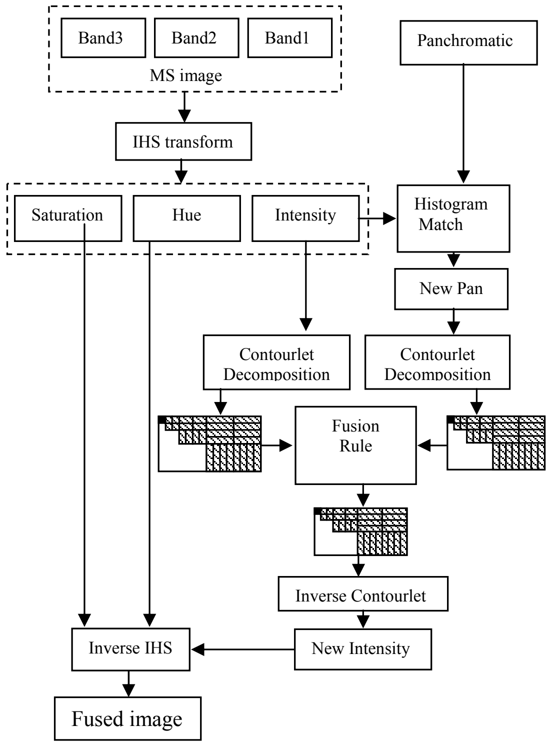

Fig. 5.Flow of the fusion based on second and third algorithms

IV. EXPERIMENTS AND FUSION RESULT ANALYSIS

In these experiments, the contourlet-based methods are computed with the parameter levels= [0, 2, 3, 4]. M. Qiguang et al [9] algorithm and our proposed algorithms are tested using synthesized and real remote sensing images.

For synthesized images, to get the synthesized PAN images we converted a true color (RGB) image into gray scale image. The MS image is created by applying a digital image filter like disk and Gaussian filters to the original color image. In case of using synthesized images as an input to the fusion process, performance of the fused image is evaluated by comparing it's results with the original color image.

The use of synthesized images has two advantages; the first is that we have an original reference image with which so we can easily compare our result, the second advantage is that the PAN and MS images are perfectly registered to each other and the problem which we may have due to miss registration of input images is completely avoided.

Figure 6 shows the fusion results of the synthesized images in which the MS image is produced by degrade the original true color image using the filter (disk).

The other tested remote sensing images consist of 0.61m panchromatic and 2.44m multi-spectral QuickBird images. Figure 7 shows the fusion results of the QuickBird images. The visual inspection shows that the fused images produced by proposed algorithms have more details than that of the algorithm given in [9].

Three different measures are used to evaluate the performance of the algorithms under investigation. These measures are: the normalized correlation, the entropy and the standard deviation. Detailed equations of these measures can be found in the literature.

Table 1 shows the results of the fusion experiment of the synthesized images where the correlation is measured between the PAN image and the corresponding gray fused image, and correlation between gray MS and gray fused image. The entropy and the standard deviation are measured for the fused images.

TABLE 1.
Performance measures for fused synthesized images

	Pan to gray fused Correlation	Entropy	Standard deviation
CT([9] algorithm)	0.948	7.5988	10.5266
CT(replace all details)	0.9638	7.5447	10.6976
CT(max energy)	**0.9866**	7.6492	**10.6401**
CT(average)	0.9835	**7.6604**	10.0121

Table 2 shows the results of the fusion experiment of QuickBird images where the correlation is measured between the PAN image and the corresponding gray fused image. The entropy and the standard deviation are measured for the fused images.

TABLE 2.
Performance measures for fused QuickBird images

	Pan to gray fused Correlation	Entropy	Standard deviation
CT([9] algorithm)	0.9356	7.6393	10.1062
CT(replace all details)	0.9482	**7.9028**	**10.6903**
CT(max energy)	**0.9917**	7.6105	10.2637
CT(average)	0.9916	7.6039	10.2744

The results indicated in the tables 1 and 2 and the visual inspection of the images of figure 6 and 7 reveal that, in general, the performances of the proposed algorithms outperform the algorithm in [9]. The normalized correlation results indicate that the max energy and the average fusion algorithms have better results than the other two algorithms.

V. Conclusions

In this paper, three different fusion algorithms are proposed. All algorithms are based on the contourlet transformation. In order to validate the performance of the proposed work, synthesized and real remote sensed images are used in the experiments. Comparisons with a recent contourlet-based algorithm show that the proposed techniques have better objective and subject performances.

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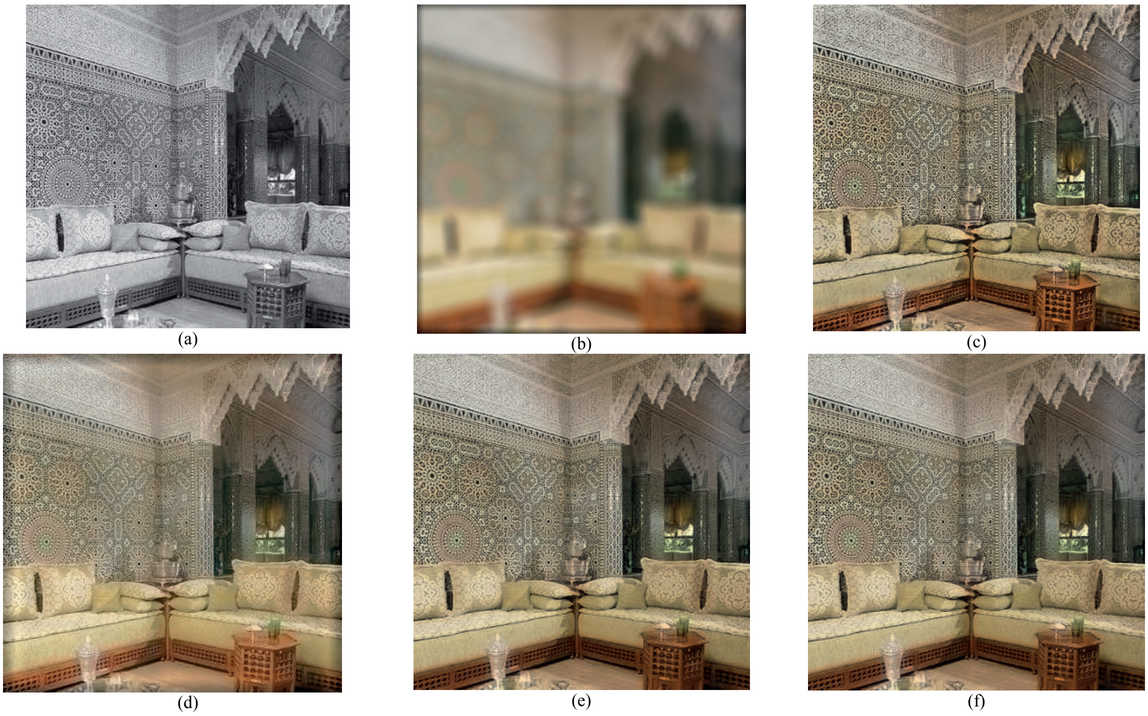

Figure 6: synthesized images (a) panchromatic (b) multispectral; (c) Qiguang et al [9] algorithm (d) replacing all detail.(e) max energy (f) averaging.

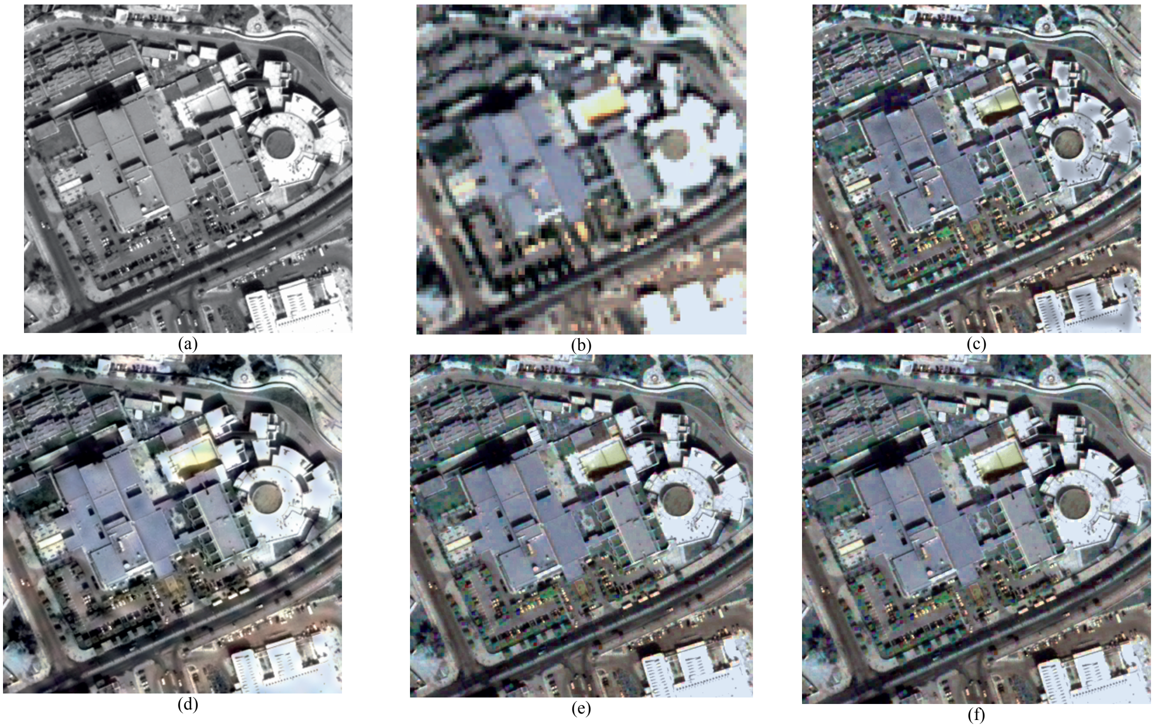

Figure 7: QuickBird images (a) panchromatic (b) multispectral; (c) Qiguang et al [9] algorithm (d) replacing all detail.(e) max energy (f) averaging.

Modeling and Simulation of Dynamic Processes with the Help of Program Package BLADIS+

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Abstract—**The problem of extension the fatigue life of high loading rotors of turbomachines may be solved by different methods. One of this methods, using mistuning effect is realised in program package BLADIS+.**

For extension the fatigue life of a machine (or its components) under variable stresses reducing these stresses is of considerable importance. Under certain conditions, i.e. under resonance phenomena, the variable stress value may exceed the fatigue point of the material that may result in plastic deformation and fatigue failure of the component. At the present time there exist a number of techniques to reduce the stresses and therefore to increase component fatigue life. With new technologies, new methods are being created. In this paper method of increasing the turbomachine bladed disc fatigue life is considered: a passive method, using mistuning effect. The mistuning effect will be considered in connection with the construction material properties changes.

The first stage of the parameter mistuning analysis is calculating eigenfrequencies and vibration mode of an ideal model. A model of a bladed disk consisting of 8 sectors [1] was used as a base; outer radius is 0.070 m; inner radius – 0.025 m; blade length – 0.030 m (the blade is trapeziform: upper base is 0.009 m; lower base – 0.011 m); blade thickness – 0.003 m; elasticity coefficient is $7.2 \cdot 10^4$ N/mm^2; density is $2.7 \cdot 10^3$ kg/m^3; the Poisson's ratio is 0.3. The model has 53 nodes and 72 plate triangular finite elements (Fig. 1).

The eigenfrequencies calculated without parameter mistuning are shown in Table 1.

In order to estimate calculation reliability the results obtained with the BLADIS+ software [2, 3] were compared with the calculations in [1] and both are represented in a diagram form on the Fig. 2.

The comparison showed that the results obtained with BLADIS+ differ insignificantly from those in [1] (not more than 3–4%).

Then, the calculation with mistuning was conducted. For this purpose the eigenfrequencies of every single (hard fixed) blade were calculated, taking into consideration the change of construction material elasticity coefficient using the law $E_i = (1 + d_i)$, where $E = 7.2 \cdot 10$ N/mm^2 [1], $i = 1, \ldots, 8$; the variables for d_i are represented in the Table 2.

The obtained eigenfrequencies subject to the elasticity coefficient are represented in the Table 3.

Then, eigenfrequencies of blades with mistuning are calculated using the BLADMIS block of the BLADIS+ software [2, 3]. The initial data for the calculation are natural frequencies from the Table 3 and the frequency of the ideal blade (without elasticity coefficient change) which is equal to 8908.40 Hz.

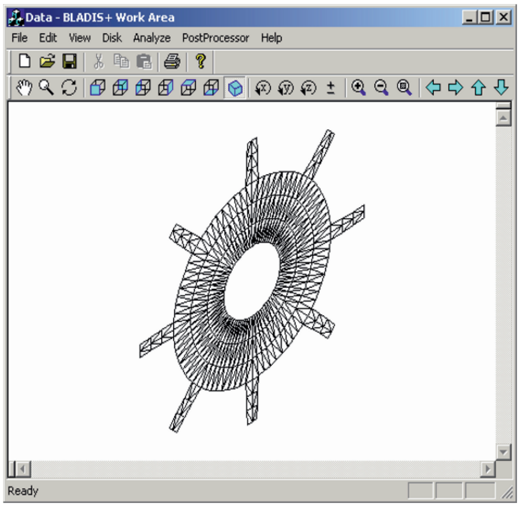

Fig. 1. Finite element model of bladed disk

TABLE I
VIBRATION EIGENFREQUENCIES WITHOUT MISTUNING

Number of nodal diameters	0	1	2	3	4
Frequency, Hz	1452	1463	1644	2028	2197

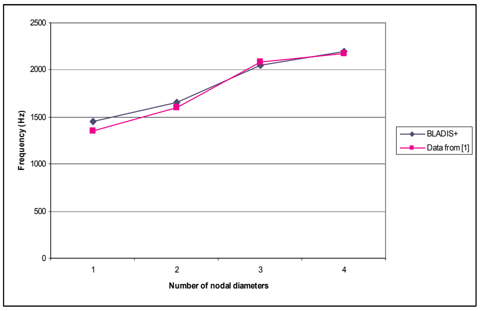

Fig. 2. Eigenfrequencies without mistuning

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TABLE 2
CHANGE OF MATERIAL ELASTICITY COEFFICIENT

№ of blade	d_i
1	-0.02390
2	0.03808
3	0.05392
4	-0.05998
5	-0.05388
6	0.02622
7	-0.01222
8	0.03136

TABLE 3
EIGENFREQUENCIES SUBJECT TO THE CHANGE OF MATERIAL ELASTICITY COEFFICIENT

№ of blade	Frequency, Hz
1	8788.75
2	9068.40
3	9137.40
4	8647.70
5	8657.60
6	9021.80
7	8849.50
8	9023.20

This frequency is used as nominal during calculations subject to mistuning. The objectives of this work are to investigate how mistuning influences the eigenfrequencies while elasticity coefficient changes and therefore to find how mistuning effects the component durability. The following eigenfrequencies with mistuning were obtained using the BLADMIST block, to compare them with the data in [1] (Fig. 3).

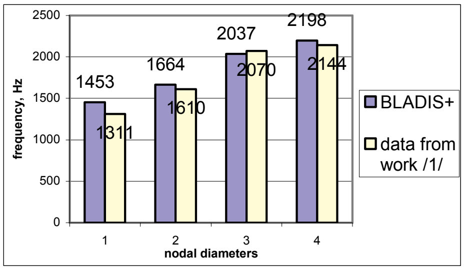

Fig. 3. Eigenfrequencies with mistuning

The analysis of the results showed that the obtained eigenfrequencies with mistuning (set by elasticity coefficient change from [1]) differ from eigenfrequencies without mistuning for 1-2%.

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A Real-Time Specific Weed Recognition System by Measuring Weeds Density Through Mask Operation

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Abstract— **The identification and classification of weeds are of major technical and economical importance in the agricultural industry. To automate these activities, like in shape, color and texture, weed control system is feasible. The goal of this paper is to build a real-time, machine vision weed control system that can detect weed locations. In order to accomplish this objective, a real-time robotic system is developed to identify and locate outdoor plants using machine vision technology and pattern recognition. The algorithm which is based on Measuring Weeds Density through Mask operation is developed to classify images into broad and narrow class for real-time selective herbicide application. The developed algorithm has been tested on weeds at various locations, which have shown that the algorithm to be very effectiveness in weed identification. Further the results show a very reliable performance on weeds under varying field conditions. The analysis of the results shows over 95 % classification accuracy over 170 sample images (broad and narrow) with 70 samples from each category of weeds.**

Keywords— Weed detection, Image Processing, real-time recognition, weed density, mask.

I. INTRODUCTION

WEEDS are "any plant growing in the wrong place at the wrong time and doing more harm than good" [1]. Weeds compete with the crop for water, light, nutrients and space, and therefore reduce crop yields and also affect the efficient use of machinery. A lot of methods are used for weed control. Among them, mechanical cultivation is commonly practiced in many vegetable crops to remove weeds, aerate soil, and improve irrigation efficiency, but this technique cannot selectively remove weeds from the field. The most widely used method for weed control is to use agricultural chemicals (herbicides and fertilizer products). In fact, the success of agriculture is attributable to the effective used of chemicals.

Agricultural production experienced a revolution in mechanization over the past century. However, due to the working environment, plant characteristics, or costs, or there are still tasks that have remained largely untouched by the revolution. Hand laborers in 1990's still may have to perform tedious field operations that have not changed for centuries. Identification of individual weeds in the field and location their exact position is one of the most important tasks needed to further automate farming. Only with the technology to locate individual plants, can "smart" field machinery be developed to automatically and precisely perform treatments.

Simple methods of weed control avoid using chemicals. They are often used by farmers. However weed control can also be achieved by the use of herbicides. Selective herbicides kill certain targets while leaving the desired crop relatively unharmed. Some of these act by interfering with the growth of the weed and are often based on plant hormones.

A real-time automatic weed control system can reduce or eliminate the need for chemicals between broad and narrow weeds.

The purpose of this paper is to investigate a real-time automatic machine vision system to distinguish weeds into broad and narrow categories.

2 . OBJECTIVE

There are only two types of herbicides used: broad leave weed and narrow leave weed, our objective is to develop an algorithm which can

Recognize the presence of weeds

Differentiate the broad leaves weeds and narrow leaves weeds.

3. MATERIALS

3.1 Hardware Design

The concept of the proposed automated sprayer system is shown in Fig.1, which includes camera, Central Processing Unit (CPU), and Decision Box which is used to control dc pumps. The images were taken at an angle of 45 degree with the ground. Using this method, the long narrow area in front of the sprayer could be captured with high resolution without increasing the image size.

The images are given to Central Processing Unit. The Decision Box is connected to the Central Processing Unit

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through a parallel port which ON or OFF the corresponding pumps, based on the type of image processed by the Central Processing Unit.

3.2. Software Development

The software is developed in Microsoft Visual C++ 6.0. A Graphical User Interface (GUI) is developed for Original image and processed image is based on Measuring Weeds Density through Mask operation.. The image resolution was 240 pixel rows by 320 pixel columns.

4. METHODOLOGY

Fig. 2 shows the Flow Chart of a Real-Time Specific Weed Recognition System which were developed to recognize the broad and narrow weed classification [2]. The algorithm was based on Measuring Weeds Density through Mask Operation.

A. Image acquisition

Our system get image as RGB color format with a resulation of 320 * 240. This mage can be a stored image or it can be areal image get from an attached camera or even in can be an image obtained from a video. If an image is not in the given resulation our system convert it into the standerd format.

B. RGB to Grayscale

In the start of the image preprocessing operation the input image is decomposed into red, green and blue components to create a binary image using the following transformation.

If G>R and G>B and G>150 then
PIMG = 1
Else
PIMG = 0
End if

Where R, G and B are the red, green and blue components and PIMG is the processed binary image.

The resulting binary image will have weeds in bright pixels and background in dark pixels.

C. Classification by Measuring Weeds Density through Spatial Masking

We first measure the percentage of weeds in the image by calculating the percentage between bright pixels in the binary image and the total image pixels.

$$\text{Weed\_Percentage} = \frac{\text{No of Bright pixels in the Binary Image}}{\text{Total no of Image pixels}} \times 100$$

To measure weeds density in the binary image we convolve a 9 x 7 mask over the entire image and then calculate the percentage of bright pixels (weed pixels) within the small mask and then depending on the percentage the value of the center pixel is set to either 1 (if Percentage >= 50) or 0 (if Percentage <50). This procedure is repeated for all pixels in the image. The resulting image (Density Image) will highlight areas with higher density (i.e Broad Weeds). A percentage value is calculated for the broad weeds in the image as:

$$\text{Percentage\_Density (Broad Weed)} = \frac{\text{Number of Bright Pixels in the Density image}}{\text{Total No of Bright pixels in the binary preprocessed image}} \times 100$$

By using the above calculated values i.e Weed_Percentage and Percentage_Density we classify the image as Broad Weed, Mixed weed or Narrow Weed.

D. Algorithm for the classification of Images

The images are classified using the following procedure.

Start

If

(Weed_Percentage >=T1 and Percentage_Density >=T2) or
(Weed_Percentage>=T3 && Percentage_Density >=T4)
Classify as "Broad Weed"
Else if
Weed_Percentage >=T5
Classify as "Narrow Weed"

Else
Classify as "Little Weed"
Endif

End

T1, T2,T3,T4,T5 are threshold values [3], [4], [5].

5. RESULTS AND DISCUSSION

The results are shown in fig-3. Which were classified by measuring weed density through mask operation. The 170 images of 85 for each class were taken. The algorithm classified 99% for little weed or no weed and 95% for narrow and braod weeds as shown in table 1 and table 2.

6. CONCLUSION

A real-time weed control system is developed and tested in the lab for selective spraying of weeds using vision recognition system. In this paper, feature extraction based system for weed classification and recognition is developed. The system shows an effective and reliable classification of images captured by a video camera. The system composes of four main stages: image capturing, image pre-processing, feature extractions and classification.

7. FUTURE WORK

In this paper weed image, which has one dominant weed species can be classified reasonably accurate. But the case of more than one weed classes cannot be accurately classified. Further research is needed to classify mixed weeds. One way is to break the image into smaller region. With smaller region, there will be less possibility to find more than one weed classes in this small region.

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TABLE 1

Weeds Type	Results %
Broad Weeds	97%
Narrow Weeds	93%
Little Weeds	99%

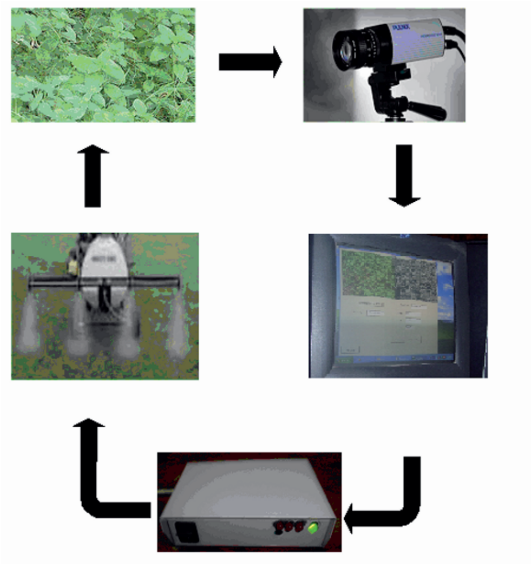
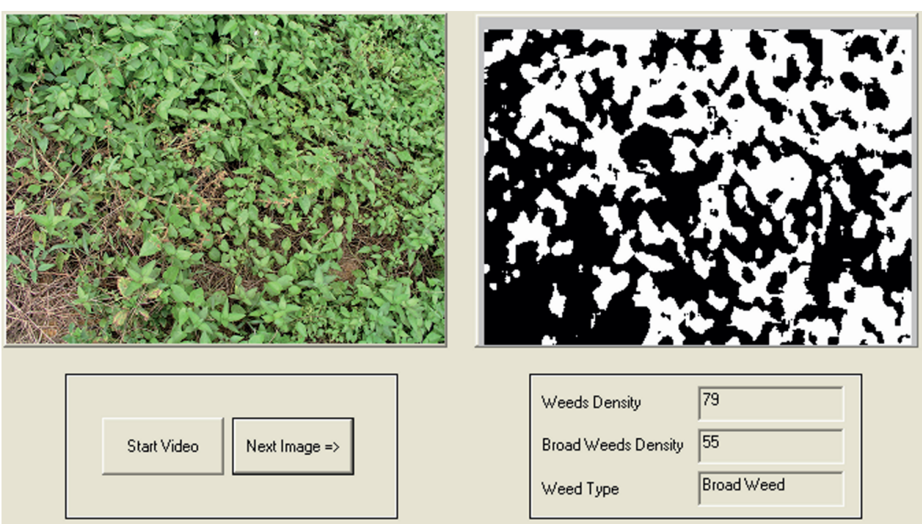


Fig 2

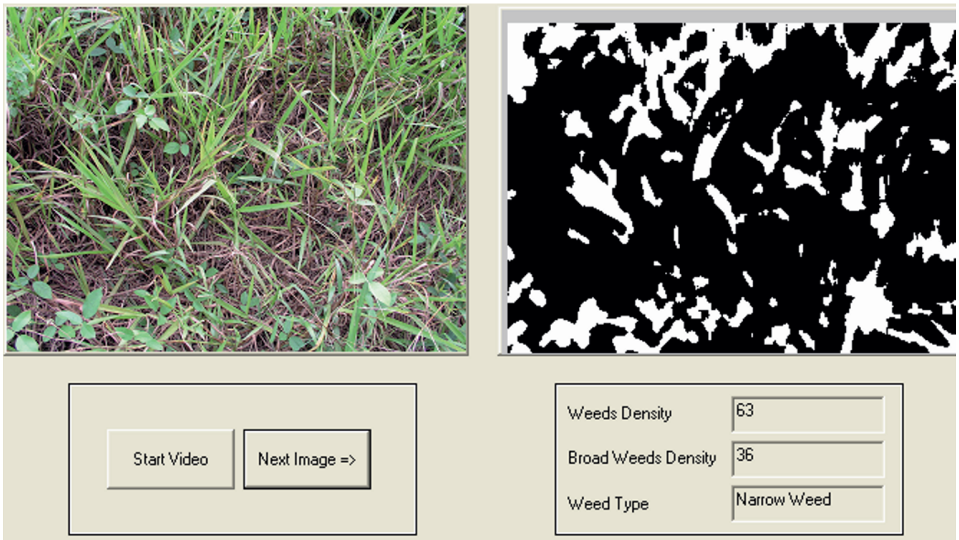
TABLE 2
RESULTS OF CLASSIFICATION OF NARROW, LITTLE, AND BROAD WEEDS

Narrow Weed		Little Weed		Broad Weed	
Weed Density	Broad weed density	Weed Density	Broad weed density	Weed Density	Broad weed density
52.09	37.73	30.42	19.49	75.93	60.54
62.95	35.20	22.27	22.34	69.39	53.43
46.95	47.64	12.67	8.99	69.39	53.43
30.42	19.49	24.02	19.46	79.88	55.69
41.95	29.17	7.74	9.05	71.98	49.65
56.72	35.69	27.76	20.19	70.09	48.61
53.80	26.09	6.29	11.78	75.80	47.91
60.66	27.03	0.29	3.10	69.39	53.43
33.74	47.04	27.90	39.05	76.64	49.80
45.41	39.82	22.27	28.49	70.09	48.61
41.80	46.71	23.51	19.04	49.93	59.21
44.65	43.37	29.29	37.41	69.39	53.43
36.84	34.66	27.90	39.05	75.80	47.91
53.84	37.55	27.08	20.17	77.65	57.40
58.36	45.78	27.99	19.49	65.20	41.73
31.92	34.96	29.82	27.67	77.31	51.05
62.95	35.20	6.67	11.71	75.80	47.91
32.50	54.70	25.19	27.28	65.20	41.73
62.95	35.20	29.63	10.47	62.85	44.90
59.62	49.51	29.99	15.26	69.29	60.65
48.08	32.56	28.96	18.25	49.93	59.21
54.91	30.77			61.46	49.01
54.81	28.44			70.67	52.04
64.04	36.17			78.36	59.51
30.42	19.49			79.88	55.69
62.95	35.20			63.40	52.46
35.74	24.27			62.64	54.16
56.72	35.69			57.65	59.82
60.88	31.10			71.29	64.00

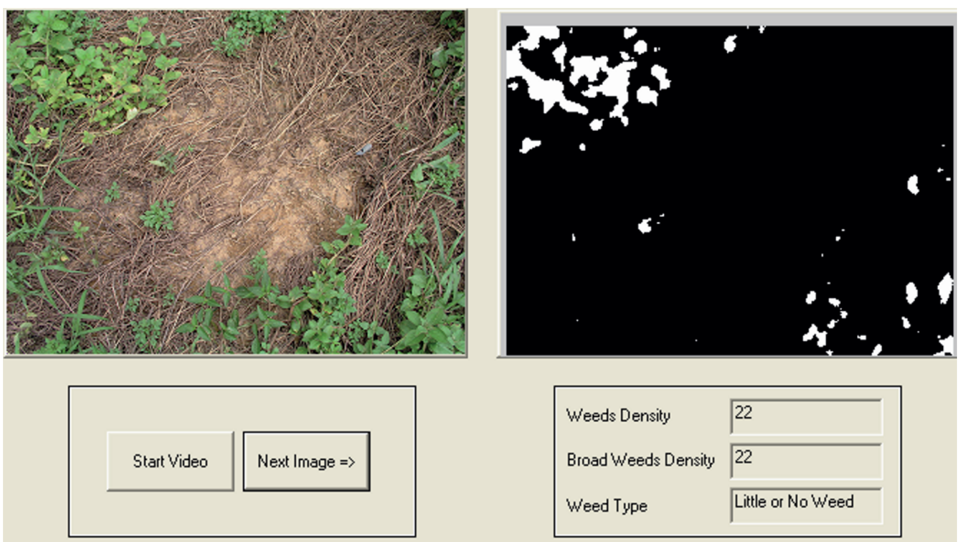


(a) Broad Weed

Fig 3 Classified Images



(b) Narrow weed



(c) Little weed

Fig 3 Classified Images

Towards a Better Robustness-Imperceptibility Tradeoff in Digital Watermarking

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***Abstract*-Robustness and imperceptibility are two essential but contradicting properties in robust digital watermarking. This paper proposes an approach for obtaining superior robustness-imperceptibility tradeoff by considering the likely set of attacks a watermark is expected to be mounted with. The suggested approach achieves this improvement by using Genetic Programming to develop appropriate perceptual shaping functions for structuring the watermark intelligently by choosing the optimum strength of allowable alteration in watermarkable DCT features, in view of a set of conceivable attacks. The developed perceptual shaping functions, which outperform the conventional ones, are generalized with respect to the cover work and are based on the watermark application.**

***Keywords*: Digital Watermarking, Genetic Programming, Bit Correct Ratio and Structural Similarity Index**

I. INTRODUCTION

The prevalence of digital data and interconnected networks has stimulated the research in the field of digital watermarking. On commercial basis, the demand of digital watermarking-based applications, systems and services is increasing in order to deter the piracy of entertainment content, better manage and measure broadcast programming, enhance the security of driver licenses, and improve the use of commercial, medical and geospatial imagery [1]. Robust digital watermarking deals with piracy control and owner assertion. The two essential but contrary properties of such a watermarking system are *robustness* and *imperceptibility* [2]. An efficient watermarking system needs to make a delicate tradeoff between these contradicting properties, i.e. minimizing the distortion of the watermarked work with respect to the original and being robust against any legitimate and/or illegitimate attacks [2].

On the other hand, different applications pose different requirements on the watermark design, since they are heavily dependent upon the set of attacks that are likely to be mounted on them [3]. A universal watermarking system that can withstand all attacks and at the same time fulfill all other requirements is difficult to exist [2-3]. Therefore, while designing a watermarking system, its intended application and thus the corresponding set of conceivable attacks are of prime importance.

In order to be imperceptible, a watermark is generally structured using perceptual models which exploit the characteristics of Human Visual System (HVS). Watson's Perceptual Model (WPM) is a pertinent example of such models as it has been extensively used in digital watermarking and image compression [4-5]. Other perceptual models may include Lambrecht *et al.*'s Gabor filters based model [6] and Kutter *et al*.'s model based on local isotropic measure and a masking function [7]. These perceptual models are able to structure the watermark in view of imperceptibility alone, and thus, lack the information pertaining to distortions caused by a set of likely attacks. Hence, they are inefficient to cater, concurrently, for both robustness and imperceptibility requirements of a watermarking system and are inapplicable in applications, such as medical imagery watermarking, that require high level of robustness along with imperceptibility.

In this work, we present a system for developing appropriate (application dependent) perceptual shaping functions that are able to structure a watermark in view of robustness as well as imperceptibility. We use Genetic Programming (GP) to develop such perceptual shaping functions by exploiting the characteristics of HVS in block-based DCT feature space along with the selection of optimum strength of watermark to be embedded. The improvement in robustness achieved is not against a single attack, as in our previous work [8], rather, it is achieved against a series of anticipated attacks, which is a far more difficult task.

II. PROPOSED SCHEME

GP uses a data structure such as a tree to encode multi-potential solutions for specific problems. Initially, a random population of such candidate solutions is created. Every candidate solution is evaluated and assigned a numerical rating, or *fitness*, using an application specific, predefined *fitness function*. The best individuals in a generation are retained while the rest are deleted and replaced by the offspring of the best individuals to make a new generation. Offspring are created by applying genetic operators. The whole process is repeated for subsequent generations, with scoring and selection procedure continued, until the termination criterion is met. In

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this way, the solution space is refined generation by generation and converges to an optimum solution.

Before representing a candidate solution with a GP tree, one needs to define suitable GP *terminal set*, GP *function set* and the *fitness function* according to the optimization problem. Watson [9] modified the visibility threshold $T'(i,j)$, which incorporates the effect of frequency and luminance sensitivity, into $T^*(i,j)$ to take into consideration the effect of contrast masking as well:

$$T^*(i,j) = \max\left[T'(i,j)\ ,\ \left|T'(i,j)\right|^{1-\omega} X(i,j)^{\omega} \right], \tag{1}$$

where, $X(i,j)$ is the AC DCT coefficients of each 8×8 block and ω is empirically set to a value of 0.7. These allowed alterations represent the perceptual mask denoted by $\boldsymbol{\alpha}$. Therefore, the GP *terminal set* consists of the current value of WPM-based perceptual mask, DC and AC DCT coefficients of 8×8 block, and the coefficient indices (i,j) of the current DCT coefficients as variable terminals. Random constants in the range [-1, 1] are used as constant terminals. The GP *function set* comprises of simple functions including four binary floating arithmetic operators (+, -, *, and protected division), *LOG, EXP, SIN, COS, MAX and MIN*. Figure 1 shows a sample GP tree representing a candidate Genetic Perceptual Shaping Function (GPSF). Note that the actual evolved GPSF may comprise of a tree with much larger number of levels.

If A denote the information pertaining to the distortions caused by the set of conceivable attacks. The functional dependency of the GPSF on the characteristics of HVS and a cascade of conceivable attacks can be represented as follows:

$$\alpha_G(k_1,k_2) = f\left(X_{0,0},\ X(i,j),\ \alpha(k_1,k_2), A \right) \tag{2}$$

Where, $X_{0,0}$ is the DC DCT coefficient, $X(i,\ j)$ is the AC DCT coefficient of the current block. They represent the luminance sensitivity and contrast masking characteristics of HVS respectively, whereas, frequency sensitivity of HVS is included in $\alpha(k_1,k_2)$.

We define the *fitness function* as follows:

$$Fitness = W_1 * F_i + W_2 * BCR_{attack} \tag{3}$$

where, $F_i = SSIM_{E.S}$ represent watermark imperceptibility measure. $SSIM_{E.S}$ denotes the Structural Similarity Index Measure [10] of the marked image. On the other hand, BCR_{attack} is the robustness measure of the watermarked work and represents Bit Correct Ratio after the set of attacks are carried out in a sequence. It is defined as:

$$BCR(M,M')_{attack} = \sum_{i=1}^{L_m} \overline{(m_i \oplus m_i')} \Big/ L_m \tag{4}$$

where M is the original, while M' represents the decoded message. L_m is the length of the message and $\oplus$ represents exclusive-OR operation.

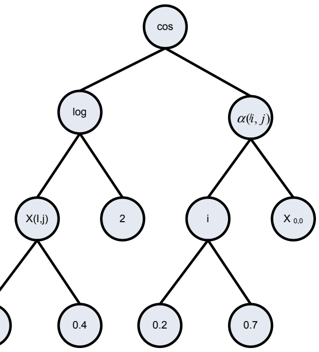

Fig. 1. A sample GPSF represented by a tree data structure.

A. *Training Algorithm*

The GP training algorithm for the proposed scheme is as follows:

FOR i = 1 to Number of Generations
    FOR j = 1 to population size
        ▪ Select an individual jth GPSF
        FOR k = 1 to Number of Training Images
        - Compute fitness corresponding to the k^{th} image using the algorithm presented in figure 2 and $GPSF_j$.
        END

$$Fitness(GPSF_j) = \sum Fitness_k \Big/ Number\ of\ training\ images$$

    END
- Rank GPSFs based on their fitness values.
- IF Termination criterion satisfied
    - ▪ Save best GPSF
    - ▪ Terminate algorithm
- ELSE
  Create next generation using the following genetic operators
    - ▪ Replication
    - ▪ Crossover
    - ▪ Mutation

END

Each individual in a population is scored against a number of training images. We choose this number to be 10 in the present work. The average fitness across the training images is the actual fitness of the individual GPSF. Figure 2 presents the block diagram of the *fitness computation module*.

An individual GPSF is presented to generate the perceptual mask α_G for k^{th} image. This perceptual mask incorporates the strength of allowable alteration within 8×8 block DCT features, and is presented to the *watermarking module*. The watermarking module uses Hernandez's watermarking technique [4] to watermark $image_k$ using the perceptual mask α_G. After the image is watermarked, its imperceptibility is computed before presenting it to a sequence of attacks in *attack module*. The attack module applies a sequence of attacks on the marked image in order to simulate the conceivable attacks the application is prone to in real world. Figure 3 illustrates an example of such a sequence consisting of intentional and unintentional attacks. A watermarked image is JPEG compressed in order to reduce its size as a normal image processing operation, thus, presenting an unintentional attack. An attacker gets hold of the image, and, in an attempt to remove the watermark, applies median filtering on it. The resultant image is then transmitted through an open network characterized by Gaussian noise. The received image is a distorted image induced with legitimate and/or illegitimate distortions in at different stages.

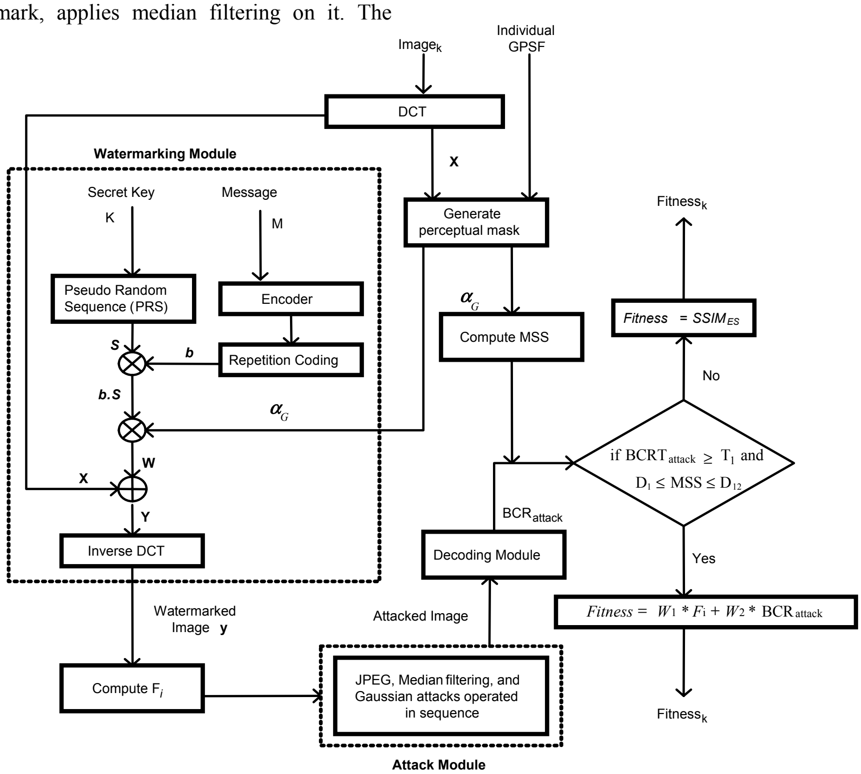

Fig. 2. Fitness computation module of the proposed scheme.

To estimate robustness of the watermark after a sequence of attacks, BCR_{attack} is computed. To estimate robustness during GP simulation, we also use watermark power, represented by Mean Squared Strength (*MSS*), given as:

$$MSS = \frac{1}{N_b N_c} \sum_{x_1=1}^{N_b} \sum_{x_2=1}^{N_c} \alpha(x_1, x_2)^2 \qquad (5)$$

where, N_b is the total number of 8×8 blocks in the cover image and N_c is the number of selected DCT coefficients in a block. x_1 and x_2 are the respective indices. If *MSS* and BCR_{attack} values lie within and under certain bounds respectively, *bonus fitness* is given which includes fitness due to imperceptibility as well. Else, the fitness equals BCR_{attack} value. In this way, those individuals which are performing well in terms of both robustness and imperceptibility are given more chance to reproduce in the next generation. Bounding of *MSS* and BCR_{attack} values helps in selecting individuals which provide sufficient watermark strength without violating the imperceptibility requirement.

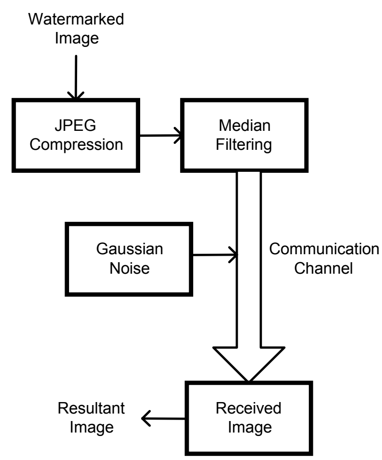

Fig. 3. Sequence of attacks performed on the watermarked work.

The best evolved expression is saved at the end of the GP simulation and is used to watermark the test images. In our previous work [8], we have considered a simple case where we evolved expressions for a single attack. As regards, the real world application requirements, a much harsh and challenging scenario is to make the watermark resistant to a set of attacks.

III. EXPERIMENTAL RESULTS

The proposed scheme is implemented in Matlab and using GP Lab toolbox for Matlab. It is tested on 200 images from everyday life.

The generated GPSF for attack sequence presented in figure 3 is:

plus(exp(myMin(myMin(DctCurrtCoef,DctCurrtCoef),cos(DctCurrtCoef))),plus(exp(mygt(mylog(plus(exp(mygt(mylog(exp(mygt(mylog(plus(exp(mygt(mylog(DctCurrtCoef),DctDC)),mylog(myle(DctCurrtCoef,DctCurrtCoef)))),DctDC))),DctDC)),mylog(myle(DctCurrtCoef,plus(exp(mygt(cos(DctCurrtCoef),DctDC)),mylog(myle(DctCurrtCoef,DctCurrtCoef))))))),DctDC)),mylog(myle(DctCurrtCoef,DctCurrtCoef)))).

For reference to the function set denotation, GP Lab manual may be found at [11].

Table 1 presents performance of the evolved expression on the training images. Table 2 presents performance measures using Watson's Model on the same set of training images. As is obvious from the table, the average BCR value attained by the generated PSF and the overall fitness value is greater than that achieved from Watson's model.

TABLE 1
PERFORMANCE OF DIFFERENT MEASURES FOR THE EVOLVED EXPRESSION ON TRAINING IMAGES

	Alpha	DWR	Wpsnr	SSIM	Fitness imp. (F_i)	BCR
	1	26.37	39.527	0.9631	0.9631	0.625
	1	28.675	40.274	0.97161	0.97161	0.4375
	1	30.906	40.903	0.96431	0.96431	0.625
	1	22.703	37.204	0.95374	0.95374	0.6875
	1	26.208	37.821	0.95862	0.95862	0.5
	1	25.989	41.114	0.95738	0.95738	0.4375
	1	28.777	41.172	0.96962	0.96962	0.5
	1	30.141	39.823	0.95848	0.95848	0.625
	1	35.606	42.195	0.96272	0.96272	0.5625
	1	33.258	41.503	0.96733	0.96733	0.5625
Average	**1**	**28.8633**	**40.1536**	**0.962691**	**0.962691**	**0.55625**
						Fitness =1.518941

TABLE 2
PERFORMANCE COMPARISON OF WATSON'S PERCEPTUAL MODEL FOR DIFFERENT MEASURES ON TRAINING IMAGES

	Alpha	DWR	wPSNR	SSIM	Fitness imp. (F_i)	BCR
	0.2021	37.035	48.112	0.99272	0.99272	0.375
	0.1921	37.49	48.154	0.99412	0.99412	0.5
	0.1701	40.237	49.443	0.99366	0.99366	0.625
	0.3241	31.544	44.397	0.98747	0.98747	0.5
	0.1501	40.128	50.176	0.99584	0.99584	0.3125
	0.3041	31.788	46.267	0.98546	0.98546	0.4375
	0.1121	42.145	53.481	0.99797	0.99797	0.375
	0.1621	41.851	48.7	0.99207	0.99207	0.375
	0.2361	40.729	47.462	0.98874	0.98874	0.5
	0.1881	39.214	47.675	0.99244	0.99244	0.4375
Average	**0.2041**	**38.2161**	**48.3867**	**0.99204**	**0.99204**	**0.44375**
						Fitness =1.435799

After generating an optimum GPSF through the training phase, it is tested using Hernandez's watermarking technique [4]. Figure 4 illustrates the performance comparison of the evolved GPSF and Watson's perceptual shaping function on 200 test images. The evolved expression outperforms Watson's in terms of average BCR, on training images as well as on test images, which suggests its generalization.

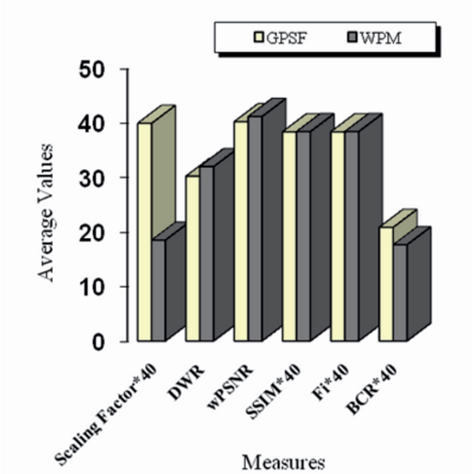

Fig. 4. Performance of the evolved expression and Watson's model on 200 images from the everyday life. Note that both perceptual shaping functions provide almost similar amount of imperceptibility. But the evolved expression outperforms Watson's perceptual shaping function in terms of robustness.

IV. Conclusion

The proposed method makes a better robustness-imperceptibility tradeoff and develops appropriate perceptual shaping functions by taking into account the attack information as well besides the allowable strength of embedding. It is able to improve the BCR after a sequence of attacks without compromising much on imperceptibility, which is mostly desirous in real watermarking applications. Experimental results show that the proposed scheme outperforms the conventional perceptual models that structure the watermark in view of imperceptibility alone.

Acknowledgements

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Conflict Expansion in an Information Rich Society: Feasibility of Corrective Actions

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***Abstract*— The probability, power, and influence of terrorist attacks are increasing with growing access to information and material resources. This paper investigates the character of this relationship. Is it evolutionary, allowing to take measures when the situation indicates that the level of resources is too high and it is time to take a more restrictive approach? Or could it be an outbreak that would leave no way back after a certain level of resources would have been exceeded? The paper presents a simulation framework and experiments designed to answer these questions. The findings indicate that the results of terrorism activities can potentially start spreading very fast with the growing amount of information and material resources in individuals' hands, allowing no point of return. These results may be useful when designing political, social and technical systems to prevent terrorism.**

***Index Terms*—Biological system modeling, Simulation, Social factors, Stability criteria**

I. The Research problem, Methodology, and overview

The Oklahoma City bombing in April 1995 killed 168 people, 19 of them children. The truck bomb was composed of ammonium nitrate, an agricultural fertilizer, and nitromethane, a motor-racing fuel. Both components are accessible to an ordinary person. Similarly, information about how to use these and other components for preparing explosives is also easily available in the Internet. The more the technology develops and information spreads, the more of such incidents one can expect – a fact that certainly worries many people.

It is clear that when the resources, both the information and materials, are on a low level, one cannot do much harm. For example, prior the twentieth century it was practically impossible to affect the living conditions on the Earth significantly by even a large number of persons. The situation has changed, and mankind has now resources to destroy itself. Today these resources are usually out of reach of an ordinary person.

Little by little, this encouraging situation may also change. Converging of potentially disruptive technologies, such as nanotechnology, biotechnology, information technology, cognitive science, and artificial intelligence, may give previously unseen power in the individuals' hands. Like in the Oklahoma event, ordinary people will have more and more information and resources available for terrorism.

Let us therefore have a look at the other extreme – the hypothetical situation of the extremely large resources being available to everyone. It seems also clear that in such a case the world would not last long. There will inevitably be people who are stressed enough, or who believe that death is the only viable option for everybody. This extreme situation is unlikely, for the governments recognize the danger and are building barriers to conflict spreading. Still it seems inevitable that the power in hands of individuals is growing, and the governments are taking more measures to prevent that power from growing too high. The question is, how high is acceptable?

This question may be illustrated in the following way. Fig. 1 depicts two hypothetical graphs of relationship between the level of information and material resources available to an ordinary person (Resources) and the probability of surviving the population (Probability of Survival). The previous discussion indicates the location of two points on these graphs: the starting point (a – low resources, survival probability 1) and the ending point (b – very high resources, survival probability 0). The real situation in the world is somewhere between them, currently probably moving little by little from left to right.

The critical question is what happens between the points (a) and (b). Will it be a linear move that would allow taking measures when the situation indicates that the level of protective measures is too low and it is time to take a more restrictive approach? For example, if a population has moved on line L from point (c') to point (d') and decides that the survival probability is dangerously low, it could move back to point (c'). "Moving back" in this context does not necessarily mean that some information should be removed after it has been released (this may be impossible or at least very difficult), but rather that access to certain kind of information as a general resource might be more restricted.

Or could it be a curve with an unforgiving turning point that would leave no way back after a certain level of resources

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K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 231–236.

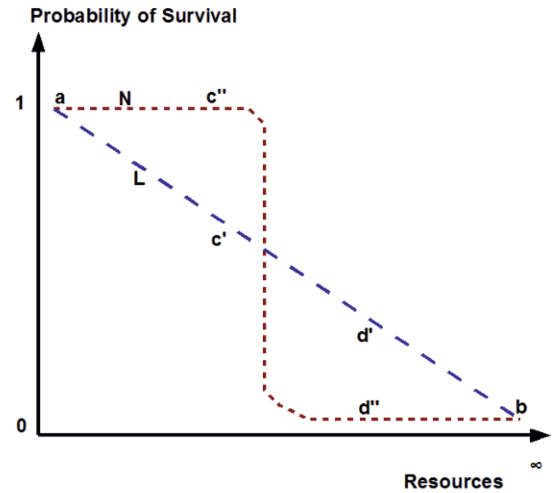

Fig. 1. Two hypothetical relationships between the level of resources and the probability of population survival: smooth (line L) and stepwise (line N).

would have been exceeded? In the case of line N, moving back from point (d'') to point (c'') may be impossible, because the population has ceased to exist.

In years following the Oklahoma event the importance of this question has been confirmed and proved worth consideration. The objective of the paper is to investigate it – to explore the character of the relationship between the level of availability of resources and the probability of population survival.

Direct examination is impossible in this case, so the methodology is based on simulation - modeling the problem area, development of a simulation environment, performing simulation experiments, and analyzing the results. It includes the following steps.

1) Development of a generic framework for the experiments.
2) Development of specific world models and associated simulation environments, based on the framework model.
3) Development of sub-models defined within the simulation environment.
4) Performing experiments with the sub-models and analyzing their results.
5) Developing further the models and the simulation environment.

Research on terrorism simulation is evolving along a number of tracks, especially after the 9/11 attacks. Important topics, such as the influence of deliberate release of infectious agents [1], protection of stationary potential terrorist targets [2], evolution of new simulation models and tools [3], simulation of the dynamic interactions between terror and anti-terror organizational structures [4], simulation of the dynamics of spread of extreme opinions in a society [5], simulation of large-scale worm attacks in Internet [6] have been studied. Criteria for a computational tool to support social simulation with examples using the language SDML are presented in [7]. To the authors' knowledge the problem brought up in this paper has not been investigated elsewhere.

In the earlier papers, the authors have developed a general framework for the investigation, including mathematical representations of the world, agents, interactions, and the lifecycle [8]. A prototype simulation environment was presented in [9] together with experiments focusing on the dependence of average aggressiveness of the agents relative to the access to information during the world life cycle (this is a different issue of the one studied in the current paper). Details of the final simulation environment, a specific sub-model, and initial experiments addressing the probability of termination of the population were introduced in [10]. In the current paper new environments and experiments are presented discussing the influence of other parameters on the relationships.

II. The Generic Framework for Simulating the Spreading of Terrorist Attacks

Due to the world complexity, the conflict situations must be modeled using incomplete models of reality. To have more trust in the results of the investigation, a generic framework has been developed which allows initiating different world models. The assumption is that if they all lead to similar results, then the overall credibility of the conclusions will be higher.

The generic framework includes a world of agents. The world has certain properties and so do the agents. The world evolves in a discrete time. The properties of each individual agent at the next time moment are established according to the world properties and the values of its own and neighbor agents at the current moment. In each world cycle, the agents go through various interactions. The world life-cycle includes initialization, iteration, and finalizing stages. In what follows, these concepts are presented in more detail.

A. The World Properties

The world is determined by its shape and size, the values for overall access to information and access to resources, initial distributions of information, resources, violence, and charity, as well as the rules for activating violence or charity acts and for changing the agent values. Among the world properties are the following:

Reproduction - reflects the probability of reproduction for an individual agent;

Expiration - reflects the probability of expiration (death) of an individual agent;

AccessToInformation - reflects the overall level of access to information in the world W. As an example from the real world, wide use of Internet has significantly increased the level of access to information;

AccessToResources - reflects the overall level of access to resources in the sub-model. As an example, widespread use of chemistry products in households has significantly enhanced preparation of explosive materials;

Influence - the greater Influence, the more objects will be influenced by surrounding objects;

Violence - reflects various influence on the ability to perform terrorist acts, for example the overall characteristics of the agent population, or countermeasures taken. If Violence = 0, no terrorist acts will be performed at all.

These properties are independent. For example, increasing AccessToInformation or AccessToResources values may be accompanied with decreasing or increasing the Violence property value.

Some properties of the world determine the properties of the simulation itself, for example the rules which determine if the world is considered to be stable or terminated in a specific experiment:

GlobalEndOfPopulation - if the ratio (number of survived objects) / (initial number of objects) is less than or equal to GlobalEndOfPopulation, the population is considered to be terminated;

StablePopulation - if the ratio (number of changed objects)/ (current number of objects) is less than or equal to StablePopulation, the population is considered to be stable.

B. *The Agent Properties*

The main properties for individual agents are:

Knowledge - level of knowledge for an agent;

Resources - level of resources for an agent;

Aggressiveness – the basic property of an agent to be aggressive in its interactions;

Violence - characterizes current aggressive aspects of an agent;

Charity - characterizes current non-aggressive aspects of an agent.

The initial property values of an agent are determined by the world properties.

C. *Interactions between the Agents*

The world evolves in a discrete time. The properties of each individual agent at the next time moment are established according to the world properties and the values of its own and neighbor agents at the current moment.

In each world cycle, the agents go through various interactions. The interactions include the following:

Birth - birth of a new agent. The new agent inherits all properties of the parent;

Death - death of an agent;

Learning - in each step, agents get more information;

Forgetting - in each step, agent information is both learned and lost;

Earning - in each step, agents get more resources;

Spending - in each step, agent resources are both earned and consumed;

SocialInteraction - agressiveness value for an individual agent is moved towards the value of average Agressiveness of neighbours, and so the more the greater the value of Influence;

TerroristAct - an agent performs a terrorist act with probability proportional to the overall violence level and its own knowledge, resources, charity, violence, and aggressiveness levels. As the result of the act, the neighbours of the agent will be removed with probability adversely proportional to the distance form it (the nearer neighbours suffer to a greater extent). The Aggressiveness level of the neighbours will increase in adverse proportion with the distance from a (nearer neighbours are more influenced);

Charity-act - an agent performs a charity act with probability depending on its own and the environment property values. As the result of the act, the violence level of the neighbours will decrease in adverse proportion with the distance from the agent (nearer neighbours are more influenced).

There may be other properties and interactions.

D. *The World Life-Cycle*

The life-cycle of the world can be presented by the following algorithm.

Algorithm *WorldLifeCycle*
    Set initial values for the world and agent properties;
    PopulationHasTterminated := **False**;
    StableNonInteraction := **False**; *Lifetime* := 0;
    InitialPopulation:= Initial number of agents;
    While Not (*PopulationHasTterminated*) **And Not** (*StableNonInteraction*)
        Update the world performing the interactions defined for the agents;
        Calculate the number *Nchanged* of objects changed during the activities;
        CurrentPopulation = Current number of agents;
        If *CurrentPopulation / InitialPopulation* ≤ *GlobalEndOfPopulation* **Then**
        PopulationHasTterminated := **True**;
        If *Nchanged / CurrentPopulation* ≤ *StablePopulation* **Then**
        StableNonInteraction := **True**;
        Lifetime := *Lifetime* + 1;
    End while
    Return (*PopulationHasTterminated*, *StableNonInteraction*, *Lifetime*, the property values)
End *WorldLifeCycle*

III. The Simulation Environment

The simulation environment provides a simulation model construction language (SMCL) and tools for executing the model defined in SMCL. The SMCL defines the simulation, world, and agent properties and interactions. Examples of properties and interactions defined in SMCL are given in Table 1.

The tools for executing the model defined in SMCL allow for visualizing, saving, and analyzing the results. Visualizing the results of the simulation is provided by the main window and auxiliary windows. An example of the main window and two auxiliary windows is given in Figure 2. The agents are represented as squares of different colors. The black color of a square depicts a dead agent. The other colors indicate the state of aggressiveness of an agent, varying from light green (zero aggressiveness level) to yellow (average aggressiveness) to red (very aggressive). The auxiliary windows provide graphs

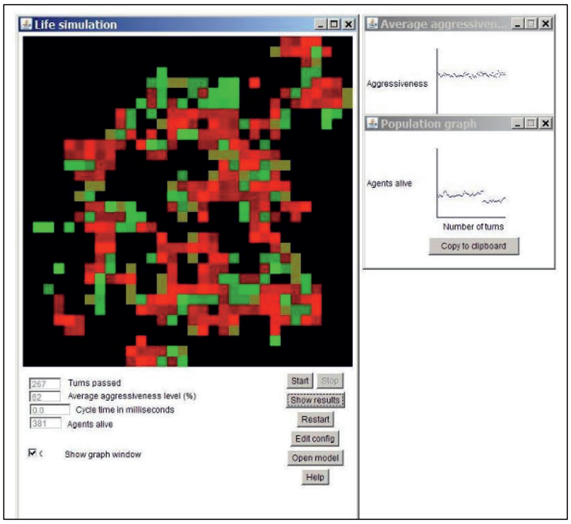

Fig. 2. Visualization of the results of simulation.

for average aggressiveness of the population and the ratio of survived agents to the initial number of agents.

At the start of a simulation run the world and the agents are initialized according to parameters given in SMCL. The rules for changing the world and agent properties during each world cycle are also given in SMCL.

At the end of each cycle, a check is performed for the end of the simulation. The simulation is finished and the final results are output in the following cases:

1) the population has survived (it has gone through the given number of iterations without termination);
2) the population has stabilized (its change has been within given small limits during a given number of iterations);
3) the population has terminated (the number of agents in the population has fallen below a specified limit).

IV. Experiments: Forms of Relationship between the Population Survival and Access to Information

Experiments have been performed using three simulation environments of different complexity. Each experiment included one or more simulation sequences. In each simulation sequence a number of sub-models were created. For each sub-model, several test runs were initiated to estimate the probability in question.

Results of a typical experiment portraying the relationship between the population survival and access to information for different levels of access to material resources are demonstrated on Figure 3. The figure depicts three simulation sequences, involving respectively high (square markers), medium (triangle markers), and low (round markers) level of material resources for individual agents.

Each simulation sequence comprises simulations using a series of sub-models, with the level of access to information in

TABLE I
Some properties and interactions defined in the SMCL

Properties	Example
Simulation general properties - end of simulation conditions, the time interval length for watching the world.	An assertion “GlobalEndOfPopulation=0.05” determines that if the ratio “number of survived agents / initial number of agents” is less than 0.05 then the population is considered to be dead.
World general properties - number of agents, access to information, cost of living, self-realization (attribute of the environment determining an agent’s potential for applying and enlarging its resources).	An assertion “Access to Information = 79” defines the overall level of access to information in a specific sub-model on a scale 0…100.
The initial distribution intervals for the agent property values - Aggressiveness, Violence, Knowledge, Charity, and Resources.	An assertion “init_[Aggressiveness]=33-66” initiates each agent’s aggressiveness to a random value in the interval 33...66 on a scale 0…100.
The rules determining change of the agent property values in each cycle.	An assertion “[Resources]=[Resources] + (([SfR] * [Knowledge]) / 100)” increases the amount of resources of an agent on each world cycle according to the value of an agent’s self-realization and knowledge.
Conditions for the agent’s rebirth - number of neighbors required for resurrection of an agent, the resurrection probability, aggressiveness value of an agent after resurrection.	Assertions “resurrection_agents=3” and “resurrection_probability=0.3” determine that if a dead agent has three neighbors, it will recover with a probability 0.3.
Condition for initiating a terrorist act.	An assertion “ aggressiveness_for_act=80” specifies that if an agent’s aggressiveness value is higher than 80 (on a scale 0...100) it will consider a terrorist act.
Condition for an agent to die if its resources are exhausted.	An assertion “set_[Resources]=affect_death” determines the agent to die if its resources are exhausted.

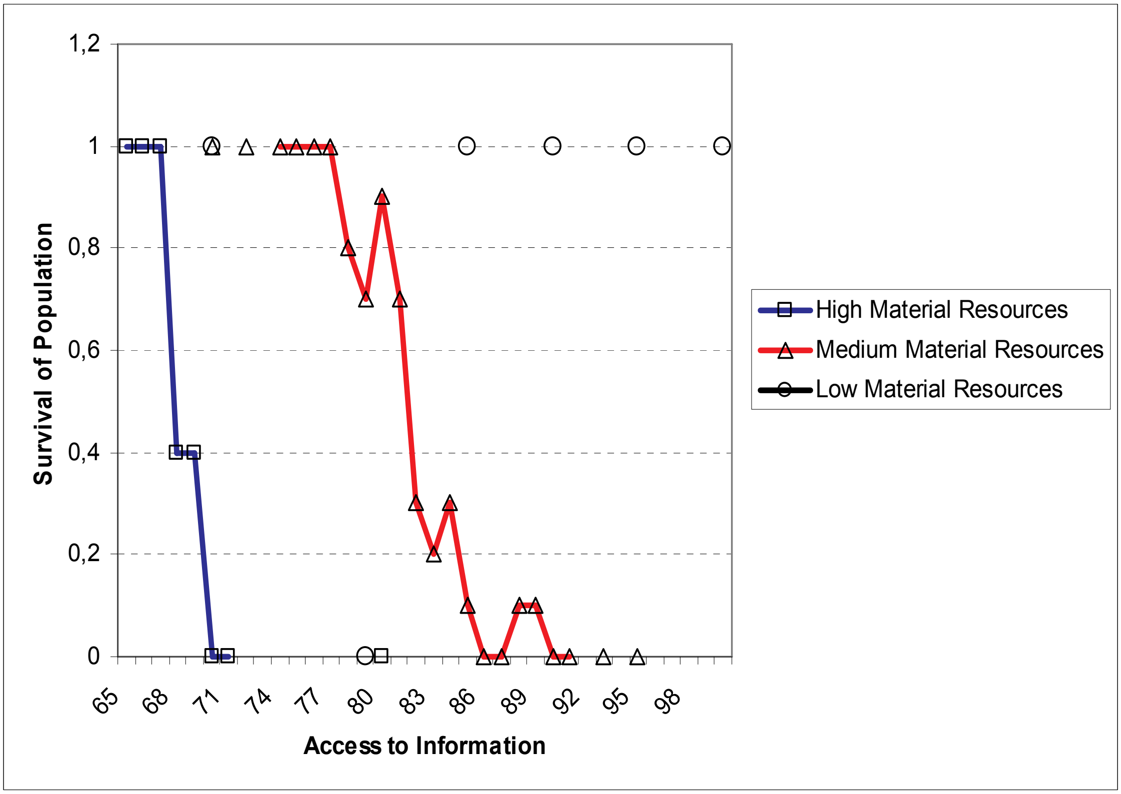

Fig. 3. An experiment with three simulation sequences involving changing level of material resources and access to information for individual agents.

each sub-model varying in the interval from 65 to 100 (on a scale 0...100). Other parameters of the simulation are kept constant. For each sub-model, five to ten test runs were performed (the number was lower in the case there was a strong tendency for the situation to remain similar in the following runs).

The estimation of the survival probability is calculated as the ratio of the test runs in which the population survived, to the overall number of test runs for this sub-model. As an example, Figure 3 shows that given a high level of material resources (square markers) and access to information value of 68, the ratio of the number of survived populations to the overall number of test runs for this specific sub-model was 0.4.

Referring back to Figure 1, the above results demonstrate that the relationship tends not to be linear like the one depicted by the line L in Figure 1. Rather, the graphs represent a stepwise (line N in Figure 1) or constant relationship for this model. Therefore the model does not necessarily lead to destabilization of the population with the growth of access to information. But in the case it does, the resulting dependency is rather step-wise than smooth.

The graphs also demonstrate the effect of material resource availability on the character of relationship. Medium (triangle markers) and high (square markers) level graphs indicate that when the material resources get higher, the survival turning point may move to left – a lower level of access to information is needed to destroy the population.

Furthermore, the graphs demonstrate the effect of shortening the turning area – the more possibilities, the narrower the turning area. Hence the population has less time to correct the situation.

Given a very low level of material resources, the population survival does not depend on access to information (line with round markers) - however clever the agents are, none of them can commit a serious terrorist act.

V. Conclusions and Further Work

This work has investigated a model of conflict and terrorism embodying some of the inherent characteristics of the modern world. Experiments have been performed using simulation environments of different complexity. The behavior of the simulated world in those experiments remained the same as far as the research question was concerned: the model does not necessarily lead to destabilization of the population with the growth of individual resources, but in the case it does, the resulting dependency is rather step-wise than smooth. These experiments allow concluding that this property may be not an incidence, but a regular behavior.

The findings indicate that the results of terrorism activities can start spreading very fast with the growing amount of information and material resources in individuals' hands, allowing no point of return. These results should be taken into account when designing political, social and technical systems to prevent terrorism.

One future research direction is to validate and improve the overall simulation model.

In the real situation, some of the world parameters can be managed while some are given. As a future research direction, the model will be restructured, in particular investigating in more details the parameters and mechanisms that can be managed and that might eliminate the threshold or make the curve smoother. Then the effect of, and need for such mechanisms could be simulated. Some examples:

1) introduction of specific "tampering centres" (analogy with government initiatives, or with critical infrastructure protection measures) that would soften the effect of terrorist attacks;
2) introduction of the effects of criminalization of the misuse of information and other technologies;
3) division of the whole population into sub-populations (like regions, individual states, or even cultures) with different characteristics (resources, aggressiveness, etc) which may restrict the influence of terrorist attacks;
4) taking into account that in the real world the availability of information and material resources also allow to define better prevention and protection strategies.

It is important that the future steps to fight with terrorism would not unnecessarily constrain free spreading and use of information. Instead focusing both on the criminalization of the misuse of information technology and on the systematic implementation of measures designed to prevent damage to critical information infrastructures should be taken into account in future simulation models.

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Ontology-based Shot Indexing for Video Surveillance System

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Indexing video data is very complex due to the enormous size of video files and their semantically rich contents and unstructured format. Therefore, the first step of video indexing begins with segmenting each video into a number of basic processing units. In general, the most widely used basic unit is a shot that is defined as collections of frames recorded from a single camera operation. Basically, there are two different approaches to index video using these shots. One is using low level features such as color, motion, or object. On the other hand, high-level features such as human interpretations are utilized for the indexing. Recently, a hybrid indexing combining these two approaches has been proposed to take advantages of both of them, and shows that dealing with both features has certain advantages. However, it is necessary that further improvements are required for which features can be extracted, and how they are identified. More importantly, further investigation is needed for the seamless integration of both levels for indexing purpose. To address these issues, we introduce ontology-based indexing for video surveillance system. Generally, video surveillance has non-predefined contents unlike movie or drama, and its main purposes are to detect and record abnormal events. Since ontology defines machine-interpretable definitions of basic concept in a domain, we can prevent semantic inconsistencies from human interpretations and apply it to handle surveillance data. Then, we can classify and index video data more precisely by using well-organized ontology. To illustrate the effectiveness of ontology based video indexing, we implement a Video Ontology System (VOS) to classify and index shots in a simple domain. Our experimental results show that the proposed approach is promising.

I. INTRODUCTION

A significant amount of multimedia data, such as image, audio and video, are being produced for many applications in the fields of education, entertainment, business and medicine due to the expansion of software and hardware technologies for data compression, storage devices and communication networks. This spurs the demand for research about content-based indexing of these multimedia data, specifically video data that is essential for effective browsing, retrieving, filtering, and summarizing. Indexing video data, however, is very complex due to the enormous size of video data and their semantically rich and unstructured format. Therefore, the first step of video indexing begins with segmenting each video into a set of basic processing units. In general, the most widely used basic unit is a shot that is defined as collections of frames recorded from a single camera operation.

Some of the existing video indexing models try to index these shots using low-level features [1, 2] such as color, motion, or objects. However, they can only provide very preliminary classification of videos and not support high-level complicated queries. The others are based on high-level features [3, 4] such as human interpretations by using meta-data and keywords. Still, this approach needs exhaustive manual operations, and may lead to semantically inconsistencies because of biased subjective content interpretations of different people. Recently, a hybrid approach combining these low and high-level features for indexing has been proposed to take advantages of both of them [5]. The approach shows that dealing with both level features has certain advantages. However, it is necessary that further improvements are still required for which features can be extracted, and how they are identified. More importantly, further investigation is needed for the seamless integration of both levels for indexing purpose. In order to address these issues, we introduce ontology-based indexing for video data. Specifically, we focus on video surveillance since contents of the data is not designed. Instead it includes mainly motions by objects.

There has been a great deal of interest in the development of ontology [6, 7, 8, 9] to facilitate knowledge sharing and database integration. In general, ontology is a set of terms of interest in a particular information domain, and shows the relationships among them. There are lots of works for using such a benefit of ontology: for example, document extracting [10], e-main classification [11, 12], photo annotation [13], and recognition [14]. Therefore, ontology-based video indexing can have the following advantages [15]. First, since it defines machine-interpretable definitions of basic concepts in a domain, we can prevent semantic inconsistencies from human interpretations. The semantic meaning in a video surveillance can be subjects, actions, places, and so on. Secondly, we can classify and index video data more precisely by using well-organized ontology. The indexed and classified data help a user query shots that include semantic meanings; for example, shots that include a man walking, or a car going out. The third benefit is an inter-operability between ontologies. In recent year, lots of ontologies have been developed in various areas. The ontology can be applied to index video data and help query semantic meaning.

To illustrate the effectiveness of ontology-based video indexing, we implement a framework called Video Ontology System (VOS) to classify and index shots in a sample domain. The proposed system can be summarized as follows. Video stream is the input data of VOS, and the input data is

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decomposed into a number of shots that are basic processing units containing meaningful data. The system consists of three components: *Video Segmentation, Meta-data Processing, and Ontology Processing modules*.

- *Video Segmentation*: This module decomposes incoming video into a number of shots using shot boundary detection technique that we have developed [16]. Then, we select candidate frames to extract key-object for each shot. The final output (key-objects) will be transferred to Ontology Processing module to find semantic meaning and index them. The decomposed segments are the basic processing unit of VOS that includes meaningful data.
- *Meta-data Processing*: Non-visual features which are called Meta-data, such as open and closed caption can be extracted and utilized to assist video indexing. Meta-data can be embedded into the video file like descriptor in MPEG-4 or MPEG-7. However, it is impossible for surveillance video to have pre-processed descriptors since it should be dealt with while it is generated.
- *Ontology Processing*: Ontology Processing module is a main component of the proposed indexing system. The results from the above two modules can be identified by matching the criteria of ontology. The characteristics of each data item are stored, and used for querying and retrieving [17,18,19].

Since this research aims at building ontology for video indexing, we implement only Video Segmentation and Ontology Processing modules. We will further combine Meta-data Processing to the system in near future. The remainder of this paper is organized as follows. Section 2 discusses the related works for shot boundary detection and object extraction to make the paper self-contained. In Section 3, we describe ontology for a set of video data, and show how to build it for indexing. In Section 4, we propose new techniques for class selection and ontology-based indexing. The experimental results are discussed in Section 5. Finally, we give our concluding remarks in Section 6.

II. Related Works

Video Segmentation phase in VOS is explained in this section. We first present our previous works [16, 20, 21] about Shot Boundary Detection (SBD) briefly to make this paper self-contained. We then discuss how to extract key object for each frame.

A. Shot Boundary Detection

To find shot boundaries, most of existing techniques compare two consecutive frames as seen in Figure 1 (a). As mentioned in our previous work [16], this approach is not effective since many video surveillance systems use stationary camera, there is no camera motion. Instead, each frame is compared with a background frame as shown in Figure 1 (b). A background frame is defined as a frame with only non-moving components.

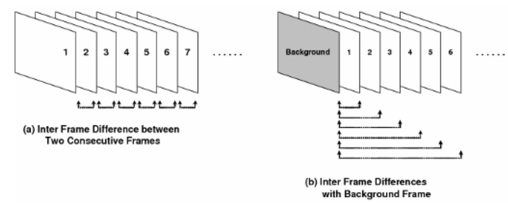

Figure 1. Frame comparison Strategies

Based on this idea, a background frame is first selected, and compared with each frame by using pixel matching. From each comparison, the difference between them is computed. A number of frames that have similar differences can be grouped into a shot by using the values of differences. An average of these differences of a shot is calculated, and then shots are classified into a number of categories according to these average values. The detail algorithm is summarized as follows. Step.1 is a preprocessing by off-line processing, and Step.2 through 5 are performed by real time processing. Note that the frame comparison can be done by any technique using color histogram, pixel-matching or edge change ratio, since this SBD algorithm is generic. We chose a simple pixel matching technique for illustration purpose.

Step.1: A background frame (F^B) is extracted from a given sequence as preprocessing, and its color space of each frame is quantized (i.e., from 256 to 64 or 32 colors) to reduce noises (false detection of motion which is not actually motion but detected as motion).

Step.2: Each frame (F^k) arriving to the system is also quantized in the same rate used to quantize the background in the previous step.

Step.3: Compare all the corresponding (same position of) pixels of two frames (background and each frame). Compute the difference (D^k) between the background (F^B) and each frame (F^k) as follows. Assume that the size of frame is c × r pixels. Note that the value of D^k is always between zero and one.

$$D^k = (Total\ \#\ of\ different\ color\ pixels) / (c \times r)$$

Step.4: Classify D^k into 10 different categories based on its value. Assign a corresponding category number (C_k) to the frame *k*. We use 10 categories for illustrated purpose, but this value can be changed properly according to the contents of video. We choose the number 10 for the illustration purpose only. The optimal number can be decided by the characteristics of video contents. We will further investigate what is an optimal number, and how to decide it in the future.

- Category 0 : $D^k < 0{:}1$
- Category 1 : $0{:}1 \le D^k < 0{:}2$
- Category 2 : $0{:}2 \le D^k < 0{:}3$
- Category 3 : $0{:}3 \le D^k < 0{:}4$
- Category 4 : $0{:}4 \le D^k < 0{:}5$

- Category 5 : $0{:}5 \le D^k < 0{:}6$
- Category 6 : $0{:}6 \le D^k < 0{:}7$
- Category 7 : $0{:}7 \le D^k < 0{:}8$
- Category 8 : $0{:}8 \le D^k < 0{:}9$
- Category 9 : $D^k \ge 0.9$

Step5: For real time on-line processing, a temporary table is maintained. To do this, we build a hierarchical structure from a sequence as mentioned in Section 1, compare C_k with C_{k-1}. In other words, compare the category number of current frame with the previous frame. We can build a hierarchical structure from a sequence based on these categories which are not independent from each other. For example, one shot A of Cat. #1 starts with Frame #a and ends with Frame #b, while the other shot B of Cat. #2 starts with Frame #c and ends with Frame #d, then it is possible that a ≤ c ≤ d ≤ b. In our hierarchical SBD, therefore, finding shot boundaries becomes finding category boundaries in which we find a starting frame (S_i) and an ending frame (E_i) for each category *i*. The following algorithm shows how to find these boundaries.

- If $C_{k-1} = C_k$, then no shot boundary occurs, so continue with the next frame.
- Else if $C_{k-1} < C_k$, then $S_{Ck} = k$, $S_{Ck-1} = k$, ... $S_{Ck-1} +1 = k$. The starting frames of category C_k through (S_{Ck-1} + 1) are *k*.
- Else, in other words, if $C_{k-1} > C_k$, then E_{Ck-1} = k - 1, E_{Ck-1} -1 = k - 1, ..., E_{Ck-1} +1 = k - 1. The ending frames of category C_{k-1} through (C_k + 1) are *k - 1*.
- If the length of a shot is less than a certain threshold value (β), we ignore this shot since it is too short to carry any semantic content. In general, this value β is one second. In other words, we assume that the minimum length of a shot is one second.

B. Key Object Extraction from Frames

To extract a key object from each frame, we use a technique developed in our previous work [21]. The technique needs only one additional step after Step.2 discussed above. First, we generate an empty image of which size is the same as a frame, and compare all the corresponding (same position of) pixels of two frames (background and each frame). If two pixels compared have the same color (value), then the corresponding pixel in the empty image becomes white. Otherwise, it becomes black. Then, we find a minimum rectangle which includes a key object in the frame. We call the rectangle as the Minimum Object Boundary Rectangle (MOBR) which is used to compute the characteristics of its object such as centroid, moving direction, and velocity of objects. More details will be discussed later.

Figure 2 shows the sample frames of some shots, and their key object extracted by the above technique. The rectangles in the images of the bottom row show the MOBRs.

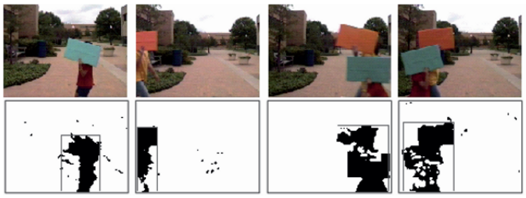

Figure 2. Examples of key object extracted from shots

III. Ontology

Ontology is a model of information in a given domain that can be used for the purposes of enterprise integration, database design, information retrieval, and information exchange via World Wide Web [6, 9]. Although the term, 'ontology' originates in the philosophical study of ontology, the usage of the term is widely spread in not only the artificial intelligent and knowledge representation communities, but most of information technology areas. In particular, ontology has become common on the Semantic Web in order to categorize large taxonomies on the web and integrate the data. The reasons for using ontology in information retrieval are mentioned in [6].

- Domain information can be sharable between human and machine.
- Domain knowledge can be reused.
- Domain knowledge can be analyzed and be more explicit.
- Domain knowledge can be separated and merged with other [22].

Ontology can be divided into two groups: generic and domain dependent. Generic ontology is not limited by a certain domain [23, 24]. On the other hand, domain-dependant ontology restricts its domain within a given area [25, 26, 27, 28]. Even if there is a restriction, we employ a domain-dependent ontology for our purpose because of the following reasons.

- Domain-dependant ontology provides a well-constructed concept that can be used to build meaningful higher level knowledge.
- The human related contents of video data can be identified and classified by well-constructed concepts and the ontology allows us to find relationship among them.

Basically, ontology can be represented as a set of classes. We describe basic elements of ontology such as class, slot, facets, and instance.

- **Class** is an abstract representation for a concept of domain. A class can have a subclass that represents concepts that are more specific than the superclass. For example, rule, student, and teacher can be classes for modeling of school.
- **Slot** is an attribute or properties of a class. For example, a class, student may have a name, identification, and degree as slots.

- **Facets** are properties of slots. For example, a slot of student class, a name should be characters.
- **Instance** is a concrete occurrence of information about a domain that is entered into a knowledge base. For example, John Smith might be an instance for a name of student class.

How to build an ontology based on above elements and its simple example are discussed in following subsections.

A. *Steps of Building Ontology*

We develop the ontology for video indexing in this paper. To do this, we adapt the process proposed by Natalya F. Noy et al. [6] with some proper modifications, which are necessary for making the process handle the video data. The general procedure is described as follows.

1. **Determine the domain of the ontology**: The starting point of the development of ontology is to define its domain. The videos that we are using were taken from a stationary camera set up in a hallway of university building with one or two object(s) of limited motions. For convenience, we call our videos as stationary videos since the camera was fixed.
2. **Define the class and the class hierarchy**: We define the classes for our domain such as 'Subjects', 'Actions', 'Places', and 'Fixed Objects'. To find class hierarchy we can use top-down, bottom-up, or their hybrid approaches. We choose the hybrid approach because of its flexibility.
3. **Define the slots (properties of class)**: We need to find properties to represent class. For example, name, identification number, and height can be the slots of a class, person. The properties inherit to the subclasses of a class.
4. **Generate instances**: Generating instances involves characterizing data items using the classes and the slots defined above.

B. *Simple Ontology for Video Indexing*

Based on the procedure discussed in the previous section, we build ontology for our surveillance video. The basic events include a subject, its motions, and a few fixed objects. The top-level ontology for our video is depicted in Figure 3. The classes of ontology consist of Subjects (moving objects), Actions (subject's actions), Places (a place of the camera shooting), Permanent Fixed Objects (wall, floor, ceiling, etc.), and Temporary Fixed Objects (desk, chair, etc.). The main goal of the ontology is detecting and identifying a subject and its actions.

The top-level classes of the ontology (see Figure 4 for details) are briefly explained in the following. In the figure, the classes with the superscripted Ms at the end have multiple superclasses.

1. **Subjects**: In general, a subject can be a car, a person, etc. as seen in Figure 4 (a). However, we are focusing on only persons in which our videos have them as subjects. This subject can be extracted from the technique that we discussed in Section 2.

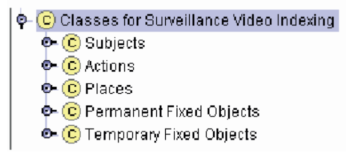

Figure 3. Top-level class of ontology for stationary video

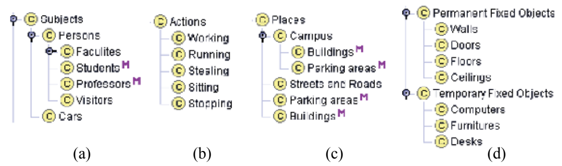

Figure 4. Classes of (a) Subjects (b) Actions (c) Places and (d) Permanent and Temporary Fixed Objects

2. **Actions**: This class indicates various actions of subjects (persons in our case) such as walking, running, and stopping as subclasses (See Figure 4 (b)). We detect these subclasses automatically.
3. **Places**: This class indicates the place where a camera is installed (See Figure 4 (c)).
4. **Fixed Objects:** We distinguish permanent and temporary fixed objects as shown in Figure 4 (d). Permanent fixed objects can be wall, floor, or ceiling. Temporary fixed objects can be desk, chair, or computer.

In this paper, we do not build the ontology for multimedia itself like video ontology [29, 30], audio ontology [31] and image ontology [32]. But, we will add multimedia ontology to the VOS in order to build more precise system.

IV. Shot Indexing Based on Ontology

In this section, we find the appropriate classes for incoming shots, and discuss how to index the shots using the corresponding classes.

Suppose that there are two simple videos: one clip has a person who is walking from right to left, and the other has the same person who is walking the opposite direction, from left to right. In this subsection, we distinguish these two videos by finding proper classes automatically. The detail procedure is described as follows:

Stage 1: As we discussed in Section 2, we partition incoming video stream into shots. Then, the key object is extracted as a MOBR (Minimum Object Boundary Rectangle) from each frame i of shot k, and represented O_i^k.

Stage 2: We compute the characteristics of frame i such as *centroid* (C_i^k), *velocity* (V_i^k), and *direction* (D_i^k).

- *Centroid* (C_i^k) is a point which indicates the center of a key object captured in Stage 1. Basically, x and y coordinate can be computed by averages of x and y values as seen in Equation (1). t_i is the total number of pixels in object region for frame i. c and r indicate the number of columns and rows of the frame.

$$C_i^k = (CX_i^k, CY_i^k) = (\sum_{p=1}^{c}\sum_{q=1}^{r}\frac{x_{pq}}{t_i}, \sum_{p=1}^{c}\sum_{q=1}^{r}\frac{y_{pq}}{t_i}) \quad (1)$$

- *Velocity* (V_i^k) is a criterion to distinguish actions such as walking, running, and stopping. The values of V_i^k are calculated for x and y directions by using Equation (2). Simply, V_i^k indicates the changing ratio of a centroid in x and y directions.

$$V_i^k = (VX_i^k, VY_i^k) = (\frac{CX_i^k - CX_1^k}{i}, \frac{CY_i^k - CY_1^k}{i}) \quad (2)$$

- *Direction* (D_i^k) shows that O_i^k moves from where to where in a shot. To decide the direction of object, we consider the centroid of frame i. If the centroid of initial frame (CX_1^k) is greater than that of the last frame (CX_i^k), the object (O_i^k) moves from right to left. Otherwise, the object moves from left to right. Similarly, we can figure out the directions of up and down.

Stage 3: To select proper classes for a shot, we use low-level features such as centroid, velocity and direction. For example, if the velocity of an object in a shot k (V^k) is greater than a certain threshold value, a class, *Running,* is assigned to the object. Otherwise, a class, *Walking,* is assigned. If V^k is zero, a class, *Stopping,* can be assigned. In this paper, we choose the threshold value as 5 pixels per frame based on our experiments. If more than one class is necessary for a shot, we split the shot into a number of sub-shots, and find a class for each sub-shot. The reason of shot split is that each action class has its own semantic meaning and the meaning can be applied to incoming shots. A split shot will be stored as an instance of class.

We index each shot using velocity and direction sequences generated above. The general procedure of indexing a new shot is described as follows.

1. When a new shot is detected from incoming video, the system creates a new empty instance.
2. The system fills up the empty fields of new instance with the class information extracted from the new shot using the steps discussed above.
3. All the information extracted from shots are stored as the ontology that we built. Actual shots are stored separately and linked to the corresponding information in our ontology.

V. Experimental Results

Our experiments were designed to assess the following performance issues:

- Performance of Class Selection

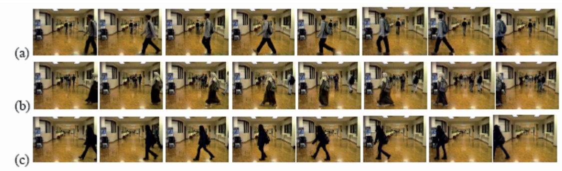

Figure 5. (a) Result Shot #47, (b) Result Shot #121, and (c) Result Shot #202 of a query, 'Shots containing persons who are walking right to left'

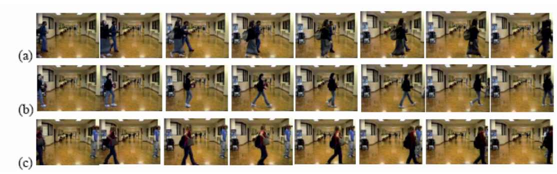

Figure 6. (a) Result Shot #32, (b) Result Shot #140, and (c) Result Shot #152 of a query, 'Shots containing persons who are walking left to right'

- Effectiveness of the Proposed Indexing

Our video clips were originally digitized in AVI format at 30 frames per second, and their resolution is 160×120 pixels. Our test set has 5 different video clips which have 256 shots. However, since 75 splits are necessary as discussed in Stage 3 in Section 4, there are actually 330 shots which consist of total 7010 frames.

A. Performance of Class Selection

The ontology was created with the Protégé ontology editor [33, 34], using JAVA as the interface language. Protégé is a knowledge base editing tool developed at Standford University. Protégé was also used for indexing and querying the shots. The implemented ontology has 29 classes and 34 properties. We use the correctness defined as the ratio of the number of correctly indexed shots over the total number of indexed shots (correctly or incorrectly) since 'recall' or 'precision' are not applicable to our case. 283 out of 330 shots are correctly indexed, which is 85.8 %. Most of the shots which are indexed incorrectly include multiple objects with multiple directions of motions, which we will investigate further.

B. Effectiveness of the Proposed Indexing

To provide more examples of the effectiveness of the proposed indexing techniques, we ran some queries using Protégé. For example, one of the queries is 'Shots containing persons who are walking right to left'. To use Protégé appropriately the above query needs to be changed such that 'The action class and its subclasses whose subject class is Person, direction start is right and direction stop is left'. The query result is the list of instances that are satisfied with all condition. Some frames of the shots included in the query results are presented in Figure 5. As seen in the figure, all the shots match the query, 'Shots containing persons who are walking right to left'.

Figure 6 is the results of another query, 'Shots containing persons who are walking left to right'. The query is translated to use Protégé such that ‘The action class and its subclasses whose subject class is Person, direction start is left and direction stop is right’. The query result is the list of instances like above example query.

VI. Concluding Remarks

We introduce in this paper two keys of ontology-based video indexing, which are how to build effective ontology and the detection of objects. All objects are identified, and their contents are extracted through the ontology processing. The main contributions of the proposed approaches are summarized as follows:

- We introduce and build a unique ontology for video indexing.
- We propose and develop an indexing technique using the ontology with semantic meaning.
- The proposed indexing approach based on the ontology can be extended into many different areas of multimedia.

Our experimental studies indicate that the proposed techniques are effective in indexing and querying video shots. But the current ontology is limited to stationary video shots. We will expand the proposed techniques to handle various kinds of videos. By using meta-data such as audio and object information, the system becomes a more complete solution. Recognition about what the object is and what the object is doing can be future works for ontology-based video indexing.

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An Adaptive Signal Processing Based on Wavelet Transforms

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Abstract: The wavelet transforms are integrated with transform domain LMS adaptive algorithm and variable step-size LMS adaptive algorithm, from which a new adaptive filtering algorithm is presented based on discrete wavelet transforms. The algorithm can reduce the self-correlation of input signals effectively and can overcome the conflict between high convergence rate and low steady state misadjustment in LMS algorithm which leads by fixed step-size. The simulation results indicate that the new algorithm has higher convergence rate and lower misadjustment noise than the traditional LMS algorithm; it can be applied effectively in the adaptive systems.

Keywords: Signal Processing; Wavelet Transforms; NDWT-LMS Algorithm

I. INTRODUCTION

LMS (Least Mean Square) has been a most extensive adopted algorithm in the approaches to adaptive filtering [1], thanks to its characteristics of simple calculation, less calculating amount and easy realization. However, its convergence rate is very sensitive to the distribution of the matrix eigenvalues of the self-correlation functions of input signals. For addressing the problem, wavelet transform (WT) has been introduced into adaptive filters [2-4].

The WT can be considered as an extension of the classic Fourier transform, except that, instead of working on a single scale (time or frequency), it can work on a multi-scale basis. This multi-scale feature of the WT allows the decomposition of a signal into a lot of scales, each scale representing a particular coarseness of the signal under study. The WT provides very general techniques which can be applied to many tasks in signal processing. One very important application is the ability to compute and manipulate data in compressed parameters which are often called features. The wavelet-based adaptive filtering conducts orthogonal transform of signals into adaptive filter. By taking the advantage of time-frequency local characteristic of wavelet, it can greatly decrease the distribution degree of the eigenvalues of the self-correlation functions of input signals and increase the iteration step-size, which in turn improves the convergence rate and stability of LMS algorithm [5-7].

In this paper, a new adaptive filtering algorithm based on discrete wavelet transforms is deduced. We propose a varying step-size LMS algorithm in the basis of power normalization processing [6] to acquire the optimal wavelet decomposition coefficients. Theoretical analysis and simulation results show that the new algorithm is more fast and stable.

II. ADAPTIVE FILTERS

Adaptive filters can be applied in a large variety of fields. The work process of adaptive filters is shown in Fig1. This is a typical block diagram of system identification using adaptive filtering. The objective is to change the coefficients of an FIR (Finite Impact Response) filter, W, to match as closely as possible the response of an unknown system, H. The unknown system and the adapting filter process the same input signal $x[n]$ and have outputs $d[n]$ (also referred to as the desired signal) and weighted signal $y[n]$.

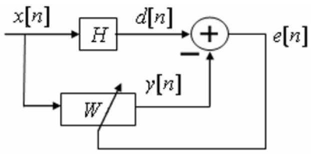

Fig1. A block diagram of system identification

LMS algorithm has been widely used to adapt the filter W. In which the error signal, $e[n]$, is first computed as $e[n] = d[n] - y[n]$, which measures the difference between the output of the adaptive filter and the output of the unknown system. On the

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basis of this measure, the adaptive filter will change its coefficients in an attempt to reduce the error.

III. A NEW ADAPTIVE FILTERING ALGORITHM BASED ON DISCRETE WAVELET TRANSFORMS

A. Discrete Wavelet Transforms Adaptive Filtering

The structure of discrete wavelet transform adaptive filter (DWTAF) is shown in Fig2.

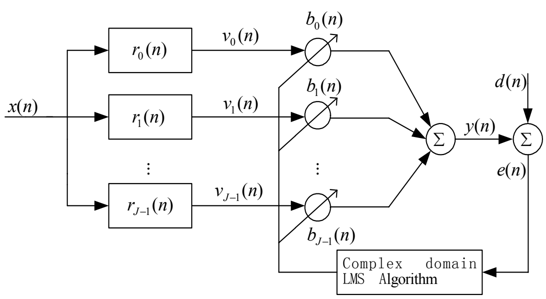

Fig2. The structure of discrete wavelet transform adaptive filter

According to wavelet transform theory, signal $x(n)$'s wavelet series reconstruction can be replaced by following finite sum.

$$x(n) = \sum_{j=0}^{J-1} \sum_{k \in Z} c_{j,k} \psi_{j,k}(n) \tag{1}$$

Formula (1) means to project the input signal $x(n)$ into corresponding scale's orthogonal subspace so as to recur signals in different distinguishing level.

To make $x_j(n) = \sum_{k \in Z} c_{j,k} \psi_{j,k}(n)$ is $x(n)$'s projective discrete form in wavelet subspace. The purpose of DWTAF is to produce the discrete reconstruction of $x_j(n)$ $(j = 0,1,\ldots,J-1)$. $v_j(n)$ (As is shown in fig.2) is the approximation of projection $x_j(n)$,

$$v_j(n) = \sum_{k \in Z} \hat{c}_{j,k} \psi_{j,k}(n) \tag{2}$$

Where, $\hat{c}_{j,k}$ is the discrete approximation of wavelet coefficient $c_{j,k}$

$$\hat{c}_{j,k} = \sum_{l} x(l) \overline{\psi}_{j,k}(l) \tag{3}$$

Formula (3) is put into formula (2), next result will be gained,

$$v_j(n) = \sum_{l} x(l) r_j(l,n) \tag{4}$$

Where, $r_j(l,n) = \sum_{k \in Z} \overline{\psi}_{j,k}(l) \psi_{j,k}(n)$.

Formula (4) is approximation formula of the $x_j(n)$. It is the discrete convolution form of input signal $x(n)$ and filter $r_j(l,n)$. These filters are consisted of wavelet $\psi(t)$'s inflation and convolution after sampling. They are band pass filters with constant bandwidth/center frequency ratio to realize the multi-distinguishing space reconstruction of input signals.

Under the supposition of time-steadiness and orthogonality, by reason of

$$\sum_{k \in Z} \overline{\psi}_{j,k}(l) \psi_{j,k}(n) = \sum_{k \in Z} \overline{\psi}_{j,k}(l-n) \psi_{j,k}(l-n)$$

Therefore

$$r_j(l,n) = r_j(l-n) \tag{5}$$

Sequentially

$$v_j(n) = \sum_{l} x(l) r_j(l-n) \tag{6}$$

The discussion above is about the signal demonstration of discrete wavelet transform. Next the discrete wavelet transform adaptive LMS algorithm is to be deduced [8].

If

$$\mathbf{V}(n) = [v_0(n), v_1(n), \ldots, v_{J-1}(n)]^T \tag{7}$$

$$\mathrm{X}(n) = [x(n), x(n-1), x(n-2), \ldots, x(n-N+1)]^T \tag{8}$$

$$[\mathbf{W}]_{jm} = r_j(m), j = 0,1,\Lambda\ J-1; m = 0,1,\ldots N-1 \tag{9}$$

$$\mathbf{B}(n) = [b_0(n), b_1(n), \ldots, b_{J-1}(n)]^T \tag{10}$$

then

$$\mathbf{V}(n)=\mathbf{W}\cdot X(n) \qquad (11)$$

Where, $\mathbf{W}$ is the wavelet transform matrix, its dimension is $J\times N$. $\mathbf{V}(n)$ is the input signal after passing through discrete transform filter. From the structure of DWTAF in fig.2, filter $r_j(n)$ $(j=0,1,\Lambda\ J-1)$ and multi-group delay line coefficient $b_j(n)$ constitute expected signal $d(n)$'s predictor. Such prediction is based on J continuous linear combination of input signal $x(n)$ [9]. The adaptive filtering output signals are,

$$y(n)=\mathbf{V}^T(n)\cdot\mathbf{B}(n)=\sum_{j=0}^{J-1}v_j(n)b_j(n)$$
$$=\sum_{j=0}^{J-1}\sum_{l}x(l)r_j(l-n)b_j(n)=\sum_{l}\alpha_l x(l) \qquad (12)$$

Where, $\alpha_l=\sum_{j=0}^{J-1}r_j(l-n)b_j(n)$

The adaptive filtering error signals are,

$$e(n)=d(n)-y(n) \qquad (13)$$

Coefficient update formula is,

$$\mathbf{B}(n+1)=\mathbf{B}(n)+2\mu\cdot e(n)\cdot\mathbf{V}(n) \qquad (14)$$

The algorithm convergence condition is

$$0<\mu<\frac{1}{\lambda_{V\max}} \qquad (15)$$

To make $\mathrm{R}_{\mathbf{VV}}$ is self-correlation matrix of $\mathbf{V}(n)$, $\mathbf{R}_{\mathbf{V}d}$ is cross correlation matrix of $\mathbf{V}(n)$ and expected signal $d(n)$. So $\lambda_{V\max}$ in formula (16) below is maximum eigenvalue of $\mathrm{R}_{\mathbf{VV}}$.

Formula (8) to (15) are discrete wavelet transform adaptive LMS algorithm (DWT-LMS).

B. Improvement to NDWT-LMS Algorithm

Solution I: The adoption of power normalization convergence step-size

The power normalization processing of transform domain algorithm [6] is introduced into DWT-LMS, A improved coefficient update formula NDWT-LMS(New Discrete Wavelet Transform -Least Mean Square) algorithm is proposed as:

$$\mathbf{B}(n+1)=\mathbf{B}(n)+2\mu\cdot e(n)\cdot\mathbf{D}_w^{-1}\mathbf{V}(n) \qquad (16)$$

where $\mathbf{D}_w=diag(\sigma_j^2)$ and its element (i,i) is the power estimation of *i*th DWT output $v_i(n)$.

This power estimation can be calculated by exponent average. That is,

$$\sigma_j^2(n+1)=\beta\sigma_j^2(n)+(1-\beta)\left|v_j(n)\right|^2 \qquad (17)$$

Where, $j=0,1,\ldots,N-1$, β is the forgetting factor which is a positive number close to but less than 1. It is between 0.99 and 0.999.

The updating formula of j th coefficient is,

$$b_j(n+1)=b_j(n)+\frac{2\mu}{\sigma_j^2}v_j(n)\cdot e(n) \qquad (18)$$

Then convergence condition is changed into, $0<\mu<1$.

Solution II: The adoption time-varying global convergence step-size

From Formula (18), the step-size value of j th coefficient is,

$$\mu_j(n)=\mu/\hat{\sigma}_j(n) \qquad (19)$$

Where, $\hat{\sigma}_j(n)$ is the maximum likehood estimation of the $\sigma_j(n)$.

From this formula, we can draw the conclusion that the adoption of power normalization helps to realize varying step-size algorithm. In addition, the power normalization technology only uses corresponding convergence step-size towards each coefficient, but fails to adopt any measures towards the convergence step-size of the whole algorithm. That is to say only $\mu_j(n)$ is time varying but μ is fixed. Thus we can suppose that if we can control μ effectively according to the principle of variable step-size LMS, then the better result will be gained.

Now the fixed global convergence step-size μ is changed into time-varying value $\mu(n)$ as

$$\mu(n) = \gamma\{1 - \exp[\alpha|e(n)|^m]\} \qquad (20)$$

where, parameter $\gamma > 0$ is to control the function scale. Reasonable augment of β can increase convergence rate if convergence conditions are satisfied. Parameter $\alpha < 0$ and parameter $m > 1$ are to control the shape of function curve. Reasonable decision of α value can decrease steady-state misadjustment. As the new introduced variable m, it should be selected according to practical application and computation amount, but never too big. That is because big m value not only adds to computation amount but also makes curve of $\mu(n) \sim e(n)$ come into the small slow varying area of $\mu(n)$ too early which will definitely ruin the convergence.

The function curves of variable step-size $\mu(i)$ and error $e(i)$ from formula (20) are shown in Fig.3.

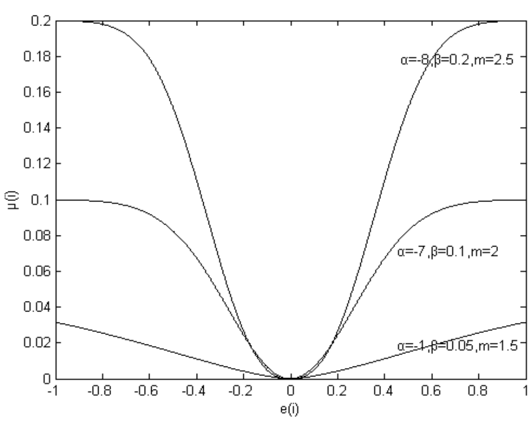

Fig.3. The function curves of $\mu(n) \sim e(n)$ from formula (20)

IV. SIMULATION RESULT

In order to verify the NDWT-LMS, we put it together with LMS algorithm into adaptive noise cancellation system and compare the two algorithms through computer simulation. All the experiments are conducted using MATLAB toolbox.

The input signals $d(n)$ in main channel are sine wave superposed signals. Parameter input signals $x(n)$ are related to main channel noises, which are set as strong correlation color noises. As we intend to compare the corresponding processing capabilities of the two algorithms. The step-size of standard LMS is selected as $\mu = 0.01$ to increase its convergence rate. The parameters of NDWT-LMS are ：

$$\alpha = -6, \gamma = 0.3, m = 15, \beta = 0.99$$

The rank of the filter is $N = 2$. Each algorithm is simulated 200 times independently to get the average. The result is shown in fig.4.

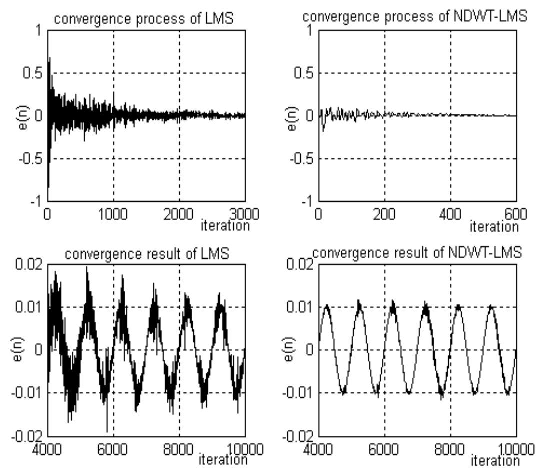

Fig.4 Simulation verification of LMS and NDWT-LMS

From fig.4, the results can be drawn that although the step-size of standard LMS is selected a bit big in order to increase the convergence rate, it does not converge gradually until the 3000 iterations while NDWT-LMS converges at about 200 iterations. It is obvious that its convergence rate is increased greatly compared to standard LMS algorithm. What's more, from the convergence result we can see that the steady-state error of NWDT-LMS is very small, the convergence wave shape looks good and the ratio of signal to noise is high. The simulation verifies the advantages of NWDT-LMS.

V. CONCLUSIONS

A new adaptive filtering algorithm based on discrete wavelet transform (NDWT-LMS) is put forward in this essay. It incorporates the advantages of wavelet transform, transform domain LMS adaptive algorithm together with variable step-size LMS adaptive algorithm, which helps maintain fast convergence rate and good steady-state effect even when input signals are highly correlated. The simulation results indicate that the new algorithm has higher convergence rate and lower misadjustment noise than LMS algorithm, it can be applied effectively in the adaptive systems.

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A Practical Approach for the Operational Level Monitoring of Executable Business Process Implemented by BPEL

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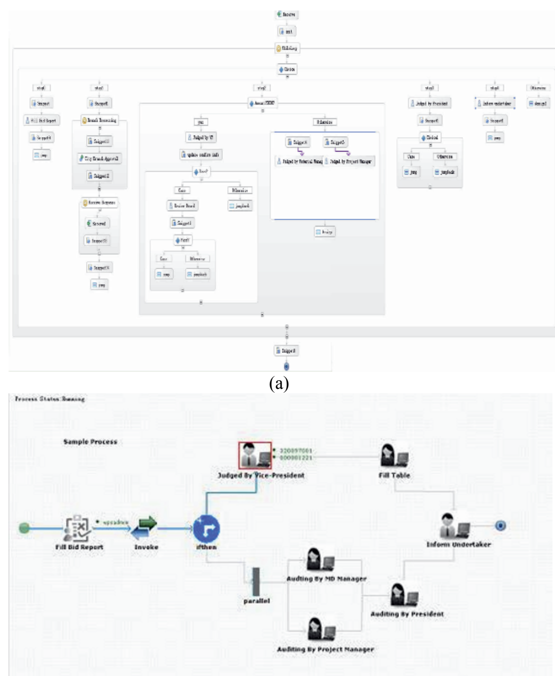
***Abstract*—Operational level monitoring of business process by business users is a key requirement for enterprises to improve business processes' visibility and efficiency. Nowadays more and more enterprise business processes are implemented by BPEL(Business Process Execution Language). Even though BPEL engine is often provided with monitoring capability, it is primarily designed to support problem determination and debugging for system administrators or developers. Business users can hardly leverage these capabilities to monitor the process at operational level. This paper introduces an approach to easily enable the operational level monitoring of business process implemented by BPEL. Operational level monitoring takes out execution status information of a process instance and renders them into an operational view for business user to track the process execution status and issue management commands. In this paper, an operational level process monitor model is developed to bridge the gap between the BPEL model and the process monitoring needs of business users, then tooling and engine extension architecture is recommended on top of a BPEL engine, finally a real customer case study is presented to illustrate the effectiveness of this proposed approach.**

I. Introduction

Operation level business process monitoring is for business end user to track and manage the business process executing in the Information Technology (IT) environment. As there are many terms being used in different context about "business process monitoring", we distinguish the term of "Business Process Monitoring" with "Business Process Measurement" and "Business Activity Monitoring (BAM)" in this paper according to Peter Kung's study [1]. Business Process Monitoring we referred in this paper is real-time monitoring of the process instance for tracking and recording purposes. Most process management systems provide the monitoring capabilities directly on the executable business process implementation, we say at the IT level. They are often provides with much IT execution information against the implemented executable process logic, which is hard for business user to understand. The Operational Level Business Process Monitoring (OLBPM) is targeted for business end users, who need to understand the status of process instance execution, audit process history, and generate reports with statistics against the operational process logic. In addition to tracking and recording, process intervention operations through the same process monitoring view are also often desired by business users.

The requirement of OLBPM often occurs at human involved business processes which are long running process that requires people to interact with process to get work done. The end users of the monitoring include business process participant (employee or manager), process owner and administrator. Intuitive graphic view of process tracking information is one of the distinct features of OLBPM solution. The process logic in the graphic view is at the operational level without many trivial IT implementation details, and has straightforward alignment with business logic. See figure1.

WS-BPEL (Web Service Business Process Execution Language) is the de facto industry standard for business process modeling and execution in the web service world [5]. It now has extension of BPEL4People [6] to support the



(a)

(b)

Fig.1. BPEL process (a) and corresponding Operational Level Business Process Monitoring view (b)

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human interaction in BPEL for human involved business process' automation based on web services. Actually before the specification was officially published in the middle of 2007, some commercial products have supported human tasks on their BPEL engines and Business Process Management System (BPMS) middleware. Lots of human intensive business processes have been developed and deployed on those platforms in SOA environment. The operational level monitoring on these processes becomes the key requirement in the BPMS solutions which has not been effectively tackled.

In this paper, we propose an approach to enable OLBPM on the executable business process implemented by BPEL. This practical approach aims to provide easy-to-use monitoring functions for end business user via bridging the Business-IT gap.

There are some papers exploring on the process monitoring work. Several concept designs of process monitoring are discussed in [1]. The operational level business process monitoring is not in the range. Josef Schiefer at al. proposed a process data store (PDS) as a data foundation providing real-time access to critical process performance indicators[2]. P .Chowdhary at al. present a Model Driven Development (MDD) framework and methodology to create IBM Business Performance Management [4]. In P. chowdhary's paper the Observe model of IBM Websphere Business Monitor was taken as the intermediate model for MDD framework. Rare of them thoroughly discuss the work about the operational level process monitor model and its generation.

Our contribution in this paper is the characterization of the operational level monitor model, showing an approach to enable the operational level business process monitoring, and the identification of the technologies and its architecture

The reminder of this paper is organized as follows. In section2, we present the recommended architecture. In section 3 and 4, we discuss the monitor model and its construction. After that, we'll introduce the monitoring engine. We also present a case study of the approach in section 6. Finally, in section 7 we present our conclusion

II. Overview of our Approach

We take a model driven approach for business process monitoring solution development and application. We have also developed a software component with modeling tool support on IBM business process management platform for

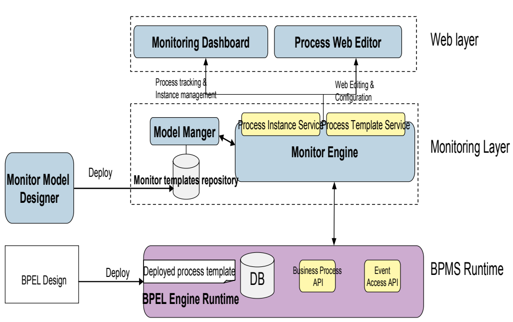

Fig.2. Architecture of our Operational Level Monitoring on BPEL Engine

the business-end-user-oriented business process monitoring approach. The architecture is shown in fig 2.

The core of this approach is the business process monitoring model. The monitor model can be firstly designed by developers via an eclipse based client tool, to establish the mapping relationship to the underlying BPEL process. After the monitor model is deployed to the runtime server, business users can modify and polish the initial monitor model on the BPMS runtime server via web browser. The monitor model will take effective immediately on end user's "Save" operation in the web client. It is a new lifecycle of monitor model that two stages of model design are involved.

The monitor model builds the correlation with the executable business process, and it is rendered to end user with a graphic view of the more understandable operational process logic. The details of model contents and its generation will be discussed in the next two sections.

Monitor model is stored and managed in runtime repository by a software component named Model Manger. A pooling cache can be used to improve the runtime performance by avoiding frequent and direct access of the monitor model repository.

The active monitor model is read by monitor engine which will attach real time instance data, and send the monitor model instance to the front web application.

The front web application is divided into two parts. One is the monitoring dashboard for business end users to track the process instance and to do some management work like sending notification, delegating tasks. The other part is the process Web editor for administrators to edit and modify the monitor model and related configuration.

It is a lightweight approach. For business end users, it is easy to use. End users can take the monitoring dashboard to have real-time intuitive and vivid operational level monitoring on processes, and utilize the web editor tool to customize and update monitor models directly online without redeployment. For solution providers, it has the least software products prerequisite besides the enterprises' already running BPMS. For developers, it is non-intrusive to the BPEL processes, requiring no modification and redeployment of them..

The distinctive feature of the approach is the monitor model and its construction process, through which the approach paves a way to bridge the gap between of the business operational level monitoring requirements and the underlining BPEL models, toward a more easy way to build monitoring solution.

III. Monitor Model

A. Contents and Concerns

There are two reasons that we need design another monitor model rather than take the BPEL model directly for monitoring. Firstly BPEL models is not sufficient for monitor model, additional parts besides process logic are needed for defining monitor requirements. We identify three types of information should be specified in a monitor model. Secondly, the operational level process logic can not directly derived from the BPEL process model as BPEL process developers should apply individual skills of modeling and programming to address the complex business process execution problems. The developed BPEL processes may have many variables or structures for control and without any obvious business meaning. Furthermore, according to their knowledge and experience, the skills they applied to model BPEL process may different. For example, some developers may use *While* plus *Switch* structure to express complex jump back logic of multi-level, department-crossing approval process. The condition expression of the *While* structure is complex and based on a variable. Each *Case* branch of the *Switch* structure then has an *Assign* activity to assign the value to the variable in the *While* condition according to the execution result and context for the next step routing. The complicated structure of the developed BPEL process, with much IT implementation details, is not intuitive for business end user to understand, not aligning with their conventional flowchart-like logic. Other developers may take use of *While* plus conditional *Link* in *Parallel Activity* structure to express the jump back logic, in which the conditions of the *Link* are very complex too.

In a word, BPEL process model is for execution, not for operational level monitoring. The model for the operational level business process monitoring should have the following three types of information:

1) Business process Logic and data

It is to define the monitoring data requirements (such as current status of human task, task status transition time) accompanying with its process logic. The process logic is necessary to be described in monitor model because it will help generate the mapping to BPEL and monitor view, making the automatic generation of them possible. The monitor process logic at the operational level is simple and easy to understand. The support on sequence, branching and loop can satisfy the most of customer requirements. The logic node in the monitor model is not for process execution control, but for monitor data acquisition and user understanding. BPEL model is not competent for this. The logic constructs of the model and their usage is not required to be rigorous and non-ambiguous.

Patterns of typical tracking requirements on different task type can be summarized to facilitate the monitor data definition. For instance, a staff task will be tracked with the status, status transition time, the potential task owner and the actual task owner. Some monitoring data are status related. For example, the data of the task owner is only available after the task created (triggered by a user's claim). It can refer [7] to define the status type of human task and related transition events.

2) Mapping to execution runtime

The mapping is for correlating the monitoring process logic and data to the underlying implementation so that the monitor engine can draw required data for the runtime process instance against the mapping. Because the logic expression method and power of the two process model is not the same, to express the same semantic logic, the two levels' process models may have different structure. For an activity in the operational level monitor logic model, there may be three cases for its mapping to the activity (s) in the underlying BPEL process. See table 1.

TABLE I
ACTIVITY MAPPING

Case	Example	Explanations
One-to -one	*Review* Activity in monitoring logic is mapped to one implemented human task named *Order Review* in BPEL	Generally a human task in monitoring logic is still implemented as the human task in the extended BPEL Process Model
One-to -many	An activity with label as: *Review the report by two managers simultaneously* in monitoring process logic need to be mapped to a parallel activity structure in the BPEL process	Activity node in monitoring logic is implemented as a process fragment of BPEL process model for detailed and complex processing
Many –to-one	Many levels *manager approval* will be implemented as one Approval Human Task in a While loop	It sometimes happens when the underlying BPEL activities included in a while structure. The example also shows the logic structures may be different of two models.

The most activity mapping between monitor process model and BPEL process is the one-to-one mapping. But the sporadic occurring one-to-many or many-to-one cases make the automatic mapping nearly impossible.

We will not discuss the complex mapping for logic nodes here. Logic node is not for routing control in monitor model. To real customer requirements, little monitoring data is required on it. We often ignore its mapping in our implementation, and let the monitor engine do some simple calculations on activities monitoring history data to achieve the process instance execution path.

3) Visualization Model

It is to define the graphic view of the monitor process model and the dynamic visualization feature for the monitoring process and data. It often needs to specify the color, highlight features according different process/node status. With contrast to BPEL process's visualization, dynamic is the key feature of the monitor visualization model. It should highlight the current running activity, color change on the executed path. It should be able to dynamic expand for the trail of the process which may be collapsed in while or sub-process logic (the sub process number may not be determined before runtime.) etc.

B. Design of monitor model

We combine the above contents into one model whose meta-model is shown in the fig 3. We integrate the mapping info into the process monitor logic rather than separate it an individual parts.

In its process model, a process consists of activities and links. The process and the activity both can have some monitor data. Monitor data can be tracking data which to record the running status and related time information, or be business objects such as purchase orders which are from BPEL process instance data. The monitor data have a property of pre-condition which is used to specify data acquisition condition. A process model is mapped to BPEL process model via BPEL process template id, whereas an activity mapped to BPEL activity via ActivityMapping. It allows one activity has multiple ActivityMappings. The logic node is one type of activity which is allowed to have no ActivityMapping.

The Visual Model is composed by visual elements. Each visual element has multiple PresentationMappings to set the visualization features, i.e. which presentation (such as text color, image, etc) applied to which monitor data (such as time, status etc). A visual element can be a node, a connection or an annotation. A node of visual element has reference to activity in the process model. The annotation is used for attached text or picture. We take DoJo Widgets to implement the visual elements which enable the extensibility and mash-up third party's service to render additional information.

When we design and construct the monitor model, we should have the two principles to ensure the monitor model be meaningful to business users.

1) Modeling the operational level process logic against the corresponding BPEL process should be in a simple and easy-to-understand way. The logic should be consistent to the BPEL process, but the structure is not required to be same. Each Activity should have its business meaning. No activity like *assign*, *java snippet* in BPEL process occurs at the monitoring model.
2) Monitor Data of the process or activity should have its business meaning. It is necessary to transform variable values into data with business semantics. For instance, to transform the user ID into its corresponding user name.

IV. MONITOR MODEL CONSTRUCTION

There are two roles in the model construction process. One is the IT developers who know the details of the executable BPEL process logic. The other is the business end user who knows the operational level business process monitoring requirements. There are two stages to build the monitor model. The work in stage one is IT-oriented, and can be supported by tools for the automation. The work in stage two is business-oriented. It lets end users to model their real monitoring requirements at the operational level process logic. The core of the process of the constructing the monitor model is to eliminate the difference between the two process models.

Stage one- initial model to establish mapping: The IT developer builds an initial monitor model via a client tool. The objective of this stage is to establish the initial mapping

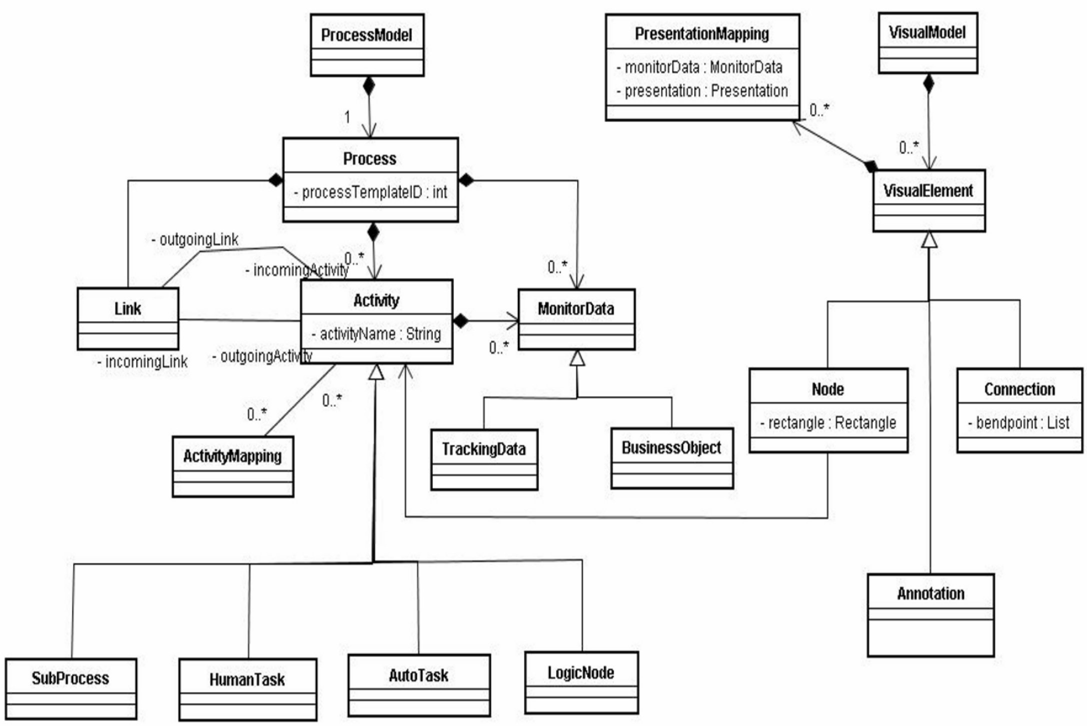

Fig. 3. The monitor model meta-model, schema level

relationship with the BPEL process. The resulted monitor model at this stage is not a final one. The customizations of the monitor requirements or operational level process logic view adjustments are left out to the next stage. Developers can build the monitor model according to corresponding BPEL process. Not all activities in BPEL process need monitoring and visible to end users. They can be ignored in the monitor model. But their branching information will be kept. To these reserved BPEL activities, the developer should create the activities in the monitor model, and create one-to-one activity mapping for them by specify the BPEL process template id and BPEL activity id. Different activity types (sub process, human task and auto task) would take different monitor data definition. After the step, an initial monitor model based on one-to-one mapping is established. Its process logic is the abstract BPEL process logic. We have implemented eclipse based tool to support the work, and also developed some components to assist the automatic BPEL mapping.

Stage two-end user editing: The model deployed by the IT developers is not the final one. It need end user's confirmation, polishing and/or some modification. Users can remove the mapped but not required tasks, add logic structures & connections between activities, and add annotation like notes to adjust the model when necessary. It should to be noted that the new added logic structures are treated as supplementary nodes, not mapped to the underlying BPEL process and will not rendered with monitoring data. Users can also custom the dynamic visualization feature of the monitor data. As the mappings established at stage 1 are one-to-one mapping, end users can adjust the mapping relationship at this step. For the case one-to-many, user can consolidate the original activities into one activity which is then with all the original AcitivityMappings, specify the activity type to apply the monitor data pattern and visualization pattern, customize the specific monitor data on the corresponding ActivityMapping. For the case many-to-one, users can copy the activity with the original one-to-one mapping to new activities which are with the same ActivityMapping, user then can add the links and some logic nodes for these nodes, customize the monitor data of each activity providing with the condition when the activityMapping can be applied.

Once the monitor model saved, it can be used by monitor engine to acquire data and rendering with user specified visualization features.

V. MONITOR ENGINE

The monitor model is the template for monitor engine to create instance model to acquire BPEL running data and render to the business end user the operational level monitoring view. Implementation of Business process monitoring may have different options on the data acquisition mechanism. One is to collect and analyze the data published by real-time business events, which is a common approach implemented with typical middleware products [3][8]. The advantage of the option is able to support the end-to-end business process monitoring, across the applications or process engines if there is more than one. It realizes the near real-time monitoring via propagating, transformation, and analyzing event data to monitor data (metrics or Key Performance Indicator called in some products). It often becomes an intrusive solution since it should add programming probes into the to-be-monitored process implementation to send required events. Several commercial middleware products have offered with the similar approach.

Another known option is to collect process audit data from process technical log [1]. It has a strong dependency to the process engine for the log file format. It also requires coding in the execution process model to log the needed monitoring data.

With contrast with the above two data acquisition mechanism, we take a non-intrusive approach that often seen in workflow monitoring to integrate with process engine to get process instance runtime data. We define a set of service provider interfaces (SPI) to regulate the specification that how we can obtain the process monitoring related data from underlying process engine. We implement the SPI for Websphere Process Server. Third parties can have their own implementation based on the process engine type. It should be paid attention on the configuration of the process engine to make sure the process historical execution data are kept or replicated. Some process engines often only keep process instance data during process execution, and clean the data after the process completed for performance or storage space consideration.

The monitor engine provides two types of service: process template and instance service. The process template service is provided to web editor for monitor model editing. The Stage 2 of the monitor model construction process will utilize the service to conduct.

The process instance service is to create the process instance model that will rendering to end user for a running BPEL process instance. To be noted, there is not any running monitoring instance triggered by the starting of a BPEL process. The entry could be a servlet request from the end user. The monitor engine will lookup the process monitor model via the necessary process template name, acquire monitoring data through SPI. The monitor engine should calculate the process execution path against the process logic of monitor model and give some validation. The calculation is based on the monitor data obtained. Here are two types of validation. One is for the task execution timestamp. The outgoing node of an activity's activation time should be no later than the activity. The other is for the process execution logic. We only check two types of logic nodes defined in process monitor model "Decision" and "Parallel". Only one path should be executed after the "Decision" while all of the activities should be executed after the "Parallel" node.

With monitor data, the process monitor instance model can be rendered as an intuitive graphic view which is has dynamic visualization features enable by web 2.0 technology.

VI. CASE STUDY

Our approach has been applied in several customer engagements. One of them is a leading China telecom company which has applied a Service Oriented Architecture (SOA) approach to operate their daily business and various kinds of business processes on a unified platform. There are nearly 100+ processes (human task intensive) running in its IT environment. It was found that there lacks of an overall process monitoring component to track the daily business and help make necessary decisions. The end users would not have much knowledge for the IT system. So we developed the OLBPM solution for the company. Human Tasks were treated as the milestone nodes in process monitoring, and in most process they are mapped to extended BPEL human task one to one.

The application is built on its BPM- platform Websphere Process Server. It has the least footprint on software products to build a lightweight and easy-to-use process monitoring application. We developed a tool for the automation of modeling work in stage one. The business partner who is the major role for the whole customer project development benefits from our approach to save much development effort for the 100+ processes' monitoring.

Web 2.0 technology enabled monitoring dashboard significantly improves user experience for business users. Further more, user can modify and update the monitor model with changes via a web based editing tool. As the customization can even be conducted by end user on the runtime, it greatly decreases the cycle time to build/customize monitoring features for business user.

The solution is implemented as an add-on component to Websphere Process Server without involving major architecture changes that reducing the enhancement cost to BPMS

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VII. CONCLUSION

Operational Level Business Process Monitoring is in great demand from business customers when they are deploying or have deployed executable business processes in IT environment. It is designed for business end user. The process logic shown in the intuitive monitoring graphic view is at the operational level, different with the underlying execution such as BPEL. It is a challenge to establish the correlation of the two levels' process logic automatically. In this paper we have proposed a lightweight approach practised on IBM BPEL to bridge the gap between the BPEL model and the monitoring needs of business users in a comparatively easy way. We define the monitor model to include three types of information, construct the monitor model from automatic BPEL process transformation and user interaction. This approach has been practised in real customer cases with proven major saving of monitoring development effort.

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Fuzzy SOFM and its Application to the Classification of Plant Communities in the Taihang Mountains of China

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Abstract

Fuzzy set theory and neural network are both attractive for ecological investigations for their powerful in analyzing and solving complicated and non-linear matters and for their freedom from restrictive assumptions. The combination of fuzzy set theory and neural network may produce better technique. This study tried to combine them in clustering analysis of plant communities in the Taihang Mountains in China. The dataset was consisted of importance values of 88 species in 68 samples of 10 m × 20 m. First, we calculated fuzzy similarity matrix of samples; second, the fuzzy similarity matrix was input to neural network; and then the self-organizing feature map (SOFM) was used to classified samples. We called this technique Fuzzy Self-organizing Feature Map (F-SOFM). The 68 samples were clustered into 8 groups, representing 8 vegetation formations. This classification result was reasonable and fully interpreted, which suggests that F-SOFM is effective method in ecological study. The F-SOFM shares both advantages of fuzzy set theory and neural network, and is applicable to all branches of science.

***Keywords**: Fuzzy relationship, neural network, clustering, quantitative method, plant community*

1. Introduction

Quantitative methods are very important and had been widely used in plant ecology since 1950s (Greig-Smith 1983, Zhang 2004). Numerical classification is one of the most common techniques in vegetation ecology. Vegetation ecologists have invented or adopted a wide variety of methods of classification in the past fifty years. Fuzzy set classification and self-organizing feature map are comparatively new in this field (Equihua 1990, Zhang 1985, Lek and Guegan 1999).

Fuzzy mathematics has been developed by numerous individuals, and is now applied in all the fields of sciences (Kaufmann 1975, Bezdek 1981). It is potentially useful in ecology because the description of ecological systems in not always possible in terms of a binary approach. Ecological communities have been shown to vary as their component species respond more or less independently to environmental gradients. Because of this, both overlapping and internal heterogeneities are important characteristics of ecological communities. The techniques based on fuzzy set theory might be more appropriate in ecology study (Boyce and Ellison 2001, Sarbu and Zwanziger 2001).

Self-Organizing Feature Map (SOFM) is based on artificial neural networks theory. Artificial neural networks (ANN) have also been successfully used in ecology (Lek and Guegan, 1999). Based on the mechanism of the human brain, the Artificial Neural Network is a convenient alternative tool to traditional statistical methods. The Kohonen Self-Organizing Feature Map (SOFM) is one of the most well-known neural network with unsupervised learning rules (Giraudel & Lek, 2001). The output of the SOFM is a low, typically two-dimensional, array in which similar samples are clustered together.

A combination of fuzzy set theory and SOFM may produce more reasonable result of classification of plant communities since they both showed advantages in research practices (e.g. Zhang 1985, 2003; Yang and Lu 1981, Equihua 1990; Zhang and Oxley 1994; Foody 1999; Bagan et al. 2005; Stuart et al. 2006). In this paper, the Fuzzy-SOFM clustering which based on both fuzzy set theory and artificial neural networks was described and applied to the study of plant communities in the Taihang Mountains, China.

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2. Methods

2.1. Fuzzy Self-organizing feature map

Fuzzy Self-Organizing Feature Map (F-SOFM) is similar to SOFM, but its input layer is consisted of fuzzy relationships between samples (Fig. 1). Therefore, first, F-SOFM needs to calculate fuzzy similarity matrix of samples (Zhang 2004), and then the neural network uses unsupervised learning and produces a topologically output that displays the similarity between the samples presented to it (Schalkoff, 1992; Foody 1999). The network consists of two layers, input layer and output layer (Fig. 1). The input layer contains a unit (neuron) for each variable in the fuzzy matrix. The input units operate in a similar way to those in other neural networks, effectively presenting the data for each sample to the network in an appropriate format. The input units are connected directly to units in the output layer or competitive layer. This output layer is also a two-dimensional array of units and each of these units is connected to every unit in the input layer by a weighted connection. Lateral interaction between units in the output layer also ensures that learning is a competitive process in which the network adapts to respond in different locations for inputs that differ (Yuan, 2000). Consequently, samples that are similar should be associated with units that are close together in the output layer while a dissimilar sample would be associated with a distant unit elsewhere in the output layer (Goodacre et al., 1994; Foody, 1999).

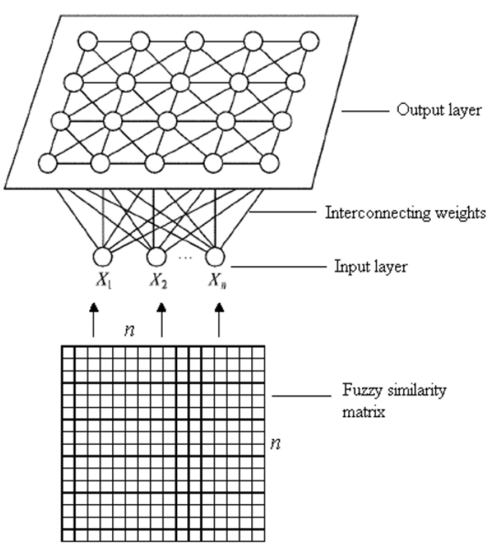

Fig. 1. The structure of fuzzy self-organizing feature map (SOFM) neural network.

2.2. F-SOFM clustering procedure

Starting with an ecological data matrix, S species observed in N samples, the F-SOFM clustering steps are:

Step 1: Calculating fuzzy relationships of samples. Fuzzy relationship refers to fuzzy similarity, and there are several formulas can be selected (Kaufmann 1975, Zhang 2004). We use correlation coefficient:

$$r_{jk} = \frac{\sum_{i=1}^{S}(x_{ij}-\overline{x}_j)(x_{ik}-\overline{x}_k)}{[\sum_{i=1}^{S}(x_{ij}-\overline{x}_j)^2(x_{ik}-\overline{x}_k)^2]^{1/2}} \qquad (1)$$

where r_{jk} is the fuzzy similarity between samples j and k; x_{ij} and x_{ik} are abundance or importance value of species i in samples j and k respectively; $\overline{x}_j$ and $\overline{x}_k$ are the mean abundance of all species in samples j and k respectively.

Step 2: Initializing. Suppose the input data vector: $P_k = (P_1^k, P_2^k, ..., P_N^k)$, $(k = 1, 2, …, q)$; The associated weight vector $W_{ij} = (W_{j1}, W_{j2}, …, W_{ji}, …, W_{jN})$, $i = 1, 2, …, N$; $j = 1, 2, …, M$.

Giving initial values of W_{ij} within [0，1] randomly ($i = 1, 2, …, N$; $j = 1, 2, …, M$), initial values of learning rate $\eta(0)$ and neighborhood $Ng(0)$, and determining total learning times T.

Step 3: Inputting a random sample unit drawn from the input dataset P_k into the network and calculating $\overline{P}_k$ and $\overline{w}_j$:

$$\overline{P}_k = \frac{P_k}{\|P_k\|} = \frac{(P_1^k, P_2^k, ... P_N^k)}{[(P_1^k)^2 + (P_2^k)^2 + ... + (P_N^k)^2]^{1/2}} \qquad (2)$$

$$\overline{W}_j = \frac{W_j}{\|W_j\|} = \frac{(W_{j1}, W_{j2}, ... W_{jN})}{[(W_{j1})^2 + (W_{j2})^2 + ... + (W_{jN})^2]^{1/2}} \qquad (3)$$

Step 4: Defining Euclidean distance between $\overline{w}_j$ and $\overline{P}_k$, and determining the minimum distance d_g, g is chosen as the winning neuron, called the Best Matching Unit (BMU):

$$d_j = [\sum_{i=1}^{N}(\overline{P_i^k} - \overline{W_{ji}})^2]^{1/2}, \quad (j = 1, 2, …, M) \qquad (4)$$

$$d_g = \min[d_j], j = 1, 2, …, M \qquad (5)$$

Step 5: Adjusting the weights (W_{ij})

$$\overline{W_{ji}(t+1)} = \overline{W_{ji}(t)} + \eta(t) \bullet [\overline{P_i^k} - \overline{W_{jt}(t)}]$$

($j = 1, 2, \ldots, M$) (6)

where $\eta(t)$ is the learning rate at t time, here we define it as follow:

$\eta(t) = \eta(0)(1 - t/T)$ $(0 < \eta(0)$ (7)

the neighborhood $N_g(t)$ is defined:

$N_g(t) = \mathrm{INT}\,[N_g(0)(1 - t/T)]$ (8)

$Ng(0)$ is initial value of $Ng(t)$.

Step 6: Increasing time t to t+1. If $t < T$ then go to step 2), else stop the training.

When the learning process is finished, a topology map of small squares is obtained and sample units can be mapped into the corresponding squares. According to this map, clustering of samples (communities) can be carried out (Yuan, 2000).

3. Vegetation data

3.1. Study area

The study area, located at 113° 06’ - 113° 48’ E, 36° 45’- 37° 17’ N, is the midst part of Taihang Mountains. The elevation varies between 1000 m and 2180 m. The climate of this area is warm temperate and semi humidity with continental characteristics and controlled by seasonal wind. The annual mean temperature is 7.3 °C, the monthly mean temperatures of January and July are -9.1°C and 21.6°C respectively. The annual mean precipitation is 560.3 mm and 63% of the precipitation falling in July to September within a year. Several soil types, such as cinnamon soil (brown soil), mountain cinnamon soil, umber forest soil and mountain meadow soil can be found in this area (Liu 1992, Zhang et al. 2006).

3.2. Sampling

Along the elevation gradient of 1 050 - 2 180 m, 12 transects separated by 100 m in altitude were set up. These transects were oriented along the contours. Four to six samples along each transect were established randomly and the number of samples for each transect depended on the length of contours. The sample size was 10 m × 20 m. The cover, height, basal area, individual number for trees, and the cover, height for shrubs and herbs were measured in each sample. The cover of plants was estimated by eye, and the heights were measured using a height-meter for trees and a ruler for shrubs and herbs. The basal diameters of trees were measured using callipers and from which basal areas were calculated. Altogether 88 plant species were recorded in 68 samples. Measurements were made of elevation (by an altimeter), slope and aspect (by a compass) for each sample (Zhang et al. 2006).

The Importance Values of species were used in F-SOFM analyses. The importance value was calculated (Zhang, 1995; 2004):

IV_{Tree} = Relative cover + Relative dominance (from the basal area measurements) + Relative height

$IV_{\text{Shrub and Herb}}$ = Relative cover + Relative height

The original data was a matrix of importance value of 88 species in 68 samples.

4. Results

Following the F-SOFM steps, the data matrix was analyzed. The output map of F-SOFM was chosen as a grid with 8 × 8 small squares. The learning rate was 0.1 for the ordinating phase and 0.02 for adjusting phase; the learning phase was broken down into 5000 steps for the ordinating phase and 50 000 steps for the tuning phase. At the end of the learning process, the topology-preserving map of 8 × 8 small squares with sample composition was obtained (Fig. 2). Based on Fig. 2, 68 samples were clustered into eight groups, representing eight vegetation formations. Their names and main composition are as follows:

I Form. *Rosa xanthina* + *Artemisia sacrorum*. Containing samples 57-59, 61-62, 64, 66-68. This type is distributed from 1050 to 1350 m with slop 5 - 15° and cinnamon soil. The common species in this formation are *Ziziphus jujuba, Scutellaria scordifolia, Lithospermum erythrorhizon, Adenopnora* sp.*, Aster tataricus, Artemisia capillaries, Angilica dahurica* and *Plantago asiatica.*

II Form. *Vitex negundo* var. *heterophylla* + *Ziziphus jujuba*. Containing samples 14, 44, 60, 63, 65. Distributed from 1400 to 1800 m with slop around 15° and mountain cinnamon soil. The common species are *Elsholtzia stauntoni, Rosa xanthina, Hippophae rhamnoides, Spiraea trilobata, Indigofera bungeana, Elsholtzia stauntoni, Carex lanceolata, Poa pratensis, Dendranthema chanetii, Cleistogenes serotina, Bothrichloa ischaemum* and *Saussurea nimborum.*

III Form. *Hippophae rhamnoides*+*Rosa xanthina*. Containing samples 42, 45-46, 50-52, 54-56. Distributed from 1700 to 1900 m with slop around 20° and brown forest soil. The common species are *Spiraea trilobata, Folium elaeagni, Lespedeza bicolor, Artemisia* spp., *Carex lanceolata, Poa pratensis, Bupleurum chinense* and *Potentilla chinensis.*

IV Form. *Spiraea pubescens* + *Vitex negundo* var. *heterophylla*. Containing samples 2, 5, 10, 37. Distributed from 1300 to 1700 m with slop 10 - 15° and cinnamon and mountain cinnamon soil. The common species are *Elsholtzia stauntoni, Indigofera bungeana, Lespedeza bicolor, Ziziphus jujuba, Cotoneaster multiflorus, Carex lanceolata, Poa pratensis, Dendranthema chanetii* and *Artemisia* spp.

V Form. *Quercus liaotungensis* - *Salix floderusii* + *Corylus heterophylla*. Containing samples 20-22, 47. Distributed from 1900 to 2100 m with slop around 30° and brown forest soil. The common species are *Corylus heterophylla, Spiraea trilobata, Artemisia sacrorum, Carex lanceolata, Polygonum viviparum, Poa pratensis, Dendranthema chanetii, Polygonum viviparum* and *Leontopodium leontopodioides.*

VI Form. *Rhamnus dahurica* + *Vitex negundo* var. *heterophylla* + *Elsholtzia stauntoni*. Containing samples 6, 15-16, 24-25, 28, 30-31, 36. Distributed from 1200 to 1700 m with slop 10 - 15° and cinnamon and mountain cinnamon soil. The common species are *Lespedeza bicolor, Ziziphus jujuba, Spiraea pubescens, Artemisia sacrorum, Artemisia lavandulaefolia, Carex lanceolata, Thalictrum petaloideum* and *Stipa bungeana.*

VII Form. *Spiraea trilobata* + *Rosa xanthina*. Containing samples 9, 17-18, 23, 26-27, 29, 32-35, 38-41, 43, 48, 53. Distributed from 1800 to 1900 m with slop around 20° and brown forest soil. The common species are *Vitex negundo* var. *heterophylla, Lespedeza bicolor, Artemisia* spp., *Carex lanceolata, Poa pratensis, Dendranthema chanetii, Sanguisorba officinalis, Polygonum viviparum* and *Fragaria orientalis.*

VIII Form. *Betula platyphylla* - *Spiraea trilobata* + *Rosa xanthina*. Containing samples 1, 3-4, 7-8, 11-13, 19, 49. Distributed from 1900 to 2000 m with slop 25 - 30°and brown forest soil. The common species are *Quercus liaotungensis, Pinus tabulaeformis, Folium elaeagni, Rhododendron micranthum, Abelia biflora, Carex lanceolata, Phlolmis umbrosa, Poa pratensis, Geranium wibfordii* and *Angilica dahurica.*

57 59 66	64	61 67	60	63 65	44	54 55	46 56
58 62 68				14			
			22		47	52	45 50 51
37	10		15	20	21		42
2 5		30 36		16	25		18 34 40
1		6		31	24 28	39	43 53
4 11 13		19					35 26 29
3 12		7 8 49		9 27 41	32 33 48		17 23 38

Fig. 2. Topology-preserving map of fuzzy self-organizing feature map (F-SOFM) of 68 samples of plant communities in the Taihang Mountains, China. The Arabic numbers refer to samples. The same color samples represent one vegetation formation.

5. Discussion

The application results of F-SOFM to plant community classification in the Taihang Mountains showed that the F-SOFM is fully usable in vegetation ecology, and it can perfectly complete community classification (Zhang & Oxley, 1994; Giraudel & Lek, 2001; Tasser and Tappeiner, 2002). It could cluster community samples into groups (formations), which reflecting ecological relationships between communities, species and environmental variables (Zhang 2004). The eight vegetation formations are representative of present vegetation in the midst of Taihang Mountains (Wu, 1980; Ma, 2001; Zhang, 2006). They are all secondary vegetation, following the destruction of the original broad-leaved deciduous forests (Zhang 2005). The variations of species composition, structure, environmental characteristics etc. were apparent among these formations, which indicated that the classification results of F-SOFM were reasonable and could be interpreted ecologically (Lek and Guegan 2000; Greig-Smith, 1983; Manly, 1994; Foody, 1999; Zhang, 2004). So ecologically speaking, F-SOFM clustering is an effective multivariate method in plant ecology (Orloci, 1978; Hill & Gauch, 1980; Gauch, 1982; Filippi and Jensen 2006; Zhang and Zhang 2007).

Theoretically, F-SOFM combines advantages of neural network and fuzzy set theory in solving non-linear problem and in studying complicated and fuzzy system (Greig-Smith 1983, Jongman et al. 1995; Schalkoff, 1992; Yuan, 2000). Vegetation ecosystem is a complexity system with various non-linear and fuzzy relations of species, communities and environmental factors (Podani, 1991; Zhang, 1994; Foody, 1999; Giraudel et al., 2000). Therefore, F-SOFM clustering should be perfect methodology in ecology study. The application of F-SOFM to plant community classification in the Taihang Mountains in this study provided a successful case comparing with former classification results (Ma 2001; Zhang ,2002, 2005).

Clustering is important in almost all scientific studies (Greig-Smith, 1983; Yuan, 2000; Wu & Huang, 2005). As a clustering technique, F-SOFM algorithm is applicable to all such studies. More case studies and applications of F-SOFM need to be carried out to prove its advantages (Robert 1986, Foody, 1999; Giraudel et al., 2000; Kaufmann 1975, Bezdek 1981).

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Dimension Reduction in Image Databases using the Logical Combinatorial Approach

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***Abstract*-The development of techniques that allow the representation space reduction is a topic which has become of special interest, since these allow us to work with less attributes and to obtain the same results. These techniques are used mainly for the treatment of big databases, in which an immense quantity of numeric data is managed, where the final objective is the data classification. This paper presents the results of the "typical testors" technique, for reducing dimensions, and the "k-means" technique, for carrying out the object classification. The main objective of making a reduction of the representation space with typical testors is to facilitate the classification, that is, the classification is a way to check the efficiency of typical testors.**

I. Introduction

The biggest existing problem when using n-dimensional data is the information redundancy, that is, in many cases different attributes describe the same object or group of objects. With the reduction of dimensions (attributes selection), we intend to describe the original information in a synthetic or summarized way. Simplicity is pursued based on obtaining a complexity reduction of the problem.

In general, the methods for reducing the number of attributes that are used for a data set representation consist on packing the original attributes in a smaller number of new ones still containing a great part of the original information, which are created by the analysis itself. This type of method is called *dimension reduction* since the original data is expressed in a space of n dimensions while as a result of the analysis we can express it in another space of dimension p, where $p < n$ [3].

One of the objectives of reducing dimensions in a group of objects is to be able to carry out the classification of these objects in a quicker and easier way. Classification is the first step for understanding complex phenomena. In some cases, the interest is on determining a group of classes as different as possible over the group of objects; this is the case of the cluster analysis. In other cases, the groups differ in a natural way and it is of interest to determine the characteristics that cause the existing separation among the groups, here we speak about the discriminate analysis [9].

II. N-dimensional analysis

The multivariate analysis or n-dimensional is "the branch of Statistics that studies the relationships between groups of dependent attributes and the individuals for which these attributes have been measured" [6].

A multidimensional-type data is one that has n-dimensions (attributes or characteristic) that differentiate it from other data or that are similar to data that share the same characteristics. From these

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 260–265.

attributes, certain techniques appear which are used for the analysis of this type of data.

These techniques try to visualize and simplify the data structure. In some cases, projecting given points in a space of smaller dimension is intended, keeping certain properties of the initial data group, especially those referred to the distances among individuals.

Among the statistical techniques that can be used for exploratory purposes, those of more utility are clearly the techniques that can be included in the multidimensional analysis, characterized by the combined study of a group of n attributes, each one of them measured on m objects [9].

In other cases, we try to make a base change. The original object observations are expressed in a base formed by the attributes that can or cannot be correlated. That is, in general it is not an orthogonal base. When transforming the initial group of attributes into a non-correlated group, a change of base is made to obtain a new orthogonal base.

Among this type of methods are the "principal component analysis", "independent component analysis", "curvilinear component analysis", "factorial analysis", and "typical testors", to mention some. In the same way that it is possible to recognize photographed three-dimensional objects within a picture, with these methods it is possible to recognize certain properties of the groups of original data.

For the case of study in this work, the typical testors' technique, which is a part of the logical combinatorial approach, was used.

With this technique, favorable results to similar problems have been obtained, as in [2][10][12], achieving an exceptional representation space reduction comparing to other techniques like ICA, PCA and more (depending on the analyzed data origin), as also shown by the authors in [9]. Because of the characteristics present in this research data, this technique was selected as a methodology for solving our case studies.

A. Testors Theory

The testors theory was formulated as one of the research areas of the Mathematical Cybernetics in the middle of the 60's, in the former Union of Soviet Socialist Republics (USSR). Its origin, in 1954-1955, is linked to the use of mathematical logical methods for the localization of damages in electric circuits [11].

In 1965, a new line of application for the testors theory opens up to solve the classic problem of Pattern Recognition [11].

According to the definition of Zhuravliov [11], a testor is a group of features (columns) that allows differentiating between two classes, because any object of class T0 cannot be confused with an object of class T1. A testor is called irreducible (typical) if no longer is testor for (T0, T1) when eliminating any of these columns.

The selection of attributes has two main uses:

- To reduce the number of features that should describe objects in an efficient way.
- To find the features that impact on the problem in a decisive manner.

The term "irreducible", reflects with clarity the idea that no more columns can be eliminated. The term "typical" refers to the sense of mathematical exemplification, and it reflects the fact that the combination of features that form a typical testor has, in a certain way, the same idea of the "tipicity" for a class of objects, that is, a group of features that in a certain way represent a class of objects and in another sense differentiate them from other classes [7].

An advantage of dimension reduction is that when applying classification techniques, they produce the same results that would be obtained before the reduction, but in less time and with less complexity.

III. Classification Techniques

The classification algorithms or *clustering* refers to the capacity of *clusters* creation, that is, classes or patterns. The only information that classification algorithms require is the previous definition of the vector of characteristics. Some of these algorithms need to know also the number of classes [8].

The classification techniques are used when there is no knowledge of the classes in which the objects of interest can be distributed, that is, where classes are not well-defined. These techniques are known as non-supervised or self-organized, for example: the chained distances algorithm, the Max-Min algorithm, etc.

Nevertheless, they are also of great practical interest in some areas in which there is a complete knowledge of the classes and therefore, supervised methods can be applied, such as: K-means and ISODATA, to mention some.

In this work the k-means classification technique was used, with the purpose of determining the importance of applying the typical testors technique to a database.

A. K-means Algorithm

This algorithm makes reference to the fact that there are k classes or patterns, being necessary, therefore, to know before hand the number of existing classes.

It is a simple and very efficient algorithm, as long as the number of classes is known previously with accuracy. This means that the algorithm is very sensitive to the k parameter. A value of k superior to the real number of classes will give place to fictitious classes, while an inferior k will produce less classes than the real ones [8].

The k-means method allows processing a limitless number of objects, but it only allows using a grouping method and requires previous knowledge of the class number we want to obtain.

The k-means method has shown to be very efficient on its results when carrying out classifications adapted in many practical applications. However, the k-means algorithm requires proportional time to the number of objects multiplied by the number of groups for iteration [1].

This method is given by a group of n points of data in a *d-dimensional* space of real type, R^d, and a k integer number, the problem is to determine the groups of k points in R^d, called *centroids*, to minimize the square average distance from each point to its nearest center [5].

IV. Implementation of Testors Theory

To develop this investigation the testors' theory was implemented. This technique is in charge of obtaining the smallest number of attributes describing an object without existing information loss. If it is important to have a concept adapted to the specific conditions of the problem to solve, it is still more important the fact of having an algorithm that allows us to efficiently calculate all the (typical) testors of a given matrix and for that purpose the algorithm denominated BT is used [7].

V. BT Algorithm for calculating Typical Testors

In general, we could say that finding all the (typical) testors of a given matrix is equivalent to carrying out an exhaustive search among all the subsets of features that are $2^{|R|}$, being R the group of features.

With the increase of the number of lines and columns or the number of classes in the matrix, the cost in time of this procedure can rise up to the point of being practically impossible. As a consequence, studies for searching efficient algorithms for the calculation of all the (typical) testors have been made. In fact, this problem itself constitutes a line of work in the Theory of Testors framework [7].

Next, the algorithm used in this implementation is shown:

1. The first non null list α of n length is generated.
2. (Condition (*)) is determined if the generated list is a testor list in the Basic matrix (MB).
3. If it is testor list, the *proposition 1.4** is applied. If it is typical testor list, α is printed and the jump proposition 2^{n-k}-1 is applied. If it is not testor list, the line of the MB that causes this fact is determined (if it is not the only one, the line with the last 1 more to the left is taken) and the corresponding proposition is applied.
4. The following list of discarded elements is generated by using step 3 and we return to step 2, only in case the resulting list of step 3 is not greater to (1,1,... ,1,1).

**Proposition 1.4.* Let α be a (typical) testor list and k the sub index of the last 1 in α then the following 2^{n-k}-1 n-arrays are testor lists but they are not typical.

Notice that before using the BT algorithm, the matrix of difference (MD) must be obtained as well as the basic matrix (BM), since that algorithm works with the last one [11].

VI. Application example

To evaluate the performance and functionality of the testors' theory a database of images was formed and analyzed, which contains 123 images represented by 21 attributes or characteristic of binary type. The data of the images are divided into three categories or classes.

The classes of the images are: SpringFlowers, images of summer flowers; The SwissMountains, air images of Swiss mountains; and YellowStone, images of the famous national park in the United States.

Some of the attributes used are: sky, flowers, bushes, water, mountains, snow, etc.

VII. Contribute of Results

In order to observe and analyze the results provided by the algorithm of typical testors, the previous example was used, which allowed us on one hand to prove the efficiency of the testors' theory and the reliability when using this type of techniques for the dimension reduction. On the other hand, the k-mean algorithm was used to carry out the data classification using only the attributes that the algorithm of typical testors considered as being the maximum attributes that would describe in a correct way the objects of the different classes.

In Fig. 1, the result of the system is shown using the database of images mentioned before.

After applying typical testors to the database of 21 attributes it was obtained that when using only five attributes the same results could be obtained, the attributes used were: 1, 2, 4, 12, and 13.

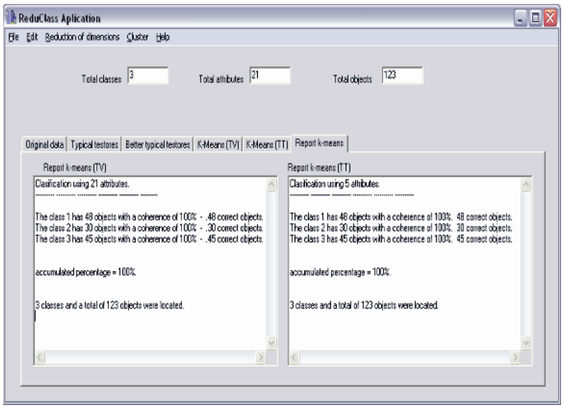

Fig. 1. Comparative screen of results in the classification of the database of 123 images.

Later, a comparative test was carried out with the "weka" tool [13], which received the same database and, as it can be observed in Fig. 2, the result of the classification is similar to ours.

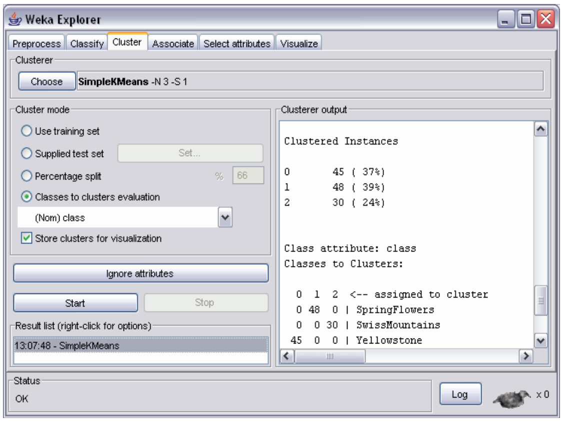

Fig. 2. Screen of result of the application "weka" in the classification of the database of 123 images.

As a last comparative case, the dimension reduction technique called "principal components analysis, PCA" was applied to the same database (technique that is integrated in the "weka" application), the results are shown in the Fig. 3.

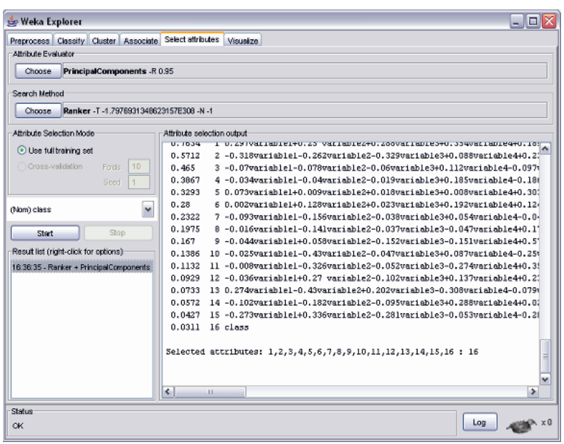

Fig. 3. Screen of results of the "weka" application when applying PCA to the database of 123 images.

As can be appreciated in Fig. 3, from the 21 attributes originally used in the database it is recommended to use 16 of them (from attributes 1 to 16). With this comparative case of dimension reduction techniques, we can suppose that the reduction carried out by "typical testors" is better than the one carried out for "PCA".

VIII. Future works

With the objective of extending this investigation the implementation of the technique called Component Curvilinear Analysis (CCA) [4] is proposed. Also, perform a comparative study of both techniques in dimension reduction to determine the advantage and disadvantage of each one of them.

IX. Conclusions

The interest for facilitating data handling for the information manipulation has become an hard task, in which a great number of investigations have been performed from which, several techniques have been derived which pursue the same goal. This is achieved by reducing the number of attributes that are used to carry out the description of an object. This method is known as dimension reduction.

The objective of this research is to prove the operation of some of these techniques. In the first part of this research project the testors theory was implemented, obtaining good results in the reduction of the representation space for the case of application that was used. This was proven when carrying out the classification of the objects with the k-means algorithm, which exhibited a negligible error in the test; however, the algorithm of typical testors is not responsible for this, since even when using all the attributes, the k-means algorithm shows the same results. Finally, the objective of this first stage of the investigation was completed: to demonstrate that the same results are obtained in the classification with all the attributes as with those used after the reduction of dimensions.

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Sensitivity Analysis of a Genetic Algorithm for a Competitive Facility Location Problem

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***Abstract* –The paper describes an application of a multi-objective evolutionary algorithm that finds Pareto-optimal solutions to a competitive facility location problem and reports a sensitivity analysis of the evolutionary algorithm model. A genetic algorithm which uses non-uniform mutation and continuous recombination is developed to find solutions to the bi-objective location problem. Results show that the genetic algorithm can find solutions close to the Pareto-optimal set in polynomial time and results from sensitivity analysis show that the mutation rate has the largest influence on the output among the input factors.**

I. Introduction

A location model is said to be competitive when the problem of locating a new facility in the market incorporates the existence of other facilities and that the new facility has to compete for its market share [1]. That is, Competitive Facility Location (CFL) models describe how facilities capture their market share and where a new facility should be located to maximize its market share. Most CFL models represent market share as a function of both location and quality of a facility, which leads to a bi-objective profit optimization problem of finding efficient solutions that maximize "captured" consumers, while minimizing the quality costs of the new facility.

The competitive location model involving bi-objective maximization of location and quality of a facility is due to [2]. They present a general profit maximizing competitive location model with different attraction types on a consumer and limit the location of the new facility within a bounded area. Plastria and Carrizosa [2] show that a maximal profit solution to the CFL problem through the determination of efficient solutions reduces to the bi-objective optimization of

$$\begin{aligned}&\text{Min } \alpha\\&\text{Max } CW(\alpha, \boldsymbol{x})\ ;\ \alpha \geq \alpha_0\ ,\ \boldsymbol{x} \in S\end{aligned}$$

where: α = unknown quality cost of the new facility at an unknown site $\boldsymbol{x}$
$CW(\alpha,\boldsymbol{x})$ = captured weight of the new facility
α_0 = minimal quality cost ($\alpha_0 > 0$)
$\boldsymbol{S}$ = closed set in the plane

A number of models exist to solve different competitive facility location problems. A survey of such models is discussed by [3]. The authors suggest fives major components in competitive facility location which are: space, number of players, pricing policy, rules of the game, and behavior of customers. The mostly used metric space in competitive facility location is the *m*-dimensional real space $\Re^m$ and, in most instances, is bounded by a convex polygon denoted by $|\Re^m|$. A pricing policy distinguishes between models with both price and location as decision variables and models that consider location as the only decision variable. Rules of the game define the general concept of equilibrium as applied to games but most competitive facility location studies in operations research do not fall under equilibrium models because competing facilities already exist. Such location models are called centroids and medianoids. The last component, behavior of customers, defines deterministic and probabilistic models. In a deterministic model, customers always patronize a single facility that they are most attracted to. On the other hand, probabilistic models assign probabilities of customer attraction to each facility.

Drezner [4] proposes a solution for the location of a new facility in a continuous planar space and in an environment where competing facilities have different levels of attraction on a consumer. Drezner's [4] algorithm solves a single-objective optimization problem that first calculates a "break-even" distance which is the maximum distance that a consumer is willing to travel to the new facility. Afterwards, the market share is computed for the new facility relative to the break-even distance. She concludes that her model guarantees a superior location based on information about consumer preferences and facility attributes. Plastria and Carrizosa [2] observe that most algorithms that solve competitive facility location problems either consider a fixed site and then attempt to maximize profit by an adequate choice of quality or consider the quality fixed and then find an optimal site. They show that profit maximization with respect to both location and the quality of the new facility may be obtained by solving efficient solutions (Pareto-optimal) in a bi-objective location problem.

The main objective of this paper is to report a sensitivity analysis of a multiobjective evolutionary algorithm (MOEA) as applied to finding solutions to the CFL problem. The paper is organized as follows. Section 2 introduces some basic concepts in multiobjective optimization and evolutionary algorithms. Section 3 presents the experimental design, a genetic algorithm for the multiobjective competitive facility location problem, and a computational example as applied to gravity-type attraction models. It shows a comparative analysis of results of the MOEA with [2] and a

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sensitivity analysis of the evolutionary algorithm. Section 4 presents the summary of the findings, conclusions, and directions for further study.

II. Multiobjective Evolutionary Algorithms

A. Multiobjective Optimization

Multi-objective optimization is concerned with the minimization of a vector of objectives $F(\boldsymbol{x})$ subject to a number of constraints or bounds and is defined as follows.

$$
\begin{aligned}
&\text{Minimize/Maximize} \quad F(\boldsymbol{x}) \\
&\text{subject to} \\
&\quad G_i(\boldsymbol{x}) = 0, \qquad i = 1, 2, \ldots, j \\
&\quad H_i(\boldsymbol{x}) \geq 0, \qquad i = 1, 2, \ldots, k \\
&\quad x_i^l \leq x_i \leq x_i^u \quad i = 1, 2, \ldots, m
\end{aligned}
$$

In the above definition, there are n objective functions $F(\boldsymbol{x})=(F_1(\boldsymbol{x}), F_2(\boldsymbol{x}),\ldots, F_n(\boldsymbol{x}))$, m decision variables, j equality and k inequality constraints. The term $G_i(\mathrm{x})$ refers to equality constraint functions and $H_i(\mathrm{x})$ refers to inequality constraints functions. The terms x_i^l and x_i^u are the lower and upper bounds that define the decision space and restrict the decision variable x_i to take a value within the decision space. If components of $F(\boldsymbol{x})$ are competing or conflicting, there is no unique solution to this problem. Instead, the concept of non-domination (also called Pareto optimality) must be used to characterize the objectives. A non-inferior or nondominated solution is one in which an improvement in one objective requires a degradation of another.

In a two-dimensional representation of an objective space and a feasible region $\boldsymbol{Z}$, the set of nondominated solutions (Pareto front) lies on the bold curve as shown in Fig. 1. The points **A** and **B** are solution points found in the Pareto front. In a minimization problem, solution **A** is better in terms of f_2 but solution **B** is better in terms of f_1 but it cannot be concluded that solution **A** is better than solution **B** in terms of both objective functions f_1 and f_2. Points A and B are referred to as Pareto-optimal solutions. In general, a solution $\boldsymbol{x} \in \boldsymbol{D}$ is Pareto-optimal if there is no other vector $\boldsymbol{x}' \in \boldsymbol{D}$ such that $f_i(\boldsymbol{x}') \leq f_i(\boldsymbol{x})$ for all i=1, …,m and $f_i(\boldsymbol{x}) < f_j(\boldsymbol{x}')$ for at least one j. $\boldsymbol{D}$ is a feasible region in the parameter space $\boldsymbol{x} \in \Re^n$.

Multiobjective optimization techniques that do no use evolutionary algorithms (EA) are termed traditional or classical in the sense that the algorithms combine multiple objectives into a single function, commonly known as aggregating functions. Different strategies are used for solving multiobjective optimization problems. On one hand, the decision-making is reduced to a single-objective function and scalar optimization is used to find the corresponding solution [5]. This approach often requires knowledge of the optimization problem and the optimal assignment of weights for each objective. Such approach entails repetition of the optimization procedure until a satisfactory solution is found.

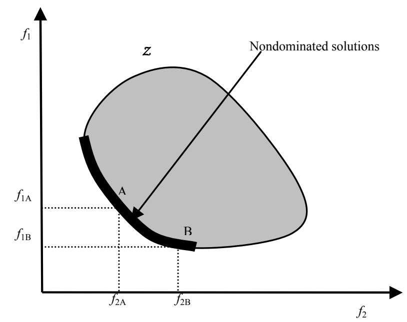

Fig. 1. Nondominated solutions (Pareto-optimal solutions)

On another hand, the decision-making is applied at the end of the optimization procedure which eventually eliminates repetition. Two strategies namely, weighted sums and ε-constraint are two of the mainstream aggregating functions used in multiobjective optimization [2].

There are two distinct goals in multiobjective optimization: i) determine solutions as close to the Pareto-optimal solutions as possible, and ii) find solutions as diverse as possible in the non-dominated front. These are somewhat conflicting goals. The first goal searches towards the Pareto-optimal region, whereas the second goal requires a search along the Pareto front. Hence, there is no single measure that can absolutely decide the performance of an algorithm.

B. Evolutionary Algorithms

Evolutionary algorithms represent a subset of generic population-based metaheuristic optimization algorithms in Artificial Intelligence. They are stochastic algorithms that make use of methods motivated by natural evolution such as, random variation, recombination, selection and competition of individuals in a population. Genetic algorithms, evolutionary programming, and evolution strategies are the mainstream computational models of EA whereas genetic programming and learning classifier systems are related evolutionary techniques.

Evolutionary algorithms work on a population of solutions and with the application of evolutionary or genetic operators. They produce improved approximations of solutions to a problem as generations progress. The candidate solutions in a population are referred to as chromosomes or individuals and depending on the EA model, individuals are coded as bit strings, real-valued vectors, trees, graphs, and matrices. A new set of chromosomes is created by selecting individuals according to their level of fitness in the problem domain. At each generation, the genetic operators lead to the creation of a population of new individuals that are fitter than the individuals from which they originated. The process of reproduction and selection continues until the algorithm satisfies an optimization criterion.

The advantages of an evolutionary algorithm in multi-objective optimization are i) it can generate a population of efficient solutions (nondominated solutions) in a single simulation run and ii) it eliminates the use of weighted parameters or aggregation functions. It has also become a preferred method for multiobjective optimization problems that are too complex for traditional techniques.

A complete discussion of multi-objective evolutionary algorithms (MOEA) can be found in [6]. Coello [5] also gives a summary of current approaches in MOEA and emphasizes the importance of new approaches in exploiting the capabilities of evolutionary algorithms in multi-objective optimization. A tutorial on evolutionary multi-objective optimization can be found in [7].

III. Experimental Design

The study uses the EA algorithm by [8], the Strength Pareto Evolutionary Algorithm (SPEA). Zitzler, Laumanns and Thiele [9] have shown that SPEA2, an improved version of SPEA, outperforms other evolutionary techniques and seems to have performance advantages for solving optimization problems with higher dimensional objective spaces. For such reasons, this paper employs SPEA2 to find nondominated solutions to the CFL problem as described in [2]. Bleuler, Zitzler, Laumanns and Thiele [10] provide a ready-to-use package which implements SPEA2 for multi-objective optimization problems. The public-domain package, Platform and Programming Language Independent Interface for Search Algorithms (PISA) can be found at http://tik.ee.ethz.ch/pisa/. The package makes it possible for researchers and practitioners to specify and implement representation-independent selection modules while allowing them to define problem-specific variation operators.

Sensitivity analysis is often defined as a local measure of the effect of a given input on a given output. This measure can be achieved most often by Monte Carlo methods in conjunction with a variety of sampling strategies [11]. Monte Carlo sensitivity analysis is based on performing multiple model evaluations with probabilistically selected model inputs, and the results of such evaluations are used to i) determine the uncertainty in model predictions and ii) identify the input variables that influence uncertainty.

SIMLAB [12] is a software designed for global uncertainty and sensitivity analysis based on Monte Carlo methods. It offers several techniques for sample generation, sensitivity analysis, and a link to external model execution. The link allows execution of complex models that can hardly be coded as simple mathematical functions such as genetic algorithms. Fig. 2 shows the schema of the external model execution in SIMLAB.

In general, a Monte Carlo sensitivity analysis involves five steps [12]. In the first step, a range and distribution are selected for each input variable (input factor). If the analysis is primarily of an exploratory nature, then quite rough distribution assumptions may be adequate. In the second step, a sample of points is generated from the distribution of the inputs specified in the first step. The result is a sequence of samples (input sample). In the third step, the model is fed with the samples and a set of model outputs is produced. In essence, these model evaluations create a mapping from the input space to the space of the results. This mapping is the basis for subsequent uncertainty and sensitivity analysis. In the fourth step, the results of model evaluations are used as the basis for uncertainty analysis. Uncertainty is characterized statistically by the mean value and the variance. In the fifth step, the results of model evaluations are used as the basis for sensitivity analysis.

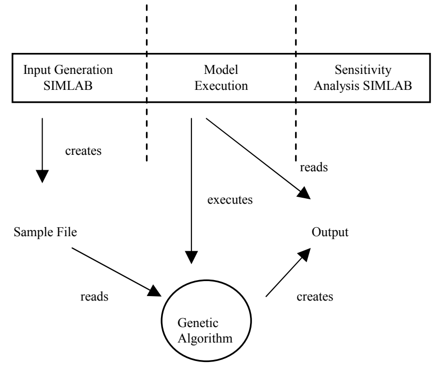

Fig 2. External model execution.

A. Genetic Algorithm Design

A chromosome ***c*** for the CFL problem is represented by real-parameters which define the location (*xy*-coordinates), the quality α, and captured weight *CW* of a new facility. Variation and selection operators are applied directly to these real parameter values. The main design problem is how to create a new pair of offspring vectors or how to mutate a facility location to a new location in a meaningful manner. The algorithm implements non-uniform mutation and arithmetic crossover (continuous recombination) although [2] argues that crossover does not always create meaningful offspring vectors in real-parameter EAs. Non-uniform mutation is generally defined as

$$x_i^{t+1} = x_i^t + p(x_i^l - x_i^u)h(t) \tag{1}$$

$$h(t) = 1 - r^{(1-t/t_{max})b} \tag{2}$$

where p is either -1 or 1 with probability of 0.5. The terms x_i^l and x_i^u are the lower and upper bounds that define the decision space and restrict the decision variable x_i to take a value within the decision space at generation t. The parameter r is a real random number in [0,1], t is the generation counter, and t_{max} is the maximum number of generations.. The parameter b is user-defined input which determines non-uniformity and has a value usually greater than 1. Equation (1) assures that mutation accomplishes uniform exploration at the first generations and search becomes local at the last generations. Equation (2) makes it possible to create a non-uniform mutation where the probability of creating solutions closer to the parent gets higher as the number of generations increase.

Recombination is implemented as the arithmetic average of both parents and is defined as

$$z_i = \frac{1}{2}(x_i + y_i) \tag{3}$$

where $\boldsymbol{x}$ and $\boldsymbol{y}$ are the parents, $\boldsymbol{z}$ is the offspring, i=1,…, m, and m is the number of dimensions. The offspring genes represent the arithmetic mean of the values of the parents.

The raw fitness function in this MOEA evaluates the real parameters α and $CW(\alpha, \boldsymbol{x})$ and is a multi-objective function that minimizes α and maximizes $CW(\alpha, \boldsymbol{x})$ as described in Section 1. SPEA2 first assigns a strength value $S(\boldsymbol{c})$, to each facility from the archive ($\bar{N}$) and population (N) representing the number of solutions $\boldsymbol{c}$ dominates. Then the raw fitness $R(\boldsymbol{c})$ of each facility $\boldsymbol{c}$ is calculated which measures the strength of $\boldsymbol{c}$'s dominators. The raw fitness acts as a niching mechanism but poorly performs when most paths in $M = N + \bar{N}$ are nondominated, i.e. the population forms new solutions in only a few clusters, in effect compromising exploration of the search space This phenomenon is called genetic drift. SPEA2 introduces a density estimator, a fitness sharing mechanism to avoid genetic drift. The density estimator is defined as the inverse of the distance of an individual in objective space to the k-th nearest neighbor. The density value is then added to the raw fitness value to give the final fitness function.

SPEA2 offers two selection procedures, environmental and mating selection. The environmental selection is concerned with choosing individuals that will have to move to the next generation archive $\bar{P}_{t+1}$ from the current archive $\bar{P}_t$ and population P_t. It is a form of an archive update operation. SPEA2 maintains an archive $\bar{P}_t$ in each generation composed of the "best" individuals with a fixed size $\bar{N}$ which is equal to the population size N. Two usual situations may occur. First, the number of nondominated solutions in $\bar{P}_{t+1}$ is less than $\bar{N}$. This case requires more individuals in $\bar{P}_{t+1}$ to make $|\bar{P}_{t+1}| = \bar{N}$. SPEA2 resolves this by adding the "best" dominated individuals from $\bar{P}_t + P_t$. Second, the number of nondominated solutions for the next generation is greater than $\bar{N}$. SPEA2 uses a truncation procedure whereby the individual with the minimum distance to another individual is truncated until $|\bar{P}_{t+1}| = \bar{N}$. On the other hand, mating deals with selecting parents from the archive population for recombination. SPEA2 implements binary tournament selection with replacement to fill in the mating pool. This type of mating selects two solutions at a time in each tournament. Their fitness values are evaluated and the better solution is placed in the mating pool. The option for replacement on the other hand, grants a solution a chance to compete in more than one tournament.

B. A Computational Example

The problem considered is taken from [2]. Specifically, a new facility must be located in the region $\boldsymbol{S}$ bounded by a convex polygon with corner points (0,0), (50,0), (50,20), (25,45), (0,45) A set of two facilities, $CF = \{xf_1, xf_2\}$ already exist and compete in the region. For each consumer a the gravity-type attraction to any facility has the form

$$A_a(\alpha, \boldsymbol{x}) = \frac{k_a \alpha}{\|\boldsymbol{x} - \boldsymbol{x}_a\|^p} \tag{4}$$

with each k_a = 1 which represents some proportionality constant depending on consumer a, some minimal quality $\alpha_0 \approx 0$, $\alpha_0 = 0.0000001$ and p = 2. Parameter p represents the sensitivity of attraction A to distance. Consumer a is captured by the existing facility yielding the highest attraction. The attraction, u_a, is determined by:

$$u_a = max\left\{\frac{\alpha_f}{\|\boldsymbol{x}_a - \boldsymbol{x}_f\|^2} \middle| f \in CF\right\} \tag{5}$$

An overview of the attraction parameters of ten potential consumers is presented in Table 1. The terms x_a and w_a represent the location and weight of each consumer a respectively.

C. Input Parameters

One difficulty in using evolutionary algorithms is its sensitivity to the input parameters such as: population size, random seed, number of generations, and probability values for crossover and mutation. Determining a balanced interaction between parameter settings to find efficient solutions is not an easy task. A number of simulation runs is usually required to come up with near optimal solutions. The experiment generated 384 samples with 5 input parameters using the Sobol method [12], which is also the method that generated the output for sensitivity analysis. The 5 input parameters for the generic algorithm model and their configuration are shown in Table 2.

TABLE 1
ATTRACTION PARAMETERS

Consumer	Location xa	Weight wa	Attraction ua	Facility F(a)
a1	(64,34)	600	0.6702	f2
a2	(60,19)	100	0.3702	f2
a3	(50,38)	100	0.9766	f2
a4	(45,55)	100	4.0000	f2
a5	(20,52)	400	2.8345	f1
a6	(27.8,7)	300	0.2830	f1
a7	(24,40)	100	1.1312	f1
a8	(20,31)	100	0.7086	f1
a9	(9,36)	100	0.8389	f1
a10	(3.8,7)	600	0.2707	f1

TABLE 2
INPUT PARAMETERS

Input Factor	Description	Probability Distribution
xyrange	Range for searching a location	Uniform(0, 10)
αrange	Range for searching quality	Uniform(0, 20)
mrate	Mutation probability	Uniform(0, 1)
xover	Crossover probability	Uniform(0, 1)
bitrate	Bit-turn probability	Uniform(0, 1)

The factor "xyrange" defines the upper and lower limit for searching a point in the polygon ***S***. The factor "αrange" defines the limits for searching the quality α of the new facility. "Mrate", "xover", "bitrate" are the mutation, recombination, and bit-turn probabilities respectively. The bit-turn probability is the probability that a gene will undergo mutation whereas mutation rate defines the probability that a chromosome or an individual will undergo mutation. The MOEA is run for 250 generations for each sample.

D. Interpretation of the Results

Table 3 shows the comparative results of captured weights and facility quality as computed from [2] and the best solution set from 384 different solution sets generated by the genetic algorithm. Plastria and Carrizosa's [2] results show the Pareto-optimal set ***P****. Clearly, the genetic algorithm was able to compute the exact captured weight *CW* of the new facilities but show slightly inferior solutions of quality values. The solutions of the genetic algorithm are near optimal because the CFLP is a real-parameter EA problem. Deb [6] explains that after a few generations, a real-parameter genetic algorithm treats a continuous search space problem as a discrete search space problem. The EA soon converges to intersections of different decision variables and the global optimum will be difficult to find. The performance of a solution set ***Q*** from the genetic algorithm is based on three parameters namely, i) the ratio of *CW* Pareto-optimal solutions in ***Q*** over the number of solutions in ***P****, ii) the ratio of *CW* solutions in ***Q*** that are members of ***P**** over the number of solutions in ***Q***, and iii) the sum of differences in quality values of ***Q*** and ***P****. The best solution ***Q**** shown in Table 3 detects all the *CW* solutions in ***P**** and the sum of the differences of quality values is 11.51 between ***Q**** and ***P****.

In terms of runtime complexity, the genetic algorithm takes $O(M^2)$ to evaluate the raw fitness of all candidate solutions per generation. The fitness assignment in SPEA2 per generation takes $O(M^2 logM)$ and the environmental selection per generation takes $O(M^2 logM)$ where M is the sum of the list of candidate solutions N and the archive size $\overline{N}$. Calculation of the quality α and captured weight CW per generation takes $O(nM)$ where n is the number of consumers. The runtime complexity of the genetic algorithm is dominated by its selection operator and is influenced by the initial population size N and not by the number of consumers n.

As regards to sensitivity analysis, Table 4 shows the Sobol first-order and total-order indices. The first-order sensitivity index shows the individual effect of an input factor on the output. More specifically a first-order index gives a measure of the direct effect of an input factor on the output variation. An input parameter having a first-order index with the least value means that it has the least influence on the output whereas a factor with the highest first-order value is most important for further investigation. If the sum of the first-order indices equal 1.0, then the model is linear. If the sum of the Sobol first-order indices does not equal 1.0 then the model is nonlinear and implies that some effects on the output are due to interactions among the input factors. First-order sensitivity measures do not capture interactions between factors whereas total-order sensitivity measures detect interactions among input factors.

The total-order indices describe the share of the output variation that is related to each input factor. This includes the direct effect as well as interactions with other factors. Removal of a factor means removal of the amount of the total-order index from the output variation. Hence, only factors with very small total-order indices can be removed to avoid significant changes in the output.

Table 4 has three columns which represent the performance metrics for an MOEA described previously. The values represent the first-order and total-order sensitivity indices. The mutation rate has the highest first-order index and causes nearly 30% of the variation in each of the output metrics. Similarly, the factors "xyrange" and the bit-turn probability share almost 60% of the variation on the third output metric-the sum of the differences in the quality α of a facility between solutions ***Q*** and ***P**** and can be interpreted as the proximity of an α solution in ***Q*** to ***P****. The parameters "αrange" and crossover probability have almost no direct influence on the output metrics.

TABLE 3
COMPARATIVE RESULTS

CW	Plastria (α)	MOEA Best Solution
600	0.0000	0.0003
900	39.8488	40.3606
1000	89.8289	90.4611
1100	135.2698	135.5118
1200	182.7161	182.7483
1300	359.5603	361.9230
1600	361.9952	362.3166
1800	440.4785	441.3086
1900	446.9055	447.3366
2000	566.0434	566.1086
2400	767.5907	768.5999
2500	1800.0000	1805.0749

TABLE 4
SOBOL FIRST-ORDER AND TOTAL-ORDER INDICES

Sobol First order indexes			
	Number of *CW* solutions in *Q* over |*P**|	Number of *CW* solutions in ***Q*** in *P** over |***Q***|	Sum of differences in quality
xyrange	0.046857	0.11457	0.298271
αrange	-0.0889	-0.0765	0.039282
mrate	0.285715	0.318398	0.292061
xover	-0.00165	0.082032	0.053537
bitrate	0.065333	-0.0698	0.207023
Sobol total order Indexes			
xyrange	0.601621	0.641919	0.407146
αrange	0.449627	0.264491	0.152863
mrate	0.611092	0.637363	0.528285
xover	0.199582	0.300252	0.137458
bitrate	0.404751	0.24301	0.372439

The sum of the first-order indices does not add up to 1.0 which means that there is interaction among the factors. The mutation rate factor still remains the highest in the total-order indices followed by "xyrange" and the bit-turn probability. The order of the remaining factors has changed implying that higher-order effects of these factors vary. The crossover probability factor shows small values and can be a candidate for removal from the list of parameters. The results also show that the mutation rate has the greatest influence on the output but what are the specific input values that generate near-optimal solutions? Further investigation reveals that out of the 384 input configurations 21 samples have generated good solutions i.e. they have detected all Pareto-optimal solutions for the function "captured weight", *CW* and have small differences in values for the function "facility quality", α. The 21 samples that generated near-optimal solutions show high values for the mutation rate, specifically an average of 0.75 and a standard deviation of 0.16. Other averages for the input factors are: bit-turn probability = 0.62, xyrange = 5.96, αrange = 6.46, and crossover probability = 0.40.

IV. Conclusion

This paper explores the performance of a MOEA in discovering a set of solutions to the competitive facility location problem and proved its feasibility in finding solutions that are near the Pareto-optimal front and are diverse along this front. In terms of the optimality of solutions, the MOEA is able to find Pareto-optimal solutions in the discrete-valued objective (captured weights) search space but shows inferior Pareto-optimal solutions to the real-valued objective (facility quality). In terms of runtime complexity, the MOEA runs in polynomial time but is computationally expensive in terms of repeated executions for sensitivity analysis.

The sensitivity analysis shows that the mutation rate causes much of the variation in the output and requires a high probability value in order to generate CFL solutions near the Pareto-optimal front. The crossover parameter on the other hand, has the least influence on the output and requires low probability values.

The interaction among parameters with high-valued indices is recommended for further investigation in order to improve the solutions in the real-valued objective. The repeated execution of the multiobjective evolutionary algorithm for sensitivity analysis is computationally expensive. A method that requires lesser input sample sizes will be beneficial in the investigation of sensitivity analyses of evolutionary algorithms as applied to multiobjective optimization problems.

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Automatic Collision Avoidance and Navigation

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***Abstract*— The main goal of this paper is to permit the growth of air traffic, whilst maintaining or increasing flight safety levels through automation. This should assist pilots and air traffic controllers and increase aircraft autonomy. MAGENTA is a PhD thesis, providing a framework for developing automatic learning tools. This framework has proven to be suitable for the aeronautic environment because of its adaptability to dynamic environments. One objective is to automate present aircraft collision avoidance systems. Collision avoidance refers here to a variety of possible occurrences, including other aircraft, geography, buildings, and so on. Another specific objective is optimize present automatic steering systems, by means of which aircraft can follow a specific or select the best route from a starting point to the head of the runway at the destination airport. These two objectives are in line with current *Free Flight* and *Free Route* plans in both the United States and Europe. This paper includes testing these automatic steering, navigation and collision avoidance capabilities with a flight simulator in a similar way to how pilots learn and are tested.**

I. Introduction

World air traffic is permanently on the increase. The density of this traffic is especially notable in Europe because of its population density and its economic level. Air traffic density and growth are especially intense in Spain on account of its expanding tourism industry and its geopolitical situation as a gateway from the continents of America and Africa to the rest of Europe. The European Reduced Vertical Separation Minimum (RVSM) and the USA En Route Congestion Project were approved with a view to meeting this need for higher operational effectiveness in the sky. This traffic density growth is permanently raising concern over air conflicts and collisions [3].

Some new features, and possibly a new philosophy, needs to be designed to maintain today's flight safety level in this growth scenario. A number of initiatives are trying to establish an automatic communications link between the controller and pilots. This communication, originally by voice only in a language which is sometimes neither the pilot's nor the controller's mother tongue, is a source of ambiguities and misunderstandings. Improvements like cockpit printers and displays and communication by keyboard via *Datalink* are moving in this direction [4]. Moving away from the traditional trend towards total and permanent control of all aircraft from the ground, new ideas are being developed to maintain the same levels of safety with higher aircraft autonomy by increasing automation and allowing more independence from the ground and more flight flexibility [8].

Free Flight was defined for the first time by the Radio Technical Commission for Aeronautics (RTCA) as "the capacity to operate in a safe and efficient way, under instrumental flight rules, where pilots are free to select the route and speed real time without need to contact ATC (controller)".The Free Route Airspace Project (FRAP) defined this concept for European countries [5].

Of these two trends, automation and delegation to the cockpit crew, the first one is clear and will lead towards a situation where the pilot will monitor automatic aircraft operation.

II. Magenta Approach

An alternative solution for collision avoidance is, as proposed in this paper, to do this automatically in case both the ATC and the pilot fail.

MAGENTA (General Framework for Learning Work) is a generalized framework for developing automatic learning processes, developed in a PhD thesis defended at the Technical University of Madrid's School of Computing in 2000. It was based on a previous USA patent by the same author *Pattern Association Central System and a Perception Learning Subsystem,* U.S. Patent Number 5,321,795 [2].

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This framework is based on reinforcement learning [6]. This way the agent modifies its action as it proceeds in the learning process. The agent first acts unintelligently with a very high proportion of errors. Later, after taking a near correct action in a specific circumstance, the agent gathers information on this action, as well as on the erroneous actions and their circumstances. The agent tries to repeat its successes and avoid its mistakes in the same or similar circumstances. The agent optimizes its performance by relating the best actions to the same or similar environments and circumstances and also by relating actions to be avoided to the same or similar circumstances [7].

A special MAGENTA algorithm implements the above with specific learning principles that specially support and speed up the learning process [1].

Mistakes and successes: provisional and final results. If we aim to build a framework that can be generalized, the information on the results has to give both positive and negative feedback and not wait for the end result.

Granularity: intensity of resulting values. One way of enriching the resulting data is to have the system know the intensity or extent of the resulting values: how positive or negative the successful or mistaken results were.

Measurement of time. When there is a time lapse between the action and the result, even if both happen repeatedly, there are other intermediate results that could be candidate effects for the combined cause (environment plus action). Taking into account these time lapses helps to establish associations that would not be detected if only the simultaneous occurrence of a perception, an action and a result were considered. Additionally, measuring such time lapses facilitates the statistical analysis and speeds up the learning association.

Back propagation of the results evaluation. The perceptions that do not have a positive or negative value of their own but are simultaneous or contiguous to others that do receive a corresponding back-propagated positive or negative value.

Multisensorial patterns. When groups of different and heterogeneous sensors are used, the input is much more easily identified and the action is therefore better.

Contiguous patterns. When there is a lapse of time between events, their elements may not be similar. A pattern comprising all these contiguous perceptions enables the identification of this reality. Additionally, there is the possibility of juggling with more actions, since there are more input combinations within one string. Each combination is potentially associated with a different action.

Direct non-learnt action. In the early stages, before the agent has learnt enough to survive, it should take some direct actions that would be triggered by some specific perceptions or stimuli.

Action probability. Actions should be taken in proportion to their probability of bringing about a successful result. The algorithm also has to include a proportion of exploratory actions that will be the ones with a higher statistical probability of success.

The algorithm has five main steps:

Step 1. A number of sensors perceive the environment to generate the above multisensorial perception pattern. This perception pattern is evaluated according to other positive and negative value sensors to output a mean value.

Step 2. An action is selected in the light of all the historical information about the positive and negative values associated with the perception-action pairs. This is done according to the probabilities of a more positive or less negative value being generated for the pairs. Nevertheless, exploration, i.e. selection of a different action than the one dictated by experience, takes place in the early stages to increase the experience level. Some direct actions, e.g. special action routines for getting out of difficult situations during the early learning period, are also taken in a few exceptional circumstances.

Step 3. The perception pattern and action pattern pairs are associated with the resulting value output during the action time cycle or later. This resulting value is associated with a decreasing weight depending on the number of cycles in which the result occurred versus the perception and versus the action, according to the formula

$$V(p_k, a_j)_i = V(p_k, a_j)_{i-1} + \alpha_{pa}^{k} \alpha_{av}^{j} v \qquad (1)$$

where V is the value of the association or pair, p is the perception pattern, a is the action pattern, i is the cycle in which the resulting value occurs and α is the discount value per cycle that is applied to reduce the resulting value to the historical value (the above measurement of time).

Step 4. Additionally, each perception pattern has a unique value that depends on the resulting value perceived later, its intensity and sign, and the number of cycles between the result and the perception. At the same time, the less certainty there is of this value being associated with the patterns, the bigger the part of the resulting value back-propagated towards previous perception patterns is. The formula for this is

$$V_{P_i} = \frac{c_{P_{i-1}} V_{P_{i-1}} + \frac{\alpha v}{2^j}}{c_{P_{i-1}} + \frac{\alpha v}{2^j}} \qquad (2)$$

where V_{Pi} is the value of the perception pattern in the present cycle, αv is the discounted value depending on the elapsed j cycles, c_{Pi-1} is the certainty of the perception pattern value in the previous cycle, V_{Pi-1} is the value of the perception pattern in the previous cycle.

The above certainty of such a value for the perception pattern depends on the actual resulting value and the number of cycles between the result and the perception (measurement of time). The formula for this is

$$c_{pi} = c_{p_{i-1}} + \frac{\alpha v}{2^j} \tag{3}$$

where c_{pi} is the certainty.

Step 5 of the algorithm is the deletion phase, when the superfluous information is deleted. This step deletes the perception patterns that are less frequent than a specific certainty factor, as well as the pair associations with a lower level of association than another constant. The perception pattern values that are higher than those factors are reduced by a specified amount.

III. Exercises/Tests

MAGENTA has demonstrated its adaptability to environments with 2D and 3D vehicles and specifically to aeronautics in a number of applications developed at the Technical University of Madrid's School of Computing.

The learning phase proceeds through a simulation of the agent/vehicle and its environment. When the learning curve has converged the system is ready for use either in a simulation or to be transferred into real life.

Some early applications of MAGENTA were tested on a 2D vehicle environment. The vehicles had to reach a target. The tests produced successful results even with moving obstacles interfering with the vehicles, with which they avoided collision after a short learning period of three hundred to four hundred trials.

To test MAGENTA's adaptability to an aeronautical environment, a 2D glide path was designed to emulate a triangle ending in the runway. The agent/aircraft learnt to head towards the start of the runway without moving out of the triangle. These results did not depend on the kind of triangle. This meant that the agent had taken into account the landing limitations, which are different for each kind of aircraft.

The same kind of aeronautic application was tested in a 3D environment. A special cone was designed with virtual walls. The walls define the maximum permitted altitude and lateral angles for approach. The aircraft can physically go through the cone's virtual walls, but the landing manoeuvre will not be correct unless it keeps inside. The start of the runway is at the end of this virtual cone. Although the glide path (e.g. for a jet) could be just a narrow tunnel, we used a path for an ultralight or very small and simple aircraft, which can approach the runway from a lateral angle. In this way, all possible errors of an unskilled / pre-learning agent or aircraft are crystal clear, since the space at the entrance of the virtual cone is bigger, and the learning process is easier to follow.

This 3D application is much more complex. There is a huge increase in the number of patterns, since they depend on the aircraft's position, orientation to the start of the runway and perception of the virtual cone walls. These complex patterns are multiplied combinatorily to produce hundreds of thousands. As an added difficulty, the agent/aircraft starts every trial from a different and randomly selected point and with a different and random orientation with respect to the start of the runway. It never knows where it is in the space. All the information that the agent/aircraft has is its orientation towards the runway and its proximity to the virtual cone walls –and this only when it is very close.

First comes a learning phase. This is, of course, a simulation phase, as it is equivalent to a pilot's first learning period. In the first trial, the aircraft makes a number of senseless movements. No limitations have been placed on these movements. Therefore, the number of possible actions is higher and mistaken actions are more likely. This makes learning more difficult, but more obvious and easy to follow when it takes place. If this were not a simulation, the first movements of the aircraft would send it crashing straight to the ground. When the agent starts to navigate with some equilibrium, we find that it does not move in any particular direction in a sort of fly anywhere navigation. Later it starts to head towards the start of the runway, but reaches the end of the glide path dangerously. This would in practice force it to abort the landing. Finally, the agent learns to navigate towards the runway, to keep to the right direction, to correct its trajectory when it is likely to move outside the virtual cone (either laterally or vertically) and to safely end at the start of the runway.

After the learning process, the successful landing levels are practically 100% and the optimum trajectory levels are 98%. The first successful landings take place after some 1,000 trials. Normally there are about 60,000 trials each with up to 600 cycles or independent actions/movements (Figures 1 and 2).

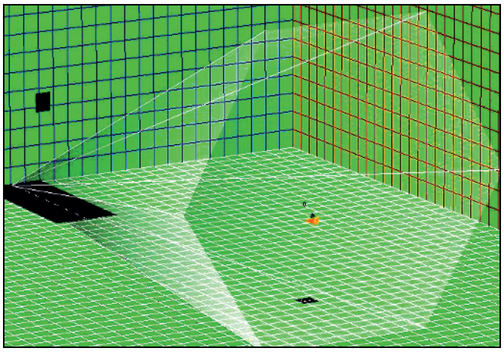

Figure 1

The aircraft does not penetrate the virtual cone walls because of the negative value or *error feeling* it gets when it does so. This reinforcement learning principle is used for collision avoidance with any other physical or virtual obstacle. Physical obstacles may be other aircraft, mountains, the ground, buildings and so on. Virtual obstacles may be the actual cone walls, prohibited national airspace, a military zone or an area with dangerous meteorological conditions. Therefore, we ran a number of different exercises with real and virtual obstacles.

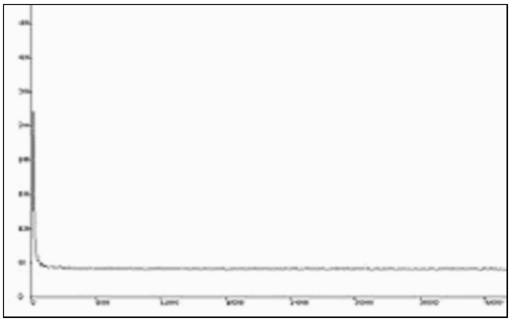

Figure 2

In the case of two or more aircraft, the system receives negative values when another aircraft approaches the agent. The algorithm is so effective, from a safety point of view, that there is never a collision even in the very early trials in the simulation.

To make it more difficult, two parallel aircraft were positioned to land on the same runway. This constitutes a permanent risk of collision. At the beginning of the learning simulation phase, there are of course negative values because they move too close to each other, as well as a lack of equilibrium, stability and deviations from the cone walls. The learning process leads to the ideal behaviour, where both aircraft keep together, proceed in parallel, optimize the two trajectories and glide paths and, finally, give each other priority. If more skills are demanded of the system and no landing priority is ever given to any aircraft, they have to reach agreement with each other and try to optimize their behaviour depending on their trajectories and positions.

Additionally, the process has been designed so that what one agent/aircraft learns is transmitted to the other for faster learning and better performance. In the most difficult exercise, the two aircraft approach the runway along two parallel trajectories, at the minimum permitted distance. They proceed; keeping this minimum distance and the due orientation until the aircraft furthest to the side is on the verge of moving out of the virtual cone. At this point the aircraft turns to correct its trajectory and is hit by the other aircraft, which goes straight on. Afterwards when the second aircraft is not allowed to hit the first one when it turns towards the centre of the virtual cone, we find that the second aircraft slows down to avoid flying at less than the minimum distance. In this skilled way they give way to each other when necessary, respect the minimum distances, establish their own landing priorities and land one after the other at the start of the runway. This exercise has been successfully run with aircraft starting points selected at random. The orientation of the aircraft was also randomly selected. Note that neither agent/aircraft knows the other's *intention*; even so, they never collide from very early on in the learning process. The learning curve converges at around 50,000 trials (Figure 3).

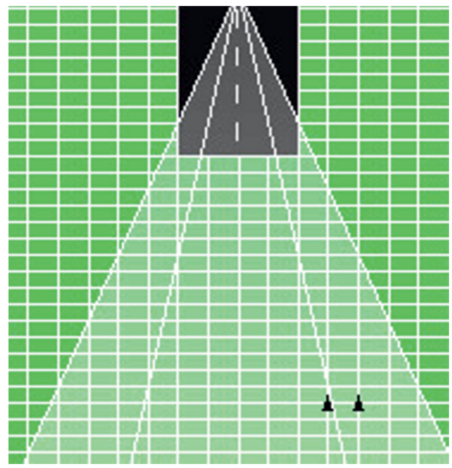

Figure 3

Other exercises have been run with real and virtual 3D obstacles of different shapes and sizes. A wide variety of ground, mountains, buildings, military areas, prohibited national airspace and other zones have been tried. The aircraft moves around the obstacle and steers again towards its previous target. The minimum distance to be kept from the obstacle can be changed by just modifying a parameter. Some special values for orientation and approaches and some special routines have been added for these exercises.

A very special obstacle exercise, which is difficult even for a human pilot, has been developed with a series of towers and buildings that the aircraft has to avoid. Several lines of towers have been positioned so that they can only be avoided by flying zigzag. Eventually the aircraft follows a zigzag flight path and goes around the building, avoiding collision with them all.

Later, an airport runway was added to the simulation environment. In this case, the aircraft has to go around the towers while descending towards the airport. The agent/aircraft moves successfully left and right around the towers and when it has successfully moved around the last tower, it steers again towards the start of the runway and finally lands (Figure 4).

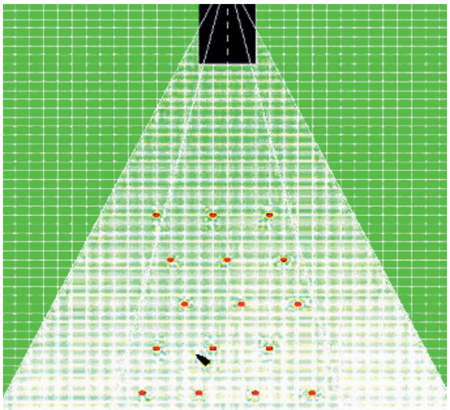

Figure 4

In all these exercises the trajectories are normalized. This means that the turns of just a few degrees are smooth. This is akin to what today's large commercial jets require. Additionally, the trajectories are optimized to a minimum within the physical possibilities of the aircraft.

Finally, the agent/aircraft is capable of navigating from one spatial point to any other. When a target is reached a new objective may be added and so on. This automatically provides a full navigation plan composed of a number of successive targets and a final objective.

IV. Future Plan

The exercises and tests that we have described here have been run on a simulator specially designed for this application. This software simulator logically accepts and receives digital data about the aircraft's environment and its action in this environment.

Most of the information that a pilot receives from and gives to the controllers and to the aircraft is also digitalized in the real aeronautic life nowadays. The MAGENTA software can use this digitalized input and output information. An input/output interface could therefore be designed for one of the real simulators used today by pilots for their own learning. In this way, MAGENTA software would take the pilot's place. The reactions of the software would be displayed on the screen as a real pilot's are today, and the learning process could be monitored.

Eventually, the fully automated navigation and collision avoidance system would be installable in the aircraft. We consider this to be a better place for the whole system for safety reasons, and mainly to achieve due reaction speed. This does not preclude the additional use of environmental information, which could be provided from ground ATC for larger areas than are detected by the aircraft sensors. It is assumed that in the future the system would pilot the aircraft, supervised by a human being acting as a *co-pilot*.

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Training Documentation Teams - A Research Plan

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***Abstract* – Technical documentation teams play a unique role in organizations as they help to build and create a public identity through end user manuals. In addition many technical documentation teams also provide information for the corporate website, as well as maintain intellectual knowledge through knowledge sharing and management. The technical communicator "*makes sense*" of complex engineering specifications by creating user-friendly manuals for the layman. The practitioner who compiles and records this complex information is a valuable resource to any organization. Therefore, on-going training for technical documentation teams is essential to stay competitive in the fast-paced technical market. This paper describes a research plan to study how organizations train their technical documentation teams.**

I. INTRODUCTION

The technical communication profession covers a broad spectrum of scientific and electronic technologies. The nature of this type of information is complex, and therefore is carefully documented for knowledge sharing with subject matter experts and the non-technical layperson. Although scientists and engineers create this complex information that impacts society and the world, the scope of their job does not entail communicating the information to the layperson. Practitioners in the field who specialize in communicating this complex information are referred to as technical writers or more recently technical communicators, exemplifying a combination of technical writing and design skills. The focus of this research study will spotlight those practitioners within technical documentation teams that create and design end user manuals for information technology (IT) organizations (i.e., computer electronics, software, and telecommunications).

Documentation teams compile and document fundamental information about an organization's product line, which is then made available for customers, clients, and the overall organization to view and critique. One of the main functions of a documentation team is to produce a comprehensive end user manual that describes a product's functionality and features in a clear and concise manner. The documentation team must have a general working knowledge of the organization's products and a broad understanding of technical terms of the desired industry in an effort to translate somewhat confusing jargon and complex terminology into plain, simple language for the non-technical user, as well as communicate on a higher level with the subject matter expert.

The average technical writer is either liberal arts-trained or has a strong technical background in electronics. A job posting for a technical writer often desires someone with an undergraduate discipline in Computer Science, but rarely does a degreed computer scientist find him or herself in the role of a technical writer. For the most part, the technical writer simply needs to have a good command of the English language, good verbal and written skills, solid analytical skills, a broad knowledge of the industry in which they work, and expert level proficiency in the personal computing environment using various office tools (i.e., word processing, graphics, project management, and spreadsheet software), desktop publishing, and documentation design software. Advanced and sought after technical writers are more valued if they can also use web-based authoring tools, such as HyperText Markup Language (HTML), Extensible Markup Language (XML), or Standard Generalized Markup Language (SGML).

II. THE PROBLEM AT HAND

There are many problems documentation teams experience that are both behavioral and structural in nature. This research study will focus on one structural problem that many documentation teams encounter. The technical communication profession is driven by technological advancements in software, presenting technical writers with new tools to create and design user manuals. The problem is twofold: 1) the documentation team's inability to keep pace with new software tools, and 2) insufficient training on

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new and existing tools that the team uses to perform their work.

The first component of the problem involves the inability to keep up with new software technology. Time constraints in an organization's project-driven development cycle are a major factor. Constant deadlines often restrict documentation teams from acquiring the necessary information on the latest software tools that could make their jobs easier and more efficient. In addition, work schedules compete with the technical writer's time to gather knowledge about how the new software tool works, how it might fit into the department's current way of electronic publishing, design, and knowledge management, how it is currently being used by others in the profession, and how the tool's features set it apart from and put it ahead of the competition.

The second component of the problem involves inadequate training on existing software tools. According to Grice and Krull (2001), technical communicators need to quickly learn new tools and drop existing ones in an effort to renew skills and keep pace with technology. Even though this may appear to be a simple matter of upgrading software, this can be quite a dilemma for technical communicators. Advancements in software technology have left many technical documentation teams in a quandary over what tools should be utilized within the department to unify the look and feel of the end user manuals, and yet not be so complicated to learn and use that it impinges upon the writer's ability to perform his or her job.

Veteran practitioners usually don't want or like to change from what they are comfortable with and have been using over a long period of time; often holding steadfast to the cliché, "if it ain't broke, don't fix it." In contrast, the more computer savvy practitioners feel frustrated and stifled at not being able to use the slick animation software and design tools for illustrations and layout or the latest timesaving tools to expedite the project. The technical publications manager is usually left with the difficult task of trying to determine a reasonable compromise, which can typically lead to discord among the team.

III. RESEARCH PROBLEM STATEMENT

The research problem under consideration is to understand what methodologies of team dynamics can be utilized to develop and implement on-going training and development to support technical documentation teams in IT organizations. In these researcher's career experiences as a software engineer, project manager, technical publication manager, senior technical writer, and contract technical writer-editor with various IT companies, the technical documentation team is vastly overlooked and undervalued. Because IT organizations don't fully understand how to best utilize a documentation team, the team often finds itself in the role of second-class citizen (Wilson & Ford, 2003). Documentation teams are usually under such time constraints to produce the end user manual that there is barely enough time to fully understand the complexity and functionality of the organization's new product deliverable, let alone become proficient on the very software tools that are supposed to make the overall writing and publishing tasks more efficient. Thus, the dichotomy of keeping abreast of new software tools and insufficient training on existing tools, has led to frustration and disharmony within documentation teams.

The researchers hypothesize that by using various methodologies in team dynamics and organizational learning, steps can be taken to: 1) motivate documentation teams to incorporate on-going training in their work schedule; 2) inspire the technical publication manager to be open to change, and foster an atmosphere of support and encouragement, so that team members can stay current with the demands of their profession; and, 3) enhance organizational learning by exposing stakeholders to the unique function and contribution that documentation teams bring to the organization. The researchers further hypothesize that the integration of on-going training for documentation teams in IT organizations will enhance individual competency and improve team performance, as well as influence new learning in IT organizations.

The purpose of this research study is to determine if the integration of on-going training in the work schedules of documentation teams would enhance individual competencies and improve team performance. In this regard, the researcher's main objective is to devise an intervention for documentation teams utilizing various mechanisms of team dynamics.

IV. DOCUMENTATION TEAM DYNAMICS

Advances in software technology have changed the face of documentation teams in IT organizations. These teams are no longer relegated to the singular task of writing user manuals. Today, documentation teams have expanded their role in IT organizations to encompass content writers, visual layout designers, usability testers, and knowledge management specialists (Grice & Krull, 2001). With the constant changes in software technology, practitioners are feeling the pressure to stay current in their field. However, in doing so technical writers are experiencing frustration, exhaustion, and burnout (Wilson & Ford, 2003). In a list serve (i.e., electronic chat) discussion group conducted by Wilson & Ford (2003), several technical communicators agreed that the desire to keep up with technological advances in software tools is difficult and leads to professional burnout sooner rather than later. Burnout or not, practitioners have a real concern about staying current with software technology because it defines their profession, and has a direct correlation with the writers' earning capacity and demand in the job market.

A study of 3,200 U.S. companies, conducted by Zemsky and Shaman, revealed that a 10 percent increase in spending for employee training and development lead to an 8.5 percent increase in productivity; whereas, a similar increase in capital expenditures lead to a 3.8 percent increase in productivity (Bennis, 1999). This study greatly emphasizes the importance of training and development for work teams. However, technical writers have little time or opportunity to sharpen or develop their skills (Gould, 1968). Grice & Krull (2001) suggest organizations support technical writers in learning new software tools with time and overhead cost dedicated to training and education. Technical communicators in the list serve discussion propose technical publication managers advocate on behalf of their documentation teams to communicate with and educate their direct managers on professional growth and the need to stay current (Wilson & Ford, 2003). Gould (1968) suggests that organizations employ in-house training programs, such as systematic training seminars, are a good approach to the problem.

Another contributing factor to the problem documentation teams experience with keeping pace with new software tools and insufficient training on existing tools is the ratio of engineers to writers in the organization. A typical documentation team in an IT organization can consist of anywhere from three to five individuals. Thus, it is not unusual for a documentation team to consist of only two individuals, depending on the size and structure of the organization. A web survey conducted by the Society of Technical Communicators Suncoast Chapter, a professional organization for technical communicators, found that the average ratio of technical writers to engineering developers is 1:20 for the IT industry (Johnson, 2006). This disparity in the ratio of engineers to technical writers can further propagate the feeling of being undervalued and overlooked amongst the documentation team. Obviously there is no need to have as many technical writers on a documentation team as engineering developers in an IT organization, but the workload generated by the engineering team can hinder opportunities for documentation team members to investigate new tools that might help the team accomplish their work more efficiently. It is no wonder why many technical writers experience frustration and burnout as they try to keep pace with the latest advancements in software tools for their trade.

V. STUDY DESIGN

Keeping pace with new software design tools and insufficient training on existing tools has led to frustration and disharmony within documentation teams in IT organizations. It is the intention of the researchers to show how the integration of on-going training for documentation teams will enhance individual competency and improve team performance, as well as influence organizational learning.

The researchers will utilize mechanisms of team dynamics and various research methodologies to collect data from three documentation teams in three different IT organizations. The planned research approach will be managed in five phases, as outlined in the following text.

First Phase – Interview Method (part 1)

The objective in phase one is to make initial contact with each organization and collect general demographic data (estimated time: 20 minutes). The researchers will conduct an interview via telephone with each documentation team manager to obtain demographic information about the teams and their respective organizations. The researchers also plan to set up site visits with each organization for the second phase of this project.

During the telephone interviews, the researchers will determine each documentation team's ability to participate in a two-week training study during the fourth phase of this project, as well as each documentation team's availability to participate in a focus group or online interactive group discussion during the fifth phase of this project.

The specific objective in this phase is to introduce the researchers and documentation team manager, explain the purpose of the research project, and gather enough data to establish criteria for the next phase of the intervention design.

Second Phase – Interview Method (part 2)

The objective in the second phase is to build relationships with each documentation team and collect data (estimated time: 1 hour). The researchers will conduct face-to-face interviews with each documentation team manager. There are several goals to accomplish in this phase: 1) visit each work site, 2) clarify the nature of the research project, 3) clarify the level of the organization's participation, 4) explain to what end the research will be used, 5) obtain permission and consent, 6) meet all team members, 7) observe teams in their environment, 8) observe the interaction with other work teams, and 9) observe the culture of each organization (i.e., artifacts, espoused values, and organizational climate).

This phase will also give each documentation team manager and the researchers an opportunity to discover and learn from one another using open-ended questions during the interview. The researchers hope to discover where the documentation team fits into the organizational reporting structure, the number of engineers in comparison to the number of technical writers, and the physical location of the documentation team in relation to other work teams.

Third Phase – Survey Method

The objective in this phase is to identify emerging themes (estimated time: 20 minutes). The researchers will conduct a survey of approximately 25 questions of various types, including multiple choice, ratings, and short answers. The survey topics will be divided into three parts to help the researchers assess the current state of each team, assess individual skills with industry standard software design tools, and assess individual training needs. The participants will consist of documentation writers in all three organizations. The researchers estimate anywhere from 10 to 20 participants will take part in the survey. Again, the objective in this phase is to identify emerging themes that will be further explored in the fifth and final phase of the intervention design.

Fourth Phase – Training Assessment Study

The objective in this phase is to measure changes in individual competencies and team performance when documentation teams integrate training into their daily work schedules (estimated time: Up to one hour per day for two weeks). The researchers will design an integrated training assessment and distribute it to each of the three documentation teams. The training will vary, and for the most part consist of participants determining what tool or area they need improvement or additional learning in, and spend time reading or practicing to develop that skill. Participants will be given a set of guidelines and log sheets, in which to record individual progress on a daily basis. Individuals will work at their own pace as they execute the training over a two-week or 10-day period. Depending on the outcome of the initial telephone interview with each documentation team manager from the first phase, the researcher will determine if all three teams will be administered the same training for this study, or if each team will be given a variation of the integrated training assessment. The desired goal is to measure changes in individual competencies and team performance when documentation teams integrate training into their daily work schedules.

Fifth Phase – Focus Group Method

The objective in this phase is to clarify themes (estimated time: one hour). Based on schedules and proximity of each organization, the researchers will determine if an online interactive discussion (i.e., chat room) or teleconference discussion is more feasible to administer than a face-to-face focus group discussion. The researchers intend to create a list of questions generated as a result of the data gathered from the survey, integrated training assessment study, and onsite interviews to stimulate a focus group discussion either online or via teleconference with members from all three organizations. The main objective of this phase is to clarify and validate themes stemming from previous phases, as well as identify new or emerging themes as an outcome of the group discussion.

VI. SUMMARY

The purpose of technical documentation is to impart instructional information or knowledge to different audiences. The information in its original form must be translated, so that different audiences can understand and make use of its intrinsic value.

The pace of technical communication is changing so rapidly that many technical communicators are having difficulty staying up to speed with some of the complex web development and design tools. The purpose of this research study is to determine if the integration of on-going training in the work schedules of documentation teams will enhance individual competency and improve team performance. The researchers will utilize various methods and techniques to gather data in an effort to verify this hypothesis. While the researcher's focus concerns problems of technical communicators keeping abreast of new software tools and insufficient training on existing tools, the researchers hope to discover themes that impact those currently in the technical communication field, as well as those embarking on a career in the field.

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Design Viewpoint on Public eServices

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***Abstract* – This paper discusses the design of public eServices on a theoretical level. Instead of providing clear answers, contribution of this paper is on encouraging discussion over defining correct questions to be asked when carrying out a design process for public online services. In this theoretical endeavor two concepts are utilized, the ease of use and usefulness of a web site, and how the outcome of these two concepts reflects the key issues and shortcomings of a design process. Basic three questions to be asked are what, how and the most importantly why.**

***Keywords* – Usability, Digital Divide, Design, Public Sector, eServices.**

I. INTRODUCTION

This paper presents a theoretical interplay of two key concepts adopted from theoretical setting used for measuring commercial websites, namely the ease of use and usefulness of a website. Here these concepts, as well as their origins, are discussed from the viewpoint of web site design for public services. Main goal is to point out the significance of successful dialogue between different interest groups involved into development project, by emphasizing the need to focus on correct questions. As such, this paper addresses issues usually related to design science, although being a theoretically oriented enterprise. The main goal of this paper is not to offer ready answers but instead try to learn to ask right questions.

Traditionally in Nordic welfare state model, many basic tasks and services have been provided by public administration. The shift from post industrial era to information society has forced public sector to re-evaluate it's status and responsibilities. Under the pressure caused by laws of market economy (or New Public Management, NPM), services must be delivered to the citizens all the more efficiently and economically. The rapid development of information and communication technology (ICT) and the rapid expansion of telecommunication network coverage have made it possible to provide public services in a cost effective manner. Because of this, the usability of services has become very important issue for contemporary realizations of eServices.

The role and status of public administration reflects the expectations being put on it by surrounding society. As a result of information society development, government and municipalities are in a situation where operational principles have to be seriously re-evaluated. Move towards information society changes society in many ways. Along with the development comes several changes; 1) the importance of information, formation of information, revision of information, further refinement of information and mediation of information; 2) while the significance of knowledge work increases, so does the significance of education, as well as innovation and research; 3) Society becomes more complex, technical and computerized. [1]

II. IMPLICATIONS OF EVOLVING PUBLIC ADMINISTRATION

Kauppi, Lähdesmäki and Ojala [2] have studied the change taking place in public services. They claim that this is happening on two different levels, which are intra-organizational and extra-organizational level. They even state that while in traditional public administration role of information technology was in computer aided administration and in computer aided services, the focus is different in modern e-governance, as suggested in Table 1.

TABLE I
CHANGING PUBLIC SECTOR AND E-SERVICES [3].

	Traditional Public Administration	**Modern e-governance**
Intra-organizational	Computer aided administration • Registers, databases	Knowledge management • Shared information systems • Strategic management
Extra-organizational	Computer aided public services • Statistics, financial information	E-services • Customer oriented electronic services

According to a definition by Ministry of the internal affairs [4], public online services are services, which are provided to citizens, companies and communities and bureaus using computer networks. These services can range from searching and verifying information to provision of inter-active services and possibilities to participate in preparation process and decision making. When considering the use of information technology in modern e-governance, it is being used in knowledge management and for e-services, like Kauppi et al. [5] suggests.

Peristeras, tsekos and Tarabanis [6] suggest in their four stage schema of IT effect on organizations, that a

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straightforward dependency can be seen between the maturity level of the IT employed by an organization and the influence that IT exercises upon it. In other words, the more advanced the IT employed by the organization, the more it affects the organization. Stages introduced are [7]:

- 1st Stage - Islands of Automation: Organizations continue to function and produce as before having just automated a small fraction of their processes. Information technology is used for automating existing operations.
- 2nd Stage - Automated Process Chains: Organizations produce the same products and services as before but with changes in the way the organization functions. At this stage some roles acquire new characteristics, while some other may become obsolete. In addition, some processes are simplified and various steps are replaced by the usage of information technology. This second phase, does not allow the utilization of the full potential of information technology as it leaves intact the foundations of the way work is organized.
- 3rd Stage - Reengineering through Information Technology: Organizations produce the same products and services as before but in a completely innovative way that affects all internal functions, information flows and structures. While the main objectives of the organization remain the same at this stage, there is a radical change in how the organization produces, but not in what it produces.
- 4th Stage - Total Reinvention: Organizations restructure the meaning of their existence and all their internal and external relations: why, what and how to produce their services or goods. During this stage, the revolutionary development of all relevant technologies has created a domino effect by introducing changes in almost every aspect of organizational life.

Peristeras et al. [8] make an important note while pointing out that in current, tumultuous environment, where information technology has become the main agent of change at all levels, organizations question not only how to produce but what and why to produce. They state that even the vision of the organization is in question, as societal needs change and new characteristics emerge in all fields. This is to say that demand for new kinds of services has arisen, while services provided until recently become obsolete, and their production must be decreased or even ceased.

Then again, is it really that simple – don't we still need traditional public services? From a Nordic welfare state point of view, the needs have anything but vanished. There are several issues of concern in service provision in western countries though, such as severely distorted pyramid of age, new public management (NPM), or globalization to name a few.

III. Digital Divide

In order to be able to provide all citizens equal access to public services one inevitably will be forced to tackle with issues concerning digital divide. Norris [9] raises into focus a fact that the global digital divide is substantial and has been growing during the first decade of the internet age. According to wordbook definition the digital divide refers to: "…(a) a division between those in favour of the extensive use of digital technology (esp. computers) and those against it; (b) (now the usual sense) the gulf between those who have ready access to current digital technology (esp. computers and the Internet) and those who do not; (also) the perceived social or educational inequality resulting from this…" [10]. The concept is used to refer to unequal access for computers and computer networks.

According to Economist Intelligence Unit e-readiness ranking [11] a critical shift is occurring in the social fabric of countries: as governments and enterprises increasingly migrate services online, those without reliable Internet access are in danger of being disenfranchised, e.g. the concrete risk for digital divide is present. Because of this both public- and private-sector organizations have spent considerable time and money trying to get economically disadvantaged communities online. Increasingly, these attempts are spreading to the socially disadvantaged as well. According to report [12] Senior-citizen access has been a particular focus for developed countries, where some 20,000 senior citizens are estimated to have been trained through various initiatives. EIU report [13] seems to combine concepts of economical- and social disadvantages as digital divide, whereas technical ability is not seen as a key factor, like in OED definition.

Since the introduction of the first visions concerning global information society the threat for unequal access for information has been brought up by skeptic voices. Apparently digital divide is a problem which is seen as a serious threat, and thus different strategies for suppressing digital divide have been included in different public programs. Undoubtedly this is an issue to be taken into account when designing public online services for masses.

IV. eGovernment and eServices

Despite that there are certain similarities; interests of commercial organizations differ from public organizations. According to Scherliz and Eisenberg [14] government can for the most part serve its immediate interests following the private sector, but they underline that there are perils in this approach. They [15] state that there are many areas of challenge where government is a demand leader, and that these are areas where it has requirements that may not be unique, but nonetheless exceed those of the private sector; ubiquity of service, trustworthiness, information access, and confidentiality.

When taking into consideration different definitions of eGovernment, certain characteristics can be seen. According to Webster's online dictionary [16], "The term (in all its uses) is generally agreed to derive from 'electronic government' which introduces the notion and practicalities of 'electronic technology' into the various dimensions and ramifications of government." Thus, the most frequent use of the term eGovernment [17] is related to the deliverance of services online, the conduct of government business where some electronic or online aspect is under consideration, or voting where some online aspect is under consideration.

The definition given by European Commission gives similar implications, although it does not concentrate that much on addressing working online, but more on the nature of the services: "eGovernment enables the public sector to maintain and strengthen good governance in the knowledge society." [18]. This translates [19] into a public sector that is open and transparent, an administration that is at the service for all and into a productive public sector that delivers maximum value for taxpayers' money. As a summarison, eGovernment in EC context is a mean to enable a more open, inclusive and productive public sector, in line with good governance.

According to DigitalGovernance.org Initiative [20]: "Digital Governance is often referred as Egovernance, E-governance or Electronic Governance. In simple terms, it refers to governance processes in which Information and Communications Technology (ICT) play an active and significant role." Basically this role is four fold and it aims to; a) improve quality of governance products and services; b) enhance participation of people in choice & provision of governance products & services; c) provide new governance services and products; and d) bring new sections of society under the governance sphere. The most important issue in eGovernment, according to DigitalGovernance.org Initiative, is not simply digitalization, or automation of governance services, but to act as a 'tool' for good governance and human development.

Information technology has become a necessity for both private and public organizations. It has turned out to be a infrastructural technology, pretty much in way Carr [21] argued in his famous article "IT doesn't Matter". In general, governments try their best in order to improve convenience of public service delivery. A tool for this is information technology, and according to van Duivenboden and Lips [22], this means changes in service provision, such as change from supply-oriented to demand oriented; from collective to tailor-made; from fragmented to integrated; from specialised to general; from reactive to proactive; from passive customer participation to active customer participation; and from one service counter to "multi-channeling". In general, it appears that the way how public services are provided is under focus. Goal appears to be a virtual "one service point" type of approach, where services are produced based on demand and offered to customers. Services are also provided in an integrated, proactive and customized manner, where focus is on getting customers involved in a totally new way than before. Obviously this puts a lot of expectations on public services, regardless of the form how these are provided.

V. Research Approach and Used Method

Used methodological approach in this paper in deduction (e.g. through axioms to theory) by utilizing existing knowledge from e-commerce research and combining it with studies of public administration. This paper encourages theoretical discussion, and as such it can be categorized according to taxonomy created by Järvinen [23], as conceptual-analytical research. Here previous theory is being enhanced through some changes and additions to make it suitable for assessing public services. According to Järvinen [24] it is allowed to take a theory from the other science. He uses as an example studies concerning impacts and implications of computers which are often "based on two legs", e.g. to computer science and to some reference science. This kind of study must be acceptable in the all sciences, not only in the primary science but also in all the reference sciences.

According to Lee and Baskerville [25] when generalizing concepts from theory, a researcher generalizes from theoretical propositions in the form of concepts to the theoretical propositions that make up a theory (specifically, a set of logically consistent propositions that, pending the results of empirical testing, could qualify as a theory). Lee and Baskerville [26] see that another form of generalizing from concepts to theory is the formulation of a theory based on the synthesis of ideas from a literature review. In this paper, theoretical concepts used for constructing a research setting are taken into use in slightly different context and through generalization are used as a lens for studying the significance of dialogue between different interest groups. There are problems in generalizations though, especially when using information technology as an independent variable in IS theory [27].

VI. Is a Useful Service Automatically Easy to Use?

Quite many services provided by public authorities are of such nature, that users really have no choice but to use those. In business environment choices are exit, voice and loyalty, exit driving out the voice option, but in government organizations exit is not (wholly) possible [28]. Here one could argue whether it makes sense to use money for usability issues, because everyone is forced to use systems anyway. Practical motivation is based on two important issues; a) systems which are reliable and easy to use will reduce the running costs of helpdesks and minimize the work of civil servants responsible for correcting errors, and b) public policy explicitly expresses these days "information society for all" type of approach.

According to action plan eEurope 2005: An information society for all by Commission of the European Communities [29], the eEurope action plan is based on two groups of actions which reinforce each other. First, it aims to stimulate services, applications and content, covering both online public services and e-business; Second, it addresses the underlying broadband infrastructure and security matters. According to the action plan, Europe should have by 2005:

A. *Modern online public services (e-government, e-learning services and e-health services).*

B. *A dynamic e-business environment, and as an enablers for these.*

C. *Widespread availability of broadband access at competitive prices.*

D. *A secure information infrastructure.*

In his classic work about usability engineering, Nielsen [30] defines factors influencing system acceptability in detail. Usefulness consists of two separate entities, namely utility and usability. Usability has several issues related to it; the system needs to be easy to learn, efficient to use, easy to remember, it has few errors and it is subjectively pleasing.

Thong, Hong, and Tam [31] present a digital library user acceptance model which is based on technology acceptance model (TAM [32]). According to Thong et al. [33], the premise of the model is that acceptance is determined by users' perceptions of the system's usability, that is, its usefulness and ease of use. Here usefulness has a direct influence on user acceptance because individuals will be more willing to use a system that provides helpful functions. And the easier it is for a person to interact with the system, the more likely that person will continue using it. According to authors, there are three categories of external factors that may impact user acceptance of digital libraries: interface characteristics, as the system interface is the door through which users access a digital library; organizational context in which the digital library operates, and the individual differences users hold in the ultimate acceptance of a digital library.

Usefulness and ease of use have been in focus also in e-commerce research. Although the concepts were originally developed for measuring transactional web sites, those concepts are suitable for assessing public online services as well. According to Aladwani [34], the concept of easiness and usefulness are difficult to measure. Aladwani proposes a web site analysis instrument, where easiness of use consists of twelve items and four dimensions and in addition another where web site usefulness consists of ten items and three dimensions [35].

When using these two concepts Aladwani introduced, and applying these for analysis on final results of public online services, we have a simple dichotomy to be used. Here service can be categorized either as easy to use, or not very easy to use (ease of use) and on the other hand into useful, or not very useful (usefulness). Tentative categorizations are represented in table 2.

TABLE 2. EASE OF USE AND USEFULNESS FROM THE DESIGN VIEWPOINT.

		Ease of Use	
		Easy to use	Not very easy to use
Usefulness	Useful	Symmetrical Objective of web site construction **A**	Content oriented asymmetrical condition of web site construction **C**
	Not very useful	Technically oriented asymmetrical condition of web site construction **B**	Symmetrical Nightmare of web site construction **D**

When considering the service function of public services in general, the online implementation of these appears to have two significant "layers". These are substance and technology. Here technology refers to IT infrastructure and practical issues, where substance refers to actual content provided. Often encountered problem in IT projects is distorted (or totally missing) dialogue between developers (engineers responsible for implementing new system) and customer side (in this case civil servant). From here tentative hypotheses can be drawn, it appears that:

A. *Symmetrical balance is most likely possible, when dialogue between technology and substance takes place.*

B. *When the design and planning of a web site is being left only on the hands of web design professionals, the risk of technically oriented asymmetry is a concrete risk.*

C. *When the design and planning of a web site is left primarily for content experts, there lies a severe risk for content oriented asymmetry.*

D. *Symmetrical nightmare is very likely to occur, when the dialogue is lacking.*

So, does ease of use precede usefulness, or vice versa? When using these two counterparts, this comparison is like comparing oranges and apples, where both are fruits, but don't have that much more in common. The setting turns out to be a lot more fruitful, when considering these two as equal viewpoints on a same phenomenon.

VII. Basic Questions Preceding Design Process

Eason [36] states, that impact research has shown surprisingly high level of systems that failed. The rate started at around 40 % and, despite vast improvements in the technology, has stubbornly refused to decrease through the many surveys conducted in the past 30 years. In other words, there is a lesson to be learned from this.

Preece, Rogers and Sharp [37] suggest that there are three key characteristics in interaction design (ID) process to be followed. First, the need to focus on users. Second, identification, documentation and agreement over specific usability and user experience goals at the beginning of the project. Third, iteration, which allows designers to be refined based on feedback. They [38] refine these characteristics into practical questions to be answered before starting the interaction design: a) Who are the users? b) What do we mean by needs? c) How do you generate alternative designs? d) How do you choose among alternatives?

When analyzing these key characteristics, it appears that focus on users is actually seeking answer to basic question how. Specific usability and user experience goals are actually about seeking answer to what question. Iteration offers a way for refining the process while engaged, but it appears that the why questions is not being given enough importance as it deserves, if any.

Every time when designing new eService concepts and realizations what and how questions should be given great importance. Although questions might sound quite simple, the first one is about substance, and the latter one about technology. When focusing on the ease of use and usefulness, it appears that these are related in a following manner, what question provides building blocks for usefulness and the how question is forming the basis for ease of use. These concepts are still missing one combining factor which is possibly the most important question of them all, namely why. Without asking the why question in design process, other questions will remain very hollow and resulting outputs will never reach desired level of quality.

Although there has been distinction between design and programming, or even more disturbingly between technical and creative, this division is actually very artificial [39]. Distinction between a web site substance and how it is technically being manifested is a totally different matter. When studying public online services this needs to be kept in mind, because the main issue is not how it looks like, but how is serves it's purpose.

When constructing public online services, the role of the designer or usability professional is important, but without working dialogue with experts over substance the endeavor is doomed to fail. In a case study of library website redesign by Tolliver, Carter, Chapman, Edwards, Fisher, Haines, Krolikowski, and Price [40], the role of usability consultant who was an outsider was reassessed, when it came to understanding that the librarians were the experts when it came to understanding the content and to knowing what users needed. In this case assuming a partnership role in the testing process brought results that were relevant, and that led to a substantive understanding of the dynamics of the website. In this case, the answer to how question (e.g. usability consultants contribution) apparently was reassessed, when finding out proper answer for why question.

According to Brajnik [41] the quality of websites is dependent of factors in three categories; (1) task-related factors affecting end users, (2) performance-related factors that affect the efficiency of end users and the economics of the website within the company running it, and (3) development-related factors that affect developers and maintainers of a website. As it appears, the relation between usability and usefulness is also correlating with these afore mentioned quality issues.

VIII. Discussion

Peristeras et al. [42] outlined basic questions related to automation of public services as why, what and how. On the other hand, from the viewpoint of web site analysis of the ease of use and usability, two questions emerge, namely what and how. In general, this refers once again back to basics, where technological view point covers both design and technical issues and social viewpoint issues cover the main reasons and motives. As such, it unfortunately appears that there still exists a concrete risk for technological determinism, unless the why question is included permanently in the vocabulary of online architects.

Public online services have many expectations to fulfill. Basically these services are more than just extension for "brick and mortar" type of governmental agencies. Underlying idea for eGovernment is to enhance public administration so that it can provide high quality services in a cost effective manner to all citizens. A way for reaching this goal is to focus on design process, so that the real intentions and motives are communicated correctly already in design phase.

As a theoretical paper, this work encourages discussion on proper questions to be asked. As this paper illustrates, the polarities in the design of public online services is a problematic domain. Reasons behind bad online implementations of working public practices might vary, but a common nominator in every case is that they are not welcome. A bulletproof way towards deficient implementation is a poor communication between actors who are responsible for defining solutions for questions what and how. In addition, this paper suggests that concepts usefulness and ease of use in public online service context can be seen as more elaborated extensions to basic questions what and how, but are more likely to be used as instruments in evaluating final outcomes of a design process.

Everyone would like to see public online services, which are both easy to use and useful. Unfortunately these two concepts are not as easy to combine as one might think. There are several reasons for this though. First, there are a great variety of users, who all have their individual preferences and needs. Second, public authorities are often in a situation, where they are exercising bounded rationality, which translates into reality as procedures which are not the most efficient, nor economical – but legal and commonly accepted. Public policy is quite often the policy of common, thus holding back the most innovative ideas of implementation. In order to play it safe, correct answers for correct questions are needed in design process.

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Surveillance in Cyberspace

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***Abstract* – This paper suggests that a new approach for measuring topography of surveillance in virtual environments is needed. In this paper both surveillance and construction of virtual profiles are studied using phenomenological approach in order to be able to provide versatile cameo of present situation of electronic surveillance. Here different types of surveillance are illustrated using real life examples, main focus being on measuring the level of surveillance using two polarities; whether the level of surveillance is absolute (as in Foucault's Panopticon model), and when it is not (as in Walden-model by Thoreau). When using this scale, calculating intermediate values is a problem, and therefore authors suggest that modal terms should be used instead. The amount of available information has increased rapidly, and at the same time the nature of it has been changing all the time to a more abstract level. Personal data is being collected and stored at every turn into different databases by both private and public organizations. Authors are concerned about the way how this data is being collected and used in constructing virtual profiles and point out apparent shortcomings in combining data from several different sources without adequate diligence. Main problems addressed are the data used and accuracy of these profiles. Authors also question the correctness of these virtual constructs by returning to an old dilemma of inter-subjectivity. This paper suggests that problem of inaccurate profiles can be approached using hermeneutic circle, e.g. utilizing the process of understanding. Virtual identity is a dynamic process in the same way as the constructing of it, and by understanding this process and what the process requires, it is possible to make surveillance transparent, thus enabling such practical applications as increasing trust in e-business, self regulation in governmental agencies or informing citizens. Basic underlying question that still remain is: How surveillance and awareness of it will affect people's privacy and trust.**

***Keywords* – Surveillance, Privacy, Virtual identity, Control, Data mining.**

I. INTRODUCTION

Contemporary western society has undergone significant changes, while giant leaps in technological development have changed the very way how and where we exist. Numerous surveillance systems are tirelessly gathering data about our everyday routines and this data is being stored in different databases. This data is being used in many ways, thus giving birth to virtual identities.

This paper suggests that in metaphorical sense it is necessary to take a step backwards in order to be able to see the whole picture about what is going on in the field of electronic surveillance. The importance of surveillance technologies has gained evermore significant meaning since terrorist attack upon the United States in September 11th, 2001. Since then numerous terrorist attempts have taken place world wide, thus legitimizing the very need of electronic surveillance in order to maintain acceptable level of security. At present there are numerous governmental agencies which are directing considerable efforts on gathering surveillance information and analyzing it. This paper suggests that there is a need for a framework for understanding the validity of gathered surveillance data. This is a two way approach, providing an approach for understanding what is going on, and for making it possible to make all the time more and more immersive surveillance transparent. This paper does not offer a comprehensive solution to the problematic field of surveillance, but instead presents several very important questions in a form of theoretical framework. These questions are important for both, the ones analyzing surveillance data and gathering it and to those who are under surveillance.

II. METHOD

In this paper authors are entwining together concepts from different philosophical studies and are applying these on a framework to be used as a synthesis. Used research approach is phenomenology, which is not a method of empirical scientific investigation, but rather a philosophy of human consciousness. Phenomenology can be called eidetics, which means that it is possible with the assistance of an eidetic reduction to reach to the essence of things and phenomena [1]. In this study the essence of control is studied from different viewpoints.

Edmund Husserl (1859-1938), the founder of the phenomenology thought that philosophy had to save the sciences from a natural attitude and careless theorizing. But at the same time we are in the middle of a life flow. When an individual is interpreting his own life, he is actually moving in the same way as life itself. Husserl calls life-world the ultimate horizon of all human achievement. Individuals as conscious beings always inhabit life-world. It is given in advance and experienced as a unity. It is the general structure that allows objectivity. In Husserl's later studies an observer or scientist (philosopher) is then (situated) in the middle of the life flow, bound to historicity and life-worlds, and somehow he is able to outline the ethical condition of culture. [2]

Husserl recognized that in all grasping of objects (or phenomenon) there are aspects of objects which are not

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directly grasped. This means the horizons in Husserl's terminology. Traditional empiricism attempts to describe the actual nature of phenomenon in terms of a presence of sensual data but has ignored the manner all perception takes place under a number of horizons which are implicit structural aspects of our original experience itself. [3] Historicity and future expectations are aspects that have to be taken into consideration when researching long-term phenomenon. Husserl was aware that perception is a temporal process; it doesn't take place only in the present but is oriented towards future experiences and at the same time is an experience of enduring or continuing from past experiences. There is also a horizon of the past. [4]

When researching surveillance and control a historical perspective is extremely important. The main phenomenon namely the control itself remains the same, but the ways it is practiced are varying. And the future must also be taken into consideration.

III. Research

A. Forms of control

Social control

Social control refers to social mechanisms that regulate individual and group behaviour leading to conformity and compliances to the rules of society. Social control is present in all societies, if only in the control mechanisms used to prevent the establishment of chaos or anarchy. Mark Neocleous emphasizes the role of the police in the fabrication of social order [5]. The police came into existence in the late fifteenth century, and its main task was to improve the legislative and administrative regulation of the internal life of a community and to promote general welfare and the condition of good order and the regimenting of social life. So the police was the equipment of control, surveillance and monitoring. However, in the traditional rural societies the control was performed without authorities – it was (and still is) social control. It requires that one is "known" and "seen". It is very difficult to break out from this kind of control, physically the only way is to move out to another controlling society. Urbanization, totally solitary life and Internet can be seen as attempts to break out from the control and to gain freedom. Social control exists, however, and it is hiding in many different practices at all stages of social development – it is not bound to rural or small scale communities. Fashion, manners, gestures, clothing, integration into the society, all these practices are a part of social control. Social control can be seen as a common feature which is present in all other manifestations of control.

Foucault's Panopticon

Jeremy Bentham (1748-1832), a utilitarian philosopher designed The Panopticon to suit the needs of total surveillance. The Panopticon is a prison and at the same time a round-the-clock surveillance machine. It was constructed (or designed) so that the prisoner could never see the prison guard - the "inspector" who conducted surveillance from central location within a radial configuration. Because the prisoner could never know if he was being monitored, it caused mental uncertainty that in itself would prove to be an important instrument of discipline. Later French philosopher Michel Foucault developed further the idea of The Panopticon [6].

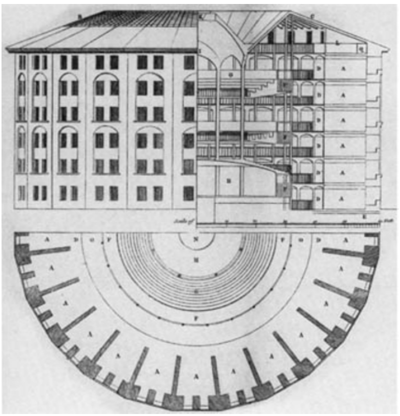

Fig. 1. The Panopticon blueprint [7].

It is possible to talk about total control and panopticism. A panoptic mechanism is a mechanism of social control based on Foucault's ideas of The Panopticon and the exertion of discipline onto the body. An apparatus for control is based on an inspecting gaze. It is possible to adapt the idea of The Panopticon for many other public buildings. The Panopticon, as a prison, is after all a physical construction (Fig. 1).

Vaasa City Library

Social control, new architecture, video surveillance, combining of different kind of data and restricting legislation, that prevents the total control of the visitors are joined together in Vaasa City library. There are 28 video cameras in the library that monitor the staff as well as the visitors. The Finnish legislation permits to monitor the employees but the customers have to be informed that surveillance is used [8]. However, he signs informing the customers are very small and placed in corners so that they are difficult to see. Customers and library loans are, of course, registered in a database. The staff is under surveillance on many levels: when they come to work, when they log in the data system and different databases and office tools, when they move in the space (video surveillance) and communicate with each other or their superiors. Vaasa is a small town with 50 000 inhabitants. Vaasa actually consists of "two towns", the Finnish speaking and Swedish speaking community. So there is also traditional social control in Vaasa. Vaasa City Library is an example of a transformation of traditional physical surveillance into virtual control, where the amount of information has increased significantly.

B. Data mining

Data mining is a term which means the utilization of artificial intelligence (AI) and pattern recognition in information retrieval [9]. Data mining means, that the "miner" tries to find the information which has got lost or is concealed in databases. With the help of data mining it is possible to reveal the figures and connections that have earlier been unknown, and use this "new" information (or knowledge) in decision-making processes. Data mining is related to research and information search that precede the formulation of hypothesis and exploration.

Web mining is an application of data mining techniques to extract knowledge from web data, including web documents, hyperlinks between documents, usage logs of web sites etc. [10]. Data mining is a developing and expanding discipline. In web mining and data mining it is possible with the help of algorithms to combine small bits of harmless information attached to private persons in a way that the final results consist of unexpected viewpoints to that person.

Data mining technology is being used for a wide variety of applications from marketing and finance to medicine and biotechnology to multimedia and entertainment. Recently there has risen an attempt to use data mining for counterterrorism applications. For example, data mining can be used to detect unusual patterns, terrorist activities and fraudulent behavior [11]. However, data mining applications can threaten our privacy. Data mining tools or the equipments are available and even naïve users can use them to extract information from the data stored in various databases and files, and in doing so they sometimes violate the privacy of the individual [12]. According to Chris Clifton et al. "when given a collection of data it is possible to learn things that are not revealed by any individual data item. An individual may not care is someone knows his date of birth, mother's maiden name, his social security number or his memberships in different associations etc. This knowledge makes identity theft possible. This type of privacy problem arises with large, multi-individual collections as well." [13]. As mentioned earlier, the final outcome may be a profile that has nothing to do with the person in question.

Let's take an example. Figure A is a freelance journalist, and he writes articles to different newspapers and journals. His motives for writing are commercial. He has a moral backbone or ground, but above all he is liberal and tolerant. To earn money he writes literary reviews on different topics, for instance books dealing with feminism, globalization, the Left, homosexuality to various journals and magazines. When he becomes an object of data mining, these things or "features" come out writ large and first of all. These facts may aptly describe Figure A as a very liberal and flexible person, but they don't tell anything about his real identity.

The possibilities of data mining increase if different commercial actors exchange customer information (cookies and web forms) in Internet. Person and customer information has been a valuable resource, although the authorities are trying to restrict its use both in legislation and in mutual agreements between companies. This is important because the lack of privacy reduces the trust of the customers/citizens. [14, 15]. Companies can make great savings by concentrating advertising, segmentation of the customers and about all by finding new target groups to their products.

According to different surveys Finnish people very easy give data about themselves to companies and to persons who can then make good use of them. Finns can be characterized as naïve and starry-eyed [16]. These features may have created a stereotype and an image about Finns as honest and reliable – aren't that kind of people a bit stupid? Too much openness doesn't fit the present-day because by means of Internet all information becomes both transparent and easy to exploit.

The surveys made among American consumers have given contrary results. The protection of privacy in Internet is seen as a very important problem in the United States. In the research made by Wright and Kakalik 9 300 Internet users were interviewed; 70 percent of all the respondents thought that privacy problems are more serious in Internet than in any other mass media. [17]. According to Benassi users appreciate privacy more than usability and easy use, expenses and security [18]. Other investigations show that two thirds of American consumers are worried about facts concerning privacy. In other words, Internet users are afraid that they lose their privacy [19]

Donald Gotterbarn wants to pay attention to the phenomenon, that information society has created new concepts: in addition to the real persons there is virtual identity and the latter is represented by virtual information or knowledge. Gotterbarn criticises the legislation which allows to citizens study the information that has been gathered about him but doesn't block effectively enough the gathering of that information. Gotterbarn quotes an incident, where a banker could use a register of cancer patients and combined that register with the database of credit status information. The outcome was a list of the customers who had both house loan and cancer. [20] Another example is a third-party motor insurance that gives extra reduction to teetotallers. Would an insurance company be interested if people were surfing in the sites of liquor stores or if they bought spirits via Internet. This would reveal their cheat. It is possible. By concluding, by making hypotheses about loose and arbitrary variables, to construct virtual identities in Internet, which are not true. Virtual information consists of loose and arbitrary conclusions, of traces and generalizations that deal with current and future behaviour, both in the society and as a consumer. Because of the huge amount of information in

Internet it is very easy and possible to collect virtual information, and often it is also legal. [21]

What relation has virtual information with real personae? It is not ethical to draw conclusions about people who are surfing in Internetnot knowing they are being monitored. People think Internet is a kind of play that has other rules than real life. In Internet it is possible to behave in a way different from real life, conduct more freely at least as a consumer of entertainment. Both parties, the surfers and gatherers of information (authorities, businessmen, secret service) should remember this and take it into consideration. People who are surfing reveal another side of themselves, they behave in a way they wouldn't do in real life. [22]

The matter becomes more complicated if those who gather information are not real people (with all the prejudices and weaknesses) but robots or espionage programs that are functioning by means of AI. They archive only the conclusions, not the data the conclusions are backed by. Machines make the work, collect the material and create a virtual profile, which is compared to the identity of real persona. This "identity" is changing all the time and the real persona doesn't know anything about the ongoing process. Maybe the real persona and the information about him never meet each other, neither has he any possibility to influence on or change this information and profile. There are many descriptions in fiction about the powerlessness and bewilderment of the man under control and surveillance, for instance The Process and The Castle of Franz Kafka.

C. *Spatial boundaries and the era of virtualization*

Although there are visions, according to which Internet technologies can provide us freedom of place and time, there are few restrictive things that must be remembered. Space and time will remain significant due to three main reasons [23]; First, cyber spatial connections and bandwidth are unequally distributed both within and between western countries, and in comparison to developing countries; Second, whilst information on-line might seem geographically dislocated, information is only as useful at the locale within which the body resides; Third, cyberspace depends on real-world spatial fixity, e.g. the points of access, the physicality and materiality of wires.

It appears that information which is being stored and used in production has not been changing just due to media used for saving data. For example, paper has certain restrictions when compared to data stored on a CD-ROM. For example the ease of copying digital data is a significant improvement, when compared to book printing technology introduced by Guttenberg. Unfortunately there are certain drawbacks in this too, for example controlling access to sensitive data is nowadays a lot harder than in the past.

Fig. 2 presents how abstraction level and boundaries of time and space are inevitably bound together when studying information produced during different development phases of modern civilization. In addition, the figure illustrates examples of surveillance at different levels and how those could be placed in scale. These examples of forms of control are previously introduced: 1) Social control; 2) Panopticon; 3) Vaasa City Library, and; 4) Different applications of data mining. Because of the different degrees of abstraction, or varying intensity of compelling dependency of spatial and/or temporal boundaries, these example cases are situated differently in the scale. Social control, as exercised in agrarian society was, (and is) dependant of social networks and is firmly dependant of both place and time, while technologies emerging at present society which utilize data mining technologies are very abstract by nature.

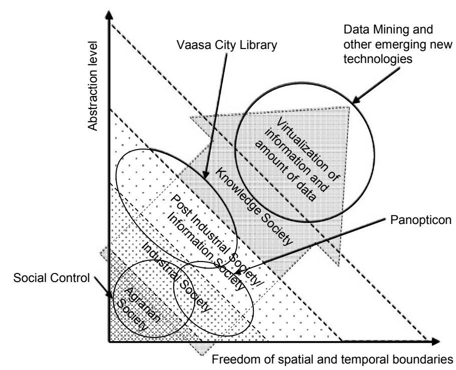

Fig. 2. Changing information and relation between concreteness and time and place.

Information stored in databases has changed to be a lot more abstract by nature, and users are actually giving a new meaning to it. When data concerning one person is collected and stored in different information systems, basically several different virtual identities are born — which all may have very little to do with original person at all. By searching information from several sources (as in data mining) about one person and combining it with information from another, it is possible to construct a virtual profile in order to try to understand this person better. As such, authors refer this as virtual profiling, but there are obvious problems related to this. To begin with, how accurate can virtual profiles be? This is a well grounded question because these are purpose-oriented constructs. Another weakness concerning virtual profiling is the data used for profiles.

D. *Virtual profiling – from data mining to virtual identities*

Personal data is being collected into different databases in a continuous manner. Depending on the systems in questions, information on personal details stored in databases varies. For example, the data contained in personal registry by IRS might vary greatly with the data stored in local video rental company's customer database. There are apparently also differences on the accuracy of the stored information. The data needs constant updating, yet there are numerous databases, which contain information which is more or less out of date.

When storing data into databases, choices need to be made about which data to collect. Usually choices are made already when designing information systems to be used in organization. In many public organizations part of daily routines is to store data into databases and to retrieve and combine data from several different sources while performing normal administrative tasks. Data mining and cross checking information from different sources is quite effective way to be able to read between the lines what is actually happening. Business Intelligence or Competitive Intelligence are good examples of this, although in those cases used information is not restricted only on electronic databases. McGonagle & Vella [24] refer to a key maxim in CI, derived from the history of governmental intelligence, where 90 percent of all information that a firm typically needs to make critical decisions and understand its market and competitors is already public or can be systematically, legally, and ethically developed from public data. There are some issues to be aware of though, which are related to the nature of collected data and on systems used for knowledge management. These issues are as follows: Data is usually quantitative, while CI often concentrates on qualitative; CI professionals should be able to have access the people who provided the data as well as the data; most knowledge management systems do not record any data which does not involve the company; sales force is rarely involved into knowledge management, although the most powerful source of in support of CI; very few knowledge management systems provide current in-formation on employees; and finally, knowledge management systems do not record decision making and the history of decisions. [25]

When combining data from different sources, for example from personal web pages, governmental databases along with bank records, the actual outputs might vary a lot depending on the way how information is being combined — or depending of the accuracy of the stored information. Fig. 3 illustrates the process, how information concerning one individual is being collected by several governmental agencies and companies into specific databases. Later on, the very same data is being used and combined with data from different databases, when trying to profile this person for varying purposes. For example, marketing people would be very interested in knowing about buying habits and economic status of this person. Police would be interested whether this person has a criminal record, problems with law or health problems before granting him/her a driver's license, while airlines would have other issues of interest, when booking an airline ticket to this person.

All these are examples of situations where a virtual identity is being born. Identity can be very narrow or very extensive and detailed one. But the main question is how correct it can be? Obvious answer would be that as long as it provides acceptable level of security, it is correct enough. But then again, what is an acceptable degree of security, and who is it to decide?

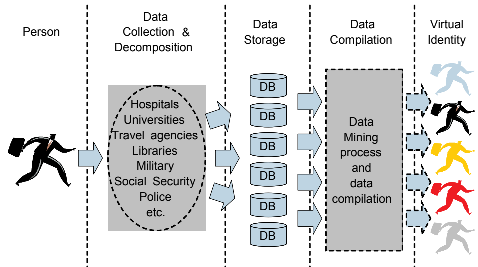

Fig. 3. Birth of virtual identity.

Problem closely related into building virtual profiles are mismatched profiles, which are based on incorrect- or incorrectly selected data. Sometimes there are serious shortages in information and when profile is constructed in such a manner that it is imperfect, practical usefulness is severely compromised. Unfortunately it appears that when combining data from several totally different databases the data is sometimes distorted in such a manner, the misinterpretations take place. Although data mining uses tools for analyzing large databases, as in competitive intelligence (CI), artificial intelligence (AI) does still need human perception in order to be as reliable as possible. Basically problem is that AI based systems can not deal reliably with all possible data and on the other hand there is always room for human error — both things, which are very unwelcome.

IV. Discussion

The authors of this paper have found it relatively difficult to describe the amount of electronic surveillance in quantitative terms. There were of course environments of non-surveillance in historical societies, where there was no electronic equipment. Electronic surveillance was replaced by social control. Not even does solitude blocks surveillance in information networks today. It is possible to gather information from different sources and combine it into different virtual identities. The difficulty (or the impossibility) of the task is the personal bulwark against the external we all have. In philosophy this is called the problem of inter-subjectivity. Therefore the virtual identities are often good or bad guesses – or hypotheses.

One possibility would be to use probability judgements that are supported by evidence. Then we are not talking about empirical research but about interpretation and hermeneutics and/or phenomenology. The starting point must, as in Immanuel Kant's philosophy, be a regulative principle: the goal is then the truth that can, however, never be gained. Then there may be many different competing interpretations – i.e. virtual identities. According to Douglas Hirsch "correctness is precisely the goal of interpretation and may in fact be achieved, even though it can never be known to be achieved". That's why the measure has to consist of scale in which the modalities are different kinds of probabilities.

How to achieve probability? In this, and in all interpretation, the method is therefore hermeneutic circle. The hermeneutic circle describes the process of understanding. It refers to the idea that one's understanding of the (virtual) identity as a whole is established by reference to the individual parts and one's understanding of each individual part by reference to the whole. Neither the whole (virtual) identity nor any individual part can be understood without reference to one another, and hence, it is a circle. However, this circular character of interpretation does not make it impossible to interpret (virtual) identity, rather, it stresses that the meaning of (virtual) identity must be found within its cultural, historical, political, commercial e.g. context.

Probability, as well as hermeneutics, is an ongoing process, which also has an ethical theme. The truth has to be achieved. The goal is the same as in all science. That's why you are never allowed to be satisfied with the results you have obtained. Virtual identity is a dynamic process in the same way as the constructing of it. And because of legal protection individuals have to know that all information stored about them is up to date and correct. In data mining and compilation the possibility of mistakes is always great. Therefore false interpretations must be eliminated by hermeneutic circle.

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Ranking Projects Using Multivariate Statistics

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***Abstract*-Organizational evaluation is a complex activity which demands a lot of effort. In order to accomplish this activity it is necessary to work with process and project indicators. Looking for project indicators it is necessary to identify good and bad projects to treat problems and to consider improvements. This paper presents a new methodology to rank projects using principal component analysis, a simple and efficient multivariate statistical method. This method aggregates project measurement indicators in just one score during the execution of the Organizational Evaluation macroactivity defined in the Organizational Software Measurement Process (OSMP) .**

I. INTRODUCTION

Concomitant to the speeded up growth of the software production market, comes an increase in the competition between the organizations that produce software. Companies are much more concerned with providing high quality products and, thus, guaranteeing the satisfaction of their customers. However, it can be perceived that to reach this objective, it is necessary to guarantee the quality with which the products are elaborated; it means that they must be concerned with improving the processes used in software development.

Measurement can be used by software engineers to help assess the quality of technical work products and to assist in tactical decision making as a project proceeds [6]. Ranking projects leads to an organizational evaluation, pointing out which projects are good and which are not. By analyzing those projects it is possible to identify strong and weak points that can be used for decision making.

This paper focuses on the Organizational Evaluation Macroactivity of the Organizational Measurement Process [3] and it describes a methodology to rank projects using principal component analysis, a multivariate statistical method.

The Organizational Software Measurement Process (OSMP) has a set of activities which must be executed in an organization in order to institutionalize a measurement process and the usage during the software project life cycle, aiming for better legibility, understanding and learning. It is used on an iterative and incremental life cycle.

This paper is organized in the following way: Section 2 accounts for OSMP overview. Section 3 presents the Organizational Evaluation Macroactivity. Section 4 presents the Multivariate statistics. Section 5 shows the principal component analysis. Section 6 is the core of the paper: here the projects are ranked using principal component analysis. Section 7 illustrates conclusions and sketches possible future developments.

II. ORGANIZATIONAL SOFTWARE MEASUREMENT PROCESS (OSMP)

The principal workflow of measurement process model proposed (Fig. 1) contains seven macroactivities to achieve its objectives [3]: (i) Organizational Planning for measurement; (ii) Plan measurement; (iii) Perform measurement; (iv) Analyze measurement; (v) Monitoring and control Measurement; (vi) Evaluate process measurement; and (vii) Organizational Evaluation.

The macroactivity Organizational Planning for measurement must be executed on organizational measurement process implantation and be repeated when there are organizational measurement process changes. The macroactivity Plan Measurement must always be executed when a new project is beginning or when there are plan measurement changes and the project isn't finished yet. The macroactivities of items (iii), (iv) and (v) must be worked at least on each iteration when an iterative incremental life cycle is used. The macroactivity Evaluate Process Measurement must be executed at the end of the project to verify process effectiveness, to capture lessons learned and point out improvements. The Organizational Evaluation macroactivity must be performed in a regularity defined in the Planning of the Organizational Measurement macroactivity.

Each one of these macroactivities contains a set of activities to be executed by specific roles. These roles are: (i) Measurement Analyst: Responsible for knowing the functioning of the measurement process well; (ii) Senior manager: responsible for business in the organization, he must know the objectives and needs of the organization very well; (iii) Project Manager: responsible for project management such as planning, problem solving and customer contact and; (iv) Relevant Stakeholders: Other people involved in the process such as the customers and the software development team.

These roles are classified as essential (the activity is carried out by the main actors) and contributing (a secondary role in which the activities are carried out by the collaborators, and in the absence of the essential role the activities are carried out by the contributing role).

The source for each proposed activity was identified [2] (CMMI-SW, ISO/IEC 15939, IEEE Std 1061, Six Sigma and PSM) if any. It was also necessary to describe in details the steps to be executed for each activity to structuralize the process adequately and to facilitate use. In/Out artifacts were generated for each activity. Some of these artifacts were taken from literature and tailored to the context of the considered process. Also some new templates were created to be filled in during the process. It shows some tips, tools and techniques to

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facilitate applicability and indicate who must work in the execution of each activity.

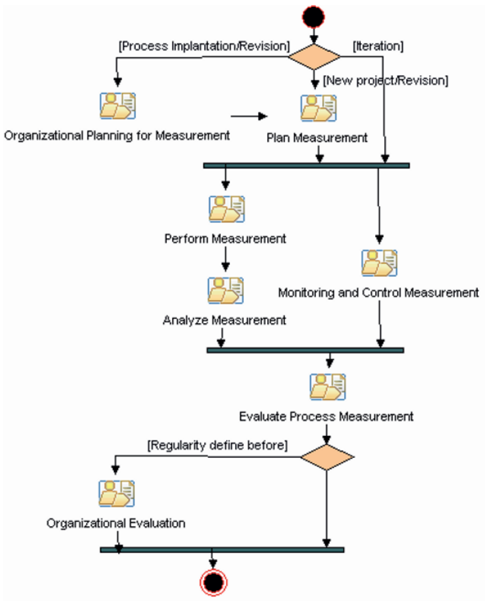

Fig. 1 – Workflow of OSMP.

III. ORGANIZATIONAL EVALUATION MACROACTIVITY

All the projects finished are evaluated based on the indicators defined in the macroactivity Organizational Planning for Measurement. The frequency is previously defined by the organization, and with the results in hand upper management can see the main deficiencies in its projects, and thus take corrective actions and consider organizational improvements.

A Activity: Evaluate the Organization based on the Measurement

The Organizational evaluation will be carried out based on the measurement. Thus, it is necessary to integrate the resulting data of all the projects finished in the previously defined period in the macroactivity Organizational Planning for Measurement, later analyzing the organizational indicators and finally considering organizational improvements based on the results.

IV. MULTIVARIATE STATISTICS

Multivariate statistics [7] describes a collection of procedures which involve observation and analysis of more than one statistical variable at a time. Sometimes a distinction is made between univariate (e.g., ANOVA, t-tests) and multivariate statistics, where univariate statistics only have one dependent variable, whereas multivariate statistics have two or more dependent variables.

There are many different models. Below some of these models are listed.

- Principal Component Analysis
- Linear Discriminant Analysis
- Canonical Correlation Analysis

In order to rank the projects the first model is used. Principal components analysis (PCA) is a technique for simplifying a dataset, by reducing multidimensional datasets to lower dimensions for analysis.

PCA is used to identify pattern in data. Since the projects performance can be analyzed by its indicators, a better approach is to use PCA to identify patterns

Applying the PCA model over the indicator values collected from the organization project is used to produce the components.

Components are a small set of new variables that explain the patter between the project variables (indicators).

V. PRINCIPAL COMPONENT ANALYSIS

Principal component analysis [1] is a multivariate statistical method which produces a lower dimensional description of observations from p-variables. The principle idea of reducing the dimension is achieved through linear combinations. Low dimensional linear combinations are often easier to interpret and serve as an intermediate step in a more complex data analysis. More precisely one looks for linear combinations which create the largest spread among the values. In other words, one is searching for linear combinations with the largest variances.

The first principal component is the linear combination with maximal variance and we are essentially searching for a dimension along which the observations are maximally separated or spread out. The second principal component is the linear combination with maximal variance in a direction orthogonal to the first principal component, and so on.

Principal component analysis deals with a single sample of n observation vectors $y_1, y_2, \dots, y_n$ that form a swarm of points in a p-dimensional space. Principal component analysis can be applied to any distribution of y, but it will be easier to visualize geometrically if the swarm of points is ellipsoidal.

If the variables $y_1, y_2, \dots, y_p$ in y are correlated, the ellipsoidal swarm of points is not oriented parallel to any of the axes represented by $y_1, y_2, \dots, y_p$. We wish to find the natural axes of the swarm of points (the axes of the ellipsoid) with origin at μ, the mean vector of $y_1, y_2, \dots, y_n$. This is done by translating the origin to μ and then rotating the axes. After rotation so that the axes become the natural axes of the ellipsoid, the new variables (principal components) will be uncorrelated.

We could indicate the translation of the origin to μ by writing $y_i - \mu$, but we will not usually do so for economy of notation. We will write $y_i - \mu$ when there is an explicit need; otherwise we assume that y_i has been centered.

The axes can be rotated by multiplying each y_i by an orthogonal matrix A as shown in equation (1)

$$z_i = Ay_i . \quad (1)$$

Since A is orthogonal, A'A = I, and the distance to the origin is unchanged as shown in equation (2)

$$z'_i z_i = (Ay_i)'(Ay_i) = y'_i A'Ay_i = y'_i y_i \quad (2)$$

Thus an orthogonal matrix transforms y_i to a point z_i that is the same distance from the origin, and the axes are effectively rotated.

Finding the axes of the ellipsoid is equivalent to finding the orthogonal matrix A that rotates the axes to line up with the natural extensions of the swarm of points so that the new variables (principal components) $z_1, z_2, \ldots, z_p$ in z = Ay are uncorrelated. Thus we want the sample covariance matrix of z, $S_z = ASA'$, to be diagonal as shown in equation (3)

$$S_z = ASA' = \begin{pmatrix} S_{z1}^2 & 0 & \cdots & 0 \\ 0 & S_{z2}^2 & \cdots & 0 \\ \vdots & \vdots & & \vdots \\ 0 & 0 & \cdots & S_{zp}^2 \end{pmatrix} \quad (3)$$

Where S is the sample covariance matrix of $y_1, y_2, \ldots, y_n$. Doing, $C'SC = D = \text{diag}(\lambda_1, \lambda_2, \ldots, \lambda_p)$, where the λ_i 's are eigenvalues of S and C is an orthogonal matrix whose columns are normalized eigenvectors of S. Thus the orthogonal matrix A that diagonalizes S is the transpose of the matrix C as shown in equation (4)

$$A = C' = \begin{pmatrix} a_1' \\ a_2' \\ \vdots \\ ap' \end{pmatrix} \quad (4)$$

Where a_i is the ith normalized ($a'_i a_i = 1$) eigenvector of S. The principal components are the transformed variables $z_1 = a'_1y$, $z_2 = a'_2y, \ldots, z_p = a'_py$ in $z = A_y$. For example, $z_1 = a_{11} y_1 + a_{12} y_2 + \cdots + a_{1p} y_p$.

The diagonal elements of ASA' matrix are eigenvalues of S. Hence the eigenvalues $\lambda_1, \lambda_2, \ldots, \lambda_p$ of S are the variances of the principal components $z_i = a'_iy$ as shown in equation (5)

$$S_{z1}^2 = \lambda_i \quad (5)$$

Since the rotation lines up with the natural extensions of the swarm of points, $z1 = a'_{1y}$ has the largest variance and $zp = a'_py$ has the smallest variance. This also follows from above, because the variance of z_1 is λ_1, the largest eigenvalue, and the variance of z_p is λ_p, the smallest eigenvalue. If some of the eigenvalues are small, we can neglect them and represent the points fairly well with fewer than p dimensions. For example, if p = 3 and λ_3 is small, then a two-dimensional representation will adequately portray the configuration of points.

Because the eigenvalues are variances of the principal components, we can speak of "the proportion of variance explained" by the first k components as shown in equation (6)

$$\text{Proportion of variance} = \frac{\lambda_1 + \lambda_2 + \cdots + \lambda_k}{\lambda_1 + \lambda_2 + \cdots + \lambda_p} = \frac{\lambda_1 + \lambda_2 + \cdots + \lambda_k}{\sum_{j=1}^{p} s_{jj}}, \quad (6)$$

Thus we try to represent the p-dimensional points $(y_{i1}, y_{i2}, \ldots, y_{ip})$ with a few principal components $(z_{i1}, z_{i2}, \ldots, z_{ik})$ that account for a large proportion of the total variance.

If the variables are highly correlated, the essential dimensionality is much smaller than p. In this case, the first few eigenvalues will be large, and the proportion of variance will be close to 1 for a small value of k. On the other hand, if the correlations among the variables are all small, the dimensionality is close to p and the eigenvalues will be nearly equal. In this case, the principal components essentially duplicate the variables, and no useful reduction in dimension is achieved.

Any two principal components $z_i = a'_iy$ and $z_j = a'_jy$ are orthogonal for i NOT = j;

that is, $a'_i a_j = 0$, because a_i and a_j are eigenvectors of the symmetric matrix S. Principal components also have the secondary property of being uncorrelated in the sample, that is, the covariance of z_i and z_j is zero as shown in equation (7)

$$s_{zizj} = a'_i S a_j = 0 \text{ for } i \neq j \quad (7)$$

If we change the scale on one or more of the y's, the shape of the swarm of points will change, and we will need different components to represent the new points.

Hence the principal components are not scale invariant. We therefore need to be concerned with the units in which the variables are measured. If possible, all variables should be expressed in the same units. If the variables have widely disparate variances, we could standardize them before extracting eigenvalues and eigenvectors. This is equivalent to finding principal components of the correlation matrix R.

Generally, extracting components from S (Covariance Matrix) rather than R (Correlation Matrix) remains closer to the spirit and intent of principal component analysis, especially if the components are to be used in further computations. However, in some cases, the principal components will be more interpretable if R is used. For example, if the variances differ widely or if the measurement units are not commensurate, the components of S will be dominated by the variables with large variances. The other variables will contribute very little. For a more balanced representation in such cases, components of R may be used.

As with any change of scale, when the variables are standardized in transforming from S to R, the shape of the swarm of points will change. Note, however, that after transforming to R, any further changes of scale on the variables would not affect the components because changes of scale do not change R. Thus the principal components from R are scale invariant.

VI. CASE STUDY

The case study of this work was carried out at IVIA, with 250 employees and ISO 9001:2000 certified since 2003, it is a software development company with a research and development area (R&D).

The measurement process of this organization was defined and institutionalized [5] from the proposed Organizational Software Measurement Process (OSMP), passing through the following stages: (i) Process specialization for the organization; (ii) Process instantiation in three pilot projects; (iii) Institutionalization of the process and (iv) Evaluation of the Organization based on the Measurement.

Organizational indicators were defined [4], which we used for comparisons among projects results. The organizational indicators are showed in table 1.

Table 1 – Organizational indicators

Indicators	Acronyms	Value
Accuracy of time estimates	ATE	95%
Deterioration of software	DS	93%
Rework index	RI	90%
Accuracy of the effort estimates	AEE	90%
Cost accuracy	CA	90%

In principal component analysis, we seek to maximize the variance of a linear combination of the variables. In our case, we might want to rank projects on the basis of their scores on properties related with ATE, DS, RI, AEE, and CA indicators. An average score would provide a single scale on which to compare the projects. Data from twelve projects were collected since the process implantation. These data were used to produce this paper. An additional project named Base was created using organizational indicators default value as showed in table 2. The Base project is used to help understanding how the projects are comparing with it.

Table 2 – Collected values for each project

Projects	ATE	DS	RI	AEE	CA
A	0.6275	1.0000	1.0000	0.5847	0.7039
B	0.8696	0.9040	0.9207	0.7001	0.9120
C	0.4043	0.8729	0.9592	-0.4961	-0.2159
D	0.7031	1.0000	1.0000	0.8068	1.0905
E	0.7656	1.0000	1.0000	0.9842	1.0739
F	1.0000	1.0000	1.0000	0.9776	1.0211
G	1.1613	1.0000	1.0000	1.2405	1.3008
H	0.8996	1.0000	1.0000	1.2156	1.2411
I	0.7182	1.0000	1.0000	1.3477	1.2737
J	0.4375	1.0000	1.0000	0.9488	0.8494
K	0.8317	1.0000	1.0000	1.6717	1.6136
L	0.4756	1.0000	1.0000	1.6155	1.4191
Base	0.9500	0.9300	0.9000	0.9000	0.9000

Each indicator was used as a variable in the PCA (Principal Component Analysis). The PCA extraction can be done using covariance matrix or correlation matrix. In order to choose the best one for our problem a comparison using both matrixes was carried out.

The two last rows of table 3 are to explain how important the component is in order to explain the variable. The first component is responsible for explaining 62.7% of the data, the second component is responsible for explaining 25.8%, and so on. As we can see in the last row, the first and the second component are together responsible for 88.5%.

Table 3 – PCA extraction using correlation matrix

Variable	PC1	PC2	PC3	PC4	PC5
ATE	-0.217	0.705	-0.664	0.066	0.1
DS	-0.517	-0.267	-0.203	-0.785	-0.065
RI	-0.367	-0.595	-0.445	0.554	0.073
AEE	-0.522	0.166	0.461	0.111	0.689
CA	-0.527	0.223	0.326	0.246	-0.711
Proportion	0.627	0.258	0.096	0.016	0.003
Cumulative	0.627	0.885	0.981	0.997	1

Table 4 shows that using a covariance matrix the first component is responsible for explaining 95.5% of the variables which is 7% greater than the first two components produced by the correlation matrix.

Table 4 – PCA extraction using covariance matrix

Variable	PC1	PC2	PC3	PC4	PC5
ATE	-0.160	0.979	0.104	-0.062	0.027
DS	-0.044	-0.041	0.094	-0.682	-0.723
RI	-0.018	-0.082	0.144	-0.705	0.689
AEE	-0.746	-0.057	-0.658	-0.075	0.034
CA	-0.645	-0.171	0.725	0.168	-0.016
Proportion	0.955	0.039	0.004	0.002	0
Cumulative	0.955	0.994	0.998	1	1

Chart 1 shows how representative the first component is using the covariance matrix and it was the reason we have chosen to use this component.

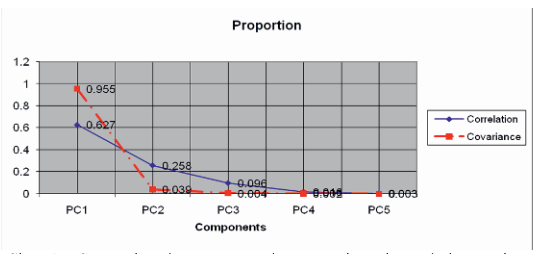

Chart 1 – Comparison between covariance matrix and correlation marix

In both cases, extracted components are negative and it means that better projects will appear at the extreme negative values as can be observed on chart 2.

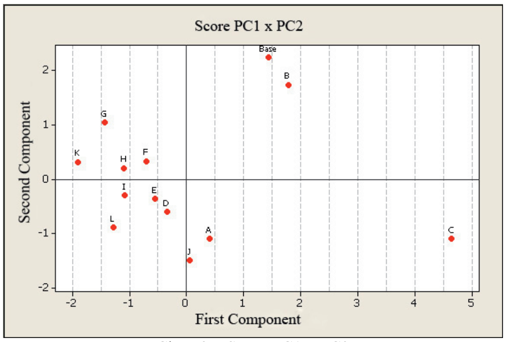

Chart 2 – Score PC1 x PC2

In order to have a better visualization we have inverted the results of the first component by multiplying it by -1. The results produced after the inversion will show better projects in the extreme positive values. The next step is to multiply the value of each variable for its corresponding PCA and sum the results in order to produce a score, where $Y_{<project>}$ represent it.

Y_A = 0.6275 x (0.160) + 1 x (0.044) + 1 x (0.018) + 0.5847 x (0.746) + 0.7039 x (0.645) = 1.052650444

Y_{Base} = 0.95 x (0.160) + 0.93 x (0.044) + 0.90 x (0.018) + 0.90 x (0.746) + 0.90 x (0.645) = 1.461020000

Doing the same to the other projects will produce a new value we called score. This score is used in the rank process. The Base project is a pattern project created with the reference values previously defined in the Organizational Planning for Measurement Macroactivity.

We have created a scale for evaluating the projects, using as a reference the Project Base. This scale is represented by the alphabet letters and represents bands of values. The tracks of values and their scores must be defined according to the organization and in general vary from organization to organization.

Before the projects classification in the scale, it is necessary normalize the data. In order to normalize the data we have used a simple calculation. First we have found a constant the we have called I and is showed in formula 8 and we have applied in formula 9.

$$I = \frac{1}{V_{Max} - V_{Min}} \tag{8}$$

$$V_{normalized} = 1 - (I \times (V_{\max} - V)) \tag{9}$$

We have defined four bands. In band A only project with normalized value over 30% of the Base project normalized score is accepted. In band B we can find all project which have normalized score above the Base project normalized score but below the band A. In band C we have put projects that theirs normalized score are below Base project normalized score but not over -10% of this score. In Band D we have put all other projects.

The result of the process is showed in table 5

Table 5 – Project sorted by score

Project	Score	Normalized	%	Band
K	2.482923764	1,000,000	36%	A
L	2.258589131	0,921888	28%	A
G	2.012263667	0,836118	19%	A
I	2.003799437	0,833171	19%	B
H	1.913255340	0,801644	16%	B
E	1.611353759	0,696523	5%	B
F	1.609905937	0,696019	5%	B
D	1.479768395	0,650705	1%	B
Base	1.461020000	0,644177	0%	B
J	1.387676572	0,618639	-3%	C
B	1.306028482	0,590210	-5%	C
A	1.052650444	0,501984	-14%	D
C	-0.389020774	0,000000	-64%	D

The final rank can be also visualized in chart 3. Projects above the base horizontal line are good projects. The ones below the base line are unsuccessful projects.

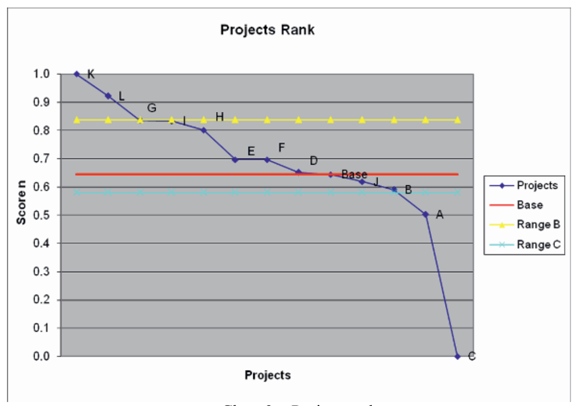

Chart 3 – Project rank

VII. CONCLUSIONS AND FUTURE WORK

This paper has showed a new methodology to rank organization projects using simple and efficient steps and producing an aggregate score from the project indicators in order to identify the best and worst projects from a pool.

Now the company can use the results to detect which projects were successfully finished and which were not according to the company baseline. Comparatives can be made between projects using theirs indicators

Successful projects can be analyzed in order to identify good practices in order to improve future projects. Unsuccessful projects can be analyzed in order to identify weak points and avoid the same mistakes on other projects.

In both cases, the lessons learned plus the result of analysis can produce a knowledge base to help improve the organizational processes.

For future work we are studying the addition of more indicators in order to increase the accuracy of the methodology.

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Modeling of Head-Related Transfer Functions Through Parallel Adaptive Filters

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Abstract-**Currently, sound spatialization techniques that utilize "individual" Head-Related Transfer Functions (HRTFs) require the intended listener to undergo lengthy measurements with specialized equipment. Alternatively, the use of generic HRTFs may contribute to additional localization errors. A third possibility that we are pursuing is the customization of HRTFs, performed on the basis of geometrical measurements of the intended listener to determine the appropriate parameters in a structural HRTF model. However, an initial step of decomposing measured HRTFs in order to reveal the parameters of the structural model must be performed. A new approach for the decomposition of HRTFs is suggested and evaluated on simulated examples. The potential of this method for the decomposition of measured HRTFs is discussed.**

Keywords-HRTF, HRIR, Sound Localization, Pinna, Signal Decomposition, Adaptive System Identification

I. Introduction

Virtual localization of sound in 3D space has been an interest of researchers for some time now. Currently, 3D audio is used in audio production, movies, aviation, video games and assistive technology for the blind.

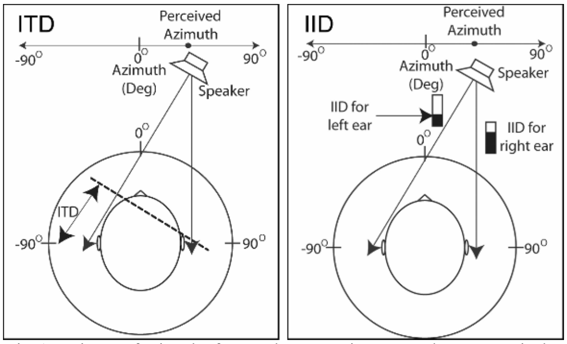

Fig. 1: Estimate of azimuth of a sound source using ITD and IID respectively.

It is well known that two key binaural cues used to localize azimuthally are the left-right time and intensity differences, called interaural time difference (ITD) and interaural intensity difference (IID) (see Fig. 1). However, these cues do not account for the elevation information of the sound source. It is believed that the cues for the elevation localization are contained in the pair of transfer functions that model the spectral changes a sound experiences as it propagates from the sound source to each of the listener's ear drums, termed Head-Related Transfer Functions (HRTFs), which depend on the anatomical features of the listener [1]. Because of the independent spectral processing that each HRTF in a pair (left, right) implements, it is believed that HRTFs contain additional monaural cues, which are the prominent indicators of elevation, distance and frequency. Furthermore, it is believed that the pinna, or outer ear, despite being "weakly azimuth dependent" contributes heavily to elevation cues [2].

If a pair of HRTFs can be experimentally measured, a monaural sound wave can then be filtered by the HRTFs and when the resulting binaural sound waves are played to the listener they would make it appear as if the sound were emanating from the desired location in 3D space, specified by azimuth (θ), elevation (Φ) and distance (r) (see Fig. 2). This is the basis for the development of synthetic 3D audio. The most accurate way of obtaining HRTFs is to measure the intended listener using specialized equipment (e.g., "HeadZap" System, AuSIM, Palo Alto, CA). These systems measure the time domain equivalent of the HRTF, which is referred to as the Head-Related Impulse Response (HRIR). In practice, these systems replace a theoretical impulse input with Golay Codes [3] or Maximum Length Sequences (MLS) [4] and record the response from microphones placed at the ear canal of the intended listener (see Fig. 3).

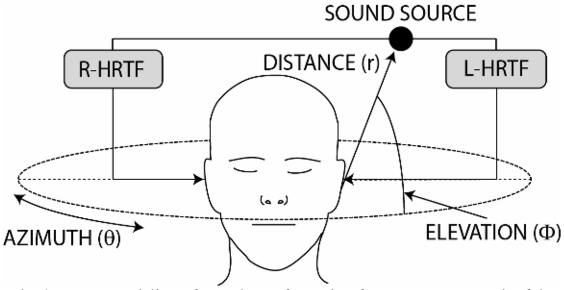

Fig. 2: HRTF modeling of sound transformation from a source to each of the listener's eardrums.

Ideally, the cues needed for elevation localization would be obtained by measuring the HRTFs form every intended listener. However, the impracticality of this has prompted researchers to seek alternative methods of generating HRTFs

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for the synthesis of 3D audio. A majority of commercial developers have resorted to using what is often referred to as "generic" HRTFs which are measured on a KEMAR mannequin with anatomical features that represent the average body type (e.g., MIT's measurements of a KEMAR Dummy-Head Microphone [4]) or a database of measured subjects (e.g., the CIPIC Database [5]).

Generic HRTFs have been successful in bringing binaural 3D audio to the general public, but it has been shown that they result in an increase in localization errors especially in elevation [1]. Numerous other methods have been developed to reduce the localization errors (see the background section of [6] for a detailed description of the available methods). In this context, our group is pursuing the development of a structural pinna model that could be used to generate customized HRTFs from physical measurements of the pinna.

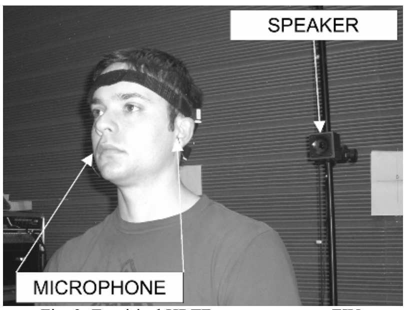

Fig. 3: Empirical HRTF measurement at FIU.

II. Structural Pinna Models

In [7], Algazi developed a spherical-head model derived from interaural time differences (ITDs) in which he successfully associated the head radius for the model to a weighted sum of three head dimensions (head height, width and length). The results were promising in that most of the testing subjects were able to localize accurately within 5º. Brown and Duda developed a structural model that accounts for the effects of the head, shoulder and pinna by cascading them together to generate HRTFs [8]. However, the parameters for the model were not defined from measurements of anatomical features of the individual users.

One recently proposed model that also pursues customization with the use of anatomical features was developed by Satarzadeh [6]. The basic idea behind this model is to replicate the magnitude response of an HRTF for a desired position in 3D space using two band pass filters and a (fixed) comb filter. The parameters for the filters were obtained from the pinna measurements of the intended listeners and the resulting HRTFs displayed a good approximation to the measured HRTFs in the frequency and time domains. However, Satarzadeh admits that his model will only work on subjects that exhibit prominent pinna reflections.

Working towards a similar goal, the model we are attempting to develop would use the pinna measurements to "tune" a database of generic HRTFs so that their characteristics would resemble those of measured HRTFs. However, one of the first difficulties faced by our group was to transform the format in which the HRTF information is obtained from the measurement systems (as 256 or more values of the HRIR sequence, which does not have a simple relationship with the pinna features).

A model that proposed a simple representation of HRTFs was developed by Batteau in [9]. He noticed that the peak of the impulse response of the HRTFs changed as the azimuth and elevation of the sound source was changed. The model consisted of a direct path (unity gain) placed in parallel with two indirect paths in which the input signal is scaled by reflection coefficients (ρ) and delayed by lags (τ). Fig. 4 shows an example of how a sound wave would enter the ear canal according to this model. On the other hand, Shaw and his colleagues in [10] identified numerous resonant modes of the pinna and the directions from which the resonances were most prominent. The first mode, which is excited in all directions of arrival, has a resonant frequency of 4.2 kHz. The other modes fall into two groups: a vertical pair (modes 2 and 3) and a horizontal triplet (modes 4, 5 and 6). Mode 4 is closest to 0^{O} elevation with a resonant frequency of 12.1 kHz. Consolidating Batteau's and Shaw's concepts, our group seeks to develop a model in which we break down measured HRIRs into partial components, likely related to physical dimensions of the pinna and how they introduce resonances and reflected paths in the overall HRIRs that are recorded experimentally.

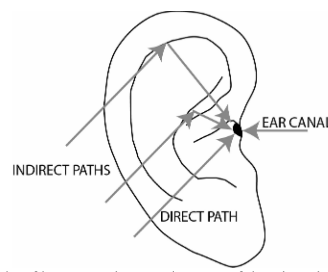

Fig. 4: Example of how sound waves bounce of the pinna into the ear canal.

Our previous models consisted of a resonator feeding into one direct and three indirect paths [11, 12]. The indirect paths modeled reflected waves bouncing off the structures of the pinna before entering the ear canal. However, the reflection coefficients (ρ) and the delays (τ) where not known. Our initial method to estimate these parameters consisted of manually windowing the HRIR and applying a signal approximation method, such as Prony, to the window [11]. It was assumed that if the window size was selected such that it only contained one resonance "echo", it could be estimated by a second order approximation, extrapolated and removed form the overall HRIR to study subsequent echoes separately.

This proved to be a tedious task that required a large amount of human interaction for the estimation of the best window size to use in each step, which also weakened the objectivity of the process. This prompted our group to develop an automated method of extracting the desired resonances and their parameters. In [12], there is a thorough description of the iterative and pole-decomposition methods developed by our group. These methods were able to decompose HRTFs into pieces that, when reassembled, were a close approximation to many original HRTFs but they did not take advantage of the knowledge available about the number and frequencies of resonances that are likely to be implicit in measured HRIRs, as reported by Shaw [10]. This prompted our group to develop an alternative HRIR decomposition method which uses that knowledge.

III. Parallel-Comb Based Pinna Model

A comb based pinna model has been previously proposed in [13]. Raykar et al. attempted to detect the spectral notches introduced by the pinna and used them to determine the characteristics of multiple comb filters arranged in parallel that make up their HRTF model. However, it was pointed out that assuming each pinna notch corresponds to a comb filter is not likely and results in a complex comb structure [14]. Despite this, the idea of using combs in parallel is appealing because it allows the implementation of the concepts suggested by Batteau and Shaw. Consider the structure in Fig. 5. This structure models two predominant resonances occurring in the ear and the two alternative paths for each resonance represent (two) different routes toward the ear canal that sound affected by each resonance may follow. Each of the paths can arrive to the ear canal at different times, τ_{ij}, and can have experienced attenuation by different reflection coefficients, ρ_{ij}. The pinna model shown in Fig. 5 only requires 12 parameters (each resonator is represented by two parameters) and can be "cascaded" with Algazi's functional head model to represent a complete HRIR.

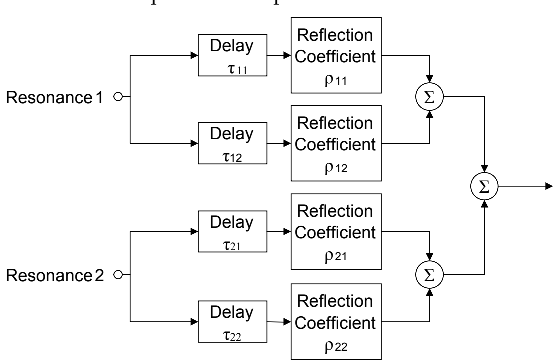

Fig. 5: Block diagram of the pinna model.

The parallel-comb structure of this model allows for the consideration of two of Shaw's modes and accounts for the reflections off various pinna structures described in Batteau's model. However, obtaining the reflection coefficients (ρ) and the delays (τ) from an HRTF is not a straightforward process. The next section will describe a method our group has developed to extract the resonances and parameters from simulated HRIRs that conform to our basic assumptions.

IV. Method Of Decomposition

To investigate the potential of the proposed method, this paper discusses its application to synthetic HRIRs created according to the model displayed in Fig. 5. This is a preliminary step to actually decomposing HRTFs and will be used to validate that our new method can extract the resonances and parameters in an ideal setting. The following describes the steps taken.

A. Simulation of Resonances and Comb Filters

The resonances used in this example where simulated using Eq. (1) to create their impulse responses, where N is the length of the signal, n = 0,…,N-1, d_i is the negative damping factor, ω_i is the digital frequency and θ_i is the phase. The frequencies for the resonances were selected according to Shaw's resonance modes. As mentioned previously, mode 1 is an omni-directional mode with a frequency of 4.2 kHz. Additionally, mode 4 (which corresponds to a frequency of 12.1 kHz) was selected as the second resonance for the simulation because it is closest to an elevation of 0^O [10]. For simplicity, the signals created in this study were artificially set to correspond to these mode frequencies. The damping factor (d_i) was selected to give the resonators a reasonable decay. Phase (θ_i) was set to zero for simplicity. Fig. 6 shows examples of the impulse responses of a 12.1 kHz resonator (left) and a 4.2 kHz resonator (right). We will refer to these resonator impulse responses as "echoes" for the reminder of the paper.

$$x_i(n)=d_i^n\cdot\sin(n\cdot\omega_i+\theta_i) \qquad (1)$$

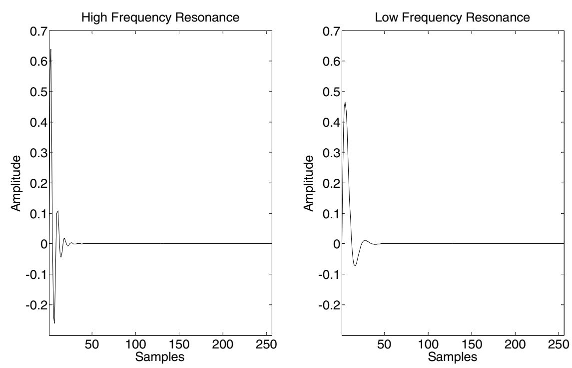

Fig. 6: Example resonances generated using Eq. (1).

$$H(z)=\rho_{i1}z^{\tau_{i1}}+\rho_{i1}z^{\tau_{i1}} \qquad (2)$$

Once the resonators have been simulated, the comb-filters are implemented using the Z-domain representation of their transfer function (see Eq. (2)) characterized by their reflection coefficients (ρ) and their delays (τ). It is not known exactly how the decaying sinusoidal echoes produced by the resonances are delayed when they bounce off the various structures of the pinna or outer ear. For example, Satarzadeh implies with his model that all the echoes of a resonance will have the same delay [6]. We believe that the echoes might enter the ear canal at different times [11, 12]. In this study we simulated several different scenarios.

B. Estimation of Resonant Frequencies

The decomposition method described in this paper starts by estimating the resonant frequencies present in the HRIR to be decomposed by application of an approximation method such as Prony or Steiglitz-McBride (STMCB). These methods, when given the impulse response of a system, approximate the transfer function coefficients. The denominator portion of the resulting transfer function contains the frequency and dampening information of the resonances. A comparison of the methods was made and it was determined that STMCB outperformed Prony in a majority of cases [15]. Hence, this study uses STMCB for the experimentation.

STMCB requires that the order for the numerator and denominator of the transfer function sought be specified in advance. However, it seems that STMCB is capable of estimating the frequencies when the orders are set to a high order. The reflection coefficients (ρ) of the model cannot be obtained through this method. The method used to estimate the reflection coefficients is described in a later sub-section.

C. Extraction of High Frequency Comb Filter

The most crucial part of this entire method is the isolation of the higher frequency resonance (in this study simulated at 12.1 kHz). The filter that worked best for the examples considered in this study was a 64th order FIR least squares high-pass filter with a cut off frequency of 5 kHz which was used to suppress the omni-directional resonance (4.2 kHz). It is known that FIR filters introduce a delay, which is dependant on the order of filter. Hence, since the filter order was 64, the result of the filter is shifted to the left 32 samples. Once this is done, a close approximation of the comb-filter for the higher frequency resonance ("Recovered Comb Filter") is obtained by allowing an adaptive transversal filter ("LMS") to adapt as described below, using the Least-Mean Square adaptation algorithm.

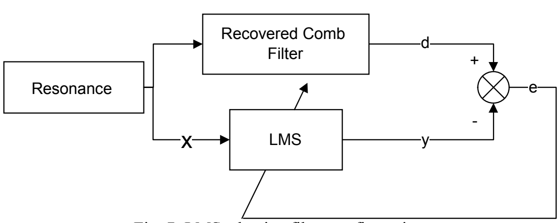

Fig. 7: LMS adaptive filter configuration.

The configuration for the LMS adaptive filter used is shown in Fig. 7. The input, x, fed into the adaptive filter ("LMS") is the decaying sinusoidal impulse response from the resonance obtained using the STMCB approximation method on the high-pass filtered HRIR signal. The signal used as "desired signal" (d) for the adaptation process is the high-pass filtered HRIR signal (Both x and d were replicated multiple times to allow the adaptation to converge). Fig. 8 shows an example result of the LMS adaptive filter. Ideally the LMS would only find two delay locations. Unfortunately, there are artifacts introduced by the high-pass filter, which cause the existence of some additional non-zero weights in the adapted LMS filter. However, as seen in Fig. 8, the dominant delay locations are much larger and visually easy to identify (in this example they are at 1 and 6). The effective echo latencies can be extracted automatically by applying two thresholds, as shown. The upper threshold is the mean of the result of the LMS plus one standard deviation and the lower threshold is the mean minus one standard deviation. Once these steps are completed, the resulting comb filter is convolved with the high frequency resonance (12.1 kHz) obtained from the STMCB approximation method (the result of this convolution will be referred to as COMB_HIGH).

It will be shown in the results and discussion section that this process detects the delays correctly. As mentioned earlier, the reflection coefficient (ρ) could not be estimated using STMCB approximation method and, due to the artifacts introduced by the filtering process, it cannot be determined by the LMS adaptive filter with enough accuracy. The next sub-section describes the processed used to estimate the reflection coefficients of the comb filter for the high frequency resonance.

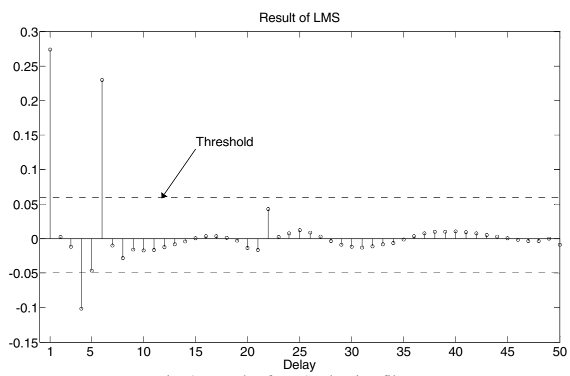

Fig. 8: Result of LMS adaptive filter.

D. Estimation of the Reflection Coefficients

The comb filter for the lower frequency (4.2 kHz) resonance is not obtained in the same way as the comb filter for the higher frequency (12.1 kHz). Initially, a low pass filter was attempted to isolate the lower frequency comb filter. However, the smoothing effect of the low pass filter seemed to disrupt the ability of the LMS adaptive filter to isolate the delay parameters. It was then proposed that a simple subtraction of the result from the previous section should isolate the comb filter for the lower frequency resonance but this depends heavily on the accuracy of the estimated reflection coefficients.

The high-pass filter used to isolate COMB_HIGH alters the true value of the reflection coefficients in an undetermined way. A method that seemed to work the best was an iterative process in which the amplitude of COMB_HIGH is raised incrementally. This incremental approach requires that the amplitude of COMB_HIGH be initially reduced to ensure that it is smaller in amplitude than the real comb filter for the high frequency resonance. A simple reduction on the entire COMB_HIGH is not sufficient because each reflection coefficient has a different value. Empirically it was determined that a close approximation to the real comb filter for the high frequency resonance is achieved if COMB_HIGH is reduced by 30% from the beginning to the first delay location and reduced by 40% for the rest of the signal.

Once COMB_HIGH has been reduced, the iterative process begins. The first step is to subtract COMB_HIGH from the original signal (ORIG) which is the result of the model in Fig. 5. The remaining signal from the subtraction is an approximation of the low frequency resonance signal processed through a comb filter, as shown in Fig. 5. An LMS adaptive filter (Fig. 7) is again used to estimate the location of the delay for the comb filter for the low frequency resonance signal. This time the 'x' signal for the LMS adaptive filter is the impulse response of the lower frequency resonance (4.2 kHz) obtained by the STMCB approximation method from the original signal (ORG) and the'd' signal is the low frequency signal estimated as ORG – COMB_HIGH. Here, also, x and d had to be replicated multiple times, to allow the LMS filter to converge. The resulting LMS adaptive filter weights are also thresholded to isolate the delay locations for the lower frequency comb filter.

The resulting comb filter is then convolved with the impulse response of the low frequency resonance (4.2 kHz) obtained from the STMCB approximation method (the result of this convolution will be referred to as COMB_LOW). Finally, COMB_HIGH and COMB_LOW are summed together to obtain an approximation of the result of the model in Fig. 5 (this will be referred to as MODEL_ESTIM). MODEL_ESTIM is then compared to ORIG using Eqs. (3)-(4) where MS is the mean square value. This iterative process will be repeated in which the amplitude of COMB_HIGH will be incremented by 5%, which was determined empirically, across the entire signal. The iterative process is stopped when the fit value starts to decrease. At this point, the parameters are kept as the closest approximation to the true parameters.

$$Error = ORIG - MODEL\_ESTIM \tag{3}$$

$$fit = 1 - \frac{MS(Error)}{MS(ORIG)} \tag{4}$$

IV. Results And Discussion

The tests performed to validate the method presented in this paper used resonances that where synthetically created but have characteristics that HRTFs are expected to possess. The sampling frequency that was simulated was 96 kHz because the AuSIM HeadZap HRTF measurement system at Florida International University has the ability to record HRTFs at that sampling rate. As mentioned in the earlier sections, it is not exactly known how the decaying sinusoidal echoes are altered as the sound wave travels to the ear canal. Hence, the parameters, such as delay and reflection coefficients, used in our simulated tests were selected to represent some possible combinations that may approximate real parameter values. Table 1 below lists the parameters and results of the tests performed, where the "fit" of the reconstructed HRIR to the original is given by Eq. (4).

TABLE I
Test Parameters And Fits

Test	τ_{11} τ_{12} τ_{21} τ_{22}	ρ_{11} ρ_{12} ρ_{21} ρ_{22}	d_1 d_2	f_1 f_2 (kHz)	fit (%)
1	1 6 3 22	1 0.5 0.75 0.3333	0.85 0.8	12.1 4.2	96
2	1 5 3 8	1 0.5 0.75 0.3333	0.9 0.85	12.1 4.2	95
3	1 6 3 22	1 0.5 0.75 0.3333	0.9 0.85	4.2 12.1	99
4	1 6 4 9	2.4767 -1.5697 1.2347 -0.7965	0.9 0.85	4.2 12.1	99
5	1 6 1 6	2.4767 -1.5697 1.2347 -0.7965	0.9 0.85	4.2 12.1	97

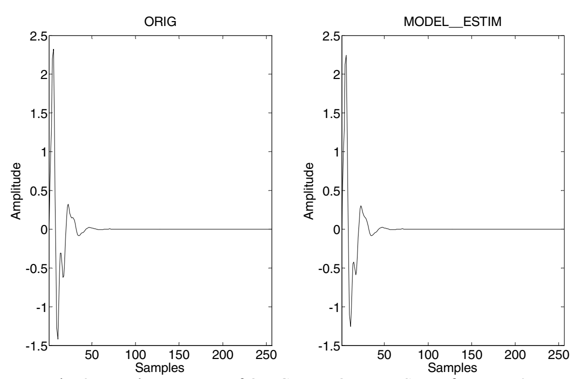

Fig. 9: Impulse response of ORIG vs. MODEL_ESTIM from test 4.

V. Conclusion

All the tests resulted in high fits (above 95%) when the impulse responses ORIG and MODEL_ESTIM are compared (see Fig. 9 and Fig. 11). Additionally, the magnitude responses of the all the MODEL_ESTIM seem to follow the ORIG magnitude responses closely (see Fig. 10 and Fig. 12). As can be seen in the magnitude responses, the peaks were detected at the exact locations. Furthermore, the method was able to determine the delay locations of the comb filters. Interestingly, even when the delay for both comb filters is identical, such as in Satarzadeh's model [6, 14], the method is still able to detect the resonances and the delay locations (test 5 was the test for this scenario).

Although the method performed well under ideal conditions, it is yet to be tested on real HRTFs. It has already been realized that additional resonances that may be present in HRTFs could influence the outcome of this method. One possible solution that will be explored is the use of multiple band stop filters, rather than a single high pass filter, to suppress unwanted additional resonances when a given frequency is being processed. Further research is needed to validate the use of this method with HRTFs.

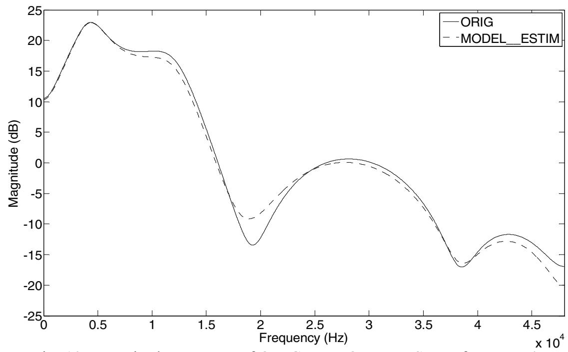

Fig. 10: Magnitude response of ORIG vs. MODEL_ESTIM from test 4.

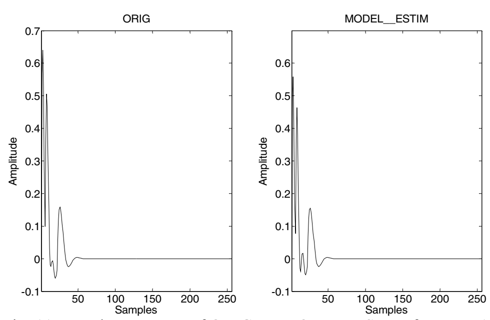

Fig. 11: Impulse response of ORIG vs. MODEL_ESTIM from test 1.

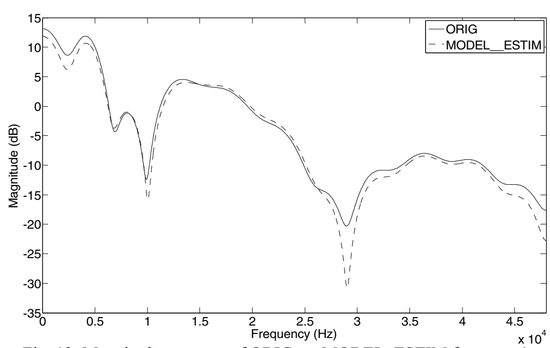

Fig. 12: Magnitude response of ORIG vs. MODEL_ESTIM from test 1.

ACKNOWLEDGMENT

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A Simple Way for Preparing to a CMMI Appraisal

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***Abstract*— This paper presents the results of a research carried out in five companies of the Brazilian telecommunications market: Vivo, TIM, Claro, Embratel and Intelig. The aim was to estimate the maturity level of the software development processes, according to the CMMI (*Capability Maturity Model Integration*) criteria. The chosen instrument was a questionnaire applied to IT (*Information Technology*) teams from each company, which made possible to estimate the current maturity level and identify improvement points, in a much faster and economic way compared to traditional appraisals, such as SCAMPI (*Standard CMMI Appraisal Method for Process Improvement*). The measured maturity levels were low for almost every team selected as a sample.**

***Index Terms*—CMMI, Maturity level, Telecom, Process management, Product management, Process quality, Process maturity, Human and technological factors**

I. Introduction

The concept originally proposed by Carnegie Mellon's Software Engineering Institute (SEI) of *process maturity* [1], based upon the principle that "the quality of the product is directly dependent on the process that has developed it" is nowadays widely accepted, in both the open market and the academy. As an example, Bradford Clarke sustained in his paper: "The results of data analysis from the 161 projects showed that PMAT is statistically significant, supporting the hypothesis that increasing process maturity decreases the effort to develop a software product" [2]. The term "maturity" refers to the *predictability* for the quality of the products created by a company, which means that companies with mature processes tend to produce with a known level of quality, while the quality of products generated by companies with immature processes may vary enormously. There are many companies willing to invest large financial resources and years of work in order to turn their processes into mature ones, but the path to maturity is long and hard, and there are some issues that could compromise all these efforts.

A formal SCAMPI appraisal is considerably expensive (over US$ 25,000.00), but is not the major cost to be considered by a company wishing to grow mature in terms of processes; in order to get ready for such an appraisal, the company should hire one or more specialized consultants, involve all its staff and invest about two years of planning and huge amounts of money to be at least able to be appraised for a single level of maturity on the CMMI scale. Hossein Saiedian and Richard Kuzara affirmed that "One nearly universal complaint is that moving from level to level can cost hundreds of thousands or even millions of dollars" [3]. This is already a challenge for a company such as a *software house*, that would benefit greatly from a recognized certification such as the CMMI one; in fact, all small and most medium companies can not afford these expenses.

There is another kind of company that could benefit indirectly from the CMMI principles, whose business is only supported by IT, such as the telecommunications segment. Getting more mature processes could mean to add quality to the products and services offered by the company, consequently adding value in the eyes of the customers, something very interesting in a highly competitive market such as the Brazilian one. Unfortunately, the cost and efforts of a Quality Journey are so high that most of CEOs (*Chief Executive Officers*) would hardly foresee a positive ROI (*Return of Investment*) on such a venture. Consequently, most companies do not have even an estimate of the gap between their current state and a mature level.

The work described on this paper has the purpose of offering a quick and economic way to estimate this gap; by means of applying a simple questionnaire to IT teams, it is possible to identify many improvement points, and therefore estimate the effort needed to possibly attain a maturity level in the future. A field research was conducted in the telecommunications sector of the State of Rio de Janeiro, Brazil, and its goal was to generate *primary data* for further studies.

II. Theoretical Framework

A. History

The CMMI is an extension from the well-known CMM, which is a capability and maturity model designed for software development. CMM has been created two decades ago and is nowadays very widespread. Hansen, Rose and Tjornehoj performed a literature revision in 2004, aiming to identify the dominant ideas regarding the subject SPI (*Software Process*

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K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 305–308.

Improvement). They found that 75% of the literature reviewed mentioned directly the CMM, and most of the rest was somehow influenced by SEI's concepts [4]. The original model was designed by SEI from Carnegie Mellon University, fulfilling a request from the American Department of Defense (DoD). The motivation was to find a way of differentiating software suppliers committed with high levels of quality from others less reliable. A group of specialists from SEI conducted a research upon the practices of companies recognized by the time as delivering high-quality services and products to their customers. Based on this research, they have built a theoretical model which they named CMM (now known as SW-CMM).

As the model spread across organizations, other maturity models were developed for other areas of knowledge, such as system engineering, product development and supplier sourcing. As each of these models were built independently from the others, soon the organizations realized that adopting one or more of these models concurrently was not a simple task, as there were many conflicts among them.

The solution was to integrate all the models into a single one, which could be adopted as a whole or as a subset by any organization. This final model was called CMMI (Capability and Maturity Model Integration).

B. Structure

There are two possible ways of representing the CMMI structure: "staged" and "continuous". Each company should select one of them, depending on its own needs. For the purpose of this paper, the staged representation [5] was chosen; therefore, the continuous one won't be described here. For further information about the continuous representation, see [6]. The staged representation has the following characteristics:

- Offers a proven sequence of improvements, beginning with basic management practices and evolving through a well-known path of successive levels;
- Allows comparisons inter and intra-organizations, through maturity levels;
- Allows an easy upgrade from CMM to CMMI;
- Offers a single score that summarizes the appraisal results, allowing comparisons among organizations.

This kind of representation, used since the original CMM (currently SW-CMM), splits the process of evolution in five levels or stages, each one indicating a degree of evolution onto the maturity scale. The first level represents a process not managed: it's the starting point for every organization. Level two indicates the presence of basic procedures of project management. Level three indicates the existence of a well-defined and integrated process, well-documented and institutionalized in the whole company, and so on. Table 1 shows the progression of maturity levels.

TABLE 1 MATURITY LEVELS

Level	Name
1	Initial
2	Managed
3	Defined
4	Quantitatively Managed
5	Optimizing

Each maturity level is composed of Process Areas (PAs). There are 25 PAs, which represent logical groupings of the good practices found by SEI at the companies considered to deliver high-quality services and products (see the "History" section above). Some few examples of PAs are: Requirements Management, Monitoring and Process Control, Verification, etc.

In order to verify its use in a company, each PA has a set of Generic and Specific Goals. The Generic Goals are the same for each PA, because they refer to the spreading and institutionalization of the PA into the company. On the other hand, the Specific Goals are different for each PA.

The model also indicates Practices and Sub practices commonly found in companies which have institutionalized some Pas; nevertheless, due to the peculiarities of each segment and company, the CMMI appraisal does not oblige the implementation of Practices to be identical to the ones suggested. Similar Practices are accepted, as long as the Goals are reached. Fig. 1 illustrates the relationships between each component of the model.

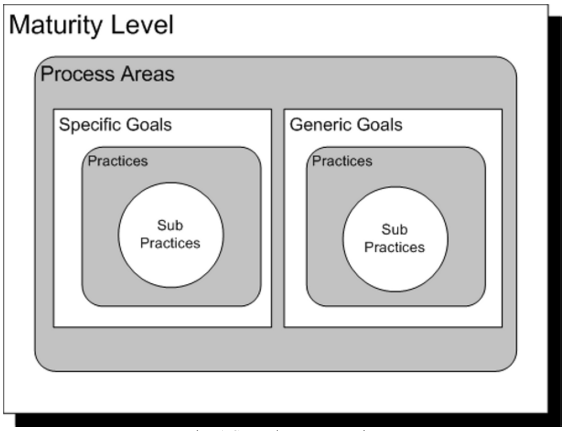

Fig. 1 Staged representation

III. METHODOLOGY

This paper is part of the research "Human and Technological Factors of Competitiveness", led by Prof. Heitor Luiz Murat de Meirelles Quintella [7]. It was developed according to a supervision method called ORIENTEL [8]. As describing the whole method is not the purpose of the present paper, in short we could define ORIENTEL as a production method of scientific work, composed of 4 phases and 12 steps. The research leader (Prof. Quintella) conducts weekly group sessions at a fixed place. They aim to integrate all researchers and promote the exchange of practices, theoretical frameworks, references and experience in general. Four big documenting tools are used: Field research notebook, Data accumulator (gathers all data in an indexed data base), Report accumulator (gathers all written material produced) and Bibliography accumulator (gathers all bibliographic references utilized). The production structure is based on the *locus*, the teaching strategy (learn by doing, focused on reality, with peer collaboration) and the co-authoring production with the research leader. ORIENTEL makes use of Karl Popper's hypothetic-deductive method [9].

The present work was done considering the universe of the research (population) as the telephony companies, both fixed and mobile, operating in the State of Rio de Janeiro. Six companies were selected as sample: TIM, Claro, Vivo, Embratel, Intelig and Oi/Telemar. Only the last one refused to take part into the research, claiming not having time to allocate employees to cooperate, due to a huge number of end-of-the-year activities.

The selection of the sample is justified by the great representativeness of the chosen companies. All of them have national range, and the three first ones (Vivo, TIM and Claro) share alone 81% of the Brazilian mobile market. The two NLD (*National Long Distance*) operators, Embratel and Intelig, are representative of another segment. Both have a strong presence at the corporative market, offering services like PABX (*Private Automatic Branch Exchange*), VPN (*Virtual Private Network*) and Internet access, among others.

In order to estimate the maturity level of the software development processes from the studied teams, a non-official questionnaire was used. This questionnaire was created and validated on a previous work from the research group ([10], [11] and [12]). It was composed of 25 questions (one for each PA) about good practices within the team. The purpose of each question was to identify if the specific goals of each PA were attained for the team being studied. In order to answer each question, the respondent had to choose a number between 1 and 5, indicating the presence of a certain good practice in the quotidian of the team, with the following meaning: (1) Very rarely or never; (2) Seldom; (3) Sometimes; (4) Often; and (5) Very frequently or always. As mentioned earlier in this document, this procedure allowed a quick and economical *estimate* of the maturity level, contrarily to an official appraisal like SCAMPI that would have been impracticable for the purpose of this research, considering the time and costs involved. This kind of experimental questionnaire based on SEI's style has already been tested with interesting results on previous works [13].

For the present work, the questionnaire was first submitted to a small group of people in order to check its clarity. After each person's feedback, some questions were rewritten for the sake of comprehension, carefully avoiding changes to the semantics of the text.

This questionnaire was then submitted to groups of volunteers of IT teams from five companies competing in the Brazilian telecommunications segment: Vivo, TIM, Claro, Intelig and Embratel. The leader of each IT team was also invited to fill in the questionnaire. An important note to observe is that no external developer (belonging to outsourcing teams temporarily hired) was included as a respondent, apart from the cases where the external developer was part of the IT team, responding directly to the leader of the team.

The questionnaire was made available on a website, protected by a pair key/password, unique by participant, e-mailed to each one. A few questionnaires were filled in by hand. The confidentiality of the data and the privacy of the participants were carefully preserved during the research. Each participant was instructed to answer without communication with their colleagues, and the names of the participants were not included in the presentation of the results, not even the companies' names.

Sixteen leaders and thirty-eight developers accepted to take part in the research, totalizing fifty-four valid questionnaires (incomplete or inconsistent questionnaires were discarded). Some open interviews were also conducted by the researcher, in order to collect some complementary data.

IV. RESULTS ANALYSIS AND DISCUSSION

A. Analysis

All valid questionnaires were consolidated on MS Excel sheets. Most part of the data was extracted from the website database and the rest was typed from the questionnaires filled in by hand, taking care of not compromising the integrity of the data.

For each question, in order to check if the objective of the related Process Area was fulfilled, the median of the answers was taken. The option for the median was because the chosen scale was qualitative and not quantitative; it would make no sense to calculate the average, but the median could indicate that at least half of the respondents had voted for or above a value in particular. A good practice was considered to be implemented only if a median of 4 or 5 (options "Often" and "Always") were found.

The result was grouped into Process Areas, and later to maturity levels. A Process Area was considered to be satisfied only if all questions related to it had obtained median of 4 or 5. A maturity level was only considered to be attained if all Process Areas that composed it were satisfied and all previous levels had been attained (according to CMMI methodology).

The maturity levels found were low (13 teams at level 1 and 3 at level 2), as illustrated in Fig. 2.

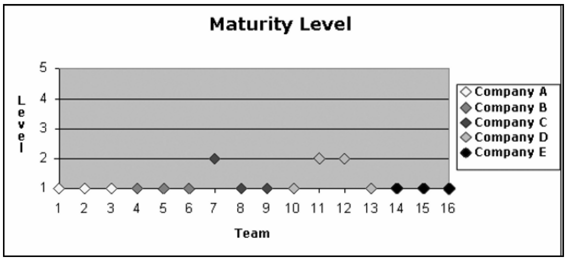

Fig. 2 Maturity levels found for 5 Brazilian telecom companies

B. Discussion

The low level of maturity found was expected and coherent, since the concept of maturity levels is not very common in the Brazilian telecommunications segment. The interviews revealed that most managers from the studied companies had heard of CMMI but never implemented it, and a few of them didn't know anything about it. There was a single exception, a team that had a direct contact with CMMI, brought by an outsourcing company ranked at CMMI level 3; after the end of the contract with this company, the team maintained many of the control practices, but without a rigorous supervision. This was one of the three teams that were classified at level 2 by the present analysis. The other two belong to a company that has adopted another quality model – COBIT – which presents some differences to CMMI, but has a lot of common practices.

V. CONCLUSIONS AND RECOMMENDATIONS

The purpose of this paper was to present a simple way for preparing to a CMMI appraisal. The methodology used allowed the following benefits:

1. It was very quick (after a few contacts and meetings, it took around fifteen days per company to fill out all questionnaires);
2. It was cheap (the researcher alone was able to perform all the processing tasks, as almost all of the data gathering was made by the employees themselves through a website);
3. It allowed the research team to get an estimate "screenshot" of the present maturity level of each IT team, as much as an idea of the maturity level for the whole telecommunications segment in Rio de Janeiro State;
4. It offered each participant the opportunity to analyze the practices and processes of their team, as well as indicating possible improvement points;
5. It allowed a simple way of turning subjective feelings from people involved directly in the development into objective data which could be used as the basis for investment discussions within the company. Many team leaders complained about having difficulty to argue the importance of investments in processes with the higher management.

The maturity levels found were low for almost all teams, what is coherent and expected, due to the characteristics of the Brazilian telecommunications segment presented earlier in this paper (IT being a secondary activity, lack of investments in IT processes, see section IV). The three teams that were detected as more evolved in terms of process maturity had good reasons to present those differences (contact with CMMI and COBIT, see section IV).

It is important to reaffirm that the methodology shown in this paper is *not* a way of appraising process maturity, but simply an instrument to help organizations interested in improving the quality of their processes to *estimate* the current process maturity and the effort needed to reach next level. An interesting information, complementary to the present research, would be to appraise the effect of the CMMI implementation, comparing metrics such as productivity or ROI (*Return of Investment*) before and after the implementation. This is left as a suggestion for further studies.

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The Application of Rough Set Theory in Worsted Roving Procedure

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Abstract

The worsted roving procedure's character has been briefly analyzed. Based on 13 variables and the importance of roving quality, it is difficult to control the quality and design new craft parameters. This paper uses the 50 groups of historical data gathering from worsted mill to realize rough set data analysis (RSDA) based on rough set (RS). After reduction by GA the parameters X_1 (moisture regain of tops), X_2 (oil content of tops), X_4 (coefficient of variability of diameter), X_{10} (top neps) and X_{12} (total drafting ratio) are kept down compared to foregoing 13 variables. According to the knowledge represent system (KRS) of roving unevenness and roving weight, it gets 35 and 28 rules respectively in Rosetta software environment. Take a batch of top as an example, the forth rule about unevenness match it completely. Therefore, when RS is introduced to Rule -Based Reasoning (CBR), the simple case storehouse is established to index the processing instance by the RS rules. It is successful to realize the union of Rule -Based Reasoning (RBR) and CBR and good guidance for the craft design and product quality control.

I. INTRODUCTION

The entire processing procedures of the worsted mill from the top to the end product generally are the tops' dyeing (neglect the piece dyeing and yarn dyeing), fore-spinning, worsted spinning, weave, after-finishing craft and so on. The corresponding half-finished product and control target are roving, worsted yarn, grey cloth and fabric. Obviously the fore-spinning is the first link of the whole processing procedure, which its quality has the very important influence to the behind working procedure. According to the actual experience, the yarn unevenness and the spinning frame end breakage have the remarkable linear relations with roving quality [1]. Controlling the half products' unevenness in fore-spinning procedures, especially the roving unevenness is an extremely important quality monitoring measure for the wool mill [2]. Therefore, controlling the roving quality becomes an important link of the yarn quality's control. While the parameters influencing the roving quality are the oil content of tops, moisture regain of tops, fiber mean diameter, coefficient of variability of diameter, fiber mean length, coefficient of variability of length and so on [3], as many as 13 variables. At present the enterprise mainly depend on the experiences, through the traditional survey and the record, the scene sight and estimate, the equipment adjustment and the manpower supplement and so on, cannot use the massive accumulation data and the new generate data to comprehensive analysis and objective judgments, also is unable to discover the problem's source and get the actual solution [4].

The rough set theory (RST) proposed by polish scholar Z.Pawlak in 1982, is a mathematical tool used for dealing with vagueness and uncertainty. In recent years it has received great attention of researchers around the world and has been successfully applied in many areas, such as AI (artificial intelligence), KDD (knowledge discovery in database), pattern recognition, fault diagnosis and expert system [5]. Based on RS, the rough set data analysis (RSDA) gets the data's correlation and dependence and generates rules. Combining with the Case-Based Reasoning (CBR), it can reduce attributes and index the case for different problems. Upon the index mechanism and the operator's practical experience, it can instruct the actual production.

II. ROUGH SET THEORY [5,6]

A. Lower approximation and upper approximation

Definition 1: the information system S contains four elements, namely S = (U, A, V, f). U is the object's finite set; $A = C \cup D$ is the attribute set, in which C is

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condition attribute set and D is the decision attribute set; V is the attribute value set, namely $V=\bigcap_{a\in A} V_a$, in which Va is the value of the attribute a∈ A; f is the information function, f: U×A→V is a single mapping, namely f(x, a)∈ Va, it appoints each object's attribute value in U. Usually a information system is expressed by the knowledge represent system (KRS), in which the line is attribute; the row is object, such as event or sample.

Definition 2: for the information system S=(U, A), if P⊆A and P≠∅, then the intersection of all equal relations are called the indiscernibility relation on P, namely ind(P).

Definition 3: about the information system K=(U, A), for each subset X ⊆ U and equal relation R∈ ind(K), then the lower approximate and upper approximate of X related R respectively are:

$\underline{R}$ (X)=□{Y□U/R|Y□X};

$\overline{R}$ (X)=□{ Y□U/R|Y∩X≠∅ }.

B. Attribute reduce and core

Reduce and core are the key contents of rough set theory. As we were known, the knowledge (attribute) in library is not equally important, even certain attribute is redundant. About the attribute reduction, it deletes the non-correlative or unimportant attribute under the condition of keeping classification ability.

Defines 4: supposing Q⊆R, if Q is independent and ind(Q)< ind(R), then called Q is a reduction for R. It is not difficult to see R has many reductions. All essential relation set of R is called R's core, namely core(R) =∩red(R).

III. ROUGH SET DATA ANALYSIS (RSDA)

A. Discretization

All the attributes' value is regarded as the qualitative data for the RSDA based on symbol, so the quantitative data must be changed into qualitative data. Therefore the discretization of the continual attribute value is the key questions of RS theory. The principle of discretization is [6]: the dimension of attribute is small as far as possible after discretization; the lost information of attribute value is few as far as possible. Usually the discretization methods are the equidistance, equifrequency, Naive Scaler, SemiNaive Scaler, Boolean Reasoning and so on. This paper focus on the craft parameters related with the roving quality: moisture regain of tops (x_1), oil content of tops (x_2), fiber mean diameter (x_3), coefficient of variability of diameter (x_4), fiber mean length (x_5), coefficient of variability of length (x_6), short fiber content (x_7), top weight (x_8), top unevenness (x_9), top neps (x_{10}), total drawing number (x_{11}), total drafting ratio (x_{12}) and roving twist factor (x_{13}). The variables related roving quality are the roving unevenness (R_1) and roving weight (R_2). The 50 groups of data gathered from the worsted textiles mill are discretized by equifrequency method. The 15 continual attributes' discretization ranges are showed in the following table (table 1).

	discretization range
X_1	(*, 1.2), [1.2, 1.3], (1.3, *)
X_2	(*, 20.5), [20.5, 22.5], (22.5, *)
X_3	(*, 18.2), [18.2, 19.6], (19.6, *)
X_4	(*, 21.6), [21.6, 22.1], (22.1, *)
X_5	(*, 65.5), [65.5, 69.6], (69.6, *)
X_6	(*, 45.6), [45.6, 48.8], (48.8, *)
X_7	(*, 11.6), [11.6, 13.6], (13.6, *)
X_8	(*, 19.42), [19.42, 19.79], (19.79, *)
X_9	(*, 1.6),[1.6, 2.0], (2.0, *)
X_{10}	(*, 1.0), [1.0, 1.4], (1.4, *)
X_{11}	(*, 49280), [49280, 537600], (537600, *)
X_{12}	(*, 739128) [739128, 843496], (843496, *)
X_{13}	(*, 4.78), [4.78, 5.02], (5.02, *)
R_1	(*, 5.81), [5.81, 5.98], (5.98, *)
R_2	(*, 0.27), [0.27, 0.29], (0.29, *)

B. Decision system

The knowledge is expressed by decision system in rough set. While the decision system is composed by the discrete data, including the condition attribute (the roving craft parameters) and the decision attribute (roving qualitative variable) after the condition attribute.

Therefore, the condition attribute set is the variables that influenced the roving quality, namely the aforementioned 13 parameters: C = {X_1, X_2, X_3, X_4, X_5, X_6, X_7, X_8, X_9, X_{10}, X_{11}, X_{12}, X_{13}}; the decision attribute set is D ={R1, R2}. So it can get KRS of the roving unevenness and weight. Table 2 shows the KRS of the roving unevenness for the frontal 6 samples.

C. Reduction

The process of reduction means removing the unnecessary condition attribute (unimportantly for decision) from KRS, getting the decision rules based on C mapping to D. In different system, the demand and expectation for reduction is different. In certain

Table 2. the KRS of the roving unevenness after discretization

U/C	condition attribute C													D
	X_1	X_2	X_3	X_4	X_5	X_6	X_7	X_8	X_9	X_{10}	X_{11}	X_{12}	X_{13}	R_1
1	1	3	2	1	3	3	2	3	3	3	2	2	2	2
2	3	2	2	2	3	1	1	2	1	2	1	2	2	2
3	3	3	3	3	2	3	3	1	2	3	1	2	2	1
4	1	2	2	3	2	2	1	2	2	2	2	2	2	3
5	3	1	2	2	3	1	1	2	1	2	2	2	2	1
6	2	1	2	2	3	1	1	2	1	2	1	2	2	3

system if some attribute is difficult to get (high cost to measure these attributes), they will be removed. Usually it is hoped that the numbers of condition attributes and decision rules are small as far as possible after discretization. Usually the reduction methods are genetic algorithm (GA) [7], Johnson, Hotel algorithm and so on. This paper adopts the GA to reduce attributes and the left attributes are X_1, X_2, X_4, X_{10} and X_{12}.

D. Decision rules

After reduction, it can get 35 and 28 rules from the KRS of roving unevenness and roving weight respectively in the Rosetta software environment. Table 3 shows the frontal 6 rules for roving unevenness.

influences the performance of solving problems directly. Especially if the case storehouse is big enough, the case adaptation becomes simple, but the search goal is difficult to reach [8]. As the many craft parameters and long working procedure, it takes many time and manpower to design the processing craft. The way of quick search the most similar case from the historical case storehouse is proved the successful craft design and has strong guiding significance [9-12].

Take a batch of top as an example, its quality parameters: oil content 1.2% (X_1), moisture regain 22% (X_2), fiber mean diameter 21.2μm (X_3), coefficient of variability of diameter 22.1% (X_4), fiber mean length 73.3mm (X_5), coefficient of variability of length 48.7% (X_6), short fiber content 12.6% (X_7), top weight

Table 3. decision rules

No.	decision rules
1	X_1([*, 1.2]) AND X_2([22.5, *]) AND X_4([*, 21.6]) AND X_{10}([1.4, *]) AND X_{12}([739128, 843496]) => R_1(2)
2	X_1([*, 1.2]) AND X_2([20.5, 22.5]) AND X_4([22.1, *]) AND X_{10}([1.0, 1.4]) AND X_{12}([739128, 843496]) => R_1(3)
3	X_1([1.3, *]) AND X_2([22.5, *]) AND X_4([22.1, *]) AND X_{10}([1.4, *]) AND X_{12}([739128, 843496]) => R_1(3)
4	X_1([1.2, 1.3]) AND X_2([20.5, 22.5]) AND X_4([21.6, 22.1]) AND X_{10}([1.4, *]) AND X_{12}([*, 739128]) => R_1(2)
5	X_1([1.3, *]) AND X_2([*, 20.5]) AND X_4([21.6, 22.1]) AND X_{10}([1.0, 1.4]) AND X_{12}([739128, 843496]) => R_1(1)
6	X_1([1.2, 1.3)) AND X2([22.5, *)) AND X_4([21.6, 22.1)) AND X_{10}([*, 1.0)) AND X_{12}([*, 739128]) => R_1(1)

IV. RS AND CBR UNION

In CBR, the state description and solution strategy of a question is implemented by case, while case itself is composed of the semantic node, frame or object in case storehouse [7]. Obviously, how to construct the appropriate case storehouse and how to search case fast and effectively are extremely important. This

20.48g/m (X_8), top unevenness 2.0% (X_9), top neps1.6 /g (X_{10}), the produced roving unevenness (R_1) is 5.9% [4]. So this instance can be expressed as a case showed in table 4-1. According to reduced attributes X_1, X_2, X_4, X_{10}, X_{12} and R_1, it can be expressed another case showed in table 4-2.

Obviously, after reduction the attributes of case storehouse are simplified, from 14 attributes (13

Table 4-1. case expressed by the eigenvector

oil content (%)	moisture regain (%)	mean diameter	diameter CV (%)	mean length	length CV(%)	Short Fiber content (%)	Weight	weight CV	neps	roving CV(%)
1.2	22	21.2μm	22.1	73.3mm	48.7	12.6	20.48	2.0	1.6	5.90

Table 4-2. case expressed by the forth rule after RSDA

oil content (%)	moisture regain (%)	diameter CV (%)	neps (/g)	total drafting ratio	roving CV (%)
[1.2, 1.3]	[20.5, 22.5]	[21.6, 22.1]	(1.4, *)	(*, 739128)	[5.81, 5.98]

parameters and 1 variable) reduce to 6 attributes (5 parameters and 1 variable). Meanwhile, putting this case into the decision rules to search, the forth rule of the 35 rules for roving unevenness completely match it. Therefore, when RS is introduced to CBR, the non-correlated and parameters or attributes can be removed out; the redundant attributes are deleted; the case storehouse is simplified; the processing instance is indexed by the RS rules. It is successful to realize the union of Rule -Based Reasoning (RBR) and CBR. Meanwhile through RST it can excavate out the character attributes and deduct rules to form case storehouse directly in the historical database. According to the rule and case storehouse forming, the craft parameters' design and the products' quality estimate and forecast can become easier.

V. CONCLUSIONS

(1) Based on the 50 groups of roving working procedure data gathered from the worsted textiles mill, using the equifrequency method to discretize them, after reduction by GA the parameters X_1 (moisture regain of tops), X_2 (oil content of tops) , X_4 (coefficient of variability of diameter), X_{10} (top neps) and X_{12} (total drafting ratio) are kept down. According to the KRS of roving unevenness and roving weight, this paper get 35 and 28 rules respectively in the Rosetta software environment.

(2) Take a batch of top as an example, two types of case is established respectively based on itself parameters and the attributes after reduction. The latter case is simple that it only has 6 attributes compared to 14 variables of first case. Therefore, when RS is introduced to CBR, the non-correlated parameters or attributes can be removed out; the redundant attributes are deleted; the case storehouse is simplified; the processing instance is indexed by the RS rules. It is successful to realize the union of Rule -Based Reasoning (RBR) and CBR.

(3) Through RS, it can excavate out the character attributes and deduct rules to form case storehouse directly in the historical database. Thereby, according to different problems, different rules and case storehouse forming, the processing instance can be indexed. So the craft parameters' design and the products' quality estimate and forecast can become easier. It is a good guidance for the craft design and product quality control.

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Adaptive Image Fusion Scheme Based on Contourlet Transform, Kernel PCA and Support Vector Machine

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Abstract: **A novel image fusion algorithm based on the combination of contourlet transform, Kernel Principal Component Analysis (K-PCA), Support Vector Machine (SVM) and Mutual Information (MI) is proposed. Image representation is improved by utilizing the Contourlet transform properties, such as localization, multiresolution, directionality and anisotropy. K-PCA is used to extract features in the low frequency subband and SVM operates on high frequency subbands to obtain a composite image with extended information. Moreover, Mutual Information (MI) is used to adjust the contribution of each source image in the final fused image. Performance of the fusion algorithm is evaluated by recently developed metric, Image Quality Index (IQI). Experimental results show that the proposed method outperforms previous approaches both subjectively and quantitatively.**

Keywords: Image Fusion, Contourlets, Support Vector Machine (SVM), Kernel Principal Component Analysis (K-PCA), Mutual Information (MI).

I. INTRODUCTION

In many real time scenarios, a single type of sensor is not sufficient to provide the complete information about the scene. Image Fusion is a special case of multisensor fusion, where the basic aim is to derive more accurate, complete and reliable knowledge through combining the complementary information from multiple source images [1].

The fused image provides superior operational performance in terms of reduced ambiguity, enhanced reliability and ease in classification and decision making [2]. Some of the application areas of image fusion include target recognition, remote sensing, medical imaging [2], camera applications, night vision applications [1, 3] and human gait detection etc.

A great number of image fusion techniques have been proposed in the past. Mostly, they fall within three categories, i.e. pixel level, feature level and decision level [4]. In Pixel level fusion, the pixels in fused images are computed from pixels in the source images [14]. In Feature level fusion, first feature extraction step is performed on the source images, followed by the fusion step that is based on some selection criteria. In Decision level, results of initial object detection and classification by individual sensors are used as an input to fusion algorithm [4]. In this paper, we will focus on pixel level and feature level fusion.

The techniques found in the literature vary in their complexity as well as in their effectiveness to perform fusion without loss of information and introduction of distortion. The simplest fusion technique is to take the pixel by pixel average of the source images. The drawback of this technique is that the resulting image will have the reduced contrast and possibly contain noise. Other common methods include Laplacian and Gaussian Pyramid [5, 7], ratio of low pass pyramid [6], and the discrete wavelet transform (DWT) [1, 2, 14]. The idea behind the above mentioned techniques is to decompose the source images by applying the transformation and then combining all the decompositions based on some fusion rules to form a single composite representation, then fused image can be obtained by applying the inverse transform. The main advantage of using multiresolution decomposition is that the features that go undetected at one level of decomposition can easily be detected at another level.

Zheng et al. [2] has proposed the technique based on DWT, Principal Component Analysis (PCA) and morphological filtering. The drawback with this scheme is that PCA is a linear reduction technique and it only guarantees the extraction of linear features. In order to cater for nonlinear features Anwaar et al. [8] have used Kernel PCA (K-PCA). However, this scheme uses choose max rule for fusing the detail coefficients. This is not a very effective way as it sometimes prefers noise coefficients over the real image coefficients especially in high contrast regions.

Many other techniques like Cross Band Window approach [9] and Activity Level Comparison approach [10] have been used for fusing detail coefficients. Li et al. [4] have used Support Vector Machine (SVM) to classify the important features in multi focus images. Adnan et al. [11] have extended their work to fusion of visual and thermal images. Their technique is based on Discrete Wavelet Frame Transform (DWFT), KPCA and SVM. Unlike DWT, DWFT is translation invariant and aliasing free. In the present work, we have enhanced this technique by using recently developed Contourlet Transform [12]. Contourlets key features include anisotropy and directionality, due to these edges and boundaries are well approximated into subbands in multiple scales and multiple directions. Liu et al. [17] has used contourlet transform in fusion of multi-focus images with region statistics. In this work we have used Contourlets together with KPCA for low frequency subband and SVM for high frequency subbands. At the end Mutual Information (MI) is applied to adjust the contribution of each source image in the final fused image.

The subsequent sections of this paper are organized as follows. Section II gives a brief overview of Contourlet

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Transform, K-PCA, SVM and MI. Section III explains the proposed scheme of image fusion, quality metric IQI used for the quality assessment [13] is introduced at the end of section III. Section IV covers the experimental results and analysis, and is subsequently followed by the conclusion, acknowledgment and references.

II. Brief Reviews of Constituent Techniques

A. Contourlet Transform

For one dimensional piecewise smooth signal, wavelet is established as an optimal tool for image representation. However, natural images contain intrinsic geometrical structures, which are actually the key features in visual information. In the case of these multidimensional signals, 2D wavelet is good at isolating discontinuities at object edges, but can not detect the smoothness along the edges [12]. Moreover, wavelets can capture only partial directional information, which is an important and unique feature of multidimensional signals. Therefore, some more powerful representation is required in the case of higher dimensions.

Recently developed Contourlets [12] can overcome the problem of multidimensional discontinuities expansion. Contourlet is an extension to wavelet transform using non-separable directional filter banks. It exhibits high directional sensitivity and high anisotropy, and therefore can exactly capture the image edges and contours (smooth edges) to different scales and frequency subbands.

Contourlet decomposition is implemented by utilizing a double filter bank scheme, including a subband decomposition and directional decomposition stage (see Fig. 1, 2). The first phase is multiresolution decomposition using Laplacian Pyramid (LP) [7]. It will generate a down-sampled lowpass version of the original image, and the difference between original and prediction, resulting in a bandpass image. The first step in applying LP is to lowpass filter the original image g_0 to obtain g_1. Similarly, g_2 is formed as a reduced version of g_1, and so on. The level to level averaging process is performed by a function *REDUCE*.

$$g_k = REDUCE(g_{k-1}) \tag{1}$$

This function is carried out for $0<l<N$ levels and nodes i, j, i.e. $0 \le i < C_l$, $0 \le j < R_l$ following the equation:

$$g_l(i,j) = \sum_{m=-2}^{2}\sum_{n=-2}^{2} w(m,n)\, g_{l-1}(2i+m,2j+n) \tag{2}$$

Here $\boldsymbol{N}$ refers to the number of levels in the pyramid, while C_l and R_l, are the dimensions of the lth level. The reverse of this operation is called *EXPAND* operation.

$$g_{l,n}(i,j) = EXPAND(g_{l,n-1}) \tag{3}$$

Its effect is to expand an $(M+1)$ x $(N+1)$ array into a $(2M+l)$ x $(2N+1)$ array by interpolating new node values between the given values.

$$g_{l,n}(i,j) = 4\sum_{m=-2}^{2}\sum_{n=-2}^{2} w(m,n)\, g_{l,n-1}\left(\frac{i-m}{2},\frac{j-n}{2}\right) \tag{4}$$

The Laplacian pyramid is a sequence of error images L_0, L_1, ….., L_N. Each is the difference between two levels of the Gaussian pyramid. Thus, for $0<l<N$,

$$L_l = g_l - EXPAND\ (g_{l+1}) = g_l - g_{l+1,1}. \tag{5}$$

It can be shown that the original image g_0 can be recovered exactly by expanding, then summing all the levels of the LP:

$$g_0 = \sum_{l=0}^{N} L_{l,l} \tag{6}$$

More details about LP can be found in Ref. [7]. In the second phase, the point discontinuities captured by the LP are linked into linear structures by directional filter bank (DFB). It has two building blocks. The first is two channel quincunx filter bank fan filters that divide a 2D spectrum into horizontal and vertical directions. The second building block of DFB is a shearing operator, which reorders the image samples. If we add the shearing operator before the two channel filter bank, and the inverse of shearing operator after the filter bank, we obtain a different frequency partition while maintaining perfect reconstruction [12].

After explaining LP and DFB, we are now ready to describe their combination in double filter bank. Fig.1 shows a multi-scale and directional decomposition using a combination of LP and DFB. Bandpass images from the LP are fed into DFB to capture the directional information. The scheme is then iterated for coarse image (downsampled lowpass image). The combined result is a double iterated filter bank structure named contourlet filter bank. It decomposes the image into directional subbands at multiple scales.

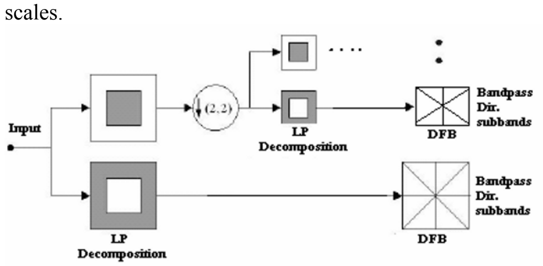

Fig. 1. Contourlet Filter Bank

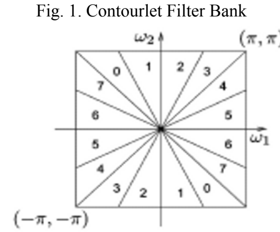

Fig 2. Frequency Partition where l=3 and we have 8 frequency subbands. Subbands 0-3 corresponds to mostly horizontal directions, Subbands 4-7 corresponds to mostly vertical directions [12]

B. *Kernel Principal Component Analysis*

Target information included in the images can be extracted using Principal Component Analysis (PCA) [2]. The main use of PCA is to reduce the dimensionality of the data while retaining most of the information. The serious limitation of this method is that it can capture only linear relationship among the original features on the criteria of their variance. The solution to deal with nonlinear features is to use Kernel Principal Component Analysis (K-PCA) [15, 8]. It can be used to extract nonlinear features in images using kernel trick.

The kernel trick is a method for converting a linear classifier algorithm into a non-linear one by using a non-linear function to map the original observations into a higher-dimensional space. This makes a linear classification in the new space equivalent to non-linear classification in the original space.

K-PCA approach the problem by mapping the data into a high dimensional feature space, where each coordinate corresponds to one feature of the data items, then PCA is performed in that feature space. Radial Basis Function (RBF), Sigmoid and Polynomial are few kernel methods. We have used RBF in our fusion technique. The RBF kernel of two input space vectors $x^{(i)}$ and $y^{(j)}$ is given as:

$$K(x^{(i)}, y^{(j)}) = \exp\left[-\frac{\| x^{(i)} - y^{(j)} \|^2}{2\sigma^2}\right] \tag{7}$$

Here, the standard deviation σ, is applied to the function. The choice of suitable kernel varies with application.

C. *Support Vector Machine*

Support Vector Machine is amongst the most efficient family of the learning algorithms. Some important characteristics of SVM include good generalization performance, computational efficiency, and robustness in higher dimensions. SVM maps input vectors to a higher dimensional space where a maximal separating hyper-plane is constructed [19].

Linear SVM for separable data is a learning algorithm that chooses the linear classifier with the largest margin. The hyper-plane, corresponding to such a linear classifier, which maximizes the margin of separation, is called the optimal hyper-plane (OHP) [16].

Given the training sample $T = \{(x_i, d_i)\}_{i=1}^{N}$, where x_i is the training example and d_i is the target value, find the Langrange multipliers $\{\alpha_i\}_{i=1}^{N}$ that maximize the objective function.

$$Q(\alpha) = \sum_{i=1}^{N} \alpha_i - \frac{1}{2} \sum_{i=1}^{N} \sum_{j=1}^{N} \alpha_i \alpha_j d_i d_j x_i^T x_j \tag{8}$$

w.r.t. the α_i 's, subject to the constraints

$$\sum_{i=1}^{N} \alpha_i d_i = 0, \; 0 \le \alpha_i \le C \quad \textit{for } i=1,2,\ldots\ldots, N \tag{9}$$

Where C is a user specified parameter. One of the key attribute of the SVM is that its performance is dependent on data separation margin, and is not attributed by dimensions of input space. For more details on SVM, refer to Ref. [16].

D. *Mutual Information*

Mutual information (MI) is considered to be a good indicator of the relevance of two random variables [4]. If we have images F and A, then $MI(F,A)$ is the quantity of information obtained about F when A is learnt.

$$MI(F, A) = \sum_{i=1}^{L} \sum_{j=1}^{L} h_{F,A}(i, j) \log_2 \frac{h_{F,A}(i, j)}{h_F(i) h_A(j)} \tag{10}$$

Here, $h_{F,A}$ is the normalized joint grey level histogram of images F and A, h_F and h_A are the normalized marginal histograms of F and A. L is the number of grey levels [3].

III. Proposed Fusion Scheme

In this section, we will describe our proposed fusion scheme, followed by the performance evaluation function we have used, i.e. IQI.

The contourlet decomposition is applied to registered images, which results in contourlet coefficients set. The contourlet coefficients can be categorized into a low frequency subband and multiple high frequency subbands in different scales and directions. The proposed scheme extracts the detail or high frequency information by using SVM with linear kernel, and for extracting the approximation or low frequency information, K-PCA with RBF kernel is used. These details and approximation are then fused to form a fused coefficients set, which is then converted into fused image by applying the inverse contourlet transform. Training and testing phases of SVM are described below in detail:

A. *SVM Training Phase*

1. For training two window pairs, each of size 32 x 32 are extracted from each original source images. The first source image has useful information in one of the pairs of the windows, while the second source image has useful information in the other pair. Fig. 3 shows the process of extraction of two windows from each source image.
2. Decompose the two source images by using the contourlet transform. After applying J level contourlet transform [17], we get following coefficients

$$A \rightarrow (b_1^A, b_2^A, \ldots\ldots, b_{J-1}^A, b_J^A, a_J^A) \tag{11}$$

$$B \rightarrow (b_1^B, b_2^B, \ldots\ldots, b_{J-1}^B, b_J^B, a_J^B) \tag{12}$$

$$b_j^X = \{d_{j,1}^X, d_{j,2}^X, \ldots\ldots, d_{j,l(j)}^X\}, \quad X{=}A \text{ or } B \tag{13}$$

Where a_j denotes the low frequency subband, b_j is the subband aggregate in scale j, $d_{j,k}$ is the k-th directional high frequency subband in the scale j. $l(j)$ shows the number of each level and is equal to 2^j [17]. As the approximation subband contains little edge information due to subsequent decompositions, hence, it is not used in the case of SVM training. So, SVM is used for fusion of detail subbands.

(a) (b)

Fig. 3. Positive and Negative training examples for SVM training. Left side window of (a) is out of focus as compared to left side window of (b). Right side window of (a) is more clear and focused as compared to right side window of (b)

3. Now, compute the activity level at each pixel location by using Coefficient based activity method (CBA) [4]. CBA is applied to detail subbands at every scale and direction. For high frequency subband aggregate at scale *j*, use CBA to compute activity level.

$$A_j^X = | b_j^X |, \; X=A \text{ or } B \tag{14}$$

Here, A_j^X is the activity level of high frequency subband aggregate at scale *j*.

4. Train SVM to determine whether coefficients from image *A* (or *B*) should be used at position (*p,q*). The activity levels obtained from step 3 formulate the input for SVM. The target values are specified based on the following criteria: if details present at (*p,q*) in image *A* is important as compared to details present at (*p,q*) in image *B*, the target value is set as 1, and -1 otherwise. In case of visible and infrared images, important means 'heat' emitting objects in an infrared image and clearly distinct objects in the visible image. In case of multifocus images, important means 'focused' part. This is an example of supervised learning. Since, we have two source images, image difference vector will be the input feature vector for SVM training.

$$\{A_j^A(p,q) - A_j^B(p,q)\} \tag{15}$$

Here, A_j^A, A_j^B is the activity level of high frequency subband aggregate at scale j of image *A* and *B*. Now, by applying input feature vector as in (15) and specifying the corresponding target values, the SVM is trained for the details at scale *j*.

Now, as the training is finished we can move on to testing phase.

B. SVM Testing Phase

1. Repeat step 2, 3 of training phase on the test source images.
2. Perform testing on whole image sequence using trained SVM. If SVM output at (*p,q*) is positive then coefficients of all detail subbands at particular scale will come from image *A*, and vice versa. Similarly we can write;

$$b_j^F(p,q) = \begin{bmatrix} b_j^A(p,q) & SVM_{output}(p,q) > 0 \\ b_j^B(p,q) & otherwise \end{bmatrix} \tag{16}$$

3. Apply K-PCA on source images *A*, *B* and select eigenvector corresponding to largest eigenvalues, denote it by α_1 and α_2.
4. Get fused approximation coefficients by using:

$$a_J^F = \frac{(\alpha_1 * a_J^A) + (\alpha_2 * a_J^B)}{(\alpha_1 + \alpha_2)} \tag{17}$$

Where α_1 and α_2 are elements of principal eigenvector, a_J^A, a_J^B and a_J^F are approximation coefficients of source images *A*, *B* and fused image *F* respectively.

5. For further processing, optionally perform any of the following methods:
   a) Consistency Verification [4] operates on binary decision map and checks that if value of the center pixel in a map comes from image *A*, while neighboring values come from image *B*, then the center pixel is switched to that of image *B*.
   b) Morphological Filtering [2], it makes use of the basic morphological fill and clean operators to remove isolated points in the binary decision map.
6. Intermediate Fused image *IF* can be reconstructed from these modified coefficients by applying inverse contourlet transform. The process is depicted as follows:

$$(b_1^F, b_2^F, \ldots\ldots, b_{J-1}^F, b_J^F, a_J^F) \rightarrow IF \tag{18}$$

7. To make the fusion process adaptive, we have used Mutual Information (MI) scheme. MI between intermediate fused image *IF* and source images *A* and *B* is determined and denoted by *MA* and *MB*, respectively. Now, compute difference between *MA* and *MB* and compare it against a minimum threshold T_h.
   a) If the difference is less than the threshold T_h then there is no need for adjustment of the contributions of the source images.
   b) If the difference is more than T_h then adjustment is required, which will be carried out by adjusting weights in (17).
   c) If MA is lesser than MB, then update α_1 by $\alpha_1 + (\alpha_1 - 1)$ and α_2 by $\alpha_2 + (\alpha_1 * \alpha_2)$ and vice versa. Now, compute the a_J^F using new values of α_1 and α_2 using (17).
8. Construct final fused image *F* using equation.

$$(b_1^F, b_2^F, \ldots\ldots, b_{J-1}^F, b_J^F, a_J^F) \rightarrow F \tag{19}$$

C. Image Quality Index (IQI)

IQI was proposed by Piella et al. [13] for image fusion quality evaluation. If δ denotes variance, the global IQI for two vectors *A, B* is defined as:

$$Q_0 = \frac{\delta_{AB}}{\delta_A \delta_B} \frac{2\overline{AB}}{\overline{A}^2 + \overline{B}^2} \frac{2\delta_A \delta_B}{\delta_A^{\ 2} + \delta_B^{\ 2}} \qquad (20)$$

Weighting procedure in calculation of Q_0 is introduced in [13]. Given the local saliencies of two input images *A* and *B*, we compute local weight λ, indicating the relative importance of image *A* compared to image *B*.

$$\lambda = SF(A) / [(SF(A)+SF(B)) \qquad (21)$$

Here, *SF(A)* means Spatial Frequency [2] of image *A*. Now, IQI of fused image can be defined as:

$$Q_F = \lambda Q_0(A,F)+(1- \lambda)\, Q_0(B,F) \qquad (22)$$

We have used (22), for evaluation of our results.

IV. Experimental Results and Analysis

In our experiments, we have compared the results of our fusion scheme i.e. Contourlets with K-PCA, SVM and MI based fusion with three other fusion techniques, i.e. simple PCA and DWT based fusion [2], Contourlet with K-PCA and MI based fusion [3], DWFT and SVM based fusion [11]. The images used in our experiments are already registered images with 256 grey levels. The experiments are carried out on more than 30 image pairs, but the results on some standard image pairs are shown here to demonstrate the effectiveness of the proposed scheme. We have applied each fusion method mentioned above to these image pairs and their fusion performance has been calculated using IQI.

In our experiments, we have used the contourlet decomposition to decompose the source images using a 9-7 bi-orthogonal Daubechies transform, where subband at each level is fed to the directional banks stage to get the sixteen directions at the finest level. In training, the patterns are extracted from the source image by using the window pairs. Target values are assigned accordingly. The training step is performed only once. For remaining images belonging to the same category, we perform the testing only. Therefore, we can distribute the trained classifier as part of software and no mundane training by the user is required. We have used Linear SVM for the classification in our scheme, as it outperforms RBF kernel based SVM.

Fig. 4 is an example of fusion of visual and infrared images obtained from different sensors. Fig. 4(a) and (b) are the visual and infrared image of the same scene. Fig. 4(c)–(f) shows fusion results using simple PCA and DWT based fusion, Contourlet with KPCA and Mutual Information (MI) based fusion, DWFT and SVM based fusion and the proposed algorithm respectively.

Fig. 5 is an example of multi focus image fusion. Fig. 5(a) and (b) are the images of the same scene, where in (a) right part of image is in focus, while in (b) left part of image is in focus. Fig. 5(c)–(f) shows fusion results using simple PCA and DWT based fusion, Contourlet with KPCA and MI based fusion, DWFT and SVM based fusion and the proposed algorithm respectively. Visually the proposed technique results in more in-focused image.

Table 1 and Fig. 6 present the IQI comparison of these four fusion techniques. It clearly shows that the proposed technique outperforms the other three approaches.

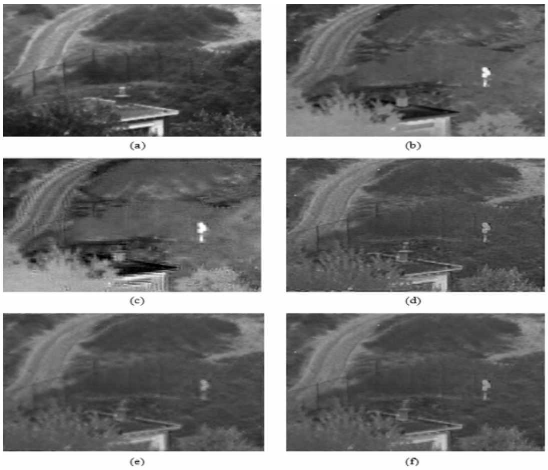

Fig. 4. Fusion of Visual and Infrared Images

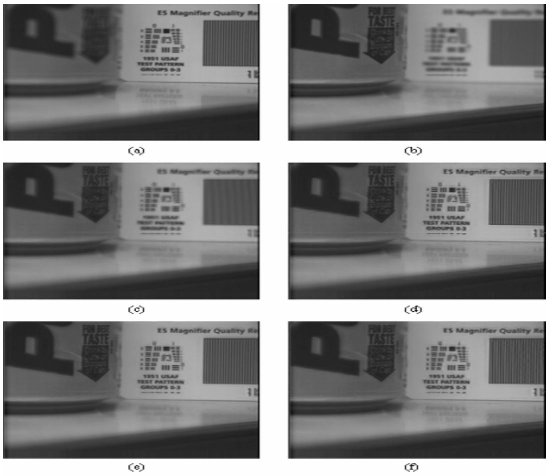

Fig. 5. Fusion of Multi Focus Images

TABLE 1
IQI VALUES FOR 6 IMAGE PAIRS

Image/Scheme	DWT + PCA	Contourlet + KPCA + MI	DWFT + SVM + KPCA	Proposed Scheme
Clock	0.6667	0.6689	0.6890	0.7058
Pepsi	0.4971	0.8117	0.7992	0.8157
Hidden Man	0.4653	0.5564	0.5233	0.5635
Toy	0.6855	0.7721	0.7772	0.7954
Disk	0.5513	0.6122	0.6165	0.6247

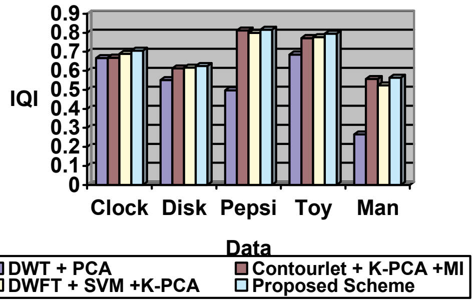

Fig. 6. IQI Values Comparison

V. Conclusion

In this paper, image fusion scheme in contourlet domain based on SVM, K-PCA and MI is proposed. Contourlet provides a flexible and robust scale-direction representation for source images. It is suitable for image fusion scheme because it provides better directional sensitivity and shift invariance. Complex fusion rules like K-PCA and SVM are used to fuse the approximation and detail subbands. The contribution of source images is adjusted by using MI. Detail subjective and objective results show that proposed scheme performs considerably well as compared to other approaches. In future, we can extend this technique to areas like high dynamic range imaging [18]. Here, the task of SVM will be to distinguish between over and under saturated regions, so that dynamic range of fused image is enhanced.

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Classification of Web Pages by Automatically Generated Categories

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Abstract-The web is a huge repository and is growing rapidly so that the access to the relevant information becomes increasingly difficult. One of the general solutions for this problem is to classify web pages. Although there are standard directories for that purpose but it is impossible to expand them manually for the whole web. On the other hands the automatic classification is considered. One approach is to use a hand-made ontology for this purpose. In this paper we firstly introduce an algorithm to select web pages related to a specified domain and then a suitable ontology is created automatically for that domain. One of the main advantages of this approach is extendibility to other domains.

Keywords: Web mining, Web Classification, Ontology

I. Introduction

One of the main challenges of the researchers is how to manage data and facilitate access to them on the web. On the other hands that information and the knowledge hidden in them are highly interested by domain experts and analysts. The primary solution was to use information retrieval methods on the web which gained high interests in the middle of 1990s. However, with the explosive growth of the web the need for approaches and tools to discover, summarize and analyze the data becomes apparent. For this purpose the data mining was the first option. Web data can be categorized in three different areas:

1-Content data: web page's contents

2-Structural data: graph structure among data

3-Usage data: those stored in web server logs.

According to this categorization as shown in the Fig. 1 we can categorize web mining approaches to three respective classes [1].

Among this three, content mining is more important. Its advantage is its correlation to text mining which is a highly researched area and many approaches have been introduced for some decades.

Some different approaches for web classification already exist but as mentioned in [8] ontology is a very useful tool for this purpose.

Fig.1.Web mining classifications

In fact, despite the good results reported in research literature, but web mining techniques are not well adopted for the practical purposes. The main limitation is that they lack understanding of contextual semantics hiding inside texts, links and other elements in web pages. We can tackle this by adding ontologies in the process of web mining [27,28]. For example when we use onotolgies for classification or knowledge extraction, it is possible to relate results to concepts in ontologies which give us a better understanding of the results.

With the introduction of Semantic Web, ontolgies plays more important roles in the web. As there is a little work to use ontologies for classification and also there is lack of ontologies for many domains, in this paper we introduce an automatic ontology creation method and an approach to use such ontologies for classification.

In section 2 a survey on related works is given. Then in Section 3 design of our classifier is explained. Section 4 discusses on our ontology learning method and section 5 shows evaluations on it. Finally a conclusion and future works are given in section 6

II. Related Works

There are many works reported on the web page classification. Some uses well-known classifiers like Bayesian Networks, SVM and the like. But there is a little work to use ontologies for this purpose.

In [8] ontologies are used for web mining of home pages of users. However, In this work the employed ontology is hand-made. Also in [6] ontology is used to do online classification of documents. To do so, for different meanings of query terms different classes are created and documents in the result list are put in appropriate classes. The used ontology in this case is

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 319–325.

Wordnet which is a general ontology. [10] also uses a ready-made ontology as well as CMU parser for classification. In it WordNet is used too. [9] Defined ontology for emails and does classification based on that. In it protégé is used for ontology definition and CLIPS is employed for reasoning on that.

As it is clear, in most of the related works, a ready made ontology is used. Such limitation prevents extendibility of the proposed methods for new domains. On the other hand in our approach the ontologies are created automatically. Although they might not be suitable for reasoning purposes but they are a good facility for extending the search for the whole web space.

III. Specification of Vector Model Classifier

In this section we first discuss about a classifier based on vector space model. Since such a classifier is the base model in our research, we tried to design it in such a way to gain a good performance. Therefore we put some efforts on one of the main factors which is the importance of tags in the html pages. It should be noted that because of the dynamic nature of the web, selection of appropriate features are highly dependent on time. Factors like the kind of tools which is used for web authoring , affects the precision of different attributes for classification along the time passage. In the next section we discuss about our experimental results in this regard.

A. Comparison of precision of tags

As explained above, because of the usage of semi-structured html, web pages are different from ordinary texts. In fact using features of this language we can visually distinguish between different parts of a document. To do so tags are used in html pages. Most important ones are:

- Tile: to specify title of a web page
- Body: for main document text
- Meta-keyword: for the keywords of the page
- Meta-description: for descriptors of the page
- H1, H2… visual tags to emphasis the important texts in a page.
- Anchor text: the text displayed on top of a hyper-link.

[3] reports about one of the earliest works which investigates the importance of these tags and presented statistics of their existence in web pages. One of the interesting results mentioned in the paper is the low performance of title tags because of the usage of general works like “home page” or “web site” in them. The data used in [3] has been the topmost category of North American Industrial sites (containing 21 categories).

The problem of tag selection is also discussed in [2], but their result is not reliable for the following reasons:

- Only three tags are considered
- The number of pages in training phase is little
- Only pages with having all three tags are considered.

Since most of the pages in web lack some tags the results can not be used for whole web

Authors of [5], in a classification framework, have done a limited study on the importance of tags.

In our experiment we use all the tags mentioned above except anchor texts (for H tags we consider only H1 to H4, since higher ones are rarely used in web pages and also their importance is very low).

Web content mining is divided to classification and clustering approaches. We use classification approach since then we can use existing directories to evaluate the performance of classifier. In some related works Yahoo directory [4] is used. In this project, however, we used Open Directory (i.e. DMOZ). It contains 600000 hierarchically organized categories and more than 700000 persons are doing the classification job.

The designed classifier is implemented in Java and works based on the vector space model [13]. In the training phase we crawled 5000 pages from each category among the top 15 Dmoz categories (totally 75000 pages). Stopwords, html tags and those words having low TF-IDF [13] value are eliminated from the pages. The size of the vectors for different tags is different. The lengths of those vectors are shown in Table 1.

In the test phase we used a collection of 1500 pages which is gathered separately from Dmoz (from each sub-directory 100 pages). After elimination of stop words for each one 8 vectors are extracted. The output of the system is such a way that for each page and each tag, a sorted list of categories is suggested. In the simplest form of evaluation we can only consider the first proposed category (Top-Rank). We also used another measure named 3-Top-Rank introduced in [4]. This measure evaluates the existence of the correct category among the top 3 categories reported.

Table.1
Size of different tag’s vector

Body	Title	H1	H2	H3	H4	Key	Desc
57890	47233	4718	5906	9531	5003	37537	15063

As shown in Fig. 2, the most precise tag is Meta-Description which is a considerable result since they are very small compared to the body texts. It seems that since they are created by the web page creators manually they are good representative texts to reveal the content of web pages. H1 tags are in the second place and show the fact that the important texts of web pages are highlighted. Another interesting result in the figure is that the H2 tags has the most progress from Top_Rank to 3-Top-Rank which means that they can correct the classification errors.

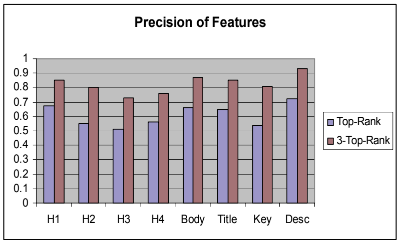

Fig.2.Classification operation accuracy base on Different tags

B. Weighting Tags

The prioritization of the tags discussed in the previous section is not enough for web mining. In fact as pointed in [11,12] availability of tags (i.e. existence of tags) is also important. Therefore in our research we also consider this parameter. The availability for the 8 mentioned tags is computed for the 75000 pages and the results are shown in Fig.3.

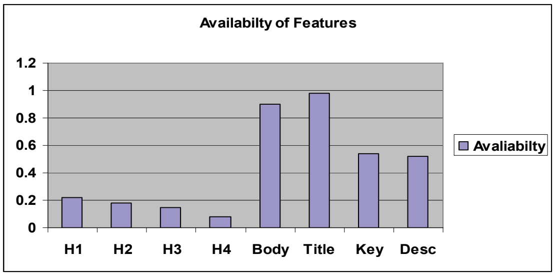

Fig.3.Different tag's availability

As shown in the figure, none of the tags appears in all the pages so we need to use a combination of them. To do so we need to introduce some weighting for them.

To do some experimentation, we introduced three weighing schemes:

- Equal: the equal weight is given to tags
- 1st Rank: the weigh is proportional to the ranks of tags discussed in section 2-1.
- Availability: the weights are proportional to availability of tags as shown in the Fig.3.

The performances of these weighting schemes are shown in table 2.

Table 2
Classification operation accuracy at first result and First three result with different weighting method

Feature	Top-Rank	3-Top-Rank
Equal	0.719	0.919
1st-Rank	0.721	0.919
Availability	0.731	0.916

According to the table, almost all of the methods have acceptable precisions and from this respect comparable to the best tag (Desc).

IV. Building Taxonomy of Categories

A.Ontology learning

Ontology is a set of concepts and relationships between them which abstracts knowledge structure for a specific domain. Ontologies can be divided to three levels according to their level of formality as shown in Fig. 4:

1-Formal ontologies: concepts and relationships are specified by definitions and axioms using mathematical logics.

2-Terminological ontologies: concepts are partially defined and sorted by is-a relationships. However some shallow form of logic may be used to specify logical statements about concepts.

3-Prototype based ontologies: concepts are specified by their instances

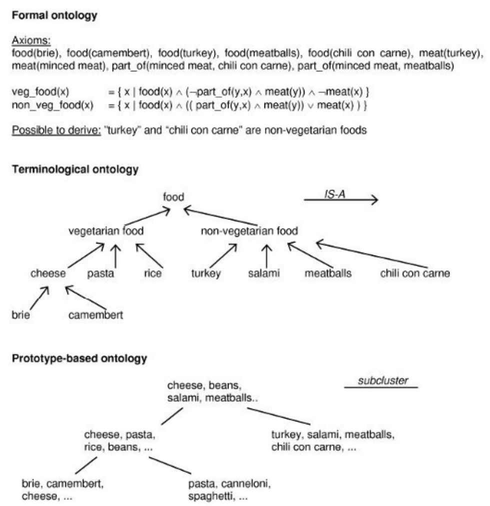

Fig.4.Ontology with different formalities level

The process of ontology learning is divided to smaller processes arranged in a layered architecture as shown in Fig.5.

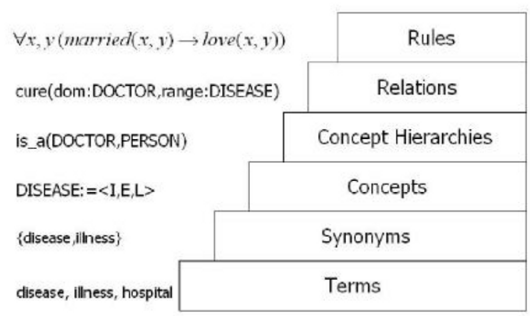

Fig.5. Ontology learning layers [16]

The examples in the figure are selected from medical domain. As shown, concepts are those like “disease”, “illness” etc. . . In an upper layer we have the synonymy relationship between disease and illness. The hierarchical relationship between doctor and person is shown in a higher level. Upper levels show the axioms for this domain. In what follows we explain about the major approaches related to each layer.

Terms Extraction

This is the prerequisite for any activity in ontology creation. Improvement of this activity has major impact on the quality of the resulted ontology. A good term should have two specific attribute:

1-Be a good descriptor for a document

2-Be a good separator to distinguish documents

Many methods for term extraction has been proposed. Most of them are based upon information retrieval techniques. In addition usage of NLP techniques is also proposed.

Extraction of Synonyms

In this process we can use the supervised and un-supervised learning methods to extract synonymy relationships.

Hierarchy extraction

There are three major approaches to build hierarchies [26]:

1- Linguistic based: including syntactic patterns to find is-a relationships like what proposed by Hearst [47]. In this approach a set of patterns are defined using regular expressions. For example a pattern like “np1 and np2 and other np” can be applied on the sentence “flu and headache and other diseases” to create is-a relationships like isa(diseases, flu) and isa(diseases, headache). In [18] such patterns are created manually but [19] proposed a method for automatically generating such pattern. The Hearst idea is extended by many researchers. For example [20] extends the patterns with some new ones and [14] applied the same idea for a specific domain. Also [14] used this idea for “is-part-of” relationships. Yet [21] used the idea to recognize cause-effects relationships. The common characteristic of these ideas is their high precision but low recall.

2- Statistical: this method like synonymy extraction is also based on Harris assumption. For example probability of having flue and headache with disease and illness is higher than with car.

3- Information retrieval: In this kind of methods taxonomy of documents are used to build taxonomy for terms.

Axiom Extraction

Works in related to this process is mainly done for medical field. The main goal of those works is to find the relationship between diseases and drugs. Such relationships are attained by analysis of large text corpuses [22].

B. Learning Taxonomies for DMIZ

To build an appropriate taxonomy we used the comparisons reported in [23] where more than 50 learning systems are studied. They also introduced a framework for categorization of the works.

According to this categorization our work is a kind of automatic learning system based on a statistical approach. Although in this approach the relationships between text components is not considered but for those applications which don’t need deduction (like information retrieval) is applicable. The main characteristic of the method is its generality, scalability, simplicity and independence of any specific domain.

Term extraction

In this process we firstly delete html tags from the document collection and then do some preprocessing like stopword elimination and deletion of the words with very low frequency.

Taxonomy creation

For the taxonomy creation we employ clustering approach. As discussed in [24] the most suitable methods are bottom-up hierarchical methods. Such algorithms need a measurable attribute for clusters which we name “signature of cluster” as well a method to calculate the similarity between clusters.

The proposed solution for signature is to use the co-occurrence property of terms. This solution is based on the hypothesis that if two terms are used together with high frequency they have some semantic similarity. To find the co-occurrences of terms we do as follows:

1-First we find all the pairs that at least occur in one document.

2-To compute coupling strength for a pair we use this assumption: two terms has high coupling if they appear in many similar documents and their frequencies in those documents are near each other. We formulate this assumption by the following formula:

$$Coupling(A,B) = \frac{(N_a + N_b)}{(N_a - N_b + 1)} \quad \text{if } N_a, N_b > 0 \quad (1)$$

$$Coupling(A,B) = \frac{(8+1)}{(|8-1|+1)} = 1.125$$

$$Coupling(C,D) = \frac{(2+2)}{(|2-2|+1)} = 4$$

Table 3
word pairs in some domains

Domain	1st Couple	2nd Couple	3rd Couple
Art	(library,university)	(drawing,artists)	(published,copyright)
Business	(accounting,tax)	(applications,offer)	(agricultural,farm)
Computers	(client,server)	(distributed,parallel)	(artificial,intelligence)
Games	(game,play)	(playing,role)	(casino,poker)
Health	(diet,weight)	(disease,health)	(clinical,treatment)
Home	(soups,salads)	(salad,sauce)	(pie,cake)
Kids & teens	(cheats,xbox)	(xbox,playstation)	(ign,fileplanet)
News	(entertainment,sports)	(journalism,reporter)	(political,iraq)
Recreation	(antique,collectors)	(rock,climbing)	(camp,trips)
References	(dictionary,words)	(library,maps)	(universities,colleges)
Regional	(burkina,faso)	(cayman,belize)	(Angola,Ghana)
Science	(laboratory,lab)	(chemistry,molecular)	(telescope,astronomy)
Shoping	(clothes,dresses)	(dresses,shoes)	(shirts,jackets)
Society	(lesbian,gay)	(united,states)	(violence,abuse)
Sports	(league,players)	(bike,bicycle)	(championships,tournament)

3-To reduce the effect of the terms which appear in many different domains we multiply the attained value in step 2 to minimum IDF values of the two terms.

4-The final value of coupling is the sum of coupling for all documents

We showed some of the most coupled pairs in table 3.

An interesting point about the co-occurrence matrix is that the number and strength of coupling for different domains does not follow a similar pattern. For example the number of coupled pairs in Art, Business and Shopping domains is much lower than Home and News domains.

After the creation of co-occurrence matrix, to reduce the dimensionality of data as well as higher the precision of taxonomy creation we apply the following steps:

- We only select the nouns among terms.
- Apply Porter stemmer to find the stems of the terms.

The list of terms coupled with a term is considered as its signature and clustering is done based on the similarity of those signatures. In each step, two clusters with highest

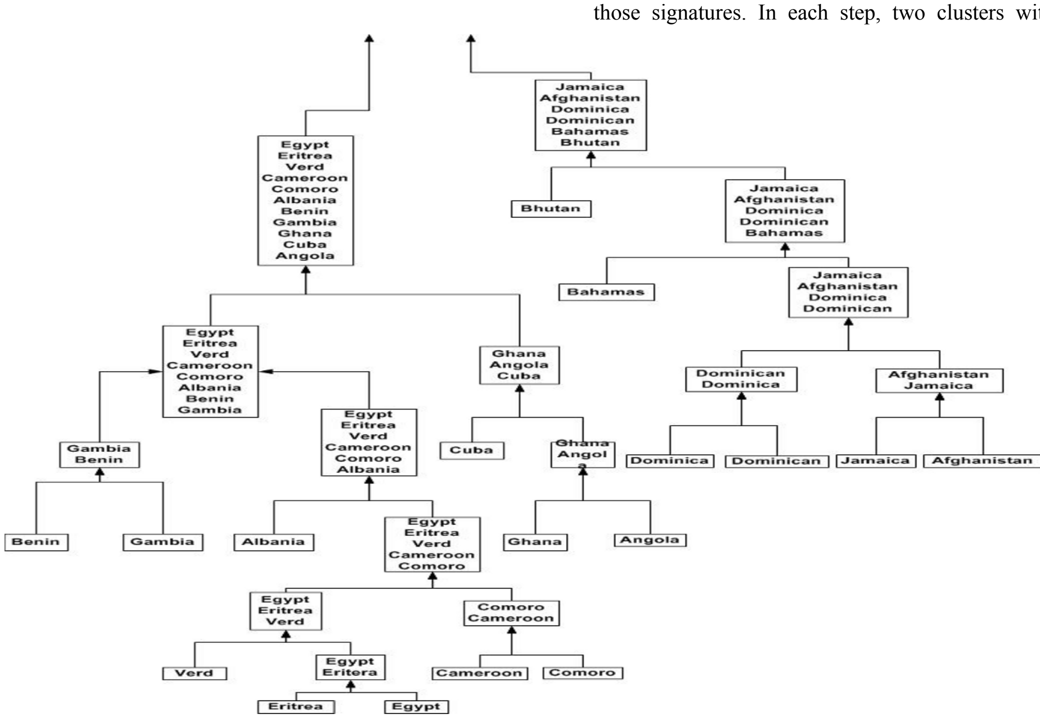

Fig. 6.Regional taxonomy

similarity is joined together and a larger cluster is created. The signature of newly created cluster is the center of gravity of its child clusters. This process is iterated until we reach to the root of the taxonomy (i.e. a single cluster). Figure 6 shows a part of the taxonomy created for Regional domain.

V. Evaluations

In one experiment we do the classification on queries while in a second experiment classification is done on documents. We compare the performance of these two experiments.

A.Evaluation Based on Query Classification

Classification by Vector Space Model

The traditional method for query classification is vector space model [13]. In this method for each class a vector is created. The user query is also converted to a vector and the cosine similarity of the query to each class is calculated.

Classification by Constructed Taxonomies

Since the taxonomy created for a domain shows the semantic relationships between terms in the domain, we can use them in query classification. For this purpose we introduced the following measures:

- Term similarity: the intersection of terms in a class and the query is calculated. So when a class and the query has more common terms their similarity becomes higher.
- Semantic coupling: in most cases the entered query has a little number of terms and therefore the classification becomes more difficult. In such cases we can use the semantic coupling between the terms in query and select a class which is the semantic bridge between those terms (i.e. the terms belong to the child of the class and the class is the lowest node in the taxonomy having this property).

Comparisons of the methods

To evaluate the performance of the explained methods we used an actual data set published by AOL. The data set contains a part of queries issued in fall 2004 by its users. A subset of these queries (20000 queries) has been classified by 10 human in 20 classes [25]. We specified the commonality of the classes with DMOZ directory: Business, Computers, Games, Health, Home, News, Regional, Shopping, and Sports.

The evaluations on the two previously specified methods are shown in figure 7.abbreviations in this figure are:

Q-Vec: query classification using vector space model

D-Vec: Document classification using vector space model

Q-Tax: Query classification using learnt taxonomies

D-Tax: Document classification using learnt taxonomies

As shown in the figure, for 4 classes among 9 the results of classification based method is better. In one class the results are almost similar and in 4 remaining classes the results by vector space model are better. Considering the low dimensions of classes compared to document vector this results is considerable.

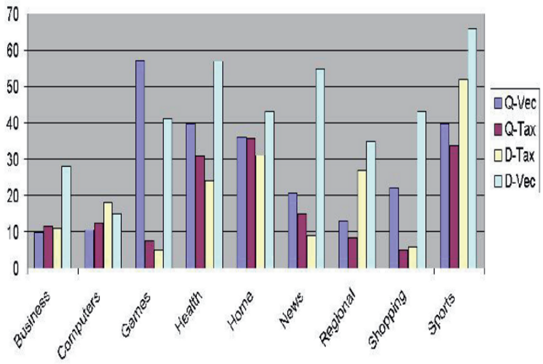

Fig.7.Performance comparisons between vector space model and learnt taxonomy for query classification

B.Classification by documents

In this experiment we used a collection of 900 documents (100 pages from each category). We selected those pages randomly from Dmoz. As a base method, we used a classifier based on vector space model explained in section 3. In the second classifier we only used the first similarity measure (because of its complexity we don't use semantic coupling measure). The results of the two methods are also shown in the Figure 7.

The figure shows that although the method based on vector space model is better in general, but the second method has also considerable results. The lower quality of the second method is natural because we did not use the semantic relationship between terms.

Closely considering the figure again we can notice interesting results. For example in classes with lower quality for classification based methods in comparison to vector space based method (shopping, game and news) we can conclude that the constructed classes are not good representative for those categories. Such a problem may be because of:

- The number of terms selected is not enough
- Co-occurrence of terms in such categories is not an appropriate measure

Therefore it seems that with manual refinement of those classes we may reach to higher precision for classification based methods.

VI. Conclusions and Future Works

In this paper we investigated on the performance of classification based methods for information retrieval. We made a classifier based on some categories of DMOZ. Then we used such a classifier for the classification of queries as well as documents. The main merit of the classification based method compared to vector space is its low dimensionality computation. In this article we only used a basic classification method. But for future works other classifiers like Bayesian Network or SVM can be considered. Also we may employ other dimensionality reduction methods [7] in our future works.

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Classification of Arrhythmia Using Rough Sets

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***Abstract*--Arrhythmia is caused due to the changes in the normal rhythm of heart with great risk of fatality if sustained over long periods of time. Many machine-learning algorithms have been developed for a fast diagnosis of cardiac arrhythmia. Also, the decision accuracies vary from method to method with Support Vector Machines (SVM) giving the highest. However, SVM cannot deal with the uncertainties in the process of diagnosis. Therefore, the authors have considered the Rough Set method for this classification. Analysis has been done with and without reducing the number of features. It has been found that the classification accuracies vary between 87% and 54% respectively. It may be emphasized that rough set reasons out all the possibilities and the decision could be as close as human diagnosis. The experimental results are presented.**

***Index Terms*--Arrhythmia classification, rough sets, principal component analysis (PCA), feature reduction, Discretization and reduction.**

I. INTRODUCTION

Characterization and classification of arrhythmia is an important step in developing devices for monitoring the health of individuals. The electrocardiogram (ECG) remains the simplest non-invasive diagnostic method for determining various heart diseases. Physicians interpret the morphology of the ECG waveform and decide whether the heartbeat belongs to the normal sinus rhythm or to the class of arrhythmia. Computerized electrocardiography is currently a well-established practice, supporting human diagnosis.

A number of approaches for arrhythmia classification which include Bayesian [1] and heuristic approaches [2], expert systems [3], Markov models [4], self-organizing map [5], Artificial Neural Networks (ANNs) [6] and SVM [7] have been reported in literature

According to published results, existing approaches generally tend to suffer from problems that result from high sensitivity to noise included in the data, and unreliability in dealing with new or ambiguous patterns.

The method based on statistical learning theory (SVM) produces a globally optimum solution and is used extensively in data mining. However, from a practical point of view the most serious problem with SVMs is the high algorithmic complexity and extensive memory requirements of the required quadratic programming in large-scale tasks. Although SVMs have good generalization performance, they can be abysmally slow in test phase, a problem addressed in (Burges, 1996; Osuna and Girosi, 1998). Also the method is more appropriate for crisp sets and Rough Set Theory (RST) is a much better classification method wherein the rhythm and morphological reasoning can be evolved from rough sets. It is also computationally much simpler than SVM.

II. ROUGH SET THEORY:

In 1982, Pawlak introduced the theory of Rough sets [8]. This theory was initially developed for a finite universe of discourse in which the knowledge base is a partition, which is obtained by any equivalence relation defined on the universe of discourse. In rough sets theory, the data is collected in a table called decision table. Rows of the decision table correspond to objects, and columns correspond to attributes. In the data set, a class label is used to indicate the class to which each row belongs. The class label is called as decision attribute, the rest of the attributes are the condition attributes. Consider that if the data set is stored in a relational table with the form Table (condition-attributes, decision-attributes). C is used to denote the condition attributes, D for decision attributes, where $C \cap D = \Phi$, and t_j denotes the j^{th} tuple of the data table. Rough sets theory defines three regions based on the equivalent classes induced by the attribute values: lower approximation, upper approximation, and boundary. Lower approximation contains all the objects, which are classified surely based on the data collected, and Upper approximation contains all the objects, which can be classified probably, while the boundary is the difference between the upper approximation and the lower approximation. Hu et al., (2004) presented the formal definitions for rough sets theory.

Let U be any finite universe of discourse. Let R be any equivalence relation defined on U. Clearly, the equivalence relation partitions U. Here, (U, R) which is the collection of all equivalence classes, is called the approximation space. Let W_1, W_2, W_3 …, W_n be the elements of the approximation space (U, R). This collection is called as knowledge base. Then for any subset A of U, the lower and upper approximations are defined as follows:

$$\underline{R}A = \mathrm{Y}\{Wi / W_i \subseteq A\}$$

$$\overline{R}A = \mathrm{Y}\{Wi / W_i \cap \neq A\Phi\}$$

The ordered pair $(\underline{R}A, \overline{R}A)$ is called a rough set. Once defined these approximations of A, the reference universe U is divided in three different regions: the positive region POS_A(A), the negative region

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NEG_A(A) and the boundary region BND_A, defined as follows:

$$POS_R(A) = \underline{R}A$$
$$NEG_R(A) = U - \overline{R}A$$
$$BND_R(A) = \overline{R}A - \underline{R}A$$

Hence, it is trivial that if BND(A) = Φ, then A is exact. This approach provides a mathematical tool that can be used to find out all possible reduces. The process of finding a set of reducts is basically a feature selection process. Feature selection is a process to choose a subset of attributes from the original set of attributes. Feature selection has been studied extensively in the past decades [9, 10, 11]. The purpose of the feature selection is to identify the significant features, eliminate the features irrelevant to the learning task, and build a good learning model. The benefits of feature selection are twofold: it considerably decreases the computation time of the induction algorithm, and increases the accuracy of the resulting model. In some cases it is possible that a decision table may have more than one reduct. Anyone of them can be used to replace the original table. A natural question is which reduct is the best if there exists more than one reduct. The selection depends on the optimality criterion associated with the attributes. If it is possible to assign a cost function to attributes, then the selection can be naturally based on the combined minimum cost criteria. In the absence of an attribute cost function, the only source of information to select the reduct is the contents of the data table [11]. From simplicity, we adopt the criteria that the best reduct is the one with the minimal number of attributes and that if there are two or more reducts with the same number of attributes, then the reduct with the least number of combinations of values of its attributes is selected.

III. PRE-PROCESSING USING PRINCIPAL COMPONENT ANALYSIS (PCA):

PCA [12] is a way of identifying patterns in data, and expressing the data in such a way as to highlight their similarities and differences. Since patterns in data can be hard to find in data of high dimension, where the possibility of graphical representation is not available, PCA is a powerful tool for analyzing data. The main advantage of PCA is that once these patterns are found, the data can be compressed i.e. the number of features (dimensions) can be reduced without much loss of information. In this paper, the original dataset consists of 452 instances each having 279 attributes. We have performed a Principal Component Analysis of the data using WEKA [13] and reduced the number of features to 103.

IV. DATASET AND ALGORITHMS:

The dataset used in this study was obtained from the archives of machine learning datasets at the University of California, Irvine [14]. The dataset is grouped into 3 broad classes to facilitate its use in experimentally determining the presence or absence of arrhythmia, and for identifying the type of arrhythmia. In the set, class 01 refers to 'normal' ECG. Classes 02 to 15 refer to different classes of arrhythmia and class 16 refers to the rest of unclassified data. There are missing values in the dataset. In such cases, probabilistic values were assigned according to the distribution of the known values for the attribute.

A. Discretization:

For discretizing the data we have used the Boolean Reasoning Algorithm. In this algorithm, the cuts found by the Naivescaler algorithm [15] are combined with a Boolean reasoning procedure for discarding a subset of these cuts. The remaining subset is a minimal set of cuts that preserves the discernibility inherent in the decision system.

The algorithm operates by first creating a Boolean function f from the set of candidate cuts and then computing a prime implicant of this function. The greedy algorithm of Johnson [16] is used to compute the prime implicant. The Boolean function f is given by

$$f = \prod_{(x,y)} \sum_a \{\sum c^* \mid c \in C_a, and, a(x) < c < a(y)$$
$$and, \partial_A(x) \neq \partial_A(y)\}$$

where a denotes condition attribute and the set C_a is computed as follows:

For each condition attribute a, its value set V_a is sorted to obtain the following ordering:

$v_a^1 < \ldots\ldots < v_a^i < \ldots\ldots v_a^{|V_a|}$

Let C_a denote the set of all naively generated cuts for attribute a, defined as shown below.

The set C_a simply consists of all cuts midway between two observed attribute values, except for the cuts that are clearly not needed if we do not bother to discern between objects with the same decision values.

$$X_a^i = \{x \in u \mid a(x) = v_a^j\}$$
$$\Delta_a^i = \{v \in V_d \mid \exists x \in X_a^i.such.that.d(x) = v\}$$
$$C_a = \left\{ \frac{v_a^i + v_a^{i+1}}{2} \mid\mid \Delta_a^i \mid > \mid \Delta_a^{i+1} \mid > 1, or, \Delta_a^i \neq \Delta_a^{i+1} \right\}$$

In essence, cuts are placed midway between all v_a^i and V_a^{i+1}, except for in the situation when all objects that have these values also have equal generalized decision values with respect to a that are singletons.

B. Reduction:

In this process, a set of reducts is obtained for each instance in the dataset. In our case, we have used the genetic algorithm [17] to compute the reducts.

The algorithm's fitness function f (B) is defined below, where S is the set of sets corresponding to the discernibility function and A is the set of condition attributes. The parameter α defines a weighting between subset cost and hitting fraction, while ε is relevant in the case of approximate solutions.

$$f(B) = (1-\alpha)\frac{\mathrm{cost}(A)-\mathrm{cost}(B)}{\mathrm{cost}(A)} + \alpha.\min\left\{\varepsilon_r \frac{|S.in.S \mid S \cap B \neq \Phi|}{|S|}\right\}$$

The subsets B of A that are found through the evolutionary search driven by the fitness function and that are "good enough" hitting sets, i.e., have a hitting fraction of at least ε, are collected in a "keep list". The size of the keep list can be specified. The function cost specifies the cost of an attribute subset. If no cost information is used, then a default unit cost is assumed, effectively defining cost (B) = |B|.

Each reduct in the returned reduct set has a support count associated with it. The support count is a measure of the "strength" of the reduct, and may interpret differently according to which algorithm that produced the reduct. For reducts computed with this algorithm, the support count equals the reduct's hitting fraction, multiplied by 100.

V. EXPERIMENTAL RESULTS AND CONCLUSION:

We have selected the SVM classifier as a baseline to compare the performance of Rough Set based classification. Though results for SVM based classification have already been reported [7] there will be differences with respect to factors like dataset used, types of features used, number of classes used in multiclass classification etc., so we have implemented our own SVM based classification (both binary and multiclass) in order to maintain homogeneity in the process of comparison and used it to benchmark the performance of Rough Sets. Out of 452 instances 292 were selected for training and the rest were used for testing. The SVM based classification was implemented using LIBSVM [18].The experimental results for SVM based classification are reported in table II. For binary classification, class 1 was taken as the normal class and all the other classes were considered as arrhythmia cases while for multiclass classification we used the 16 classes as described in the dataset [14].We have used the Radial Basis Function (RBF) kernel for experiments. The other parameters used in experiments were determined by cross-validation of training dataset.

Table II: SVM based classification results

Classification Criterion	Accuracy
Binary	98.12%
Multiclass	94.37%

The Rough set based classification has been implemented in Rosetta [19].The experimental results for Rough set based classification (using reduced features) are reported in table I and table IV. We also analyzed the dataset without reducing the number of features and the classification accuracies were found to be lower and unsuitable for any practical purposes. Experimental results for classification without reducing the number of features is reported below:

Table III: Classification using Rough sets without feature reduction:

Classification Criterion	Accuracy
Binary	61.22%
Multiclass	54.54%

Table IV: Confusion Matrix for binary classification using Rough sets (using reduced features):

	1	2	
1	74	5	93.67%
2	4	66	94.28%
	94.87%	92.95%	93.95%

Tables I and IV also indicate classification accuracy for different classes as well as the overall accuracy. As indicated, the overall classification accuracy for binary classification was found to be 93.95% whereas for multiclass case the overall classification accuracy was 87.24%.

VI. CONCLUSION

Possibility of a classification algorithm for Arrhythmias using RST is explored. A dataset was used to demonstrate the accuracy and effectiveness of this method. It is acceptable, though not outstanding accuracies are obtained. Also, the ability of RST to handle uncertainties in data renders it more suitable for the diagnosis of ambiguous patterns in the data. Since many rules and reasonings can be generated, many new types of classification of Arrhythmia are possible, which means that it is scalable. The computational efficiency of RST makes it the system of choice for portable ECGs. In reasoning, unlike Fuzzy sets it does not assume any fixed distributions and same distributions for the different variables. This makes Rough set approach more suitable for such medical diagnosis. Rough Set Theory may be combined with other classifiers like SVM to enhance the accuracy and efficiency of the present system. However this will increase the computational complexity of the system.

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Table I. Confusion Matrix for multiclass classification using Rough sets: (using reduced features)

	1	2	3	4	5	6	7	8	9	10	14	15	16	
1	72	0	0	0	0	3	0	0	0	4	0	0	0	91.13%
2	2	23	0	0	0	3	0	0	0	0	0	0	0	92%
3	0	0	6	0	0	0	0	0	0	1	0	0	0	85.71%
4	0	0	0	3	0	0	0	0	0	0	0	0	1	75%
5	0	0	0	0	1	1	0	0	0	0	0	0	0	50%
6	0	0	0	0	0	12	0	0	0	0	0	0	1	92.3%
7	0	0	0	0	0	0	1	0	0	0	0	0	0	100%
8	0	0	0	0	0	0	0	3	0	0	0	0	0	100%
9	0	0	0	0	0	0	0	0	2	0	0	0	0	100%
10	1	0	0	0	0	0	1	0	0	6	0	0	3	54.54%
14	0	0	0	0	0	0	0	0	0	0	2	0	0	100%
15	0	0	0	0	0	3	0	0	0	0	0	1	0	25%
16	0	0	0	0	0	1	0	0	0	0	0	0	1	50%
%	96	100	100	100	100	52	100	100	100	54.5	100	100	16.6	87.24%

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Cooperative Trust Model for a Decentralised Peer-to-Peer E-Market

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Abstract- **This article describes a trust model for use in a distributed peer-to-peer e-market with no central controlling node. The issue of trust in this kind of network is vitally important to its success as users are less likely to participate in a market with untrustworthy members. With no form of centralisation and control, peers must cooperate to determine which peers are trustworthy. The goal for this trust model is to make it hard for an illegitimate peer to participate in the network while making it easy and safe for legitimate peers to participate. The trust model presented in this article allows peers to work together to determine another peers trust value through cooperative message passing. It also provides a fair method for recognising potentially malicious peers through the responses they provide. Finally, we describe constraints which ensure the trust model maintains the scalability of the peer to peer network.**

I. INTRODUCTION

A peer-to-peer e-market is an approach to online commerce which utilises the benefits if the peer-to-peer network design. Each participant in the system holds equal power and influence making the system more robust by avoiding the client-server approach of traditional e-commerce. The peer to peer approach allows the market to avoid the strict rules usually imposed on electronic transactions [1]. This can allow buyers to make non-standard requests and compare offers from vendors. Vendors can cooperate to meet a buyer's request, allowing high flexibility in the way items are bought and sold. The flexibility and openness of a peer-to-peer e-market leaves it open to malicious activities from participating peers who seek to further their own interests. We introduce a way for peers to work together cooperatively to discover potentially malicious peers and prevent them from participating in the market.

The issue of trust in a peer to peer community is extremely important, particularly when those peers are participating in economic transactions. In a peer-to-peer e-market, a high level of trust is needed to encourage use of the network. The more trust that exists in the network, the more use it is likely to get [2]. Vendors within such a community can improve their appeal to customers through building upon their reputation. An example of a reputation based trust model currently in use is the online auction site eBay® [3]. By allowing users to rate the performance of a vendor, other users are able to use this information to decide whether or not to trade with that vendor. This system makes use of a secure centralised database which controls reputation information. In a peer-to-peer e-market there is no central authority, with all peers having equal rights and responsibilities in the network. This presents several problems to be solved in order to implement a trust model for a decentralised peer-to-peer e-market.

The trust issues that must be dealt with by a trust model include impersonation, fraudulent actions, mis-representation, collusion and unknown peers [4]. In dealing with these issues it is important that the trust model does not create too much network traffic or require large amounts of local storage. If this occurs the system will not cope as the network grows large and the need for reputation information increases. One approach to this is the use of groups, which delegate trust responsibility to a leader [5] [6]. In an ideal e-market however, all peers should be equal, making this approach undesirable.

The first problem to solve is that of how to represent trust in such a system. In other words, what does it mean to say that A trusts B? In this paper we propose two levels of trust, transaction trust and information trust, earned through direct transactions and the sharing of accurate information respectively. At this stage we represent these trust values as a function of the number of positive and negative experiences involving that peer.

The second problem to address is how to build these trust values in a peer-to-peer e-market. The approach taken in this paper lets peers cooperate to communally develop trust values for other peers. By cooperation we mean that peers work together to build trust values for other peers in the network. Peers can either ask for trust information from the network or report trust information. Approaches which are similar to this include EigenTrust [7] and TruthRep [6]. The method in this paper differs from these examples by recognising and punishing peers which provide malicious feedback. This recognition was looked at by Mekouar et al. [8] but not in a fully distributed environment.

The trust model we propose in this article is intended to be implemented on an unstructured peer-to-peer overlay network. Examples of this type of network include Gnutella[9] and Kazaa[10]. These networks are reliable, handle peer transience well and provide flexible querying [11]. Structured overlays such as Pastry [12] and Chord [13] provide faster searching and resource discovery guarantees, however they

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require precise knowledge of the searchable resource and high overhead to maintain their structure [15]. Peer-to-peer e-markets make use of keyword searches for resources with unspecific names, while a search may return more than one result. Hence, unstructured-overlays are more appropriate for a peer to peer e-market. For this reason our trust model is designed for such an unstructured-overlay.

In this paper we develop a method which operates in a fully distributed environment, allows peers to co-operate, recognises malicious activity and is scalable with the size of the e-market. In section 2 we describe the issues involved in developing this kind of trust model. Section 3 describes the operation of the trust model and a brief conclusion is given in section 4.

II. TRUST ISSUES

Trust is essentially derived from the outcome of a commercial transaction. At the conclusion of the transaction, if the buyer is satisfied with what they have received, they can then say that the transaction was a positive one. Similarly, if the seller is satisfied that they have received their payment then they have had a positive experience with the buyer. The more transactions that the seller completes satisfactorily, the higher the associated level of trust it gains. Trust then becomes a commodity for the seller as it makes them more attractive to buyers than other sellers with a lower level of trust. Another important point is that one bad transaction should not disproportionately affect an agent's trust status. Thus, any trust model must allow the agent to gain trust, improve trust and loose trust.

With these issues in mind we can say that a successful trust model for a decentralised peer-to-peer E-Market will need to meet the following conditions:

- Provide peers with accurate trust information on other peers
- Does not require any central authority
- Does not flood the network
- Does not require large amounts of local storage
- Is fully scalable
- Can deal with groups of malicious peers
- Considers positive and negative feedback
- Allows users with no prior information to enter the network

III. NETWORK STRUCTURE

As mentioned in the introduction, this trust model is designed for use with an unstructured peer-to-peer overlay-network. This basically means that resources, such as buyers and sellers of products, can be located at any peer in the network. This compares with structured-overlays such as Pastry [12] or chord [13], which locate resources at specific peers and require more work to maintain the network structure. The main problems which need to be addressed include search flooding and unguaranteed search results. Therefore we need to design the trust model with these issues in mind.

The problem of query flooding is caused as messages propagate through the network. This happens when one peer sends a message to all of its neighbours who in turn send it to all of their neighbours and so on. To reduce the amount of messages propagating through the network we set a hop count (as used in [1]) and a maximum neighbour count on each message.

A hop count is intended to limit the depth of a message by setting a limit on the number of hops a message can take. A 'hop' represents how many peers a single message has passed through. At each hop the message is sent to multiple peers, meaning that the number of messages increases exponentially for each hop that it takes. A hop count limits this by setting a maximum for the number of hops which can be taken.

We combine the hop count with a limit on the number of neighbours which get sent the message at each hop. This is a restriction on the breadth of each search, which can be increased to improve the quality of the results or decreased to improve the performance of the network.

Unstructured peer-to-peer overlay-networks also require local storage of information to allow peers to communicate with their neighbours. Our trust model requires local storage of trust information stored in a trust table (see Fig1.). This trust table could be implemented as part of a neighbour table used by the e-market. It is important to ensure that the size of the trust table does not become prohibitively large, such that it does not require large storage and processing power. To prevent this table from becoming too large, a maximum limit for its size should be set. When the limit is reached, new entries can only be made after another entry has been removed.

The steps described in this section will allow the trust model to operate within the framework of a peer-to-peer e-market. By imposing constraints on message passing and local storage, the trust model will fit in with an unstructured-overlay, while not causing too much burden on the network or individual peers.

IV. DECENTRALISED TRUST MODEL

In order to fit in with the concept of the decentralised market, the trust model needs to be implemented in a dynamic and decentralised way. There are several features of this trust model:

1. Each agent maintains a trust table of known associate peers' trust levels

2. Each agent makes decisions about the trust level of its associates
3. Agents can tell others about their trust of another agent
4. An agent can ask others for trust information about a particular agent
5. Agent's responses are analysed to determine quality relative to other responses.
6. A mechanism to develop initial trust

A. Transaction and Information Trust

This trust model makes use of two forms of trust; transaction trust and information trust. Each trust type is represented via two numbers: number of positive experiences and number of negative experiences.

Transaction trust is earned through direct transactions between two peers. If one peer meets the conditions of a transaction then the number of positive experiences of their transaction trust is incremented. If the conditions are not met then the number of negative experiences of transaction trust is decremented.

Information trust is used to represent the quality of information one peer provides about other peers. It can be built over time as more and more 'good' information is sent to other peers. The quality of the information is judged relative to other peers' responses. If 99% of peers say they have high trust for another peer and 1% say they don't, there is a higher chance that that 1% is providing false information. If information provided is satisfactory then the positive experiences of the information trust is incremented, otherwise the number of negative experiences of information trust is incremented. This process is described in more detail in Section C.

The reason for having an information trust and a transaction trust field is because, if there were only one trust value, an anonymous peer could gain trust through information sending alone. By joining anonymously and building a trusted list of peers which it passes trustworthy information to, it would itself gain a high level of trust. It is foreseeable that a malevolent peer could gain trust before even attempting a transaction. This trust could be used to enter into fraudulent transactions. Transaction trust should only be gained by satisfactorily completing transactions, meaning that trust is built up as you complete more and more transactions.

In this model peers cooperate by sharing their trust values (t), rather than the number of positive and negative experiences(n_+ and n_-). The first reason for this is that it distributes the processing more evenly through the network. Each peer calculates the required trust value, which is preferable to one peer calculating many trust values, as would happen if positive and negative experience counts were sent. The second reason is that it prevents a malicious node from skewing the results by sending a limitless value for either positive or negative experiences. As described in part B, trust values are limited between -1 and +1 restricting the ability of a malicious peer to skew the results.

B. Trust Valuations

As mention above, the trust values will be represented based on positive and negative experiences. If n_+ is the number of positive experiences with a subject, Beth et al. [14] calculated the trust purely based on positive experiences using

$$t_+ = 1 - (l_+)^{n_+} \tag{1}$$

where l_+ is the learning rate of positive experiences. Similarly, trust resulting from negative experiences can be written as

$$t_- = 1 - (l_-)^{n_-} \tag{2}$$

We fuse the positive trust and negative trust as

$$t = t_+ - t_- \tag{3}$$

These trust values range from -1 to +1 with +1 indicating fully perfect trust and -1 indicating fully distrust. Naturally negative learning rate is higher than the positive learning rate. The learning rate for transaction trust and information trust may differ as well. We use a higher learning rate for transaction trust and a smaller learning rate for information trust.

C. Peer Trust Table

To keep trust data, regarding other peers, each peer will maintain a trust table with the attributes shown in Fig. 1.

This table will be used by peers to make decisions about information they receive about other peers. Peer Id is used to distinguish between peers, information trust is used to indicate the trustworthiness of the information from a peer while transaction trust is used to signify the trustworthiness of the peer when dealing with transactions. The following algorithm, used to maintain the trust table, is triggered by a peer that requires trust information about another peer (for example, after receiving an offer of goods from peer A):

1. Advertise to a random group of peers for transaction trust or information trust rating of peer A via a trust request message
2. Wait for peers to respond with their trust rating of peer A from their own trust table
3. Store each response including the identity of the responding peer as well as the trust value they provided
4. Wait until a statistically significant sample is received
5. Statistically analyse results to gain a trust rating (See section D)
6. Add peer A to trust table with associated trust rating

See appendix for an example of this algorithm

Trust Table				
Peer Id	Information Trust		Transaction Trust	
	n_+	n_-	n_+	n_-

Fig 1. The attributes stored in a Trust Table

Once this table has been established it can be used to identify preferred informants as those with high trust values. A high trust value can only be gained by providing continually trustworthy information.

D. Statistical Analysis

After a statistically significant sample has been received the responses can be statistically analysed. The first purpose of this analysis is to determine a trust rating for a peer based on the responses of other peers. The second purpose is to identify peers which have provided poor information which could be deemed to be malicious. The information to be used in this analysis is collected in the algorithm described in section C. In this article we determine mean and standard deviation from the received responses. If the standard deviation is small enough the results can be used to identify potentially malicious individuals. In future work this statistical analysis could be further developed to handle more complex situations and gain more information from the data collected.

The statistical analysis algorithm is triggered once a statistically significant sample has been received from the network. The algorithm would then follow these steps:

1. Calculate the sample mean and standard deviation of the trust values.
2. If the standard deviation is too large then discard outliers and recalculate
3. If standard deviation is still too large take mean as trust value and exit
4. If standard deviation is small enough
   a) For each response OUTSIDE 2 standard deviations: increment the number of negative experiences of the peers information trust.
   b) For each response WITHIN 1 standard deviation: increment the number of positive experiences of the peers information trust.

When this algorithm exits, execution is returned to the main algorithm described in part C.

As discussed in Section IV, Part A, this statistical analysis is performed on trust values rather than positive and negative experience counts (n_+ or n_-). The trust table however is designed to hold n_+ or n_- to represent trust, meaning that we need to derive these values from the trust value found during statistical analysis. To do this we find a value for n_+ or n_-, which satisfies the trust value found in the above algorithm as follows:

For a given trust value t:
If t = 0 then
  Then set n_+ and n_- to 0
Else if t > 0 then
  $n_- = 0$
  $n_+ = \ln(1 - t) / \ln(l_+)$
Else if t < 0 then
  $n_+ = 0$
  $n_- = \ln(1 - t) / \ln(l_-)$

These values can then be entered into the trust table, while allowing the statistical analysis to be performed on trust values alone.

F. Reporting

The final job of this trust model is to allow one peer to report its trust value of another peer to the community. This would occur after a transaction between two peers or after running the cooperative trust algorithm described above. It is important to be able to communicate this information otherwise there is potential for non-malicious peers to be branded as malicious.

The trust value is sent to its neighbours who can update their own trust tables or ignore the information. This depends on whether or not they trust the peer that sent the trust value. By this method, peers can cooperate to reduce the amount of trust calculations which need to be performed at each individual peer. As peers build trust, the amount of trust calculations should reduce as peers are more likely to accept the work of their neighbours in calculating trust.

The following steps are used in the reporting of trust information between peers:

1. Peer (*A*) would calculate a transaction trust value for another peer (*X*)
2. *A* selects a group of neighbour peers to inform
3. Set hop-count and time-to-live values and send trust value to those peers

Now, when a neighbour receives this transaction trust information it would follow these steps:

1. Look-up trust table to find trust value for peer *A*
2. If information trust value for *A* is high then:
   a) add transaction trust value for *X* to trust table, increment information trust of *A* and send information to neighbours
3. If information trust value for *A* is low then:
   a) If *X* is already in trust table:
      i. If value for *X* is similar to that of *A*, increment trust value for *A* and send to neighbours

ii. If value for X is dissimilar to A, discard message, decrement trust value for A

4. If information trust value for A is neutral or non-existent then seek the help of the community to make a decision by initiating the algorithm from section C.

G. Other Considerations

New Entrants: This model provides support for new entrants into the network. Before a peer has any detailed knowledge of other nodes, they could send a broadcast message to a much larger sample of the network than would normally occur. The large result set could be statistically analysed for good and bad responses leading to a trust table being initialised. Several more initialisation searches could be performed to make the truth table more robust. From here the new peer could start to interact, receiving and sending information, building trust and transacting. The truth table will continually change as requests and responses arrive, gaining strength and integrity over time.

Incentives: A feature of peer to peer systems is that peers, being essentially equal, can be expected to act rationally[15]. This basically means that they act in their own self interests. This trust model utilises the rationality of peers as it is beneficial for them to cooperate with each other. When a peer receives a request for trust information it is in their interest to respond as it will improve their trust value at the requesting peer. This makes it more likely that 'good' information is sent about them through the network. This in turn makes it more likely that they will receive information about other peers as rational peers would favour communicating with highly trusted peers. So an individual peers desire to build trustworthy relationships, which allow mutually beneficial information to be shared, ensures that peers will be motivated to participate with this trust model. By the same logic, malicious peers will be discouraged from participating as 'bad' information about them will be sent through the network, limiting their ability to participate in E-market functions.

V. FAIRNESS

The decentralised trust model described in part IV is designed to be fair to all peers, making it hard for a peer to be unfairly labelled as untrustworthy and for a malicious peer to continually provide untrustworthy information. Fairness is provided by; minimally punishing one-off bad responses, only punishing when there is strong statistical evidence and allowing peers to inform others of trust ratings.

With this trust model it is possible for a peer to provide a bad trust value with no malicious intention. It would be unfair to punish this peer to the extent that they could not participate in the network due to a bad reputation. We guard against this by incrementally punishing bad responses. One bad response only results in that peers negative experience count being incremented. If the same peer has previously provided many good responses then the negative response will be relatively insignificant. However, we do state that the negative learning rate should be greater than the positive learning rate, meaning that one bad response is more significant than one good response. This prevents malicious nodes repairing their reputation with the occasional good response.

In selecting potentially malicious responses through statistical analysis, our trust model requires that they are outside two standard deviations of the mean, the standard deviations are not too large and the sample is suitably large. In other words, malicious responses are only acted upon if the sample is statistically significant. This reduces the chance of a peer being unfairly punished due to statistical errors in the response.

The ability of peers to tell others about trust values they have found is important in ensuring fairness. If one peer discovers an untrustworthy peer it can tell other peers about it. Peers can accept this information if they trust the sending peer enough. If the peer is trustworthy the information will pass through the network. If the peer could not inform its neighbours of trust values they may be isolated as the only peer with a certain trust value. This could result in that peer being recognised as malicious.

These measures help to ensure that peers are not identified as malicious through statistical error or isolation. If they are somehow wrongly identified as suspicious they are not severely punished for one-off offences. If a peer is identified as providing suspicious responses they can be informed so they know they have been identified this way. At the same time malicious peers will find it hard to operate once they have been well established.

VI. CONCLUSION

This paper has presented the trust issues which we see as being relevant to a decentralised peer-to-peer e-market. The model put forward to address these issues makes use of the resources of its neighbouring peers to build a trust table which will gain strength over time. It takes advantage of the size of the network by statistically analysing the responses of peers to determine fraudulent responses from potentially malicious peers. This could be a failing point if more responses were to come from malicious peers than genuine peers, however, the size of the sample should rule out this possibility unless the network itself consists of a majority of malicious peers. We feel that if this were the case the network would fail anyway which would not be in anyone's interest.

The trust model effectively takes an 'innocent until proven guilty' approach to identifying malicious peers. This ensures that peers who mistakenly provide a bad trust value will not be overly punished or excluded from the network. If a peer continues to provide poor responses then they are justifiably identified as malicious. In this way the trust model works fairly for participating peers while still recognising and punishing malicious peers.

When combined by other security measures such as digital signatures, the trust model will provide an important function within a peer to peer e-market. This type of market will be far more likely to succeed if participants feel confident in the trust of potential trading partners. Cooperation allows users to be involved in the process of building trust, giving them a deeper understanding and confidence in the market.

An area for future work with this model is the development of the way trust is represented. In this paper we have used a simplistic single-value representation of trust. In reality, trust is influenced by many factors, meaning it is overly simplistic to say that trust is a binary value. The use of trust ontologies [16] would further improve this model by giving a deeper meaning to the value of trust.

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APPENDIX

A very simplistic example of this trust model:

Peer A searches for a product and receives an offer from Peer X. Peer A does not have a trust value so decides to ask for help from its neighbours B, C and D.

Initially Peer A's trust table looks like this:

Peer ID	Transaction Trust		Information Trust	
	n_+	n_-	n_+	n_-
B	0	0	10	0
C	0	0	10	0
D	0	0	10	0

After Peer A asks B, C and D for a transaction trust value for Peer X it gets the following responses:

Peer B: 0.4
Peer C: 0.4
Peer D: -0.7

After performing statistical analysis, Peer A determines that Peer D provided a poor response which is excluded, giving a mean of 0.4 from the other responses. Since this is positive, peer A determines a value for n_+ for peer X as:

$n_+ = \ln(1 - 0.4)/\ln(l_+)$ where l_+ is the positive learning rate

If $l_+ = 0.9$, then n_+ will approximately equal 5.

So the process has found a transaction trust value for peer X as well as two 'good' responses from B and C plus a 'bad' response from peer D. The trust table would then be updated to look like this:

Peer ID	Transaction Trust		Information Trust	
	n_+	n_-	n_+	n_-
B	0	0	11 (+1)	0
C	0	0	11 (+1)	0
D	0	0	10	1 (+1)
X	5	0	0	0

Note: even though peer D was deemed to have provided a bad response it has not changed its history of good responses. This prevents peers from being blacklisted based on one-off bad responses. We do however acknowledge that the impact of one bad response is greater than one good response, which is reflected in the fact that negative trust experiences have greater influence on total trust values.

Development of Quality Expectations in Mobile Information Systems

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***Abstract*— In a typical study on mobile information system (IS) adoption and use the established methods and theories like TAM have been extended to the mobile context. The holistic views of these theories unfortunately fail to give detailed advice on how to influence usage through design and implementation. Some scholars integrated quality concepts into the acceptance models. Quality is however a complex and multidimensional concept including not only the perceived but also the expected quality dimensions. In this paper the development of quality expectations related to mobile information systems is studied in a four year period (2003 – 2007). The analyses are based on the quality framework that consists of three parts of mobile IS supply chain. The results of the study highlight the importance of the quality expectations and they support the use of original intention determinants of TAM also in the mobile context. Further studies are still needed to integrate TAM with quality and satisfaction concepts in the mobile IS adoption and use studies.**

I. Introduction

Factors and processes affecting user perception of information technology (IT) have received a lot of interest among IT researchers. Scientists have several theories to explain why and how users accept or reject some technologies and services. Well-know adoption theories include e.g. Innovation Diffusion Theory [1], Technology Acceptance Model [2], and Theory of Reasoned Action [3], Task Technology Fit [4] and Unified Theory of Acceptance and Use of Technology [5]. Although these theories have their differences they all try to explain the reasons and processes of affecting information system acceptance and success. These classical models have also been applied in many ways to mobile information systems (IS) but according to many scholars they should first be modified or extended according to the special characteristics of mobile systems.

Classical acceptance models have provided valuable information about the adoption and use processes. They have however given only limited guidance about how to influence usage through design and implementation [6]. Many scholars have pointed out the importance of quality as a key determinant of user satisfaction and system success (e.g. [7], [8]). Quality is an important factor for mobile information systems, not only for acquiring customers by satisfying them during their first time use, but also for achieving customer loyalty by making the customers use them again and again in the future [7].

In this paper a quality framework for mobile IS is developed. The model is then used to analyze the development of the quality expectations during a longer period of time. The aim of the paper is to find out how users' opinions on the importance of different quality determinants in mobile systems and services have changed during the last four years. During that period of time technological development in mobile networks and devices has been fast and new features and applications have been introduced (e.g. build-in cameras, 3G networks, Multimedia Messaging). The analysis is based on two surveys carried out in 2003 and 2007.

The structure of the paper is as follows. In Chapter 2 a short literature review on technology acceptance studies in the mobile context is presented. Technology Acceptance Model (TAM) and its use in the mobile context are analyzed in detail. TAM qualifies as a starting point of the study because it is one of the most influential and commonly employed IS models [9], [10]**.** In Chapter 3 the different dimensions and aspects of quality are discussed and a new quality framework for mobile IS is developed. The surveys and their results are discussed in Chapter 4 and the final conclusions are made in Chapter 5.

II. Technology Acceptance in Mobile Context

A. Technology Acceptance Model

Technology Acceptance Model (TAM) [2] is a popular theory of explaining the acceptance of technology amongst individuals. TAM is based on the psychological models of Theory of Reasoned Action (TRA) [3] and Theory of Planned Behavior [11]. TAM uses TRA as a theoretical basis for specifying the causal linkages between the key features: the perceived usefulness, the perceived ease of use, the users' intentions and the actual use of the system.

According to TAM the perceived ease of use and the perceived usefulness are the sole determinants of behavioral intention to use the system, which in turn predicts actual behavior as shown in Fig. 1.

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K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 336–341.

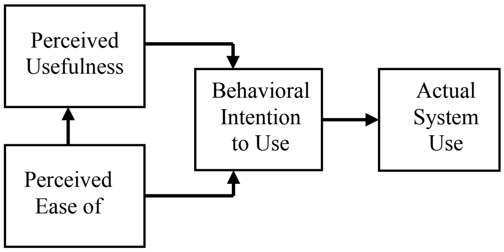

Fig. 1. Technology Acceptance Model (TAM) [2]

Many modifications and enhancements to the model have been suggested. A typical extension suggests either an introduction of new antecedents of intention or a context specific set of determinants for the perceived ease of use and the perceived usefulness [12]. Examples of this kind of studies include [13], [14] and [15]. The detailed analysis of TAM and its extensions is not included in this paper. A comprehensive review on TAM studies can be found e.g. in [16].

B. *TAM and Mobile Information Systems*

TAM has been applied to different kinds of systems and environments including mobile systems. The special characteristics of mobile contexts have required scholars to modify the original model before they have been able to apply it. Some examples of these extended models are introduced below.

Based on the results of domestication research Pedersen [12] has suggested adding the perceived self-expressiveness as additional independent variable in TAM. Self-expressiveness here includes both the social expression of identity and self-identification. Both concepts are important elements in the adoption and use of mobile services.

In their Compass Acceptance Model Amberg et al. [17] have four dimensions that focus on subjective perception. The first two, usefulness and ease of use, come from the original TAM. In addition, two new dimensions are added: the perceived mobility and the perceived cost. These new dimensions have their origins in the concepts of *benefits and efforts* and *service and general conditions of service.*

In her model Kaasinen [18] introduces two new constructs: trust and the perceived ease of adoption into the original TAM. She also suggests that the perceived usefulness should be replaced with the perceived value and special attention should be paid to the time of users taking the system into use.

Because information systems are not only used for work but also for entertainment some scholars have suggested that the entertaining features should be added into the TAM. Following this stream of research Cheong and Park [19] have added the perceived playfulness as the third determinant of intention or attitude towards the mobile Internet. Table 1 summaries the antecedents of intention used in the TAM models modified for mobile IS.

TABLE 1
ANTECEDENTS OF INTENTION IN SOME MOBILE TAM MODELS

Perceived antecedent	Pedersen [9]	Amberg et al. [14]	Kaasinen [15]	Cheong & Park [28]
Usefulness	X	X	X*	X
Ease of use	X	X	X	X
Self-expressiveness	X			
Mobility		X		
Cost		X		
Ease of adoption			X	
Trust			X	
Playfulness				X

* Renamed as the perceived value

The short literature review here indicates that scholars have various views of how the model should be extended and applied to the mobile context. Interestingly all extended versions still have the two intention determinants of the original TAM model: the perceived ease of use and the perceived usefulness. This highlights the importance of these aspects and thus supports the original TAM model.

III. QUALITY IN MOBILE CONTEXT

A. *Different Dimensions of Quality*

Although the satisfaction and usage constructs have been well studied in the information systems literature, there has been only limited attention to the information and system quality over the past decade [20]. However, quality is an important factor for information system success. Quality is not a one dimensional phenomena but a more complex structure. For example, the quality model of ISO-25000 [21] presents a wide scope and variety of general characteristics of a software product. In addition, it divides software quality characteristics into three categories: internal quality, external quality and quality in use attributes.

The ISO-25000 analyzes quality from the product perspective. The alternative approach is to view quality from the service perspective. The basic definition of the Quality of Service (QoS) can be found in [22]. It defines QoS as: "*the collective effort of the service performance, which determines the degree of satisfaction to the end user*". Naturally there are also other definitions. For instance the European Union's R&D in Advanced Communications technologies in Europe (RACE) program defines [23] QoS as: "*a set of user–perceivable attributes of that which makes a service what it is. It is expressed in user–understandable language and manifests itself as a number of parameters, all of which have either subjective or objective values*".

According to this definition, QoS covers both subjective and objective aspects of a service. This means that there are several different definitions and viewpoints from which to analyze quality. In recommendation G.1000 [24], the International Telecommunications Union – Telecommunication (ITU-T) has presented four viewpoints of QoS as shown in Fig. 2.

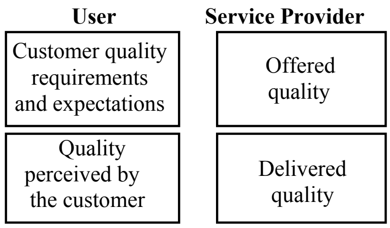

Fig. 2. Viewpoints of Quality [modified from 24]

According to Fig. 2 users' quality expectations and requirements are also important in the overall quality construct. Earlier studies have highlighted the importance of the quality expectations in mobile IS. Some of the main reasons why mobile Internet users get disappointed are the limitations that distinguish mobile devices from conventional desktop PCs [25].

B. *Quality Frameworks for IS*

Providing high quality for customers is the key success determinant of the mobile information services. In their seminal work on IS success DeLone and McLean [26] distinguished two quality dimensions: the system and the information quality. Later they also added a third component the service quality to their model [27] as can be seen in Fig. 3.

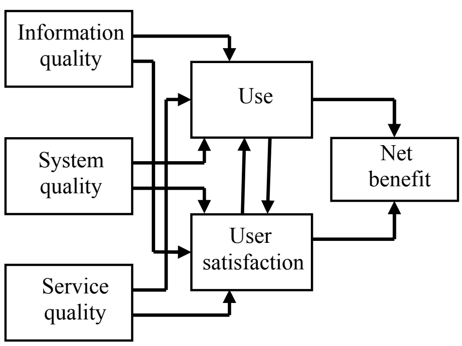

Fig. 3. Revised DeLone and McLean IS Success Model [27]

Nelson and Todd [20], on the other hand, have developed a model consisting of nine fundamental determinants of quality in an information technology context. Four of them determine the information quality (the output of an information system) and five of them describe the system quality (the information processing system required to produce the output). Their empirical examination with data warehouse users from seven different organizations suggested that these determinants are indeed predictive of the overall quality.

Based on the original DeLone and McLean IS success model Cheong and Park [19] included two quality constructs, the content quality and the system quality, in their research model on the mobile Internet acceptance. The perceived content quality is similar to the information quality used by DeLone and McLean because information is often regarded as content in the Internet context.

In their quality framework for mobile Internet services Chae and Kim [7] have four major dimensions: the connection quality, the content quality, the interaction quality and the contextual quality. This framework differs from others by pointing out the importance of networks and device characteristics in the overall quality.

In Table 2 the quality dimensions of the introduced frameworks are listed. The influence of the original DeLone and McLean IS success model is strong and the information and the system quality dimensions are strongly supported by other scholars.

TABLE 2
DIMENSIONS OF QUALITY IN SOME FRAMEWORKS

Quality Dimensions	DeLone & McLean [27]	Nelson& Todd [20]	Cheong& Park [19]	Chae& Kim [7]
Information Quality	X	X	X*	X*
System Quality	X	X	X	
Service Quality	X			
Connection Quality				X
Interaction Quality				X
Contextual Quality				X

* Renamed as the content quality

C. *Quality Framework for Mobile IS*

These quality models described above omit one important aspect of mobile systems. In the mobile network environment the overall quality is created in multiple steps and phases. Mobile IS consists of three components: a mobile device, a mobile network and the information service itself. The mobile device offers the user interface to the information service that can be accessed through the mobile network. The overall quality is a combination of all three that together form the mobile service supply chain. The challenges of mobile IS quality are increased by the fact that typically these components are offered by three different companies, i.e. mobile device manufacturers, network operators and information service providers.

The successful adoption of information technology is largely based upon understanding the linkages among quality, satisfaction, and usage. However, the perceived quality is not limited to the external factors like product/service characteristics but it is also affected by a user's individual quality expectations. When the elements of the quality viewpoints of Fig. 2 and mobile IS supply chain are combined the general quality model (see Fig. 4) for mobile IS is created.

The model gives us a systematic framework for analyzing the quality aspects of mobile IS. In typical QoS studies the main area of interest has either been in the perceived or delivered quality. Users' expectations have not received as much attention even though changes in the quality expectations can be the main reason to abandon a service that was earlier actively used. The empirical part of the paper analyzes the development of the quality expectations in mobile IS during the last four years.

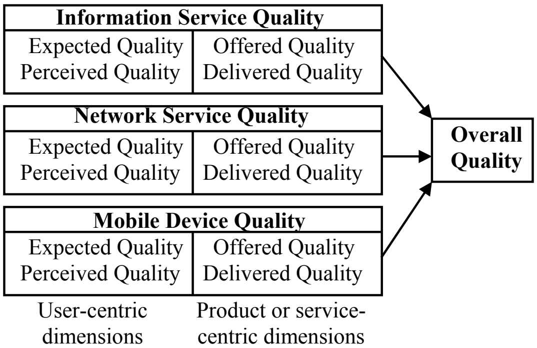

Fig. 4. Determinants of the perceived quality in Mobile IS

IV. Survey on Development in Quality Expectations

A. Method

During the last years we have witnessed a fast technological development in mobile IS and cellular networks. Latest features in end-user devices like build-in cameras, mp3 players and 3G network technologies have created new applications and services. But how have these developments affected users' quality expectations?

To answer this question, data were gathered by means of two surveys (in January 2003 and in September 2007). In the questionnaire (see Appendix) the users were asked to select two most important quality dimensions related to the device, network and information service quality. The network quality dimensions used in the questionnaire were derived from the UMTS bearer service attributes [28]. The service quality dimensions have their roots in TAM studies and the device quality dimensions are derived from the usability studies on mobile devices and especially from [29].

The participants of the test were IT students of a Finnish university. Some statistics about the participants are shown in Table 2.

TABLE 2
Statistics of the Participants

	N	Male	Female	Average age
2003	83	64	19	23.1 years
2007	93	84	9	22.2 years

B. Results

The results were first analyzed by taking into account only the most important quality dimension in each category. Table 3 shows how many percent of the participants considered the feature the most important quality dimensions in the category (column 1st). Columns 1st+2nd show how many participants listed the feature either as the most or the second important one.

Users value usability on mobile devices highly. This can be seen from high ratings of ease of use in both studies. The problems of small display sizes and cumbersome input methods have not been solved in this four year period and therefore these features get similar ratings in both studies.

TABLE 3
The Most Important and Two Most Important Quality Dimensions

Category	Quality Dimension	2003 1st	2003 1st+2nd	2007 1st	2007 1st+2nd
Device	ease of use	60.2 %	77.1 %	48.4 %	63.4 %
	display properties	20.5 %	56.6 %	21.5 %	54.8 %
	input method	10.8 %	45.8 %	17.5 %	47.3 %
	type of client SW	8.4 %	20.5 %	12.9 %	34.4 %
Network	Network speed	75.9 %	89.2 %	63.4 %	84.9 %
	error rates	6.0 %	34.9 %	25.8 %	63.4 %
	delay	12.0 %	53.0 %	6.5 %	32.3 %
	priority	6.0 %	22.9 %	4.3 %	20.4 %
Service	ease of use	45.8 %	71.1 %	26.9 %	63.4 %
	usefulness	32.5 %	66.3 %	50.5 %	74.2 %
	playfulness	3.6 %	18.1 %	2.2 %	9.7 %
	security	18.1 %	44.6 %	20.4 %	52.7 %

The results indicate that network speed is the dominant network quality parameter both in 2003 and 2007. During that four year period the importance of error rates has increased and the importance of delay has decreased. The results in Table 3 also suggest that, at the service level, the original TAM model constructs ease of use and usefulness are the most important factors. On the other hand playfulness gets really low ratings both in 2003 and 2007. The security issues seem to gain more interest among the participants now than four years earlier.

C. Discussion

It must be remembered that the results of the study are more qualitative than quantitative in nature. Although all participants were IT students the determinants of quality are not easy to analyze and describe. For this reason one should not make too strong conclusions based on the results shown in Table 3. They do, however, reveal some interesting trends and signals of development. First, the results indicate that users mainly value the same quality dimensions in mobile networks today as they did four years ago. The relative importance of ease of use has become lower both on mobile devices and services. There is no simple explanation why but one reason could be that the user centered design methods and increased interest in usability issues have had a positive effect on ease of use. Mobile devices still have their limitations when it comes to the input methods and display sizes. These features still need further research and development before the expectations of the users can be fully met.

Second, the results of the study strongly support the constructs of the original TAM model. The perceived ease of use and the perceived usefulness are the dominant variables in the service quality expectations. Extensions of the model recommended by some scholars did not get similar support. The perceived playfulness in particular was not considered an important feature in either of the studies.

Third, users are becoming more aware of the security issues related to mobile IS. The threats typically associated with stationary systems such as viruses and privacy violations are becoming more important also in mobile contexts.

V. Conclusions

Technological development not only changes the systems and services but it also affects on users' expectations. In this paper the relationship between fast development in mobile devices, networks, and services and users' quality dimension preferences was analyzed in a four year period. The findings of the study suggest that expectations change but not at the same speed as technology.

The results also indicate that the original determinants of intention used in TAM are valid also in the mobile context. Similarly the original TAM model was also supported in the literature reviewed in Section II as well as in King and He [10]. They found in their meta-analysis of TAM that despite of its recent extensions (e.g. the TAM2 [30] and revisions (e.g. the UTAUT [5]) primary the classical model is of high reliability and explanatory power and obtains high levels of robustness. The original TAM model can thus be still considered strong and it makes a suitable starting point for integrating technology acceptance, quality and satisfaction into the common framework. The integrated model was beyond the scope of this paper, and further studies are needed before this kind of model can be constructed.

This study has some limitations. First, although young people are, according to several studies, early adopters of mobile services, the results of this study cannot be generalized to cover other population groups. Second, the relatively simple questionnaire can be criticized. A simple format was chosen to avoid users' problems of distinguishing complex quality dimensions (see e.g. [31]). The simple structure and limited number of alternatives were designed to help the participants' task and to consequently give more reliable results.

This study revealed some trends in the development of the quality expectations during the last four years. Without any doubt the relationship between the quality expectations and acceptance will have to be studied in more detail. There is an urgent need for strong quality expectation metrics and measurement methods.

Appendix – Question used in the Surveys

1. *Personal information*
   - Age
   - Sex
   - Nationality
   - Do you have a build-in camera in your mobile phone?
   - Do you have a 3G mobile phone?
2. *Network quality*

Please, select from the list below the TWO most important network parameters affecting to the QUALITY in mobile Internet. Give number 1 to the most important parameter and number 2 for the second.
   - Network speed (bit rate)
   - Error rate
   - Transmission delay
   - Traffic handling priority

3. *Service quality*

Please, select from the list below the TWO most important service parameters affecting to the QUALITY in mobile Internet. Give number 1 to the most important parameter and number 2 for the second.
   - The perceived ease of use of the service
   - The perceived usefulness of the service
   - The perceived playfulness of the service
   - The perceived security of the service

4. *Device quality*

Please, select from the list below the TWO most important device parameters affecting to the QUALITY in mobile Internet. Give number 1 to the most important parameter and number 2 for the second.
   - The perceived ease of use of the mobile device
   - Display properties (like resolution and screen size) of the mobile device
   - Input method (like keyboard or touch screen) of the mobile device
   - Type of the client software used (eg. WWW or WAP browser)

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A New Heuristic Algorithm for the 3D Bin Packing Problem

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Abstract- **3D bin packing is a classical NP-hard (Nondeterministic Polynomial-time hard) problem where a set *N* of 3D boxes is to be packed in a minimum number of containers (bins). 3D bin packing is used in many industrial applications; hence computer scientists are challenged in designing practical and efficient approaches for the problem. This paper presents a new heuristic algorithm called Peak Filling Slice Push (PFSP) for 3D bin packing. The algorithm recursively divides the container into smaller slices and then fills each slice with boxes before pushing them to minimize the wasted space. Experimental results showed significant performance improvements over other current approaches.**

I. INTRODUCTION

Many industrial applications use packing. Examples include scheduling television programs, stacking cargo in a semi-truck, to loading airplanes and placing chips on circuit boards [1]. 3D bin packing is one type of packing problems where we are given a set N of 3D boxes and an unlimited number of containers (bins). The problem is to pack all the boxes into the minimum number of containers. 3D bin packing is a classical NP-hard (Nondeterministic Polynomial-time hard) problem; therefore exact solution cannot be achieved in polynomial time [1, 2]. Thus designing practical and efficient approaches to the problem is a challenge. The performance of 3D bin packing algorithm is largely affected by the strategy of packing boxes and the techniques used in minimizing wasted space. This paper presents a new heuristic algorithm for 3D bin packing. Experiments were conducted to test the performance of the algorithm and are compared with current tools like Robot packing [3].

II. LITERATURE REVIEW

Several methods have been used to solve 3D bin packing. Depending on the problem requirements, the techniques may attempt to minimize wasted space, minimize number of containers, maximize profit or stabilize the balance of containers. Being a combinatorial problem, 3D bin packing is usually solved using either optimization or heuristic algorithms. Optimization algorithms try to deliver an optimal solution [4], while heuristic algorithms deliver a good (acceptable) solution in a relatively acceptable time (that is linear time with respect to the input size) [4]. Wang presented an approach to two-dimensional rectangular packing by successively "gluing" together pairs of rectangles to produce a set of feasible sub-solutions [5].

For the non-rectangular packing, the geometric complexity of placing the pieces directly onto the stock sheet is generally prohibitive. Adamowicz and Albano and Israni and Sanders proposed an approach to first nest the pieces into regular modules [6, 7]. The wall-building approaches (George and Robinson, 1980, Bischoff and Marriott, 1990) are the common methods to deal with 3D cuboids packing problems [6]. Sections of a container across the full width and height are packed first. Identical items are grouped together to develop layers. An ordering of boxes based on decreasing volume, introduced by Gehring et al. is also used to develop layers [8].

Z. Dai and J. Cha [9] proposed a heuristic algorithm for the generation of 3D non-cuboids packing. An octree representation was used to approximate the geometry of the components. The packing algorithm is based on the idea of matching the octree nodes to identify the proper order and orientation of the components. The objects are packed into the container sequentially, depending on the number of items involved.

III. PROPOSED ALGORITHM

In 3D packing problem, we are given a set N of rectangular boxes and an unlimited number of containers. The task is to pack the N boxes into the minimum number of containers. Each box b_i has width w_i, length l_i, and height h_i. The containers are of the same size and have width W, length L, and height H. We present a heuristic algorithm called Peak Filling Slice Push (PFSP). This new approach benefits from the slicing mechanism introduced by Sweep [3]. The proposed algorithm has two main steps. First, the container is divided into slices having same height and width as the container. Fig. 1 shows a graphical explanation of the slicing mechanism. Each slice is filled using Peak Filling technique. Second, the filled slices are pushed in order to compress the boxes and minimize the wasted space. For efficiency purposes, the boxes are sorted in decreasing order by height, width, length. Slicing the container has a considerable influence on reducing packing time. It also makes the algorithm easily parallelizable.

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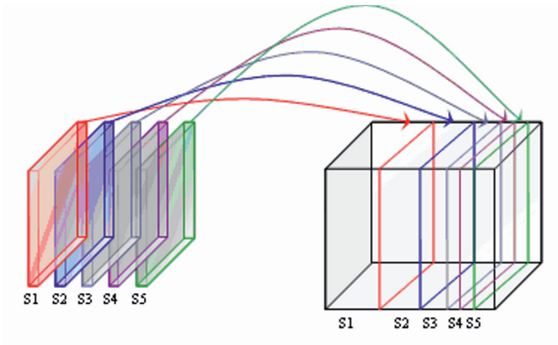

Fig. 1. Slicing mechanism.

Peak filling method is a recursive divide-and-conquer procedure. Each time a box is placed in a slice, a new sub-slice is created on the top of the placed box as shown in Fig. 2. When no more sub-slices can be created, the algorithm starts backwards and fills the left sub-slices until no usable space is left. The algorithm works by placing bigger boxes at the bottom of the container. In upper direction filling, the placed boxes are smaller or of same size as the initial box placed at the bottom. In backtracking, the chosen boxes are of smallest size.
For each slice, a stack is used to keep track of the remaining unfilled sub-slices. A reduction from 3D to 2D is done as illustrated in Fig. 3. Whenever a sub-slice is created, the dimensions of the bottom surface as well as the coordinates of the upper left corner are pushed on the stack. This is essential for the backtracking step. When the top of the slice is reached, backtracking step pops from the stack and fills the remaining unfilled space using smaller boxes. The backtracking step stops when the stack is empty meaning that there is no usable space left in the slice.

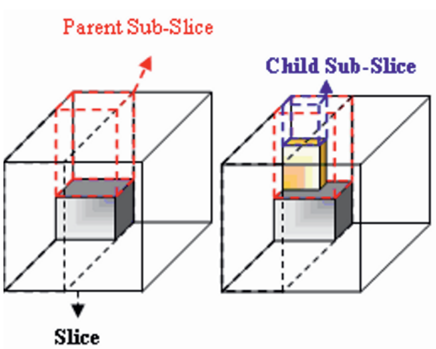

Fig. 2. Creating new sub-slice on the surface of the placed box.

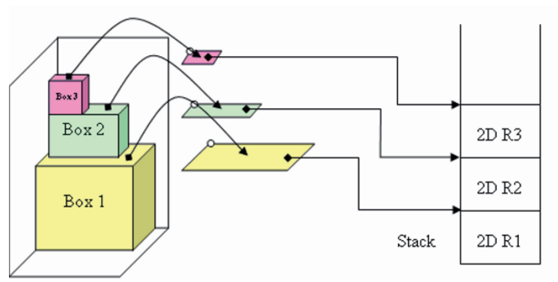

Fig. 3. Reduction from 3D to 2D.

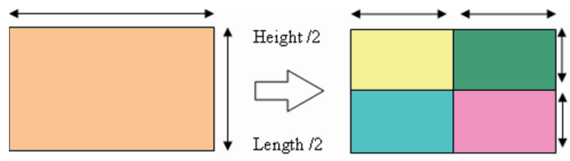

Fig. 4. Division of certain sub-slice.

Filling each slice (or sub-slice) is done by dividing the slice into four rectangular slices and using different combinations of the resulting sub-slices to determine the best arrangements of boxes inside the given slice. We have chosen to divide the slices with the ratio r = 0.5. Thus applying the ratio, we divide the slice into four equal sub-slices as shown in Fig. 4, even though different divisions can be used for the slice.

The algorithm then considers different combinations of resulting rectangles and tries to fill them with the available boxes. The combinations should result in rectangular shapes only thus some invalid combinations will be removed. The combination that gives the minimum wasted area is selected. Fig. 5 shows the possible combinations for the sub-rectangles. The aim of the algorithm is to reduce internal fragments thus minimizing space. Whenever a new slice is filled, a "Push" method is called. This method pushes the boxes of the new slice into the old slice without overlapping. Both slices are then combined into one larger slice. Fig. 6 shows the push mechanism.

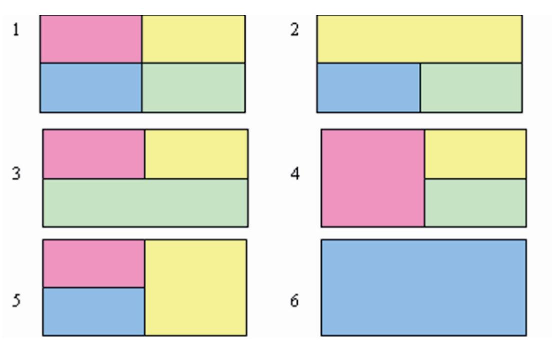

Fig. 5. Possible combinations.

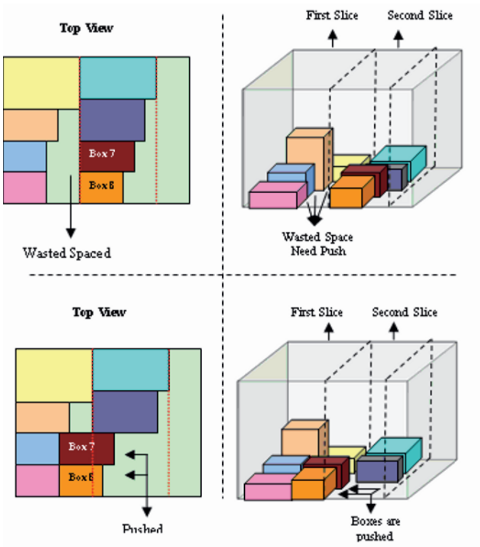

Fig. 6. Push mechanism.

The pseudo code of the algorithm can be summarized as the following:

```
Input:   Boxes Array = (Box 1, Box 2, …, Box n)
         Container = (x, y, z)
Output:  Packed Boxes Array = (Box i, Box j, … , Box m)

While (Container.FULL = False) {
         Stack.Clear()
         Temp Box = GetFirstAvailableBox(Box Array)
         Slice = CreatSlice(Temp Box)
         If (Slice = Null)   Container.FULL = True
         Rectangle = GetRectangle(Slice)
         Stack.Push (Rectangle)
         Packed Boxes = PeakFilling(Box Array, Container,
         Slice, Stack)
}
Packed Boxes = SlicePush( Packed Boxes)
```

The complexity of this algorithm is in O (N^3). The algorithm has a main "while loop" and two nested "for loops" in the peaks filling function. The result is three nested loops. It also has 2 nested loops in the function slice push. After adding them all together it would generate the following linear formula:

$$SUM = N^3+N^2+C$$

The C is a constant that represent the total additional operations. The algorithm must also take into consideration the sorting time. The sum would be:

$$SUM = N*\log(N)+N^3+N^2+C$$

The calculation of the complexity is the following:

$$\begin{gathered} O(SUM) = \\ O(N*\log(N)+N^3+N^2+C)= \\ O(N^3+N^2+C)= \\ O(N^3 +C)= \\ O(N^3) \end{gathered}$$

IV. EXPERIMENTAL RESULTS

The algorithm has been implemented using Visual Basic programming language. It is compiled with Microsoft Visual Studio .NET v2005 Professional Edition (Microsoft Corporation, Redmond, WA, USA) and Microsoft framework 2.0 (Microsoft Corporation, Redmond, WA, USA). The graphical part of the implementation used DirectX 9.0c. The algorithm was tested on Intel Pentium® 4 CPU 3.00 GHz, 32 KB L2, 512 MB of RAM, and PM800 Motherboard. The operating system that has been used was Windows XP Professional SP2 (Microsoft Corporation, Redmond, WA, USA). All the results have been calculated in time. The operating system and other processes time has been calculated within the testing time. The sorting time is not calculated within the experiments. The experiments used a sorted box ready to be packed

The proposed algorithm was tested using four test bunches. Table 1 summarizes the information about the test bunches. The test bunches ran on a Pentium 4 processor having 256 MB of RAM. The dimensions of the boxes are chosen randomly. A box whose dimensions are larger than the container is discarded. The container dimensions can be specified by the user.

TABLE 1
TEST SPECIFICATIONS

Test Bunch	Number of Boxes	Dimension of Container (H, W, L)
1	50	(10, 10, 15)
2	100	(10, 10, 15)
3	150	(10, 10, 15)
4	200	(10, 10, 15)

Table 2 shows the results obtained. On average, the bins are filled up to 85%, which is a satisfying result. The proposed algorithm runs in an acceptable time; less than 0.5 seconds with large input sets like 100 boxes. The results show an improvement over current tools like Robot. The percentage of filled volume from each container is shown in Table 3.

Tables 3 and 4 show best and worst case scenarios with respect to solution times (in seconds) for the general approach and Robot tool. It is obvious that our approach outperforms these two approaches by a large factor. Packing 50 boxes require at least 12.97 seconds in general approach and 11.09 in Robot, while in our approach, it takes about 0.019 seconds. Also both approaches do not pack more than 50 boxes, where as PFSP can pack up to 200 box in less than 0.5 sec.

TABLE 2
RESULTS OBTAINED

Test Bunch	Average Wasted Space	Average Time (sec)
1	16.08 %	0.019063
2	15.95 %	0.034375
3	15.14 %	0.037708
4	15.38 %	0.046953

TABLE 3
SOLUTION TIMES IN SECONDS FOR THE GENERAL APPROACH

Number of Boxes	Best Case Scenario (sec)	Worst Case Scenario (sec)
10	0.01	0.07
20	0.01	0.21
30	0.08	57.14
40	0.10	5247.60
50	12.97	55402.25

TABLE 4
SOLUTION TIMES IN SECONDS FOR ROBOT APPROACH

Number of Boxes	Best Case Scenario (sec)	Worst Case Scenario (sec)
10	0.01	0.07
20	0.01	0.21
30	0.07	37.54
40	0.11	50593.91
50	11.09	30719.51

TABLE 5
PERCENTAGE OF USED VOLUME IN CONTAINER

Number of Boxes	Best Case Scenario	Worst Case Scenario	Average
50	96 %	63 %	83.92 %
100	97 %	63 %	84.05 %
150	98 %	62 %	84.86 %
200	98 %	63 %	84.625 %

In the best case scenario, the container is filled up to 96% which is a very satisfying result according to industrial demands. Worst case scenario fills more than half of the container while on average the container is filled up to 84% which is an acceptable trade off between time and performance.

V. DISCUSSION AND CONCLUSIONS

Peak Filling Slice Push (PFSP), a new heuristic algorithm, is presented in this paper to solve 3D bin packing problem that is faced in many industrial applications. The implementation of PFSP using the Visual Basic programming language shows improvements in both performance and average time over other methods currently in use by the industry. PFSP presents a potential for saving time and money for numerous industries. As any new algorithm, PFSP has some limitations. PFSP can not load balanced containers like ships. It does not have the weight factor included. It also limited to use in a large cargo, the opportunity of using it is narrowed. On the other hand, it is practical in filling small volumes with different boxes' size of a cargo. Future work will include improvements to the algorithm like the ability to rotate the boxes for further efficiency, adding different ratios to divide a sub-slice and balanced packing. These improvements are expected to increase the performance and efficiency of the algorithm.

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Dynamics of Large Rings of Coupled Van der Pol Oscillators

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Abstract- **The dynamic behavior and synchronization of a large ring of mutually coupled Van der Pol oscillators is studied by extending a methodology previouly reported by another authors, in which the original equations are linearized around the unperturbed limit cycle. The linearized equations are trasformed into Hill's equations and from here conditions for stability and synchronization are derived. Numerical simulations corroborate the validity of the stability analysis for a ring of 10000 van der Pol oscillators.**

I. Introduction

Dynamics and synchronization of arrays of coupled Van der Pol (VDP) oscillators have been widely studied and reported by many researchers, both theoretically and experimentally, in the last decades. This is due to the fact that coupled VDP oscillators represent a wide variety of phenomena in many scientific and social disciplines. A great diversity of techniques and methods have been employed to analyze the dynamical behavior of coupled VDP oscillators, for example: nonlinear mode analysis [1], perturbation [2], averaging [3], topological [4], linearization [5], experimental [6], quasilinear approximation [7], numerical [8,9], and so on. Recently, the analysis of the synchronization of a ring of four identical mutually coupled VDP oscillators by means of the Floquet theory was reported in [10]. In [10], the four nonlinear coupled equations of the VDP ring were linearized around the unperturbed limit cycle and then, by introducing diagonal variables, the linearized equations were changed into a second order homogeneous differential equations. Thereafter, by a variable transformation, these equations were converted into a group of canonical Hill's equations. Finally, the stability analysis and the synchronization of the VDP ring was carried out by applying the Floquet theory. In this work the results reported in [10] are extended to the analysis of the stability and synchronization of a large ring of N coupled identical VDP oscillators. Numerical simulations show that for a ring up to 10000 oscillators synchronization is feasible whenever the stability condition is satisfied.

II. Mathematical Modelling and Stability

A ring of N identical coupled VDP oscillators can be represented by the following set of nonlinear second order ordinary differential equations [10]:

$$\ddot{x}_i + k_o\left(x_i^2 - 1\right)\dot{x}_i + x_i = k_c\left(x_{i-1} - 2x_i + x_{i+1}\right) \qquad (1)$$

where x_i is the state of the i^{th} oscillator, $\dot{x}_i$ and $\ddot{x}_i$ are the first and second time derivatives of x_i, respectively. Besides, k_o is the oscillator constant, and k_c is the coupling constant.

In order to analyze the stability of the VDP ring, from now on the methodology described in [10] is employed. Eq. (1) is linearized assuming that for small values of the oscillator constant the limit cycle of a VDP oscillator is described by $x_0(t) = A\cos(\omega t - \varphi)$, where A is the amplitude, ω is the angular frequency and t is time. Then, the following definitions are established: $x(t) = \xi_i(t) + x_0(t)$, $\tau = \omega t - \varphi$. Thereafter, diagonal variables are introduced. In the aforementioned methodology, the most severe constraints on stability and synchronization are imposed on the N^{th} oscillator. The following N^{th} diagonal variable is defined:

$$z_N = \sum_{i=1}^{N} \delta_i \xi_i \qquad (2)$$

where δ_i is equal to 1 if *i* is even and equal to –1 if *i* is odd. N is the number of VDP oscillators. When the diagonal variables are substituted into the linerized equations, a second order homogeneous linear differential equation arises:

$$\ddot{z}_i + a_{1i}(\tau)\dot{z}_i + a_{0i}(\tau)z_i = 0 \qquad (3)$$

where, for the N^{th} diagonal variable

$$a_{1N}(\tau) = k_o\left(A^2\cos^2\tau - 1\right) \qquad (4)$$

$$a_{0N}(\tau) = 1 + 4k_c - k_o A^2 \omega \sin 2\tau \qquad (5)$$

In accordance with [11], an equation such as Eq. (3) can be transformed into another one with canonical form of the Hill's equation:

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$$\ddot{y}_i + p(\tau) y_i = 0 \tag{6}$$

by applying the following tranformation:

$$z_i = y_i \exp\left(-\frac{1}{2} \int_0^{\tau} a_{1i}(s) ds \right) \tag{7}$$

where

$$p_i(\tau) = a_{0i}(t) - a_{1i}^2(\tau)/4 - \dot{a}_{1i}(t)/2 \tag{8}$$

For the N[th] diagonal variable the following expression is obtained:

$$p_N = 1 + 4k_c - k_o^2 \left(A^2 \cos^2 \tau - 1 \right)^2 /4 - k_o A^2 \omega \sin 2\tau /2 \tag{9}$$

In the Hill´s equation, the stability analysis can be done by means of the periodic or the non-periodic terms. In this context Eq. (6) is written in this way [12]:

$$\ddot{y} + \left[\lambda + Q(\tau) \right] y_i = 0 \tag{10}$$

where the non-periodic term is given by

$$\lambda = 1 + 4k_c \tag{11}$$

and the periodic term is given by

$$Q(\tau) = -k_o^2 \left(A^2 \cos^2 \tau - 1 \right)^2 /4 - k_o A^2 \omega \sin 2\tau /2 \tag{12}$$

Assuming that the periodic term Q(τ) vanishes due to the small value of the oscillator constant, the stability of the coupled ring of VDP oscillators will depend on the sign and value of the non-periodic term λ. If $\lambda > 0$ the ring is oscillatory and becomes synchronized. If $\lambda = 0$ the ring is not oscillatory, and if $\lambda < 0$ the ring is unstable. Then, in order to keep the stable oscillations and synchronization of the VDP ring, and considering only the non-periodic term, the following condition must be satisfied: $k_c \geq -1/4$. In [10] three domains of stability are discussed and numerically tested. In one of them $k_c \in [-0.25, -0.0011] \cup [0.004, +\infty)$ and in this case all four oscillators are fully synchronized or phase locked, i.e. $x_1 = x_2 = x_3 = x_4$. This is the case considered in this work.

III. Results and Discussion

Numerical simulations were carried out in order to test the validity of the above ring stability condition for small values of the oscillator constant and for a large ring of coupled VDP oscillators. Two rings were considered: one consisting of four coupled VDP oscillators, and another one with 10000 coupled VDP oscillators. For each ring, different values of the oscillator and coupling constants were employed in order that only some of the rings satisfy the stability condition (SSC). Table I shows the cases considered with their corresponding number of VDP oscillators, and the values of the oscillator and coupling constants employed in the numerical simulations. In Cases 1-4 the coupling constant is greater than –0.25 and the stability condition $k_c \in [-0.25, -0.0011] \cup [0.004, +\infty)$ is satisfied. In Cases 5-8 the coupling constant is less than –0.25, therefore the above stability condition is not satisfied. In Cases 9-10 the coupling constant satisfies the stability condition, however the oscillator constant is very large.

TABLE I
CASES CONSIDERED.

Case	k_o	k_c	N	SSC?
1	0.1	-0.1	4	Yes
2	0.32	5.0	4	Yes
3	0.1	-0.1	10000	Yes
4	0.32	5.0	10000	Yes
5	0.1	-0.5	4	No
6	0.32	-10.0	4	No
7	0.1	-0.5	10000	No
8	0.32	-10.0	10000	No
9	5.0	-0.1	4	Yes
10	5.0	-0.1	10000	Yes

Integration of the set of N nonlinear equations given by Eq. (1) was carried out by means of the 4[th] order Runge-Kutta algorithm. Double precision and a time step of 0.0001 were employed in the computer simulations in order to prevent parasite undesirable dynamics induced by the numerical solution. To perturb the ring, the initial condition of the first VDP oscillator was fixed at $x_1(0)=0.2$, $x_1'(0)= 0.2$. For the remaining oscillators, the initial conditions were as follows: $x_i(0) = 1.0$, $x_i' = 1.0$.

In order to monitor the stability and synchronization of the rings, the dynamic behavior of some oscillators was tracked. In those figures in which rings of 4 oscillators are considered, namely Figs. 1-2 and 5-6, it is shown the dynamic behavior of the first, the second and the fourth oscillator. In those figures in which rings of 10000 oscillators are considered, namely Figs. 3-4 and 7-8, the dynamic behavior of the first, the 5000[th] and the 10000[th] oscillator is shown.

As it was mentioned before, Cases 1-4 represent situations in which the stability condition is satisfied, namely $k_c \in [-0.25, -0.0011] \cup [0.004, +\infty)$ and $\lambda > 0$, whereas Cases 5-8 correspond to situations in which the stability condition does not hold. Particularly, Cases 1 and 2 correspond to rings of four VDP oscillators and to values of the oscillator and coupling constants employed in [10] and [6], respectively. In both cases the respective authors report stability and full synchronization, as was obtained in this work and shown in Figs. 1 and 2. On the other hand, Figs. 3 and 4 show the results for rings of 10000 VDP oscillators employing values of the coupling constant that match the stability condition. Stability and full synchronization are observed for these large rings.

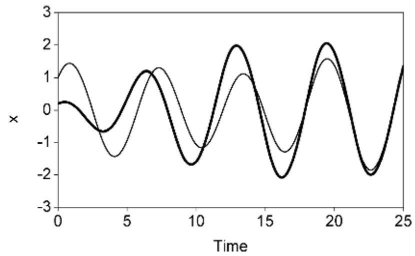

Figure 1. Time series of Case 1. N = 4. k_o = 0.1, k_c = -0.1.
1^{st} , -2^{nd} and 4^{th} oscillators.

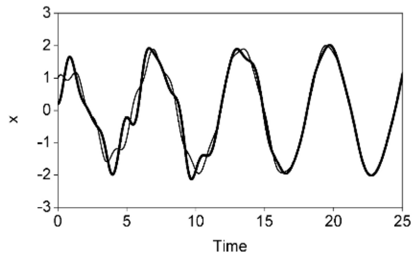

Figure 2. Time series of Case 2. N = 4. k_o = 0.32, k_c = 5.0.
1^{st} , -2^{nd} and 4^{th} oscillators.

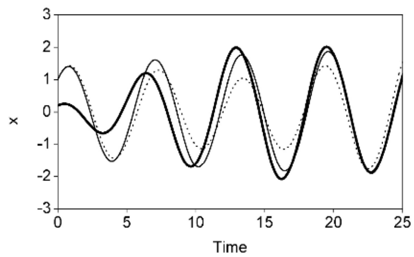

Figure 3. Time series of Case 3. N = 10,000. k_o = 0.1, k_c = -0.1.
1^{st} , -5000^{th} and --10000^{th} oscillators.

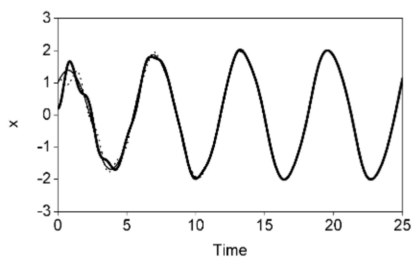

Figure 4. Time series of Case 4. N = 10000. k_o = 0.32, k_c = 5.0.
1^{st} , -5000^{th} and --10000^{th} oscillators.

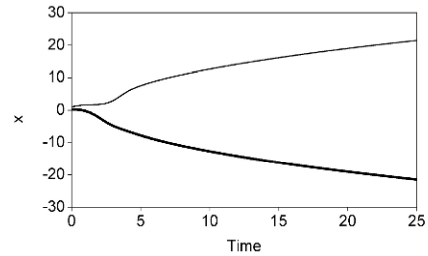

Figure 5. Time series of Case 5. N = 4. k_o = 0.1, k_c = -0.5.
1^{st} , -2^{nd} and 4^{th} oscillators.

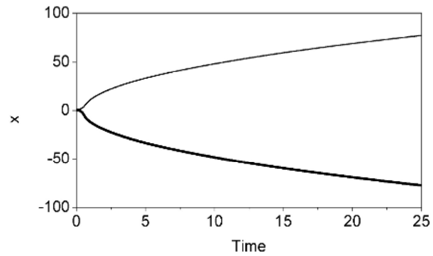

Figure 6. Time series of Case 6. N = 4. k_o = 0.32, k_c = -10.0.
1^{st} , -2^{nd} and 4^{th} oscillators.

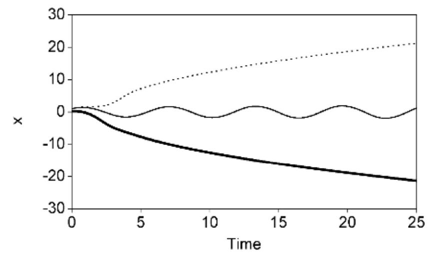

Figure 7. Time series of Case 7. N = 10000. k_o = 0.1, k_c = -0.5.
1^{st} , -5000^{th} and --10000^{th} oscillators.

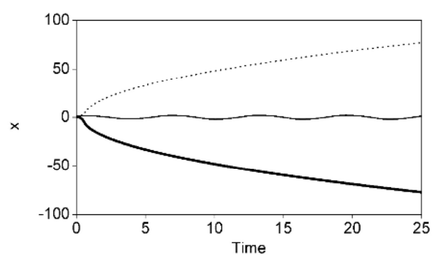

Figure 8. Time series of Case 8. N = 10000. k_o = 0.32, k_c = -10.0.
1^{st} , -5000^{th} and --10000^{th} oscillator.

Cases 5-8 correspond to coupling constants that do not match the stability condition. In those cases $\lambda < 0$, and the analysis of Hill´s equation predicts that the rings will be unstable with the state variables going to infinity. These can be clearly seen in Figs. 5-8 for rings with 4 and 10000 oscillators.

An interesting dynamic behavior of the VDP rings occurs when the stability condition is satisfied but the oscillator constant is very large, as in Cases 9-10. If $k_o >> 1$, the periodic terms given by eq. (12) in the Hill´s equation become significant and are no longer vanished. In theses cases the stability is preserved, however the synchronization is disrupted, as can be seen in the phase portraits of Figure 9 for a ring of 4 oscillators. Synchronization between the first oscillator and oscillators 2-4 is disrupted but it is preserved between 2 and 4. In Figure 10 are shown the phase portraits for the Case 10. As in Case 9, in Case 10 stability condition is satisfied but synchronization between the first and the 5000th oscillators is disrupted. However, synchronization is mantained between the first and the 10000th oscillators.

IV. Conclusions

The linearization methodology employed in this work is adequate to analyze the dynamic behavior of a large ring of coupled Van der Pol oscillators, at least for small values of the oscillators constant. The derived stability condition predicts properly the stability and synchronization of the ring. When the oscillators contant is greater than one, the periodic terms of the Hill's equation become significant and this methodlogy loses its predictive power.

Acknowledgements

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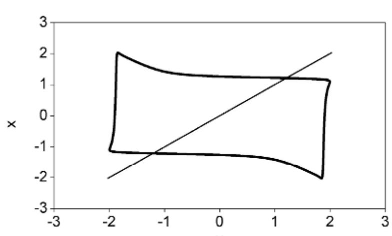

Figure 9. Phase portraits of Case 9. N = 4. $k_o = 5.0$, $k_c = -0.1$. 1st and 4th oscillators, — 2nd and 4th oscillators.

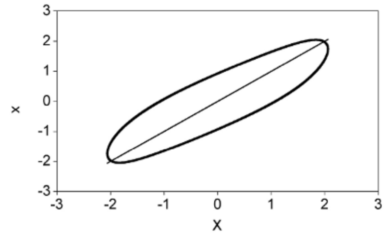

Figure 10. Phase portraits of Case 10. N = 10000. $k_o = 5.0$, $k_c = -0.1$. 1st and 5000th oscillators, — 1st and 10000th oscillators.

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The Mobile Message Receiver System

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***Abstract*-Wireless communications is a dynamic and in an accentuated development field. More and more applications rely on wireless networks. Ad hoc networks are a special type of wireless networks in which the topology of the network is continuously changing in accordance to the current communication needs. Mobile Ad Hoc Networks (MANETs) are part of the ad hoc networks. This paper introduces a new type of MANET and describes an application for sending messages to mobile destinations from a delimited space (building or group of buildings) without communication costs. The application is called Mobile Message Receiver System (MMRS). It consists in a server with a data base, from which users with different attributes send messages to all or part of mobile nodes, implemented with embedded systems, through partially mobile nodes, called base nodes, implemented with embedded systems directly connected to Internet. Each mobile node is in the coverage area of a base node. The system is scalable because the base nodes and the mobile nodes can be added or removed without affecting the rest of the system and modular because the mobile nodes are logged to base nodes. The general structure and the software of the MMRS are described.**

I. INTRODUCTION

Wireless communications is a dynamic and in an accentuated development field. More and more applications rely on wireless networks. In classical wireless networks data is routed through special nodes, called switches, routers, hubs. Such a network is structured some of the nodes having predetermined roles.

In contrast, ad hoc networks are a special type of wireless networks in which all the nodes have the possibility to forward the data thus the topology of the network being continuously changing in accordance to the current communication needs. The ad hoc networks are divided in Mobile Ad Hoc Networks (MANET), Wireless Sensor Networks and Wireless Mesh Networks, [1], [2].

Several of the MANET features are:

- any node or group of nodes can enter or leave the network, as a consequence the network topology is continuously changing;
- any node can communicate with any other node, possible restrictions being given only by the connectivity and the security;
- the nodes have the same features; yet, they can fulfil different functions at the same moment;
- any node can transmit packets to any other nodes;
- the communication between nodes is not transitive.

Many MANET based applications are developing such as: content distribution, information dissemination, data fusion, file sharing, distributed games, voting systems and so on. And more important, the MANETs are strongly sustaining the development of the ubiquitous and pervasive computing which is considered the new wave in the computing area.

This paper introduces a new type of MANET and describes an application for sending messages to mobile destinations from a delimited space (building or group of buildings). The application is called Mobile Message Receiver System (MMRS) and consists in a server with a MySQL data base, from which users with different attributes send messages to all or part of mobile nodes, implemented with embedded systems, through partially mobile nodes, called base nodes, implemented with embedded systems directly connected to Internet. Each mobile node is in the coverage area of a base node. The topology of the network is continuously changing because any node can leave the coverage area of a base node and enter the coverage area of another base node. The system is scalable because the base nodes and the corresponding mobile nodes can be added or removed without affecting the rest of the system and modular because the mobile nodes are logged to base stations.

The next section presents similar work, the third section describes the general structure of the MMRS, the fourth section details the software and the last section outlines the conclusions.

II. RELATED WORK

There is a consistent literature about MANETs, their structure and applications.

Reference [3] describes an infrastructured ad-hoc network, in which part of the nodes has fixed positions, with fixed power supplies and the other nodes are totally mobile. The nodes keep their facilities and the equal roles. This separation was done in order to ensure reliable power supplies for part of the nodes.

Reference [4] introduces a hierarchical structure for ad hoc networks that classifies nodes in two categories: Backbone Capable Nodes (BCNs) and Regular Nodes (RNs). BCNs have more features than RNs: their hardware is more complex, they can work at multiple power levels and employ multiple radio modules. Each BCN covers RNs. A protocol is developed for maintaining a Mobile Backbone due to the mobility of the nodes, to the topological changes and to the traffic flow variations.

In reference [5] a new protocol for mobile ad hoc communications, called Virtual Base Station (VBS) infrastructure creation protocol is introduced. In this protocol a mobile station is choused from a group to act as a temporary base station within its zone. A VBS is a mobile station and tracks other mobile stations in the ad hoc network. A routing protocol based on VBS is also proposed.

In reference [6] a new architecture for ad hoc networks was proposed. It is cellular based and focuses real time multimedia communications. The cellular network operates

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 350–354.

in two modes: infrastructure for intracell and ad hoc for intercell communications. Its performances were compared with conventional peer to peer ad hoc networks.

Reference [7] approaches the problem of the traffic at the center of an ad hoc network due to its scalability. It is obtained that the expected packet traffic at the center of a network is linearly related with the network size.

In reference [8] MANETs are introduced in a classical entertainment field, namely the games. The new mobile devices have sufficient resources to play multi-player games in mobile and wireless networks. As the classic centralized game server design is unsuited for mobile ad hoc networks, the alternative is the distributed game server implemented by a group of mobile nodes. An algorithm is proposed for server selection.

This paper presents the Mobile Message Receiver System based on a modified MANET. The system is structured on three levels: a server connected to a data base (level 0), partially mobile nodes, called base nodes (level 1) and totally mobile nodes (level 2). Different users can send messages to all or part of the mobile nodes through the base stations which are directly connected to Internet. The application is useful for announcing peoples from a delimited area (building, group of buildings) such as: hospitals, companies (for announcing the employees), universities, schools, and entertainment parks without communication costs.

III. General Structure of the MMRS

The subject of the present paper is a network of embedded systems (ESs) for wireless message communications, in a delimited space. The users can transmit those messages through a web interface offered by the system and can receive and visualize the messages with personal mobile embedded systems. The system is made by the following elements, as it is shown in Fig. 1: the Message Management Centre made by the HTTP server and the TCP/IP client, the Data Base Management Centre, base nodes implemented with fixed ESs and mobile nodes, implemented with mobile ESs.

The Message Management Centre is made by a HTTP server which runs a PHP application giving the user interface through which the messages are introduced by the user who wants to send a message. The message is memorized in a data base together with all the information necessary for sending it (the destination address, the id message, the hour and the date etc.). Near the HTTP server which collects the messages and memorizes them in a data base, there is also a service which takes the messages from the data base and transmits them to the base node. This service works as a TCP/IP client. The features of the system software component, at the Message Management Centre, are:

- ensures the transmission of the messages from one or more users to maximum 65536 base stations;
- only the users having an account can transmit messages;
- the accounts will be assigned by a general administrator of the system; he will have also the task to assign the mobile devices to the mobile destinations (persons);
- the message will contain an id of the user who sent it;
- an hierarchy of users will be created in the sense that the number of the mobile destinations accessible by a user will depend by its access rights.

The base node is implemented with an Embedded Internet type fixed ES, i.e. a microcontroller based system directly connected to Internet, having a wireless interface for communicating with maximum 256 mobile devices. The features of the system software component, at the base station level, are:

- communicates with the Message Management Centre through the Internet;
- receives the messages from the centre and transmits them to one or more of the mobile destinations being in its application area.

The mobile node is implemented by a mobile microcontroller based ES, having a LCD display and a wireless interface for communicating with a base station. The power will be supplied by batteries. The features of the system software component, at the mobile node level, are:

- ensures the identification and the communication with a base node;
- receives and visualizes the messages from a user, arrived through a base station which controls it with the wireless interface;
- automatically transmits to the Message Management Centre, through the base node, a message receiving acknowledge.

The hardware of the base node consists in an Ethernut, [9], board. It is a low dimension board (80 x 100 mm) based on the Atmel Atmega 128 RISC microcontroller and on the Ethernet Realtek 8019AS controller. For the wireless communication with the mobile devices the ZigBee solution was implemented. Among the different ZigBee modules, the RF Xbee PRO module was chosen. It is connected to the Ethernut board through one of its UART serial interfaces and communicates with the similar RF Xbee PRO modules connected on the mobile devices. Its main features are:

- the coverage area in an internal environment (e.g. buildings) is up to 100 m and in an open environment (without walls) up to 1500 m;
- the transmission power is 100 mW (20 dBm) and the reception sensitivity is 100 dBm;
- the RF data transfer rate is 250 000 bps;
- the current consumed at transmission is 214 mA and the current consumed at reception is 55 mA;
- transmits acknowledge signals;
- each direct sequence of channels has more than 65 000 network addresses available;
- it is compliant with other IEEE 802.15.4 devices;
- has AT and API command modes for configuring the parameters of the modules;
- needs a 2.8 – 3.4 V power supply.

The hardware of the mobile node is based on the low power and low dimensions LPC 2103 microcontroller. It is recommended for implementing gateways and protocol converters. A LCD display is connected to eight I/ O lines of the microcontroller for data transfer and more three lines for commands. The information is displayed on two 16 character lines. The LCD displays the messages and is a support for future developments, such as a menu for finding the already received messages. The wireless communication is achieved with a RF Xbee PRO module connected to one of the UART interfaces of the microcontroller. The power is supplied by a 4 V Nokia battery from which a 1.8 V level is obtained for the microcontroller, and 3.3 V and 5 V levels are obtained for the rest of the board.

For locating in reasonable time duration the mobile nodes which are destinations for the messages and the base nodes which cover the searched mobile nodes, the following algorithm was implemented: at well established time moments, generated by a timer, a message transmission session is launched. In such a session the messages are sent, from the data base, in the order, in which they were memorized, to the first base station from a list. If the base station succeeded in sending the message to the mobile destination, it will send back an ACK message. In this case the Message Management Centre will set the message as "sent" and continues with the next message. If the base station responds with an ERR message, the Message Management Centre understands that the base station did found the mobile destination in its action area but this one can not receive the message, sets the message as "not sent" and passes to the next one. If the base station answers with a NACK message the Message Management Centre understands that the mobile destination is not in the action area of the base station and repeats the described process with the next base station from the list.

IV. Software of the MMRS

Fig. 2 presents the software diagram of the MMRS.

A. *Web Module*

The Web part of the system consists in nine *.php programs and a MySql server. All the *.php programs collaborate for creating the data base called pajer. It is located on a MySql server and different accesses to it are done through queries.

The init.php program creates the *pajer* data base but first it clears the already existent data base. The data base contains two tables: persons and messages. The table with persons contains the following fields: first name, last name, account, password, address and type and the table with messages has the following fields: sender, receiver, text, sending date, reception date, type and flag. This program also creates two types of users: the administrator, with special rights and ordinary users.

The login.php program creates the recording form of a user who wants enter the system and sent a message. This user is already recorded in the data base and has a user name and a password .

The user.php and message.php programs permit to a user to create and send a message to one or part of mobile destinations recorded in a list.

The sqlfunct.php program contains several functions used by the other programs, achieving the following operations: the interruption of the connection to the server, the generation of a query, the login and a validity test for the user name and the password.

The administrator.php program allows the administrator to record new users in the *pajer* data base.

B. *Client Service*

The Client Service achieves the connection to the MySql server, takes the messages from the data base and sends them to the base node. The communication media is the Internet.

The Client Service consists in several programs and libraries. There are MySql++ programs for communicating with the *pajer* data base. The program SocketClient.cpp creates a socket through which data are sent/received to/from the base node. Other programs create functions such as: start connection to the server, stop connection, sending and receiving data, message formatting.

The PagerService.cpp program implements the thread for obtaining the message from the data base, for creating the connection to the base node by calling the function for creating the socket and for sending the message to the base node. Next the ACK message sent back by the base node is interpreted and the message for stopping the connection with the base node is created.

The function for installing the service is as follows:

```
void InstallAsService()
{
    SC_HANDLE schService;
    SC_HANDLE schSCManager;
    TCHAR szPath[512];
    if ( GetModuleFileName( NULL, szPath, 512 ) == 0 )
    {
        printf("Unable to install %s\n",SERVICENAME);
        return;
    }
        schSCManager = OpenSCManager(
                NULL,
                NULL,
                SC_MANAGER_ALL_ACCESS
                );
        if ( schSCManager)
      }
          schSrervice = CreateService(
                schSCManager,
                TEXT(SEVICENAME),
                TEXT(SERVICEDISPLAYNAME),
                SERVICE_ALL_ACCESS,
                SERVICE_WIN32_OWN_PROCESS            |
SERVICE_INTERACTIVE_PROCESS,
                SERVICE_AUTO_START,
                SERVICE_ERROR_NORMAL,
                szPath,
                NULL,
                NULL,
                TEXT(DEPENDENCIES),
                NULL,
                  NULL);
        if ( schService )
        {
        _tprintf("%s installed.\n", SERVICENAME);
        CloseServiceHandle(schService);
```

```
            }
            else
            {
            _tprintf("CreateService failed\n");
            CloseServiceHandle(schSCManager);
        }
        else
        _tprintf("OpenSCManager failed\n");
}
```

C. *Base Node's Software*

Due to the complexity of the base node's software must fulfill the NUT/OS real time operating system was implemented. Its main features are: open source, modular design, easy portable, cooperative multithreading, queries for events, management of the dynamic memory, I/ O functions. It contains a TCP/IP protocol which offers: open source, automated configuration via DHCP, HTTP API with access to the files system, CGI functions, TFTP, FTP DNS and sockets API TCP and UDP for other protocols.

The applicative program consists in the pajero.c program and has the following tasks: assumes the messages from the Message Management Center through Internet, sends them to the mobile destinations and reports the correct sending of the messages.

Two serial interfaces are controlled by the pajero.c program: for debug purposes, UART0 and for sending the messages to the mobile nodes through the RF Xbee PRO module, UART1. For using the RF module it must be first initialized. The operation is achieved entering its Command Mode. The command is interpreted by the module and if it was correct the OK message is sent back and if not the ERROR message is sent back to the UART0 interface. After this operation the 64 bit unique address of the RF module is read. Next the sending of the message takes place. It starts with the verification of the destination and the response is sent back to the UART0 interface. After that memory space is reserved for the message and the message is composed and sent. An acknowledge is waited from the destination. If it is OK it means that the message was received and displayed at the destination. If it is ER it means the message reached the destination but was not assumed because of various causes (for instance the buffer from the mobile node is full). The Client Service will be notified and it will try to send the message again. If during 5 ms the OK message will not arrive the NACK message will be written at the UART0 interface meaning the mobile destination was not found or it could not receive the message.

D. *Mobile Node's Software*

The software implementation consists in C programs and library type files, with extension *.h, helping them. The C programs are: serial.c, time.c, lcd.c and blinky.c.

Serial.c is made by two programs, one for writing and the other for reading a character to/ from the serial interface. The program time.c achieves the following functions: initializes the timer/ counter 0 of the microcontroller (it will be used at the RTC interrupt), generates 1 ms RTC interrupt and serial interrupts servicing. The last function verifies if a message arrived, reads it in affirmative case and generates an acknowledge.

The lcd.c program is made by several functions referring the lcd display: the lcd initialisation, a several seconds wait operation, the writing of a 4 bit configuration on the data bus, the writing of a byte on the data bus, the differentiation among the data and control bytes sent on the same data bus and the display of a 32 characters string on the lcd. The blinky.c program communicates with the user of the mobile device and with the Xbee PRO module.

V. CONCLUSIONS

A new type for ad hoc networks was proposed in this paper. It consists in a server from which different users can access mobile nodes, implemented with mobile embedded systems through base nodes, implemented with embedded systems, directly connected to Internet. The proposed structure is modular and scalable being limited only by the Internet access possibilities and the selecting software.

Future development directions can be:

- the development of the mobile nodes by adding them more features such as: possibility to locally memorize more messages, implementation of a menu for finding already received messages, phonic announcement of the arrival of a message etc.
- the improvement of the communication base node – mobile nodes by sending more information as acknowledge or by offering the possibility to sent messages from the mobile nodes to the central server or to other mobile nodes;
- the data base: more tables may be created allowing more complex access rights to the users.

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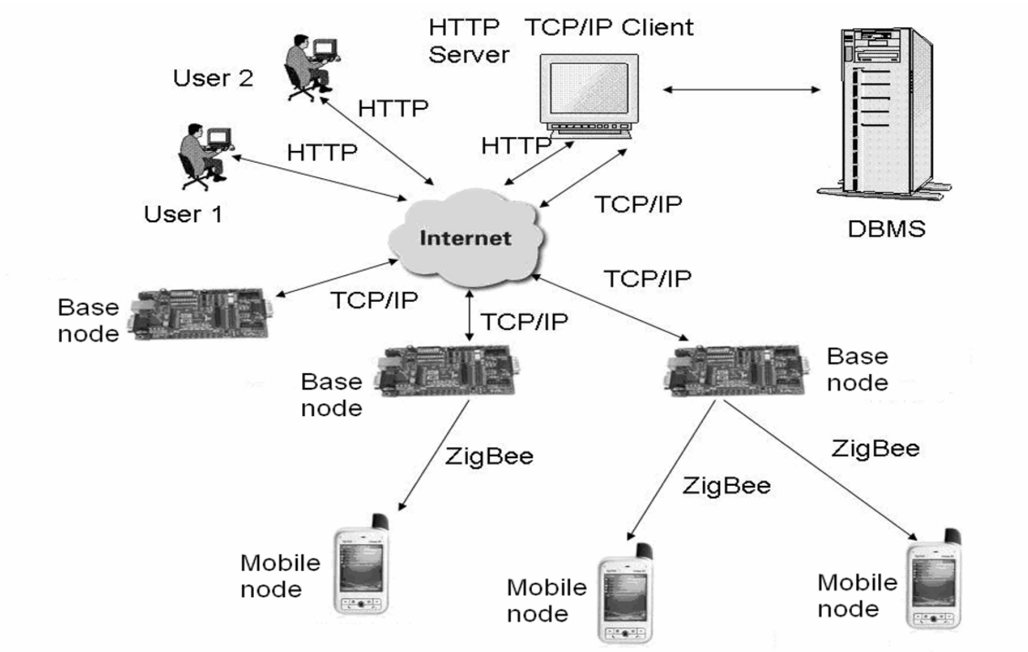

Figure 1. The Mobile Message Receiver System

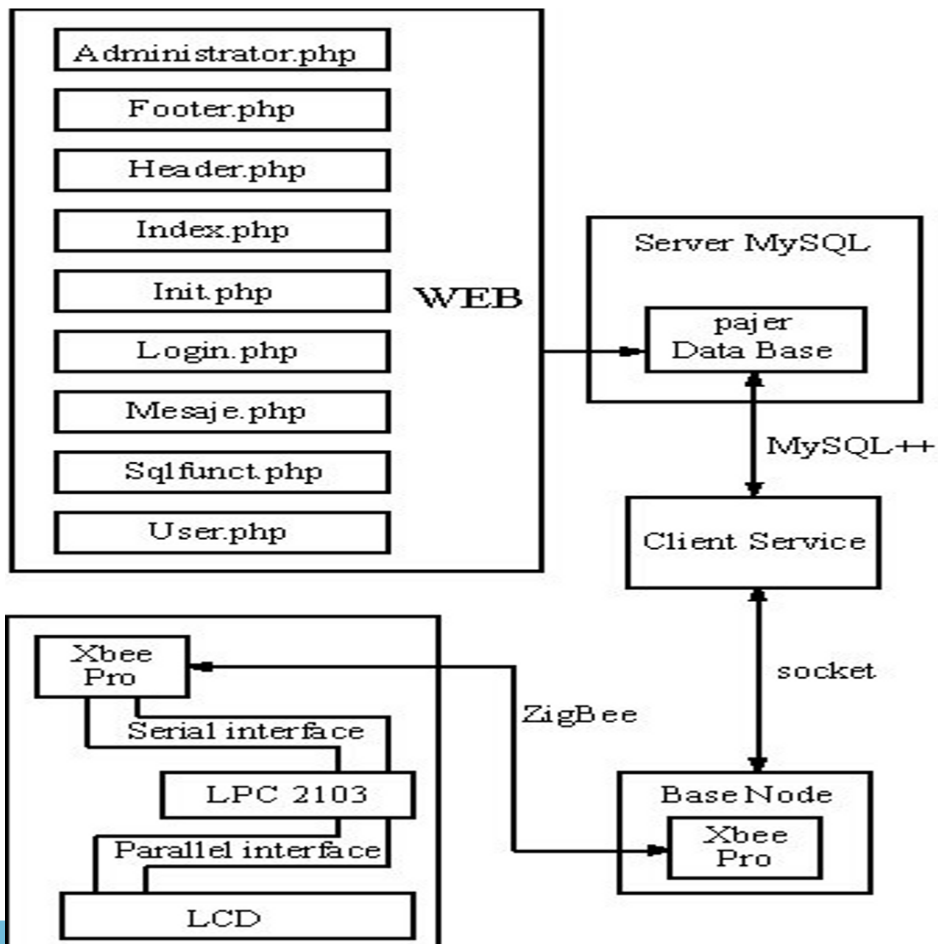

Figure 2. Software diagram of the Mobile Message Receiver System

Fast Editing of Images and Video on the Fractal Compressed Domain

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***Abstract*-Fractal coding is based on the existence of similarities between different parts of the same image (self-similarities). These similarities are unique for each image, and can be expressed in the form of transformations that match between the different parts of the image. This paper deals with the question of editing an image, and presents a new scheme which enables us to edit a given image or a video sequence on the fractal (compressed) domain. This scheme replaces the lengthy process of searching for self similarities in an edited image, with the fast adjustment of the original compression parameters. Moreover, these methods emphasize an important aspect of the fractal compression scheme, an aspect that makes this scheme a good candidate to be used in editing-related tools. In addition, we will suggest another way of searching for self-similarities in a given image, based on randomized search.**

I. INTRODUCTION

In this section we shortly describe the development of fractal image compression, and portray the nature of the fractal compression-decompression process. We also briefly summarize fractal approaches to video compression.

A. *Fractal Based Compression*

The mathematical background of the fractal transform is based on the contractive mapping fixed point theorem and on the collage theorem[1]. Combining these two results leads to the process of producing a collage of a given image by using smaller copies of the image itself.

The term IFS was coined by Barnsley [1] and is an abbreviation for 'Iterated Function System' - an extension to the work done by Hutchinson [2]. The IFS of an image is basically the set of transformations which were used to create the collage of that image.

Jacquin described for the first time an automated method of finding the IFS of a given image [3]. The algorithm divides the image into non-overlapping fixed-size *range blocks*. Each range block is then matched against all possible *domain blocks* for best fit. The comparison utilizes 3D affine transformations of the form:

$$w(x,y,z) = \begin{pmatrix} a & b & 0 \\ c & d & 0 \\ 0 & 0 & s \end{pmatrix} \begin{pmatrix} x \\ y \\ z \end{pmatrix} + \begin{pmatrix} e \\ f \\ o \end{pmatrix}$$

Where the z coordinate is the intensity of the pixel in the coordinate (x,y). We define:

$$t(x,y) = \begin{pmatrix} a & b \\ c & d \end{pmatrix} \begin{pmatrix} x \\ y \end{pmatrix} + \begin{pmatrix} e \\ f \end{pmatrix}$$

And so, $w(x,y,z) = (t(x,y), s*z+o)$. We call s the contrast scalar and o the brightness scalar (in the literature they are sometimes referred to as scaling and offset scalars).

For the mapping to be contractive we require t to be contractive, and $|s|<1$. This condition is sufficient but not essential, and a discussion on some less restricting conditions and eventually contractive mappings, see [4].

The pool of *domain blocks* is comprised of blocks of exactly twice the size of the range blocks. For each range block R_i, we search through all possible domain blocks of twice its size, and apply one of the predefined affine transformations until a good match is found. And so, each range block R_i will be associated with a domain block D_i and a mapping w_i. The mappings are of the form: $w(x) = 0.5A(x)+t$, where A is one of a few predefined matrices (usually between 4 to 8), which are capable of making simple transformations (i.e. a 90/180/270 degrees rotation, and a reflection on the x or the y axis). As explained earlier, we add the contrast and brightness scalars, so that the distance between the blocks will be minimized[2].

We determine whether $w_i(D_i)$ is close enough to R_i, by measuring if $d(w_i(D_i), R_i) < \varepsilon$, for some threshold ε. For the practical implementation we will use the RMS metric, which produces good results and is computationally feasible. In the RMS metric, the distance between the two blocks of same size A and B, $d_{RMS}(A,B)$, is given by: $\sqrt{\sum_{i=1}^{n}(a_i - b_i)^2}$, where n is the number of pixels in each block, a_i and b_i are pixel intensities in the same relative coordinates, $a_i \in A$, and $b_i \in B$.

Since the blocks are of known size, we may store for each range block R_i, the coordinates of the chosen (closest) domain block D_i (e.g. the lower left point of that block), the transformation's number (e.g. from 0 to 7), and the contrast and brightness scalars. The collection of all the domain block positions and chosen transformations, stored in a succinct manner as described above, is the heart of the fractal image compression. Of course, entropy (lossless) compression may be applied as well.

According to the collage (and the fixed-point) theorem, the fixed point x_f of w (the collection of the affine transformations), will serve as a good approximation to the original image.

Remembering that $x_f = lim_{n\to\infty} w^n(x)$, decompression will consist of applying the transformation to each R_i in some arbitrary initial image, and then repeating the process for several

[1] See [1] for details and for a more comprehensive background.

[2] The actual calculation of s and o depends on the chosen metric.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 355–361.

iterations (usually up to 10, as no visible changes occur later), to receive the approximated image.

B. *Video Compression with IFS*

The fractal approaches to video compression may be divided into two classes: dependent and independent.

In the latter approach we simply encode each frame of the sequence separately, using regular fractal-techniques. Thus, each frame is independent on other frames, and the compression/decompression process is identical to the process described above.

The dependent approach tries to use domain blocks from the previous frame. In this method [5], a standard fractal-based encoding is applied on the first image, and the following frames use blocks from previous frame as the pool of domain blocks. To be more accurate, if we wish to compress frame number k, k>1, we apply the usual range-blocks segmentation, but the domain blocks are taken from frame number k-1 (It is in a way, a similar practice to that of regular motion compensation methods). Contrary to the iterative fractal-based decoding, in this scheme the decoding is simply done by applying the chosen transformation on the domain block (from the previous frame) only once, yielding a very fast decoding process.

In a variation of that method [6], standard fractal-based compression is done to key images (every few frames), and the following images may use a previous key frame as a base image. After an image is segmented into range blocks, each block is compared to the corresponding block (a block of same size and position) in the base image. If the residual difference between the two blocks is small, then the block to be compressed may be taken as is from the base image (or in another variation- may be predicted by saving differences-only in some succinct manner). Compression is done only to blocks for which the prediction process failed, and the domain blocks in this case are taken from the same frame.

Other approaches were suggested, among them 3D range blocks, in which time serves as the third dimension [7], hybrid methods combining both DCT-based and wavelets together with fractal compression [8], and motion compensation using fractal methods [9].

C. *Organization of the paper*

This paper is organized as follows: In section II we present a new method, based on a probabilistic search, which could be used to accelerate the process of compression. In section III we introduce and discuss a method for performing editing operations on the compressed domain. This method can reduce dramatically the processing time of image or video compression. We consider here editing operations of a sequence such as fade-out, and geometric editing operations like rotation. Then, in section IV we consider implementations and results of our methods. We conclude and consider future work in section V.

II. A QUICK PROBABILISTIC SEARCH

This section proposes an extension to the compression process, aimed at allowing a faster compression. Searching through all possible domains for a given range block is quite an intensive process, even after classifying the blocks and only comparing blocks from similar classifications [10,11]. Several attempts were made to reduce the search in some heuristic manner (see, e.g., [12]), yet, this task is still far from being optimized.

Speeding up the process of compression can be done on the expense of the image quality. One way to do it is to increase the threshold of the comparison between blocks. It this way, blocks which are only remotely close might be matched, hence leading to a decrease in the image quality. However, reduction in the time complexity is not guaranteed since worst case scenario may lead to an exhaustive search.

Another way for speeding up compression time is to increase the range blocks' size. That is, instead of using blocks of say 8x8, one might use block size of 16x16. This usually leads to a noticeable degradation of the image quality, since the larger the blocks are, the more difficult it is to find a reasonable match. For that reason the time complexity reduction is usually not significant - the pool size of the domain blocks is indeed decreased, but typically, a search through all the blocks will be necessary.

The method proposed here **guarantees** significant time reduction, and tests show that the quality reduction is much less meaningful compared to the methods described above.

The basic idea is to use a probabilistic search; that is, to search through part of the blocks, in some random order. How much to search may be left to the user's decision - to increase the speed of the process, the user may limit the search to half, quarter, or even one tenth of the blocks (by setting a 'speed factor' to 1/2, 1/4, or 1/10 respectively). The randomness of the process allows us to do a quick search in different parts of the image in order to locate the best matching block. Also, if a reasonable match was found, we may continue to search a few more blocks in that vicinity, in order to make sure that the best representative of that area was chosen. This way, the degradation of the image quality is inconclusive and depends on the image self-similarity. The more self-similar it is, the more chances it has to be well-covered using relatively small amount of compared domain blocks.

Moreover, this technique does not necessarily have to be associated with quality loss, but can rather be used regularly. Standard search methods simply go through all the domain blocks in some predefined order (e.g., row by row, spiral, etc.), which may cause a search in areas which should otherwise be skipped. Using a randomized order of the blocks – even if not reducing their amount, i.e. using a factor of 1 – allows us to visit in most areas of the image quite early in the process. This allows us to use a quick filter and focus on areas which are more probable to yield a valid match. In most cases it will result in a faster compression process, without any cost in image quality.

III. IMAGE AND VIDEO EDITING IN THE COMPRESSED DOMAIN

In this section we shall describe a way to avoid the time-consuming process of **recompression** of a compressed image or video sequence that was altered due to common editing operations, by using a new approach. We shall first explain the motivation for this approach, and then discuss its application on some of the more important editing operations. The discussion will be both on video and image editing as they are closely related.

Common image and video editing tools enable the user to load and display the file he wishes to edit. Displaying involves decompressing the file and storing it as raw uncompressed data in the computer's RAM (in the case of video sequence, several frames from different parts of the sequence may be decompressed, depending on the user's request). Then, the user is able to apply some changes to the image or sequence (e.g. change of brightness or perform a 90º rotation). The naïve way to save the file after editing is to simply recompress the data regardless of the original compression; thus ignoring and overwriting the old data. This approach could introduce some visible delay to the user that might be quite long when using fractal-related schemes.

A better solution is to try and avoid the recompression process, by evaluating what changes are made to the compressed data by the naïve approach, and to implement those changes directly on the compressed data. In other words, we wish to implement the editing operations directly on the compressed file or the compressed domain.

With this approach we may reduce the amount of processing time dramatically. For example, if we wish to change the brightness of a one-hour video (compressed using fractal methods), using the naïve approach will result in several hours of work (compression), while applying only the necessary changes could take just a few seconds!

This approach was suggested with regards to other compression schemes such as JPEG and MPEG [13,14], but its necessity in fractal compression is more apparent due to its time consuming nature.

In the rest of this section we will discuss several common editing operations, including its application for image and video. The analysis presented deals with compression using the algorithm of Jacquin, with a further discussion of the implications for other implementations. For video, two schemes are being examined: dependent (on previous or future frames), and independent (where each frame of the sequence is compressed separately).

A. *Description and formulation*

Suppose for simplicity that we are dealing with a grayscale image (or video sequence), hence each pixel is given by its intensity. We will use the notation (x,y) for the pixel's coordinates, and $p(x,y)$ or p, for the pixel's intensity.

Now, suppose that the image is compressed by Jacquin's algorithm using the RMS difference to measure distance. This algorithm associates with each range block R of size 8x8, a domain block D of size 16x16, for which a contractive mapping w was found. The mapping consists of a transformation t which is one of eight predefined shrink-by-two affine transformations (capable of matching the block in four possible orientations, with and without flip), and of scale and offset scalars (s and o) that minimize the distance between the blocks.

To be more formal, let us focus on an arbitrary range block R. This block is expressed by a transformation w on a domain block D, which satisfied $d(w(\mathrm{D}), \mathrm{R})<\varepsilon$, for some pre-defined threshold ε. This transformation w may be expressed as: $w(x,y,z) = (t(x,y), s{*}z+o)$, where (x,y) are the pixel coordinates, z is its intensity, and t is the contractive affine transformation applied to the coordinates of the domain block pixels (one of eight).

The distance between the range block R and the transformed domain block t(D) (both of size 8x8 since t is shrinking the domain block by two) is given by:

$$d_{RMS}(\mathrm{R},t(\mathrm{D})) = \sqrt{\sum_{i=1}^{64} [(s*d_i+o)-r_i]^2}\ ,$$

where $d_i \in t(\mathrm{D})$, and $r_i \in \mathrm{R}$, are the intensities of the pixels in the same position within these blocks.

In the computations to follow, we will compare between two RMS distances. For simplicity we will omit the square root from the calculations, and compare their squares. Thus, we will write (on the expense of script accuracy):

$$d_{RMS}(\mathrm{R},t(\mathrm{D}))= \sum_{i=1}^{64} [(s*d_i+o)-r_i]^2$$

Therefore, every 8x8 range block may be expressed by $(\langle x,y\rangle, t, s, o)$, where:

- $\langle x,y\rangle$ are the upper left coordinates of a 16x16 reference (domain) block
- t is the number of the contractive transformation (range from 0 to 7)
- s is the scale (contrast) scalar by which the pixels in the reference block are multiplied ($0 < s < 1$)
- o is the offset (brightness) scalar to be added to the pixels in the reference block (usually -128 to +128, but other values may be used)

B. *Editing operations*

The nature of the editing operations that we will discuss is that they apply **linear transformation** to the pixel intensity or coordinates. Examples to such operations are still or video sequence Fade-Out (Contrast), Brightness, and Invert. Implementing them involves a manipulation of each pixel by means of scalar addition and/or scalar multiplication.

General scalar addition and multiplication

We first examine the case of still images and independent video sequences. The following proposition depicts our approach for performing editing operations on the compressed domain.

Proposition:
Suppose that I is an image compressed using the fractal based method which associates with each range block R of an arbitrary size, an appropriate domain block D, and uses the RMS metric to measure distance.
Suppose that we wish to construct an image I' such that each pixel in I' is given by $p'(x,y)=c*p(x,y)+b$, where $p\in I$, $c\in(-1,1)$, $b\in(-255,255)$.
Then, given the transformation t, and the parameters s and o which are associated with the domain block D and satisfies $d_{RMS}(R,t(D))<\varepsilon$, the parameters $t'=t$, $s'=s$, $o'=c*o+(1-s)b$, and the corresponding domain block D' in I' satisfies $d_{RMS}(R',t(D'))<\varepsilon$.

Proof:
Let n be the total number of pixels in R (and therefore in $t(D)$), let w be the mapping defined by t, s, and o, let w' be the mapping defined by t, s, and $o'=c*o+(1-s)b$.
Now, the RMS difference between $t(D)$ and R under w is:

$$d_{RMS} = \sum_{i=1}^{n}(s*d_i+o-r_i)^2 < \varepsilon, \text{ where } d_i\in t(D), \text{ and } r_i\in R.$$

Each pixel in D' is expressed by $d'_i = c*d_i+b$ and in R' by $r'_i = c*r_i+b$.
And the RMS difference between $t(D')$ and R' under w' is:

$$\sum_{i=1}^{n}(s'*d'_i+o'-r'_i)^2 = \sum_{i=1}^{n}(s*d'_i+c*o+(1-s)b-r'_i)^2 =$$

$$\sum_{i=1}^{n}(s(c*d_i+b)+c*o+b-s*b-(c*r_i+b))^2 =$$

$$\sum_{i=1}^{n}(s*c*d_i+s*b+c*o+b-s*b-c*r_i-b)^2 =$$

$$\sum_{i=1}^{64}c^2(s*d_i+o-r_i)^2 = c^2*d_{RMS} < \varepsilon$$

since $d_{RMS} < \varepsilon$ and $c^2\le 1$ ($|c|\le 1$). ▪

Notes:
- Some operations require the multiplication by a scalar c greater than 1. In such case, we may continue with this implementation, and initiate a new search only for range blocks that exceed the threshold distance from their original covering domain block
- Throughout this discussion we ignored the issue of re-sampling the pixels in the domain block upon downscaling it (each block of 2x2 pixels in the domain block is first shrunk into a pixel, due to the application of the affine transformation). However, it is easy to show that the re-sampling of pixels, either by selecting a representative or by averaging the pixels, does not affect the above calculations.

Editing a sequence with dependencies
In this case, we should study the relations and dependencies between consecutive frames and apply the changes accordingly. Suppose that we want to apply fade-out on a video sequence $S=S_1S_2\ldots S_n$, in which the first frame is compressed independently by Jacquin's algorithm, and all consecutive frames are compressed by using blocks from the previous frame. In other words, S_1 is independent, and $S_i = f_i(S_{i-1})$, for some contractive transformation f_i, $2\le i\le n$.
S_1 is independent, and may be treated by the above mentioned methods. Let us now examine what changes should be made to S_k, $2\le k\le n$. The uncompressed image of S_k would need to have all its pixels faded by a scalar c_k to get S'_k. In the original compression, the corresponding range block R in S_k was expressed by a domain block D in S_{k-1}, using a contractive 2D transformation t, a contrast scalar s, and a brightness scalar o. For the representation of R' we will choose the corresponding domain block D', the same contractive 2D transformation t, a contrast scalar $s'=(c_2/c_1)*s$, and a brightness scalar $o'=c_k*o$. Let us test our approach:
The RMS difference between t(D) and R, $R\in S_k$, $D\in S_{k-1}$, is:

$$d_{RMS} = \sum_{i=1}^{n}(s*d_i+o-r_i)^2 < \varepsilon, \text{ where } d_i\in t(D), \text{ and } r_i\in R, \text{ and n}$$

is the number of pixels in R. Each pixel in D' is expressed by $d'_i = d_i*c_{k-1}$, and in R' by $r'_i = r_i*c_k$. Thus, the RMS difference between t(D') and R' $R'\in S_k$, $D'\in S_{k-1}$, is:

$$\sum_{i=1}^{n}(s'*d'_i+o'-r'_i)^2 = \sum_{i=1}^{n}\left(\frac{c_2}{c_1}s*d'_i+c_2*o-r'_i\right)^2 =$$

$$\sum_{i=1}^{n}\left(\frac{c_2}{c_1}s*c_1*d_i+c_2*o-c_2*r_i\right)^2 =$$

$$\sum_{i=1}^{n}(c_2)^2(s*d_i+o-r_i)^2 = (c_2)^2\sum_{i=1}^{n}(s*d_i+o-r_i)^2 =$$

$$(c_2)^2*d_{RMS}\le d_{RMS}<\varepsilon$$

For a discussion of other linear editing operations, their application to a dependent video sequence, and their extension to different partitions see [15].

Geometric editing operations
The following operations belong to a different family and may be categorized as geometric operations (transformations), as they apply changes to the position of the pixel, rather than to its intensity.

Rotation- The first geometric editing operation that we will discuss is the rotation of an image by 90º, 180º, or 270º clockwise[3]. This is a quite common operation that arises from the need to view an image that was scanned not in its natural orientation (vertical vs. horizontal, etc.).
Let us first consider the rotation of an image of size $X_{img}*Y_{img}$ by 90º. Suppose that the coordinate system of the image is ranging from 0 to $(X_{img}-1)$ on the x-axis, and 0 to $(Y_{img}-1)$ on the y-axis (where (0,0) is the upper-left pixel of the image). The center point of the image $P_c(x_c,y_c)$ is given by:
$x_c=(X_{img}-1)/2$, $y_c=(Y_{img}-1)/2$.

[3] The limitation to multiples of 90º arises from the nature of the square-based partitions we examine. However, other partitions (e.g. triangular) may accommodate for a more general kind of rotation

The rotation by 90º for each pixel p(x,y) of the X_{img} on Y_{img} image about its center point $P_c(x_c,y_c)$ is given by:

$$\begin{pmatrix}0 & -1\\ 1 & 0\end{pmatrix}\begin{pmatrix}x - x_c\\ y - y_c\end{pmatrix}+\begin{pmatrix}x_c\\ y_c\end{pmatrix}=\begin{pmatrix}(y_c+x_c)-y\\ (y_c-x_c)+x\end{pmatrix}=\begin{pmatrix}(Y_{img}+X_{img})/2-y-1\\ (Y_{img}-X_{img})/2+x\end{pmatrix}$$

Now, suppose that before the rotation, a range block R was covered by a domain block D, and suppose that the upper-left coordinate of R is (x_r,y_r), the lower-left coordinate of R is (x'_r,y'_r), the upper-left coordinate of D is (x_d,y_d), and the lower-left coordinate of D is (x'_d,y'_d). The compressed file would indicate that the block starting at (x_r,y_r) is covered by the block at (x_d,y_d).

Now, The rotation of the image will affect both blocks and results in R being rotated so that its upper right coordinate is $T(x_r,y_r)$, its upper-left coordinate is $T(x'_r,y'_r)$, and D will be rotated so that its upper right coordinate is $T(x_d,y_d)$, and its upper left coordinate is $T(x'_d,y'_d)$. In other words, R is mapped into a block R', D is mapped into a block D', and both new blocks' coordinates are known. In order to compress the rotated image, we may mark R' covered, by using D' and the transformed coordinates.

It is easy to see that $d_{RMS}(D', R') = d_{RMS}(D, R) < \varepsilon$.

This scheme is shown in figures 1 and 2. It can be extended of course to 180º and 270º rotations.

For a discussion of other geometrical operations see [15].

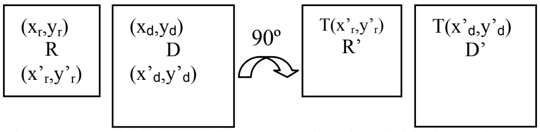


Figure 1- Range block R and its corresponding domain block D. After a 90º rotation, the new positions of R' and D' can be easily calculated.

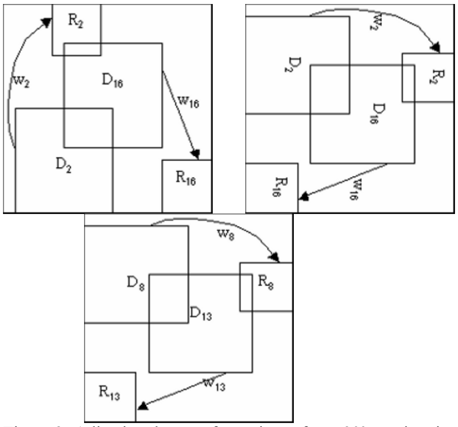


Figure 2- Adjusting the transformations after a 90º rotation, instead of recompressing

IV. IMPLEMENTATION AND RESULTS

We implemented a program based on the ideas and principles described in previous sections. The following report is based on results that were produced by this program, and portray several editing operations enabled by it.

A. Probabilistic method

The method was tested on several images of different sizes and textures. There is of course a reduction in image quality compared to the exhaustive search, but given the naïve implementation of the method (only random search with no focusing on zones) the results are promising; a massive reduction in time is achieved – about 75% less than exhaustive search, with only a relatively small reduction in image quality comparing to traditional methods (see figure 3).

One disadvantage that this method has is that although the reduction in quality is small, there is no reduction in file size, unlike conventional methods which achieve higher compression ratio at lower image quality.

B. Degradation of the image in regular save

First we note that in the standard fractal method, simply saving an already compressed image (even without any changes to the image itself) leads to a degradation of the image quality. Unlike DCT based compression schemes, degradation in quality is introduced **each** time the image is compressed. The reason for that is as follows: each time we compress, we mount a search on an approximation of a certain image. So, if the distance between a matched domain and range blocks was ε_1; the next time we search for a match we might find a domain

Figure 3 – Results of the probabilistic search method. (a) – Fractal compression, exhaustive search, low threshold, small range blocks, maximal compression time (b) – Probabilistic search, same threshold and range blocks, 75% time reduction, small reduction in image quality (c) Time reduction by using the traditional method with larger threshold, massive quality reduction (pixelization) (d) Time reduction by using the traditional method with larger range blocks, massive loss of details (e.g. the distortion of the eyes)

block which is in distance ε_2 from the new range block, and the total distance from the original block could be $\varepsilon_1+\varepsilon_2$. After few more iterations we might end up with an error far greater than intended (see figure 4). Contrary to that, our scheme does not lead to further degradation since it makes use of the original parameters.

C. Editing results

Several images were tested, by editing and saving using the standard fractal method, and the proposed method. The results show that no degradation of quality was introduced (see figure 5). The compression time (recompression) was ~0.1-0.2 seconds, regardless of image size. This should be compared with several seconds needed to compress a reasonable size image in the standard way, and with the loss of quality introduced by it. The results are even more meaningful for large images (e.g. 1027x768). Table 1 gives a comparison between the old method (i.e. recompression from scratch) and the proposed one (adjustment of the original parameters), as was tested on several images of various sizes.

The new method is much faster and maintains the original quality, while the old method introduces quality loss, even with settings aimed at achieving high quality.

For a more detailed discussion of those results see [15].

V. CONCLUSIONS AND FUTURE WORK

In this paper we proved that it was possible to edit an image which has been compressed using fractal methods, and to save the changes with minimal time complexity, despite the long compression time typically associated with fractal compression. Using the nature of the fractal transformations we managed to adjust the partitioning (and mappings) of an originally fractal compressed image (or video), that have undergone certain changes due to common editing operations. By using this approach, the edited image is readily available in its compressed form, avoiding the long process of compression. The time complexity is significantly reduced from $O(n^3)$ (in the case of traditional Jacquin) to time linear in the file size, which is at most O(n). Moreover, all of these methods can be easily adapted to other fractal partitioning schemes (for example, our implementation uses a Quadtree partitioning).

Such an implementation could compete with existing editing tools based on standard compression methods. Furthermore, this approach is not limited to editing tools only, and any software which requires the manipulation of images (for analysis or other needs), could combine these ideas in its implementation.

Other editing operations may be investigated and new approaches could be devised. Examples of such operations could be: non linear editing, morphing, blurring, sharpening, and other convolution-based or filtered-based operations.

New approaches could consider other aspects of the compression (e.g. the classification, the distance between the domain and range blocks, etc.) or the features of the blocks (average, quadratic sum or other statistics, etc.), and to exploit them appropriately.

Figure 4 – Degradation of image quality due to repeated (10) saving

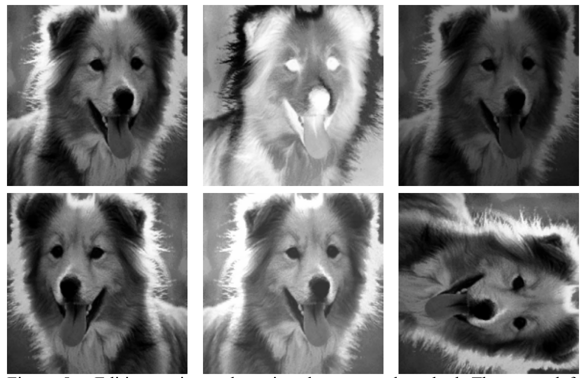
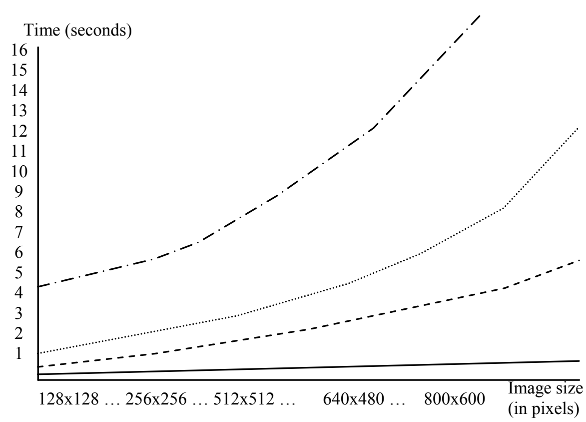


Figure 5 – Editing an image by using the proposed method. The upper left image is the originally compressed image. The following images were edited and save by using the proposed method. We can see that image quality is maintained.



Old method; parameters are set for minimal quality loss; results in low speed compression.

Old method; parameters are set for faster compression; visible quality loss is introduced.

Old method; parameters are set for fast compression and low quality.

New method; quality is always maintained, compression is almost instantaneous.

Table 1 – Old method vs. proposed method. Average times were obtained on a 1.7 GHz single core processor.

For example, suppose we want to facilitate a filter-based operation to be performed on the compressed domain. Most likely that after the operation is performed, the original domain block will not be suitable for covering the same range block as before. However, we may calculate or estimate the effect this operation had on a certain characteristic of the block (say, its average, its norm, its spanning functions in the frequency domain, or any other quality that we have established for this block using some analysis tools), and to use this information to, for example, reclassify the blocks. In this case, even though we might need to compress the image again, it would be safe to assume that the original classification combined with the new information, will lead to a much faster block search, hence a faster compression. Also, different partitions schemes may accommodate for other operations on the compressed domain (for example, triangular partitioning may facilitate the rotation of an image by angles other than multiplications of 90º). Of course, hybrid methods could be devised as well.

Final note – Although fractal compression still has its limitations compared to DCT based compression, advancement such as proposed here could make it more competitive. Moreover, while we know exactly what to expect from DCT based compression, it seems that the fractal based compression would benefit from further attention, as there could be improvements not only to the speed of the process, but also to the compression ratio; and it could be possible to realize other properties that would better define the nature of the fractal approach.

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An Optimal Job Selection Method in Load Balancing Algorithms of Economical Grids

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Abstract

A computational grid is a widespread computing environment that provides huge computational power for large-scale distributed applications. Load balancing, has a considerable effect on the grid middleware performance. Load-balancing algorithm and job selection are two parts of any load balancing methods in the grid. In the previous work we had proposed a Load-balancing algorithm for the grid environment. In this paper, we intend to complete the load-balancing algorithm by proposing an optimal job selection algorithm. As selecting jobs is a multi-criteria problem in nature, we define the job selection algorithm as an optimization problem and solve it using GA. The performance and the optimality of the method is proved by relevant simulations and experimental results.

Key Words:
Grid computing, Load balancing, Job Selection, Genetic Algorithms

1. Introduction

Grid computing enables users to gather geographically distributed computational resources to create a huge virtual supercomputer that can be applied for executing computational-intensive programs [1, 2]. An ideal grid environment should provide access to all the available resources seamlessly and fairly.

A resource manager is an important infrastructural component of a grid computing environment. The overall aim of resource management is to efficiently schedule applications that need to utilize the available resources in the grid environment [3, 4]. An ideal resource manager should exploit all of the grid resources uniformly [5, 6]. Considering the largeness, dynamic resources, stochastic users and other specifications of the grid, it is impossible to use resources in equilibrium, unless using load balancing/ sharing methods [7].

Load balancing algorithms try to equalize the workload among all available nodes, while load sharing algorithms just try to assure that no node is at rest whereas elsewhere processes are waiting for service [7].

Generally, each load balancing and load sharing algorithm can be defined by three rules: the location rule, the distribution rule and the selection rule [8]. The location rule determines which domain of resources will be included in the balancing operation. The domain may be local, i.e. inside the virtual organizations (VO), or global, i.e. between different VO. The distribution rule decides how to redistribute the workload among available resources on the domain. The selection rule decides on the picking proper jobs from overloaded node for reassign to the underloaded. So far, many contributions have been done to provide a good load balancing/sharing method for the Grid. A good survey on these algorithms can be found in [9].

In our previous works [3, 4], we have proposed new mechanisms for load balancing in the Grid. However, like most of other contributions, our mechanism had not any intelligence in picking proper jobs for transfer. In other words, they only propose how to find overloaded and underloaded nodes in the grid environment.

In this paper, we aim to provide a new job selection algorithm in which an optimal set of jobs are selected for transmission to underloaded node.

The rest of the paper is organized as follows: Section 2 contains a literature survey on job selection methods. In Section 3, the new job selection algorithm is described. The performance metrics and simulation results are included in Section 4. At the end, we present the conclusion of the article as well as the future works which can be done in the same direction as the inclination of this research.

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2. Problem Definition

As mentioned above, Load balancing methods are designed essentially to spread the load on resources equally and maximize their utilization while minimizing the total task execution time [8]. Selecting the optimal set of jobs for transferring has a significant role on the efficiency of the load balancing method as well as grid resource utilization. This problem has been neglected by researchers in most of previous contributions on load balancing, either in distributed systems or in the grid.

In preceding mechanisms, jobs, in the overloaded node, were simply selected from the end of ready queue until it fills the capacity of underloaded nodes [12]. However, such selection mechanism, in one hand, may consequences to a high communication overhead, lengthening the execution time and decreasing efficiency while, on the other hand, in economical grids, where resources receive money on executing other jobs, it may even result in losing money.

Such reasons have made us to propose a job selection method in which jobs are selected not only based on their position in the ready queue, but other factors like communication overhead and proficiency are considered.

Genetic Algorithm (GA) is a search algorithm rooted in the principles of natural genetics. GAs join the exploitation of past results with the exploration of new areas of the search space. By using *survival of the fittest* techniques, a GA can imitate some features of a human search. A generation is a group of artificial creatures (strings). In every new generation, a set of strings is produced using information from the prior ones. GAs are randomized, but they are not simple random walks. They use historical information to guess on new search points with expected enhancement [8, 10, 11].

In the next section, the selection algorithm is posed as an optimization problem and genetic algorithm is employed as a heuristics to combat the problem.

3. Algorithm Outline

As stated earlier, in this paper, we intend to put forward an algorithm which determines the optimal subset of jobs for transfer in a load balancing method.

At the first step, criteria based on which optimality is defined should be determined.

Considering difficulties of the grid environments, like communication overhead and dynamic bandwidth of routs, it seems that smaller jobs are better choice for transmission. In other words, larger jobs take longer time for transmission. Even, sometimes it is better to execute job locally instead of send it to another node and execute there. Therefore, job size is a decisive criterion for picking or not picking a job.

Since financial issues are important in the practical grid and different VOs compete with each other based on economical profitability [13], hence, in load balancing condition, the sender node would prefer to select less profit jobs for transmission.

Waiting time is the other decisive factor for selecting jobs to be relocated. Obviously, preventing starvation is a crucial goal of each load balancing/sharing algorithm. Therefore, the sender would rather to displace jobs which have more waiting time.

Now, we are going to select jobs from the overloaded nodes based on the combination of the aforementioned optimality criteria. It goes without saying that, selecting jobs from the end of ready queue wouldn't satisfy these criteria at all. On the other hand, if there are N jobs in the ready queue, then, we should examine 2^N different conditions and select the best (optimal) set of jobs that satisfies all of above criteria. Unarguably, this brute force algorithm is NP-Complete and the time complexity is not acceptable. Hence, it is better to use heuristic methods to find the optimal or near optimal solution in a satisfactory time.

Since the problem is a combinatorial optimization problem, GA is a good heuristic that help us combat the problem [12]. The technique needs a coding scheme that can represent all legal solutions to the optimization problem. Any possible solution is uniquely represented by a particular string which is called chromosome [14]. Here, the chromosomes are in decimal form and each gene shows a job number. Figure.1 shows a sample chromosome. These chromosomes are manipulated by crossover and mutation operators until the algorithm converges on an optimal or near optimal solution.

10	0	0	0	7	41	0	44	38	3

Figure.1. A sample chromosome. Each entry of the border indicates a gene and shows a job number. Job numbers are unique in each chromosome except zeros. Zeros show that there isn't any job in that cell.

A method that assigns a quality value to each solution string (*fitness function*) guarantee that this operation proceed in correct order. We formalize the proposed criteria in the form of the following fitness function for each chromosome C_k:

$$f_{C_k} = \sum_{i=1}^{L} (W_i * \frac{1}{S_i} * \frac{1}{P_i}) - \left(\sum_{i=1}^{L} T_i - \alpha \right)^2 \quad (1)$$

Where 'L' is the chromosome length, W_i indicated the waiting time of each job i (gene i) in the ready queue, S_i shows the size of each job, P_i is the profit attain from execution of each job, T_i is the time needed for executing a job, and α is the amount of time the receiver can give to the coming jobs. Actually, α is a constraint from the receiver that should be satisfied by the sender. This constraint is fulfilled by the penalty function:

$$\left(\sum_{i=1}^{L} T_i - \alpha \right)^2 \quad (2)$$

GA starts from an initial population and uses simple cross over and mutation operations in each generation it to maximize the fitness function.

4. Experimental Results

In this section, the simulation environment is described then we investigate several common statistics to illustrate the performance of the mechanism described.

We have used Genetic algorithm library of Matlab for simulation. In this simulation we assume the maximum number of jobs waiting in the ready queue is 100 and the maximum number of jobs could be selected are 50. A structure is used to indicate each job. α, which is the capacity of the receiver, could be differs time to time.

We have also implemented two other methods. The first one is the method which selects jobs from the end of ready queue. We briefly call this method '*MaxWait*'. The second method, which is implemented, builds all possible subsets and, finally, selects the best one. We call this method '*SubSets*'.

In the first experiment, fitness attained from each method is reported. As illustrated in Figure.2, the *SubSets* method results in the best fitness while the *MaxWait* method has the poorest fitness. The fitness reached through our *GA* method, after 50 and 100 generation, is near to the optimal one (*Subsets* method)

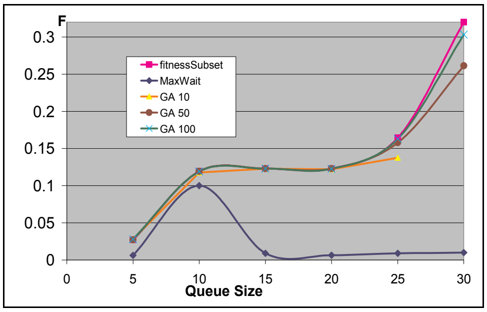

Figure.2. Comparing the fitness of selected jobs (F) for different Queue Sizes in each of three methods.

Time consumption of proposed *GA* method, with 50 and 100 generation, versus *Subsets* is compared in the second experiment. The NP-completeness of *Subsets* method could be easily understood from Figure.3. However, as it can be seen in this figure, the new method eliminates this unacceptable time. As figure 3 could not clarify the time difference between various methods, results of this experiment are also presented in Table 1.

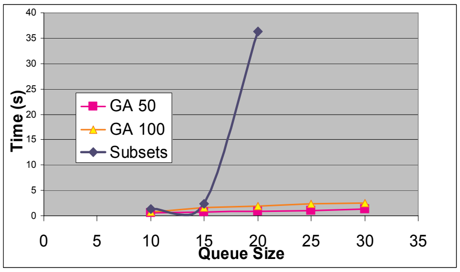

Figure 3. Time needed for selecting jobs for our new method (*GA*) (base on seconds) versus the optimal algorithm (*Subsets*)

Queue Size	Subsets	GA (50 Generation)	GA (100 Generation)
10	1.401	0.591	0.800
15	2.402	0.665	1.681
20	36.252	0.812	1.854
25	519.547	0.988	2.298
30	8304.21	1.310	2.532

Table 1. Precise time consumed (base on seconds) for selecting jobs in queues with different sizes in each method.

The third experiment elucidates the Efficiency of each method. Efficiency is computed based on its standard rule i.e. Performance/ Cost. Here, performance is the fitness achieved and cost is the time needed to reach that fitness. Since *MaxWait* method behaves completely random, we do not count on it in this experiment. However, the efficiency of other methods is illuminated in Figure 4.

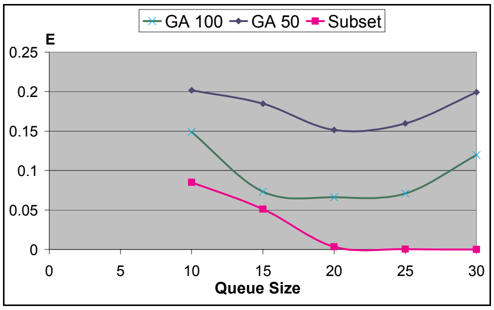

Figure 4. Efficiency (*E*) achieved in each method for queues with different sizes.

As grids are going to be economical [13], it is very important for any load balancing algorithm to get rid of less benefit jobs. In our new method, we have considered this factor and try to lessen the profit of submitted jobs. Hence, as our last experiment, profit loss of each method is revealed in figure 5. To achieve this experiment, a random number is assigned to each job as its profit.

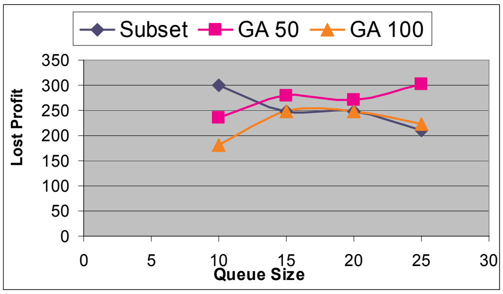

Figure 5. Lost profit caused by job selection algorithm for different queue sizes.

5. Conclusion and Future Works

As described in this paper, selecting an optimal set of jobs for transmission in overloaded nodes is critical for load balancing in the grid. In this way, with respect to grid specifications, an algorithm which acts based on genetic algorithms is proposed in this paper to meet the challenges of picking proper jobs for transmission. Experimental results, which are shown in preceding section, prove the appropriateness of genetic algorithm for solving this problem.

In future work, we plan to implement this method in a more realistic environment. Promoting intelligence and more adaptation, establishing billing contracts among resources as they exchange their extra loads, as well as overcoming security concerns are other future work.

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Weed Classification Using Erosion and Watershed Segmentation Algorithm

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Abstract- **Computer has application in every field in our society. It is also used in agriculture for automation, especially in the automatic spray for herbicides. In this paper an algorithm is developed for automatic spray control system. The algorithm is based on erosion followed by watershed segmentation algorithm. This algorithm can detect weeds and also classify it. Currently the algorithm is tested on two types of weeds i.e broad and narrow. The developed algorithm has been tested on these two types of weeds in the lab, which gives a very reliable performance. The algorithm was applied on 200 images stored in a database in the lab, of which 100 images were taken from broad leaf weeds and 100 were taken from narrow leaf weeds. The result showed over 89% results.**

I. INTRODUCTION

Over the last three decades, there has been a great deal of concern over the impact of agriculture on the environment. Modern conventional agriculture is heavily dependent on chemical inputs to promote high crop yields. Conventional agriculture has tended to ignore spatial variability within fields. Machine vision and the spectroscopic methods are best solutions to achieve spatial variability and thus can be used as a best replacement of conventional agricultural methods.

Two approaches can be used in a machine vision system. One is based on the size, shape, texture and or other statistical features of an image to detect and classify weeds. But machine vision system based on these features is usually slow in the real time environment because of mathematical complexity [8]. Reference [1, 2] used shape features, geometric invariants and statistical methods to discriminate between crop and weeds. In Reference [3, 4], the image is classified using color information and other has classified it using non-morphological analysis based on texture of the image as in [5].

Second method is Spectroscopic method; in which spectral reflectance or absorbance patterns used to discriminate between weeds and crops [6, 7]. The discrimination between plant and soil can be done easily using this method; however discrimination among plant species is difficult [9].

In practice, there are two methods used to accomplish variable herbicide application rates, Offline and Online [10]. In the offline, an application map is generated in one operation and then variable rate spraying is realized in further operations. Such systems have been developed and practiced by [11,12]. Second approach is online or real-time systems, in which weed detection and variable rate spraying are achieved in a single operation.

The overall objective of the present study was to develop image processing algorithm that are based on image texture in order to classify weeds into broad and narrow weeds. This paper describes algorithm developed based on online or real time approach, which is applied on different images and the results

II. OBJECTIVE

Since in practice there are only two types of herbicides used: for broad leave weed and narrow leave weed (grass), the objective of this paper is to develop an algorithm that can

- Recognize the presence of weeds
- Differentiate the presence of broad leaves weeds and narrow leaves weeds.

III. MATERIALS AND METHOD

A. Hardware Design

The initial design of the machine-vision-controlled sprayer included camera vision system is shown in Fig. 1. Images are taken with the help of camera before being processed by the system. The resolution of an image is 240 by 320 pixels. The camera is positioned at four meter and at a angle of 45 degree from the ground in front of the sprayer boom. This is because; the long narrow area in front of the sprayer could be captured with higher resolution without increasing the image size.

B. Proposed Method

The color images are converted to gray scale image for easy and fast processing. An image segmentation step is conducted to divide the image into two classes i.e plant and background (soil).The gray scale image A of the plant class is eroded by a structuring element B is defined as

$$A \Theta B = \{z \mid (B)_z \cap A^c \neq \phi\}$$

The MATLAB code of erosion is as

```
B = strel(shape,parameter)
```

strle is a morphological struturing element.The shape used here is ‘arbitrary’ and paramter is ‘eye(5)’

```
A = imerode(A,B)
```

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Here A is the input image and B is structuring element.
After eroding the image A, the watershed segmentation algorithm is applied whose MATLAB code is as

L = WATERSHED(A)

It computes a label matrix identifying the watershed regions of the input image A. The elements of L are integer values greater than or equal to 0. The elements labeled 0 do not belong to a unique watershed region. These are called "watershed pixels." The elements labeled 1 belong to the first watershed region, the elements labeled 2 belong to the second watershed region, and so on.

By default, WATERSHED uses 8-connected neighborhoods for 2-D inputs. For higher dimensions, WATERSHED uses the connectivity given by

CONNDEF(NDIMS(A),'maximal').

The sum of all pixels of the resultant image is calculated as

result = SUM(SUM(L)).

The value of 'result' is then compared with the selected threshold (T) for classification of weed in to broad and narrow weeds. This threshold T is selected after performing various trials on the images stored in the database.
This algorithm is developed using MATLAB 7.0 running on P.IV, Processor 3.06 GHz and 512 Mb of RAM.

IV. RESULTS AND DISCUSSION

Figure 2 show the classification images of broad and narrow weeds. These images are processed by the proposed algorithm. The algorithms gave 100% accuracy to detect the presence or absence of weed cover. For areas where weeds are detected, results show over 89 percent classification accuracy over 200 sample images with 100 samples from each class as shown in Table 1.

V. CONCLUSION

An algorithm based on erosion and watershed segmentation algorithm is developed for real-time weed classification and tested in the lab on a database of 200 images. Color images are taken which are then converted to grayscale image. Image segmentation step was conducted to remove background from the plant and then the proposed algorithm was applied to it. The system shows an effective and reliable classification of images captured by a video camera.

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Table I

Results of The Weeds in Fig 2 Using the proposed algorithm

Weeds Type	Results found correct %
Broad Weeds	90%
Narrow Weeds	89%
No or Little Weeds	100%

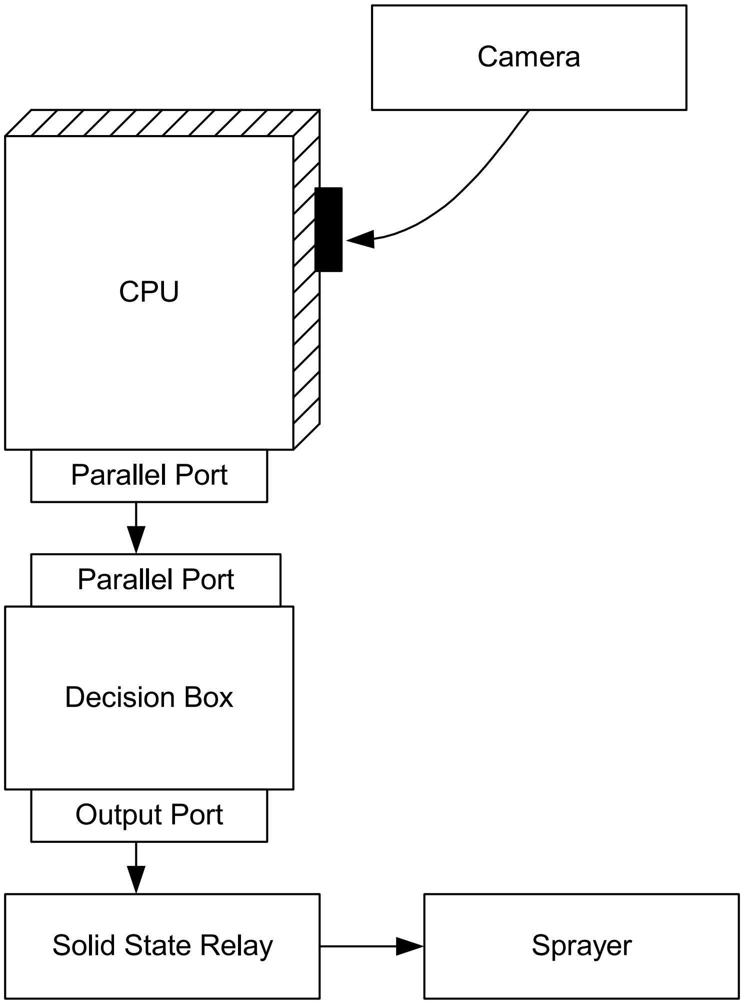

Fig. 1. The concept of a Real-Time Specific Weed Sprayer System

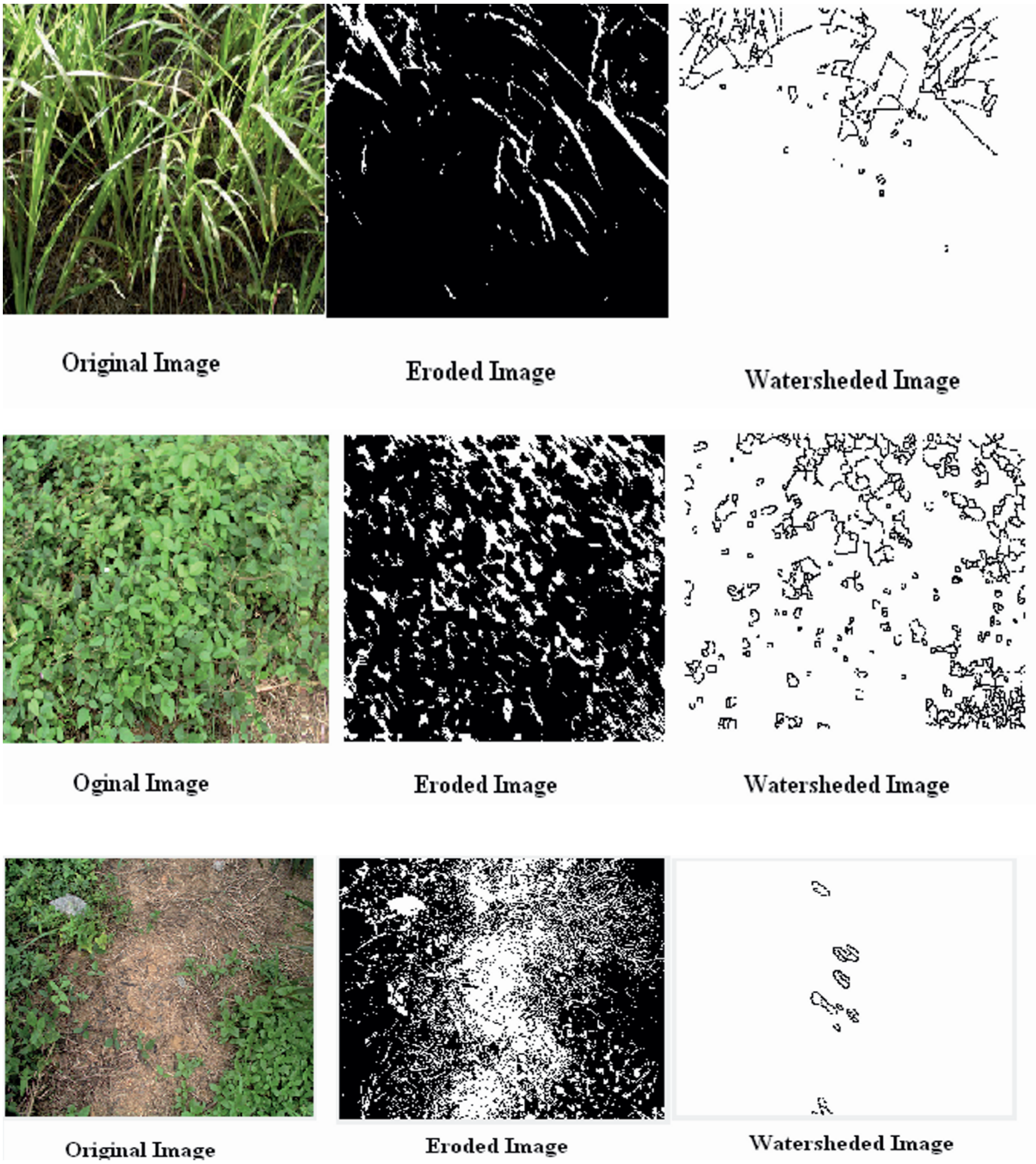

Fig.2. Resultant Images of Weed Classification Using Erosion and Watershed Segmentation

Leakage Power Mitigation in Static Cache Memories Using Threshold Voltage Modulation

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Abstract -Over the last few years, leakage power dissipation component in static cache memory modules has increasingly proliferated leading to higher level of static power dissipation. The trend has been caused by the aggressive shrinkage of CMOS transistor feature size and supply voltage reduction required by the CMOS process scaling to meet the growing need for high-performance computation and low-power consumption in microprocessor-based systems. This paper proposes a framework to monitor the processor cache cell at its finest granularity, reads out its threshold voltages, incorporates that into a popular cache power estimator and then optimizes leakage architecturally in early design stages. The associated effects of leakage optimizations on performance have also been quantified and evaluated thoroughly.

I. INTRODUCTION

Ideally, SRAM cells experience no static current. Yet, leakage current has increasingly become a major component of retention current due to subthreshold conduction that strikes CMOS transistors. This, in turn, contributes to higher levels of static power dissipation in the new generations of cache memories. Over the last decade, several threshold-related schemes have been devised by semiconductor researchers to combat the problem. Due to it to simplicity and no silicon area overhead, threshold-based techniques have particularly become appealing. Mutoh et al. [1] proposed dual threshold mechanism where devices in the critical path of the targeted circuits are driven with lower threshold voltage while those lie off the critical path are made less leaky-and therefore slow- by hiking the threshold voltage level. The concept has since then been adopted widely. Kuroda et al. [2] proposed a scheme where leakage is controlled by varying threshold voltage using the substrate bias voltage. Powell et al. [3] devised a model to optimize leakage proliferation based on supply voltage gating. The concept was then employed by Agarwal [4] to build the DRG-cache, a low-leakage cache memory.

As transistor counts in cache systems continue to grow along with circuit complexity, the need to explore design space at higher levels of hierarchy to save power, has become more important than ever before.

The existence of a wide range of threshold voltage variability has motivated the authors of this work to exploit the use of threshold voltage to conserve leakage dissipation. It proposes an architectural framework that parameterizes the threshold voltage and interfaces it into cache simulator [5] to achieve leakage optimization without upsetting the stored data in the CMOS based cells. At the microarchitectural level, the paper showcases the trade off between optimizing leakage power and performance in the cache system of a contemporary microprocessor.

The rest of this paper is organized as follows: section II studies and analyzes the vulnerability of the basic SRAM cell to leakage proliferation. In addition, it highlights the impacts of using the cell MOSFET device threshold voltage to mitigate leakage. Section III proposes our framework. In section IV, we implement the scheme in an experimental vehicle and present the results. The paper concludes in section V.

II. SUBTHRESHOLD LEAKAGE CONDUCTION IN BASIC SRAM CELL

This paper focuses on the family of six-transistor static memories that formulate the core of most of today's cache systems. The basic building block of such a standard CMOS cell structure is shown in figure 1. This category of cells tends to draw the least sub-threshold conduction current and therefore has been widely used in most cotemporary embedded and desktop computers including the one used as vehicle for this work. At its finest level of granularity, and regardless of the cell operational status, the cache MOSFET devices experiences static dissipation in form of off and on leakages. The former, however, comprises great portion of leakage budget in today's nanoscale regimes.

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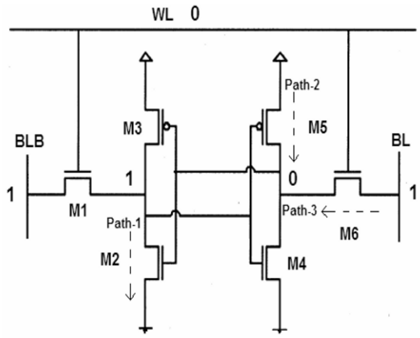

Figure 1 Subthreshold Leakage current component (dotted) in a CMOS based SRAM Cell.

When such a cell becomes idle – i.e. when WL is deasserted isolating its bit lines- either M3 and M4 or M5 and M2 operate in the weak inversion region depending on whether '1' or '0' is being stored. In either event, streams of subthreshold currents flow across the cell and add up to form the overall subthreshold current. The magnitude of each current component can be approximated mathematically by the following formula [6]:

$$I_{sub} = I_o \exp(V_{gs} - V_{th})/n\, V_T \quad (1)$$

where V_{gs} is the transistor gate to source voltage, V_{th} is its threshold voltage, n is the subthreshold coefficient and I_o , a process-rated current ,is given by:

$$I_o = \mu_o\, C_{ox}\, (W/L)\, V_T^2 \exp(1.8) \quad (2)$$

where μ_o is known as zero bias mobility. C_{ox} is the gate oxide capacitance. W/L is the width to length ratio of the MOSFET device. V_T is the thermal voltage which amounts to 26 mV at T=300K.

These formulas reveal the strong dependency of subthreshold current on threshold voltage.

Over the last few years, the supply voltage has been lowered to meet device technology scaling. The supply voltage scaling is normally accompanied with similar scaling on threshold voltage to maintain lower delay. This makes the cell vulnerable to subthreshold conduction in form of elevated leakage current. On the other hand, while raising the threshold suppresses leakage, it adversely increases the circuit delay and weakens its current drive which degrades its performance substantially. Exploring the variability of the cell threshold voltage does, however, provide great opportunities for computer architects to exploit this circuit parameter to combat leakage architecturally at the design time and project leakage savings levels early on. The next section focuses on the mechanism of achieving such a goal in more details.

III. FRAMEWORK STRUCTURE

The proposed framework allows the integration of the threshold variability into the process of leakage evaluation and optimization. The system can well be described by figure 2. It consists of the following modules:

1) Cache power estimator: This is the widely used architectural simulator, cacti [5]. It has been targeted and modified to become sensitive to threshold voltage variations in the CMOS cell building blocks. This is meant to influence leakage calculus architecturally as spelled out by preceding equation (1). That becomes possible by supplementing 2 pieces of firmware scripted using the original c-language semantics.

2) Threshold modulator: This module accepts the offset threshold voltages as a user input and interfaces it into the original estimator. These voltages are latched to this module in form of heuristically populated lock up tables (LUT). The degree of variability is design such that it does not hinder the CMOS cell functionality. A working set of these voltages is shown later in Table 1.

3) Optimizer: This unit performs data analysis of the results obtained by the modified estimator and attempts to calculate the minimum power estimates in coordination with the validator. Basically, it consists of a swap-like function that compares the previous threshold modulated leakage estimate with the current estimate and returns the lower values. The lower value is then tested for validation before output.

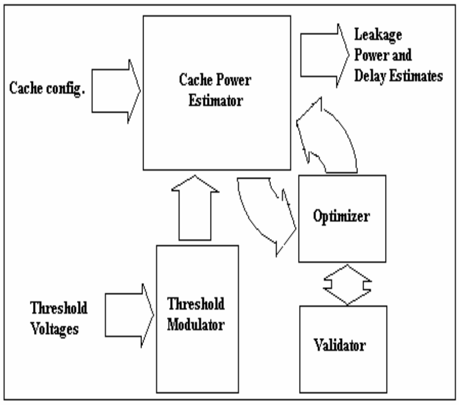

Figure 2 Block diagram of the composite system

4) Validator: This unit cross-checks the instantiated threshold voltage validity against the actual CMOS cell circuit technology-based datasheets. It excludes any possible out-of- range threshold voltage(s) that are either impractical or malfunction. In addition, the unit checks for possible erroneous estimates such as negative or imaginary power values. When a violation is encountered, the unit flags the optimizer to disqualify the current returned estimate and return previous values.

Algorithmically, an iteration of system operation can be illustrated figure 3 below:

Algorithm
Input: Parameterized cache configurations and Threshold Voltages
Output: Threshold-sensitive Leakage Power and Delay Time Estimates.
begin
set Technology parameters;
set Cache Organization parameters;
set = Lowest_Threshold;
Threshold= Lowest_Threshold;
for each (Threshold) **do**
Threshold= Thresh_Mod(Threshold);
if (Validate (Threshold)) then
Leakage_New= cacti_leakage (Threshold);
Leakage = Optimize(Leakage_New);
Time_Delay_New=cacti_Time(Threshold);
else
Leakage_New= Err_leakage (Threshold);
Time_Delay_New= Err_Time(Threshold);
endif
Threshold= Next_Threshold;
endfor
Leakage= Leakage_New;
Delay =Time_Delay_New;
end

Figure 3 Algorithmic skeleton of the proposed framework.

In compliance with the original cache simulator coding, the subsystems 2 through 4 are scripted in C and were compiled and tested using gcc-based commercial compiler.

The upper and lower bounds of the system power and time estimates were compared to closely published peer works [7, 8, and 9] for validation purposes. A moderate relative accuracy has been observed. The uniqueness of each the compared works can only allow apples-to-oranges comparisons.

IV. EXPERIMENTAL RESULTS

In an AMD-like modeled microprocessor, we have heuristically subjected the nominal threshold voltages of level-1 and level-2 cache transistors to 5% incremental changes. These changes are shown in Tables 1. The inter-steps have been chosen carefully to reduce leakage current through the cell without affecting the circuit functionality or upsetting the stored data. To architecturally showcase the impact on leakage dissipation, we employed the framework on the cache system of the processor. The validated leakage power and access time measurements in the two levels of hierarchies were also included in Tables 1.

Table 1 Threshold voltage various settings in NMOS and PMOS transistors of the SRAM cell and the associated leakage dissipations and access time.

Cache Level	V_{tn} (mV)	V_{tp} (mV)	Leakage Power (W)	Access Time (pS)
L-1	190.2	213.0	0.492	696.601
	192.1	215.1	0.471	696.609
	199.7	214.1	0.339	696.609
	209.2	234.3	0.325	696.609
	218.7	244.9	0.254	696.609
	228.2	255.6	0.204	696.609
	238.7	266.3	0.163	719.224
	247.3	276.9	0.124	719.228
L-2	190.2	213.0	1.848	1479.63
	192.1	215.1	1.768	1476.68
	199.7	214.1	1.498	1479.61
	209.2	234.3	1.219	1479.68
	218.7	244.9	0.951	1486.25
	228.2	255.6	0.762	1486.26
	237.8	266.3	0.610	1486.25
	247.3	276.9	0.488	1486.25

Results show that the static power dissipations in both levels of cache hierarchy have shown a pattern of asymptotical decreased while linear-like increases in the cell access time. The leakage seems to approach an optimal level at about 30% increase above the nominal threshold voltage. This trend can be observed in figures 4 and 5.

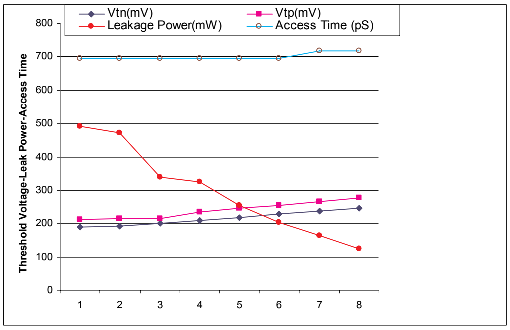

Figure 4 L-1 cache leakage dissipation and access time sensitivity to threshold voltage variation. X-axis represents the measurement number.

The results shed light on the effects of threshold voltage modulations on performance. Increasing the threshold voltage does show to increase cell access time which, in essence, slows its accessibility. This is due to the resulting weak current drive of the basic cell which marks a clear tradeoff between leakage savings and system speed.

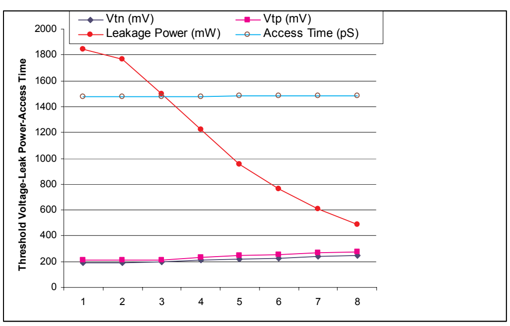

Figure 5 L-2 cache leakage dissipation and access time sensitivity to threshold voltage variation. X-axis represents the measurement number.

To quantify performance, the simplescalar [10] out-of-order simulator has been modified and configured to loosely mimic AMD Athlon X2 3800 [11], a commercially well-known contemporary processor; with level-1 64KB data cache, Level-1 64KB instruction cache, both host 32B block size, directly mapped and 1 cycle hit latency. The lower cache level is unified 512KB, 4-way, 64B block size and 6 cycle hit latency.

Table 2 Simulated processor configuration

Processor core	
Technology	70nm
RUU size	16 Instructions
Issue width	4 Instruction/cycle
Fetch width	4 Instruction/cycle
Decode width	4 Instruction/cycle
Commit width	4 Instruction/cycle
LSQ size	8 Instructions
Functional Units	4 Int ALUs,4 FP ALUs, 1 Multiplier/Divider
Cycle Time	100 cycles
Memory System	
L-1 Instruction Cache	64KB,32B block size,DM,WB
L-1 Data Cache	64KB,32B block size,DM,WB
L-2 Unified Cache	Unified,512KB, 64B block size,4-way,WB
Memory	100 cycles latency
TLB	128-entry,30 cycles Miss Penalty

The main memory latency is set to 100 cycles. The simulation cycle time is 1 µs. These characteristics are summarized in Table 2.
Miscellaneous integer and floating point benchmarks [12] have been elected randomly from several of applications.
Simulations were run for 1 billion instructions after fast-forwarding through the first few millions of instructions. The performance is measured in terms of instructions per cycle. The performance statistics are summarized in the bottom of Table-3.

According to all benchmarks, while increasing the threshold voltages in both types of transistors in the CMOS cells can yield substantial decrease in subthreshold leakage, an increase in the execution cycle per instruction demonstrates a clear performance overhead. The optimal performance degradations are normalized in figure 6.

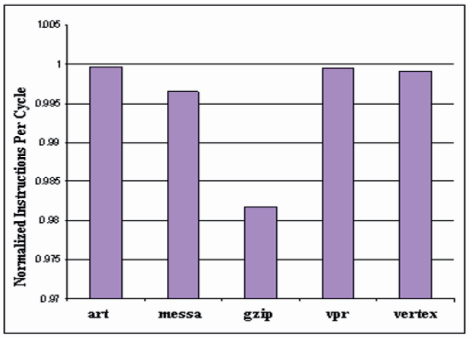

Figure 6 Normalized instructions per cycle measure with respect to unity baseline. The performance overhead, as reflected by different benchmarks, is due to threshold hike and corresponds to optimal leakage savings.

Results have also demonstrated a disparity in leakage power savings across the two levels of cache hierarchy. Approximately, up to three and four folds of magnitudes can potentially be saved in the first and the second levels respectively if the optimal performance were to sacrifice. This prospect is illustrated graphically in figure 7.

Apparently, unlike the upper level, the lower cache is dominated by static power due to L-1 near perfect hit rates. This strongly suggests a better overall performance-leakage edge, should only L-2 cache has been subject to threshold voltage boost.

When dynamic energy per cell access is factored in, level-1 has shown slight decrease with respect to the baseline.

The energy conserved by the effects of lower current drive of the high-threshold cells seems to mildly outweigh the extra energy drawn by the soaring switching activities of the instruction and data caches. That is not the case, however, with respect to the lower level: virtually no savings has been achieved.

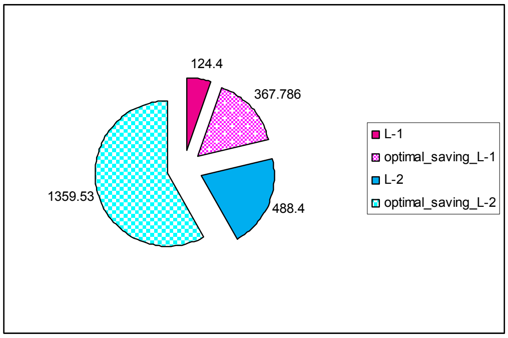

Figure 7 Optimal Leakage power savings (in mW) in cache system

Table 3 Summary of experimental results: threshold voltages, leakage power, dynamic energy statistics. The bottom of the table includes the execution cycle per instruction as yielded by SPEC2000 benchmarks.

%Vth		0	1	5	10	15	20	25	30
Leakage Power (mW)	L-1	492.186	470.848	339.00	324.70	254.13	203.64	163.02	130.45
	L-2	1847.75	1768.13	1497.90	1219.30	951.42	762.02	610.33	488.40
Dynamic Energy Per Access (pJ)	L-1	9.71	9.71	9.71	9.71	9.42	9.41	9.421	9.41
	L-2	69.23	69.23	69.22	69.22	69.22	69.22	69.22	69.22
CPI_art		0.3788	0.3788	0.3788	0.3788	0.3788	0.3789	0.3789	0.37889
CPI_messa		0.6647	0.6668	0.6668	0.6668	0.6668	0.6669	0.6669	0.6670
CPI_gzip		1.9662	2.0004	2.0008	2.0012	2.0016	2.0020	2.0024	2.0028
CPI_vpr		0.5825	0.5825	0.5825	0.5826	0.5826	0.5827	0.58267	0.58272
CPI_vertex		1.2963	1.2965	1.2967	1.2968	1.2970	1.2971	1.2974	1.2975

V. CONCLUSION

This work has proposed a framework that integrates the threshold voltage into an existing cache power estimator and employs it to study the effectiveness of using the threshold voltage to mitigate and optimize leakage power dissipation in static cache memories of an embedded microprocessor-based system. The resulting piece of firmware is also capable of providing the time statistics that correspond to the threshold adjustments. This allows the ability to study the associated implications on the host system performance. The tool can greatly helps computer architects to explore the use of the threshold voltage as a cheap measure to speculate and budget subthreshold leakage dissipation at the early stages of design cycle.

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A Multi-Agent System for Optimization of Object Selection in Relational Database

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Abstract- This paper describes a project designed to show how a multi-agent architecture can be used for optimization of object selection within cache memory, in relational database applications. A multi-agent system was designed using JACK^{TM} (Agent Oriented Software Group), integrated with a SQL Server relational database and interfaced it with a Java Builder application. The system was tested by triggering agent actions. The agents used in this multi-agent system are intelligent agents. They model reasoning behaviour according to the theoretical Belief Desire Intention (BDI) model of artificial intelligence. Intelligent agents compare LRU algorithm and three improved alternatives: LRFU, ARC and LRU-H, and choose the optimum algorithm for replacement of objects from cache. LRU, LRFU and ARC algorithms are well-known in the literature, and LRU-H algorithm [7] is a personal contribution, using a heuristic function for an optimum determination of the object to be removed from cache. The multi-agent system enables to obtain a significant decrease of the time needed to extract data from a database, using an object cache extraction.

Keywords: intelligent agents, multi-agent system, optimization, object cache, relational database

I. INTRODUCTION

Cache offers support for memory management in database applications, web servers, file systems, processors, operating systems; and it is faster than an auxiliary storage. Both memories, the cache and the auxiliary storage, handle uniformly sized items called pages. Requests for pages are first directed to the cache and, if the page is not in the cache, then to the auxiliary memory. In the latter case, a copy of the page is saved into the cache. If the cache is full, one of the pages in the cache must be eliminated. A replacement policy determines which page is evicted. LRU (Least Recently Used) is the policy of choice in cache management. LRU algorithm replaces the least recently used page from the cache.

This paper describes a multi-agent architecture, that can be used for optimization of object selection within cache memory, in relational database applications. The multi-agent system contains two intelligent agents (in the Jack application), which co-operate to find a method of optimization of object selection within cache memory, in the relational database application (Java Builder and SQL Server). An agent chooses an algorithm, taking into account every parameter that impacts performance, for a number of executions of the server application: average values of parameter H (hit ratio), average number of objects found in cache and the size of cache. It uses (about 92% from time) the LRU-H algorithm – the LRU improved algorithm, based on a heuristic function which finds the object to be replaced from cache. The heuristic function depends on one parameter and minimizes the sum among the number of references and the value of a Hash function, which is associated with the object. As a result, the heuristic function relocates this object to the last place within the list LRU-H. The time of the last reference of the object in cache and the number of the effective stored objects in cache shall be considered for choosing the Hash function.

The paper is organized as follows:

- the section *Replacement of Objects from Cache Problem* presents LRU algorithm and three improved alternatives of LRU algorithm, for creation of a cache of objects
- the section *The architecture of the multi-agent system* describes the types of applications used for optimization of object selection in relational database

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 376–380.

- the section *Belief-Desire-Intention Agents* shows the types of agents that compose the system, the responsibilities of each agent, and the communication between agent types
- the section *Evaluation of the multi-agent system* presents the interaction diagram method and tests of the program executions, for various size of cache.

II. Replacement of Objects from Cache Problem

For the optimization of the access of data of relational database can be used a cache for temporary storage of table consisting of frequently referenced and result of queries objects. The implementation of a cache of objects [1, 2, 3, 4] means issuance and administration of cache of stored objects and setting the strategies for selection of the elements to be removed to free space within cache.

Multi-agent system uses LRU algorithm and three improved alternatives of LRU algorithm, such as:

LRFU (Least Recently/Frequently Used) [5] combines LRU and LFU. It assigns a value $C(x) = 0$ to every page x and, depending on a parameter $\lambda > 0$, after t time units, updates $C(x) = 1 + 2^{-\lambda} C(x)$, if x is referenced and $C(x) = 2^{-\lambda} C(x)$ otherwise. LRFU replaces the page with the least C(x) value. If λ tends to 0, then C(x) tends to the number of changes of x and LRFU tends to LFU. If λ tends to 1, then LRFU becomes LRU. The performance depends on λ.

ARC (Adaptive Replacement Cache) [6] maintains two LRU lists of pages, L_1 and L_2 and the parameter p, $0 <= p <= c$. If the cache can hold c pages, L_1 contains p pages, which have been seen only once recently and L_2 contains c-p pages which have been seen at least twice recently. Initially, $L_1 = L_2 = \varphi$. If a requested page is in the cache, it is moved to the top of L_2, otherwise, it is placed at the top of L_1. L_1 is partitioned into T_1 (containing the most recent pages in L_1) and B_1 (containing the least recent pages in L_1), and L_2 is partitioned into T_2 and B_2.

LRU-H (Least Recently Used - Heuristic) [7] uses a heuristic function for optimizing the selection of object in cache, when it is necessary to remove an element. We have defined a heuristic function that determines within cache the object having parameter p=0 and that minimizes the expression consisting of the sum among the number of references and the value of a Hash function, which is associated with the object. As a result, the heuristic function relocates this object to the last place within the list LRU-H.

The implementation of a cache of objects problem can be viewed as a problem to find an optimum algorithm for replacement of objects from cache. Formally, an implementation of a cache of objects problem consists [8] of n variables $t_1, t_2, \ldots, t_n$, whose values are taken from finite domains D1, D2, . . ., Dn, respectively, and a set of constraints on their values. The following shall be considered:

Input data: a domain $D = ((D_1, D_2, \ldots, D_n)$, D_i – range of values, $i = 1, \ldots, n$, for $n \in N^*$; a relation $R(c_1, c_2, \ldots, c_n)$, c_i attribute, $c_i \in D_i$, $i = 1, \ldots, n$, c_1 - primary key of relation R; a cache object C - the set of tuples obtained after an interrogation upon R relation: $C = \{ (t_1, t_2, \ldots, t_n) / t_i \in D_i, i = 1, \ldots n\}$ and a Hash function defined as follows: $h: D_1 \rightarrow N^*, k \rightarrow h[k]$

Goal: find $t'_{i,p}$ $i \in \{2, \ldots, k\}$, $k = |C'|$, so that:

a) $t'_{i,p}$ is the least recently used element, where
   - t_{n+1} - a counter equal to the number of references of the tuple $t = (t_1, t_2, \ldots, t_n)$
   - $C' = \{ (t_1, t_2, \ldots, t_n, t_{n+1}, p) / (t_1, t_2, \ldots, t_n) \in C, t_{n+1}, p \in N^*\}$
   - $ti' = (t_1, t_2, \ldots, t_n, t_{n+1}, p) \in C' - \{ t'_1\}$
   - p parameter, $p \in N^*$, p assigned to t:
     Case 1: at addition of one tuple t* to the set C: p = 1
     Case 2: for $\forall t \in C - \{t^*\}$, $p = p - 1$ (if $p <> 0$).

b) $h(t'_{i,1})$ is minimal, where h - Hash function, depending on the time of the last object reference in cache and the number of objects really stored within cache:
   $h: D_1 \rightarrow N^*$, $h(t'_{i,1}) = (\text{current\_time} - \text{last\_reference\_time}(t'_{i,1})) \bmod k$, $k = |C'|$.

c) $t'_{i,p} = 0$ (parameter p for object $t'_i = 0$)

The constraints are the following**:** $t'_{i,p}$ is the least recently used element and it can be deleted - condition a); h - a Hash function for optimization, defined by the relation b) and the object $t'_{i,p}$ is not the first in cache – relation c)

The objective function determines the object to be removed from cache: $f : C' \rightarrow N^*$, $f(t'_i) = \min(t'_{i,n+1} + h(t'_{i,1}))$, $(\forall)$ $ti' = (t_1, t_2, \ldots, t_n, t_{n+1}, p) \in C' - \{ t'_1\}$, $t'_{i,n+1} = t_{n+1}$, $i \in \{2, \ldots, k\}$, $k = |C'|$, $t'_{i,p} = 0$, $t'_{i,1} \in D_1$ The objective function minimizes the sum among the number of references and the value of a Hash function, which is associated with the object

III. The Architecture Of The Multi-Agent System

The application for Optimization of Object Selection in Relational Database contains three layers:

- the client application created using JACK™ Developer Environment, includes the multi-agent system. The client sends a specified data object to the server (and receives the object back again) a specified number of times. In Figure 1 it is shown the architecture of the client-server application:
- the server application which is a Java Builder program that uses a SQL Server database. The server receives JACOB objects sent to it by a client, and sends the objects back to the client. The JACOB™ Object Modeller (JACOB) may be used to transport data between a JACK Agent and a Java class.
- data transfer procedure between JACK™ Intelligent Agents and Java Builder application uses JACOB Objects, with *.dat files and input / output ASCII stream. JACOB object structures contain objects that are defined in dictionary files, using the JACOB Data Definition Language. A dictionary file defines the structure and fields of one or more objects. Populated objects are sent and received between the processes via a socket. The Network protocol stack assigns to the socket the user-specified port number and host.

Jack application contains two BDI agents: Information Agent and Decision Agent, which use the following conceptual statements:

- *Events* (both internal and external) that the agent is prepared to handle
- *Plans* that the agent can execute
- *Events* the agent can post internally (to be handled by other plans)
- *Events* the agent can send externally to other agents
- The *capability* concept [9] is a means of structuring reasoning elements of agents into clusters that implement selected reasoning capabilities. This technique simplifies agent system design, allows code reuse, and encapsulation of agent functionality.
- Beliefset - a rational agent [10] has beliefs about the world and desires to satisfy, driving it to form intentions to act. An intention is a commitment to perform a plan.

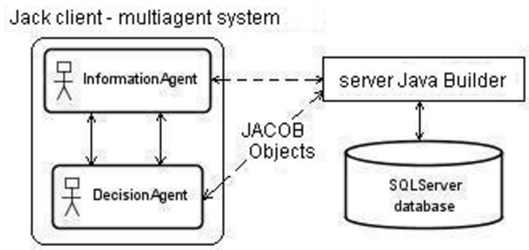

Fig. 1. The architecture of the client-server application

IV. BELIEF-DESIRE-INTENTION AGENTS

Beliefs, Desires and Intentions are called the mental attitudes (or mental states) of an agent. Rao and Georgeff [11] have described a computational model for a generic software system implementing a BDI agent. In reaction to an event, for instance a change in the environment or its own beliefs, a BDI agent adopts a plan as one of its intentions.

A step of a plan [12] can consist of adding a goal (that is, a desire to achieve a certain objective) to the agent itself, changing its beliefs, interacting with other agents, and any other atomic action on the agent's own state or the external world.

The multi-agent system created using JACK™ Developer Environment contains two BDI agents: Information Agent (Robot2) and Decision Agent (Robot1) which have the following capabilities:

- Information Agent (Robot2) which statistically determines the effectiveness of cache, by calculating a hit ratio H, a time of reading in cache T, and number of objects found in cache, for a number of executions of the server application. Hit ratio depends upon implementation of model of objects cache administration and means the percent of objects interrogations that the cache administrator is obtaining to. Information Agent handles external event "Information Request", sends event "ChooseAlgorithm" and it sends data to Decision Agent, using the plan "SendCommand". The figure 2 shows the design graph of Information Agent:

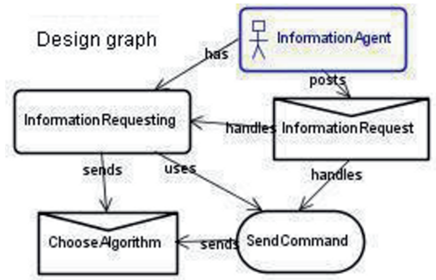

Fig. 2. Design: Information Agent

- Decision Agent (Robot1) receives data from Information Agent and chooses an optimal algorithm for replacement of objects from cache. This algorithm will be executed within the Server application (using Java Builder, with SQL Server database connection). Decision Agent uses plans: “Plan_LRU”, “Plan_LRFU”, “Plan_ARC”, and “Plan_LRUH”. Average values of parameter H, average number of objects found in cache and the size of cache, determine the effectiveness of implementation of objects cache. In addition, the execution of “Plan_LRFU” depends on parameter λ, 0 < λ < 1, and “Plan_LRUH” uses a parameter “heuristic”.

V. Evaluation of the Multi-Agent System

As an evaluation method of the multi-agent system we have chosen the interaction diagram method. The Agent Interaction Diagram is a useful tool both for analysing and debugging communication between agents. In Figure 3 it is presented an example of interaction diagram, which illustrates the communication process between agents and message flow.

The main benefits of the agent-based approach adopted for r*eplacement of objects from cache problem* are given by the analysis of the exchanged messages flow between agents with the interaction diagrams.

The program execution has been checked using tests applied for various size of cache. The results of tests enable to determine the average values of parameter H, average number of objects found in cache and the size of cache and suggests the effectiveness of implementation of objects cache, using „Plan_LRUH” due to:

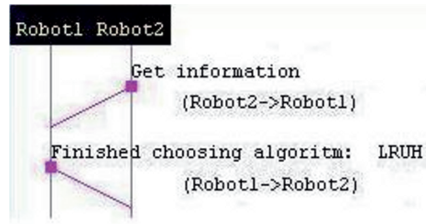

Fig. 3. Agent Interaction Diagram

- hit ratio for LRU-H algorithm is higher than for LRU, LRFU and ARC algorithms (table 1): LRU-H provides a hit ratio about 14.37% higher than LRU algorithm, 10.84% higher than LRFU algorithm and 7.37% higher than ARC algorithm
- LRU-H algorithm finds more objects within cache than LRU, LRFU and ARC algorithms (table 2).
- “Plan_LRUH” provides minimum values of the objective function.

TABLE 1.
Performing Characteristics Of Algorithms

cache size (number of objects)	hit ratio (H - %)			
	LRU	LRU-H	LRFU	ARC
10	14.79	30.31	17.77	22.18
13	15.77	30.49	19.41	23.1
15	20.19	32.42	23.82	28.16
17	20.97	34.91	24.28	28.48
20	22.85	38.3	26.97	29.51

TABLE 2.
Average Number Of Objects Found In Cache

cache size (number of objects)	LRU	LRU-H	LRFU	ARC
10	4.8	7.33	5.67	6.2
13	4.88	9.25	5.75	6.88
15	5.3	7.7	6.1	7.1
17	4.57	9.43	5.14	6.43
20	4.6	9.4	6.0	7.4

LRU-H algorithm provides a hit ratio about 14.37% higher than LRU algorithm, 10.84% higher than LRFU algorithm and 7.37% higher than ARC algorithm. The multi-agent system enables to obtain a significant decrease of the time needed to extract data from a database, using BDI agents.

VI. Conclusion

The paper presents the multi-agent system for optimization of object selection in relational database. We have described the architecture of a multi-agent system for r*eplacement of objects from cache problem*, and discussed about the BDI agents that compose the system. We can conclude that the multi-agent system uses an objective function for an optimum determination of the object to be removed from cache

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Shape Indexing and Retrieval: A Hybrid Approach Using Ontological Descriptions

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Abstract— **This paper presents a novel hybrid approach for visual information retrieval (VIR) that combines shape analysis of objects in image with their indexing by textual descriptions. The principal goal of presented technique is applying Two Segments Turning Function (2STF) proposed by authors for efficient invariant to spatial variations shape processing and implementation of semantic Web approaches for ontology-based user-oriented annotations of multimedia information. In the proposed approach the user's textual queries are converted to image features, which are used for images searching, indexing, interpretation, and retrieval. A decision about similarity between retrieved image and user's query is taken computing the shape convergence to 2STF combining it with matching the ontological annotations of objects in image and providing in this way automatic definition of the machine-understandable semantics. In order to evaluate the proposed approach the Image Retrieval by Ontological Description of Shapes system has been designed and tested using some standard image domains.**

Index Terms—**Image processing, ontological descriptions, shape indexing, visual information retrieval.**

I. Introduction

Actually, in well-known searching engines the seeking and retrieval of documents in Web is based on textual queries. The same approaches are used for retrieval of documents with images assuming that those images have textual annotations. In general, these approaches are not efficient for visual information because documents in Web frequently have not any textual descriptions. In order to overcome this problem the alternative techniques are used [1], [2]. They provide searching, retrieval, indexing, and classification of visual information on base of analysis of low-level image features, such as a color, texture or shape comparing them with characteristics of reference image or textual query [3], [4], [5]. Using a query that defines what user is looking for, the textual annotations are interpreted as image features. Thus, computing similarity between a query and images, which are candidates to be retrieved, is achieved on the level of image features. Unfortunately, this type of systems does not define understandable semantics associated with automatic indexing and classification of visual information [2].

There are some well-known visual information retrieval VIR systems which may be used as prototypes for development of novel content based retrieval techniques. One of them is Query by Image Content system QBIC, which provides retrieval of images, graphics, and video data from online collections using image features, such as a color, texture, and shape [6]. AMORE (Advanced Multimedia Oriented Retrieval Engine) and SQUID systems provide image retrieval from the Web using queries formed by keywords specifying similar images, sketches, and SQL predicates [7]. CIRES (Content Based Image REtrieval System) is another facility providing retrieval of features, such as a structure, color, and texture of image combining them with user specifications of importance of these features in a query [8]. Even though the contributions of these systems to field of VIR were important, they do not provide mechanisms to represent a meaning of objects and semantics of whole image. That is why, we propose to apply a novel approach for design of Content-Based Image Retrieval (CBIR) systems using the both an image semantics and its low-level features.

The possible applications of the proposed image retrieval approach are systems for supporting digital image processing services, high performance exchange of multimedia data in distributed collaborative and learning environments, medical and biological researches, digital libraries, and others where information is presented in visual form.

The paper organization is as it follows: after analysis of well-known approaches in CBIR area the proposed technique for shape analysis is discussed in section the II; the section III and IV present a novel technique in proposed approach using ontological description of the object in image and describe the block diagram of designed image retrieval IRONS system; finally, in section V and VI the description of testing the approach, experiments, obtained results, and evaluation of the system are presented.

II. Proposed Approach For Shape Analysis

The extraction of low-level image features is important step in CBIR systems. Among these features the most useful are color and shape. A color extraction is quite fast operation and permits to reduce a quantity of images to be analyzed during a searching process. A shape is more informational feature because it has a meaning by itself. In this research we consider a shape as geometrical pattern, including a set of points, curves, surfaces, solids, etc. [9]. The shape matching is considered as one of the most difficult aspects of CBIR due to a common shape needs a lot of parameters to be represented explicitly.

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A. Shape indexing by curve evolution and tangent space

The well known methods for global shape description, such as Elasticity correspondence [10], Curvature scale space approach [11], B-splines and chain case codes [12], Fourier and wavelets spectral descriptors [13], etc. sometimes are too complex for fast processing, and they are sensitive to spatial variations.

Usually, obtained shape is described as a closed polygon with a great number of vertices, that require a lot of time for their processing [14], [15]. In order to reduce a quantity of polygon vertices to a subset of vertices containing relevant information about the original outline, the discrete curve evolution process is used as simple and efficient approach. The shape complexity is evaluated by assigning a relevance measure to each vertex. The least important vertex may be removed. Once a vertex is removed, its neighboring vertices must be connected. This process is repeated until we obtain the desired simplification of a shape. The relevance measure K is defined as it follows.

$$K(S_1,S_2)=\frac{\beta(S_1,S_2)l(S_1)l(S_2)}{l(S_1)+l(S_2)}, \qquad (1)$$

where $\beta(S_1,S_2)$ is the normalized angle in radians between two segments S_1,S_2, and $l(S_1)$, $l(S_2)$ are the length functions for segments normalized with respect to the total length of the polygonal curve C. The lowest value of $K(S_1, S_2)$ corresponds to the least contribution of arc $S_1 \cup S_2$ to the curve C.

Fig. 1 shows the results of applying this technique that preserves main visual parts of a original polygonal curve and obviously the amount of information to be processed is decreased significantly.

However, the polygonal representation of a shape is not a convenient form for calculation how similar is that shape to another. An alternative representation, such as Tangent Space Representation (TSR) is used. Applying TSR the curve C is represented by a graph of the step function; the steps on x-axis represents the arc length of each segment in C, and the y-axis represents the turning angle between two consecutive segments in C.

Fig. 2 shows the example of TSR obtained from polygon simplified by curve evolution process. The TSR is sensitive to rotation, scaling, and translation because variation of segments size or their position produce completely different step functions.

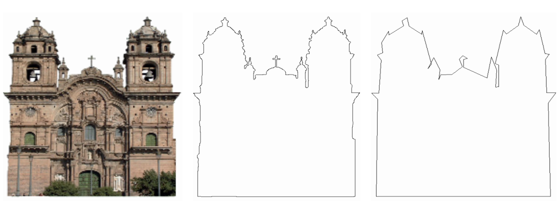

Fig. 1. The original image and results of curve evolution process with 3135 and 50 vertices respectively.

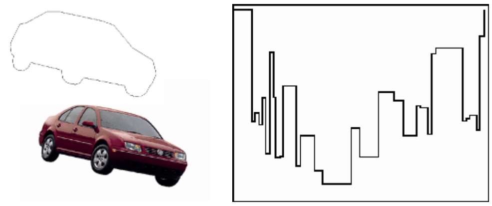

Fig. 2. The TSR of polygon corresponding to original image

B. Shapes matching with 2STF

In order to solve the principal problem of TSR sensitivity to spatial variations, we propose to compute a similarity between polygons using Two Segments Turning Function [12]. Our approach for representing a shape is related to proposal of Arkin in [16], where a step function calculated from a silhouette of object represents a polygon with some disadvantages regarding to invariant features. 2STF solves these problems by a simple strategy, which operates with turning angle between two consecutive segments of the curve C. In 2STF the steps on x - axis is defined by the normalized arc length of each segment, and the y - axis represents the turning angle between two consecutive segments in C. Fig. 3 shows the angle taking into account for construction of 2STF. The angle θ is defined by segment S_2 and the imaginary line that pass through the segment S_1. The measurement of this angle has an intuitive reason: the angle θ measures the rotation of the second segment with respect to the first one. It is clear that the angle values are in the interval $[-\pi, \pi]$. Thus, the proposed approach is invariant to rotation because it is built taking into account the relative angle between consecutive segments. That allows getting the same representation for a set of shapes even though they are placed in different positions or has been reflected or rotated. Additionally, 2STF uses the normalized length with respect to total perimeter of shape C. As consequence, this approach is invariant to scaling too. In Fig. 4 the 2STFs is presented for similar but not equal polygonal curves. The second image in Fig. 4 has been rotated and scaled. The both 2STFs have the similar characteristics.

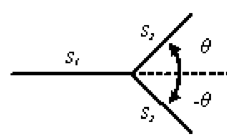

Fig. 3. Normalization of angle between segments S_1 and S_2

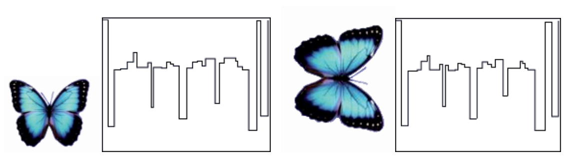

Fig. 4. The 2STFs of similar images with rotation and scaling

The matching between two polygons is obtained by comparing their 2STFs applying first of all the scaling factor *sf* and then computing the area between those functions. The scaling factor is found as $sf = \frac{l(C_1)}{l(C_2)}$, where $l(C_i)$ is the length of *i-th* curve and $l(C_1) > l(C_2)$. In Fig. 5 the shaded area shows difference between two 2TSFs.

In the proposed approach the feature vector described an image has two components: a color (used for reduction of a set of images to which shape analysis will be applied) and shape itself represented by corresponding 2TSF. In practice, the difference between two vectors is obtained by applying the Euclidian distance equation (2).

$$d_{EC}(f_1,f_2) = \sqrt{\Sigma(f_1 - f_2)^2} \qquad d_{ES}(f_1,f_2) = \sqrt{\Sigma(f_1 - f_2)^2} \quad (2)$$

The d_{EC} and d_{ES} are Euclidian distances for color and shape between two images f_1 and f_2 respectively. The color filter is implemented using analysis of image histogram. Even though a color is important feature in retrieval process, it has less impact than a shape. Thus, the results computed by (2) have a different relevance. Experimentally, the well acceptable results for total Euclidian distance d_T is that, when a shape relevance has a double value than a color:

$$d_T = 2d_{ES} + d_{EC} \quad (3)$$

The disadvantage of 2STF representation is significant time that it takes to find the best correspondence between two curves. We determined empirically that this time is about some seconds for polygons with tens segments processed on personal computer of 2GHz and RAM of 1GB [1]. This time may be reduced by decomposition of the curve *C* into groups *G* in order to obtain the same smaller number of arcs in both polygons. This process consists in grouping the consecutive largest convex or concave arcs forming groups of segments covering the whole curve *C*. The number of operations for definition of groups is $N_G = \left(2^{M_a}\right)^2$, where M_a is the number of largest arcs of the curve. Taking into account restrictions for grouping either concave or convex arcs the number of operations will be less, reducing at least in ten times the duration of matching procedure.

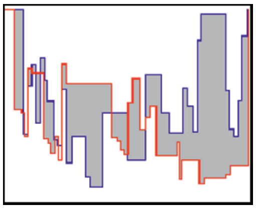

Fig. 5. Computing similarity between two 2STFs.

III. Ontology in Content Based VIR

The ontology as a part of content theory in artificial intelligence provides the identification of classes and properties of objects defining the relationship between these objects and logical concepts in a specific domain of knowledge [17]. Gruber in [18] defines the ontology as a formal explicit specification of a shared conceptualization, where the conceptualization means an abstract model of phenomenon with precise mathematical description. Applying it to VIR systems design, we identify some essential useful aspects of ontology, which permits to describe semantics that establishes a common and shared understanding of a domain and facilitates the implementation of user-oriented vocabulary of terms and their relationship with objects in image.

According to Guarino [19], an ontology as the formal distinction for information organization is defined as a subset of all possible options *AO(L)* in description of any concept and includes the set of language conceptualizations *C(L),* such as $C(L) \subset ONTOLOGY \subset AO(L)$. For VIR systems we use the language *L* as a vocabulary for ontological descriptions of shapes. The language conceptualization *C(L)* in our approach represents a set of image classes, which satisfy the user's description (query), and *AO(L)* is a set of all possible images in collection that may be queried by language *L*. Therefore ontology establishes the relationship between the object and its formal explicit definition on base of previous knowledge about features of particular domain of images.

The ontology is described by a directed acyclic graph; each node has a feature vector that represents the concept associated with that node. Concept inclusion is represented by the IS-A inter-relationship made by manual indexing of the preprocessed images. For example, particular elements of buildings, churches, etc. correspond to specific concepts of shapes defining these buildings, churches. If the query describes an object using this ontology, the system would recover shapes that contain windows, columns, façades, etc. even though, those images have not been labeled as geometric figures for the retrieved object. In order to support the ontology management, the Resource Description Framework (RDF) language has been used. It defines a syntactic convention and a simple data model to implement machine-readable semantics [20]. Using RDF it is possible to describe each Web resource with relation to its object-attributes-value based on metadata standard developed by the World Wide Web Consortium.

IV. Design of Image Retrieval IRONS System

In order to evaluate the proposed approach, the Image Retrieval by Ontological description of Shapes (IRONS) system has been designed and tested. Its block diagram is shown in Fig. 6. The input for the system may be an image, its shape, or a keyword, which describes the object in image to be retrieved. The retrieved images will be those which have a high degree of matching in color/shape and ontological annotations defining the content of an image in query [4].

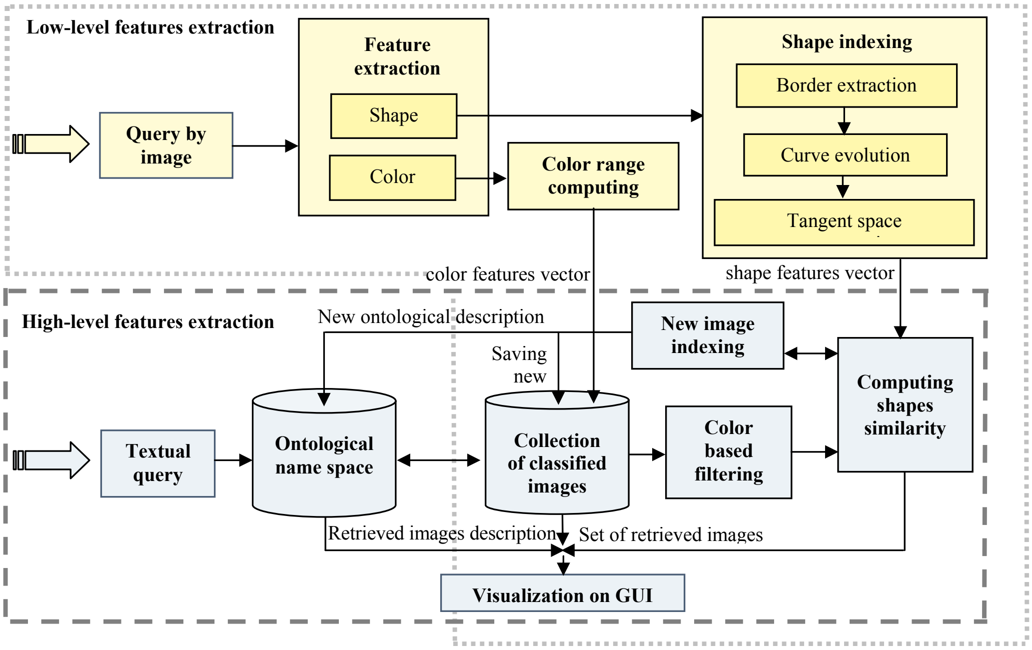

Fig. 6. IRONS system for shape indexing using the ontological descriptions

The system consists of the module of low-low feature extraction used for visual queries and high-level feature module for textual ones. The previously preprocessed images and their manually defined descriptions are organized in specific ontological structure. They are stored in collection of classified images and in ontological namespace respectively. In case of textual query the color/shape feature vector does not used. The retrieved images are those, which textual description is better matched with user's query.

The feature vectors of each node in the ontology name space consist of keywords linking the previously classified images to the characteristics of new shape represented by 2STF. In this way the meaning of an image may be obtained in textual form as a set of descriptions for each object related to a particular ontology. Additionally in IRONS system a user may define his proper user-oriented annotations for new input images or sketches adjusting the ontological name space according his perception of objects and scenes in image. The indexing and the ontology annotation processes may be quantified for computing of the final grade of similarity d_{FG} on base of color/shape and ontological description feature vectors using the Euclidian distance as it follows in (4)

$$d_{FG}(d_T, d_O) = \sqrt{\sum[(d_{T1} + d_{O1}) - (d_{T2} + d_{O2})]^2}\,, \qquad (4)$$

where d_{Ti} and d_{Oi} denote total color/shape and ontological description vectors of i-th image respectively. The sum $(d_{T1} + d_{O1})$ in (4) represents the set of two vectors of *1*-st image compared with *2*-nd one.

The values of the final grades of similarity for retrieved images are presented in GUI of IRONS system.

V. Experiments and Discussion

For test of system we divide experiments in two groups. First one is *Candidate Images Selected by Means of Structural Features experiment,* which consists in applying our shape retrieval approach over sets of previously classified images in collections (300 images are grouped into 15 classes of 20 images in each one). The purpose of this experiment is to observe how well the proposed approach is able to choose the images that belong to the same class corresponding to the particular query without applying ontology.

In Fig. 7 the results of the first experiment are presented on GUI of the IRONS system that retrieves images in downward order taking into account only color/shape features vector. That is why, in the retrieved set there are images from different classes, such as fruits, trees, cars, geometrical patterns, etc. Some of retrieved images have high grade of similarity in color/shape features but they are not correspond to input visual query "green apple", which appears as the first image in the GUI.

The second *Candidate Images Selected Using an Image Ontology experiment* consists in pruning the search three using both color/shape features and ontology. The application of ontology allows us to reduce the semantic gap problem retrieving relevant images only from the same class to which the particular query belongs. In Fig. 8 the retrieved images correspond to querying textual description "gothic church".

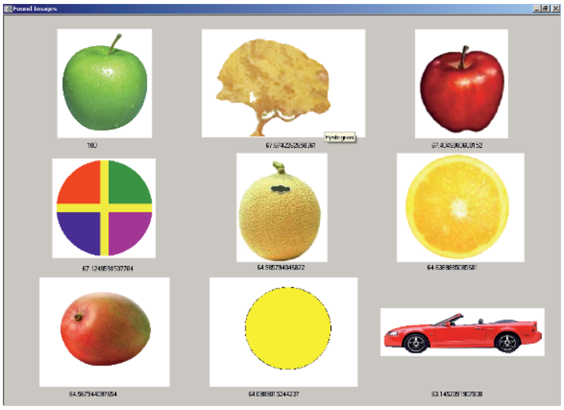

Fig. 7. The image retrieval without ontology

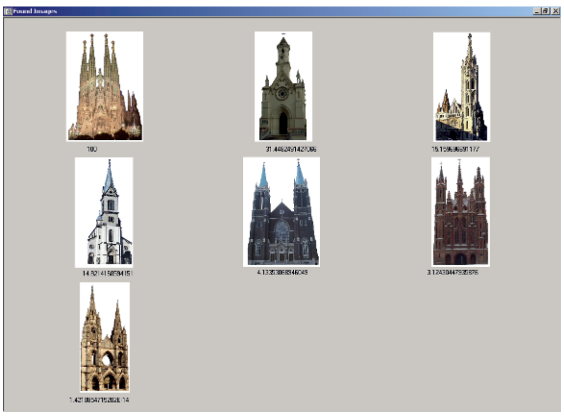

Fig. 8. The image retrieval with ontological descriptions

It possible to appreciate that the IRONS system retrieves images from class "Church" and does not retrieve images from other classes, for example, fruits, cars, animals, etc. reducing a set of irrelevant retrieved images and number of iteration needed to obtain a satisfactory result.

The evaluation of CBIR system is non-trivial task. This is because there is an amount of subjectivity involved into query interpretation by user. The similarity between a query and a set of retrieved image depends on individual perception of user. Nevertheless, there is a standard way of judging the obtained results. This technique consists in calculation of two metrics associated with VIR systems, such as Precision and Recall. Recall measures the ability of system to retrieve relevant information from all image collection and precision is the proportion of the number of relevant retrieved images to the total number of relevant ones in collection. They may be computed according the following equation

$$PRECISION = A/B, \quad RECALL = A/C, \tag{5}$$

where A is a set of relevant images retrieved by system, B is a set of relevant and irrelevant images retrieved by system for particular query, and C is a set of all relevant images in collection for particular query taking into account that $A=|B \cap C|$.

Fig. 9 a) and b) show the average recall and precision for the first experiment without ontology. The x-axis represents the number of tests with 20 images in each one; y-axis shows the average recall and precision computed according to (5). When proposed approach is applied over a small set of images, recall and precision values are low. When the number of images increases recall/precision also increases because there are more possibilities for applying the similarity metrics over a major number of shapes that belong to the image query's class. The average recall/precision values for this experiment lie in the intervals, such as (0.5÷0.75)/(0.28÷0.4) respectively.

In the second experiment extraction of the subset of images is provided by applying the textual descriptions, which permits to choose the images belonging to the same class as particular query using ontological name space improving the recall/precision as it shown in Fig. 9 c) and d). As a consequence, the average recall/precision have higher values which lie in the range of (0.7÷0.9)/(0.35÷0.55) respectively.

VI. Conclusion

The evaluation of the proposed approach and testing of the designed IRONS system show that 2TSF is quite acceptable for color/shape features indexing if there is not time restriction for image retrieval process. Satisfactory automatic retrieval of expected images is achieved faster due to the lower number of iterations in a searching process with ontology. The most important contribution of this research is proposed hybrid approach combining the advantages of low-level image features extraction with textual description of image semantics. The ontological annotations allow simple and fast estimation of meaning of a shape and of a whole image. The proposed image retrieval approach is robust to partial occlusion, rotation, and scaling of objects.

The disadvantages of the system are the presence of errors during spatial sampling and generation of the image feature vectors as well as the required amount of system memory. It is important to mention the restrictions for input visual queries, which must have small number of well-defined and separated objects. Additionally, significant occlusions between objects, week borders or complex background in image are not recommended in this application. The analysis of factors like tolerance to occlusion and deformation, robustness against noise, and feasibility of indexing are considered as possible extension of the proposed approach. From the obtained experimental results we can conclude that the approach could be considered as alternative way for the development of visual information retrieval facilities.

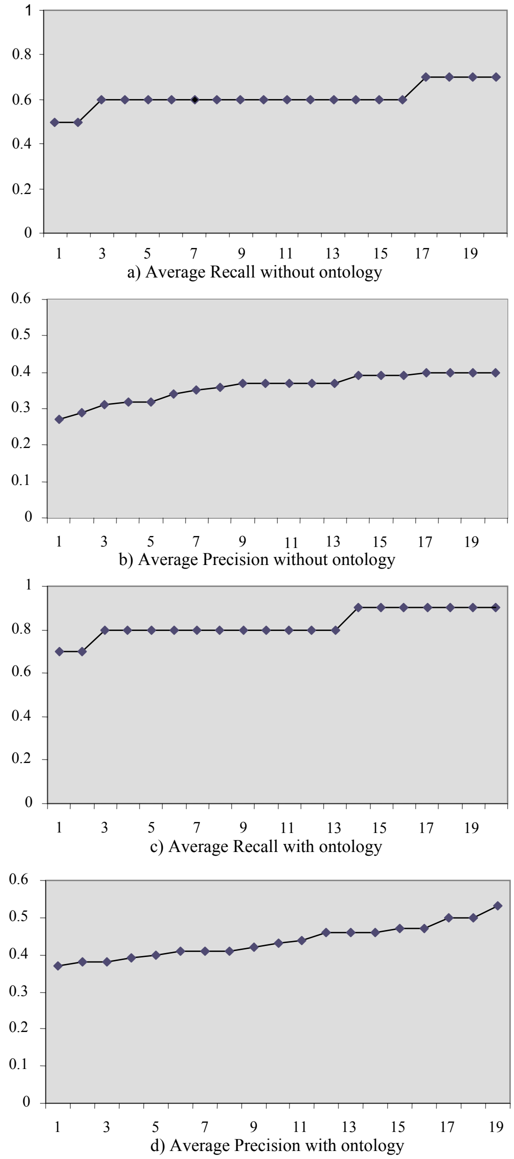

Fig. 9. Recall and Precision metrics of IRONS system

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A Block-Diagonal Dynamic Fuzzy Filter for Adaptive Noise Cancellation

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***Abstract*— This paper presents a dynamic fuzzy filter with internal feedback, for adaptive noise cancellation. The cancellation task is transformed to a system-identification problem, which is tackled by the use of the Dynamic Block-Diagonal Fuzzy Neural Network. The filter is a novel generalized Takagi-Sugeno-Kang fuzzy model, where the consequent parts of the fuzzy rules are Block-Diagonal Recurrent Neural Networks. In order to underline the effectiveness of the proposed fuzzy noise canceller, it is applied to a benchmark noise cancellation problem, and a comparative analysis with a series of other dynamic models is conducted.**

I. Introduction

A benchmark problem in the area of signal processing is the extraction of an information signal buried in noise [1]. Noise cancellation is encountered in many cases, including interference in electrocardiographs and periodic interference in speech signals. The most common method of signal estimation is to pass the noisy signal through a filter, which suppresses the noise and leaves the signal relatively unchanged. Adaptive filters attracted a lot of attention during the last decades and there exists a variety of filters in literature. They have the ability to adjust their parameters automatically, requiring little or no prior knowledge of the signal or noise characteristics.

Recently, neural networks and fuzzy systems have been established as effective tools for adaptive filtering, exhibiting promising results [2]-[4]. In all cases, however, the suggested structures are static and the series-parallel identification approach is followed. Therefore, these models provide insufficient signal estimations when noise passes through nonlinear dynamic channels.

In an attempt to alleviate this problem, a number of recurrent neural and fuzzy-neural models have been suggested as dynamic adaptive noise cancellers [5]-[7]. These models are capable of effectively model the dynamics of a channel and exhibit superior cancellation performance compared to the aforementioned static filters.

In this work an alternative recurrent fuzzy structure is proposed as a noise cancellation filter. The novelty of the proposed model lies in the consequent parts of the fuzzy rules, which are small block-diagonal recurrent neural networks [8], thus introducing dynamics to the overall network. The proposed model is entitled Dynamic Block-Diagonal Fuzzy Neural Network (DBD-FNN).

The rest of paper is organized as follows: In the next section the transformation of the noise cancellation problem to a system identification problem is given. In section III the proposed dynamic fuzzy model is described. The simulation results are hosted in section IV, where a comparative analysis with other recurrent neural and fuzzy models is, also, conducted. In the last section conclusions are drawn.

II. Transformation of the Noise Cancellation Problem to a System Identification Problem

According to [1], a typical structure of a noise cancellation system is shown in Fig. 1. Additive noise, *n(k)*, corrupts the information signal, *s(k)*, resulting in the noisy signal, *d(k)*. *d(k)* and *s(k)* are assumed to be uncorrelated.

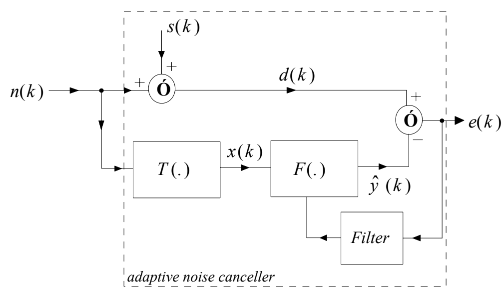

Fig. 1. The problem of adaptive noise cancellation

The principle of noise cancellation is based on the assumption that both the noisy signal, and a filtered or distorted measurement of the noise, named reference noise *x(k)*, are available. Noise *n(k)* is considered to pass through a channel with a transfer function *T*(.). Under the assumption that the inverse of the filter noise distortion can be estimated, the noise corrupting the signal can be identified and cancelled. In this perspective, the problem of noise cancellation can be transformed to a system identification problem [9], as shown in Fig. 2.

Let *F*(.) denote the transfer function of the system. It is derived from Fig. 2:

$$\hat{y}(k) = F(x(k)) = n(k) = \hat{T}^{-1}(x(k)) \qquad (1)$$

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and

$$e(k) = d(k) - \hat{y}(k) = s(k) + n(k) - \hat{y}(k) \rightarrow s(k) \qquad (2)$$

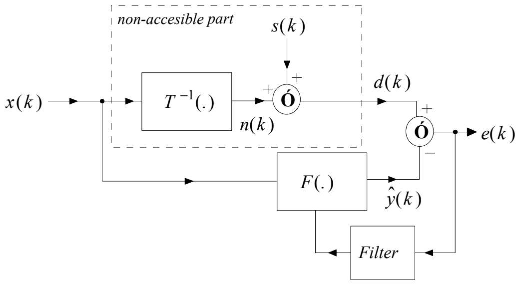

Fig. 2. Adaptive noise cancellation as a system identification problem

Let *x(k)*, *d(k)* be considered as the desired input and output, respectively, of the system *F*(.). According to (1) and (2), the error *e(k)* will correspond to the information signal, which can be regarded as noise, additive to the output of the system $\hat{y}(k)$, as shown in Fig. 2. It is derived from (2):

$$e^2(k) = (d(k) - y(k))^2 \Rightarrow E\{e^2(k)\} = E\{s^2(k)\} + E\{(n(k) - \hat{y}(k))^2\} + 2E\{s(k) \cdot n(k)\} - 2E\{s(k) \cdot \hat{y}(k)\} \qquad (3)$$

The information signal is statistically uncorrelated to the noise $n(k)$ and its estimate $\hat{y}(k)$. Therefore the last two terms of (3) are equal to zero and (3) becomes

$$E\{e^2(k)\} = E\{s^2(k)\} + E\{(n(k) - \hat{y}(k))^2\} \qquad (4)$$

Applying an optimization method, the parameters of the adaptive filter should be adjusted such that an error measure is minimized. Since the power of the information signal remains unchanged, minimizing the error measure leads to minimization of $E\{(n(k) - \hat{y}(k))^2\}$ and, according to (4), to minimization of the quantity $E\{(e(k) - s(k))^2\}$. Thus, minimization of the total output power of the adaptive model leads to the optimal mean squared estimate of the information signal.

III. The Dynamic Block-Diagonal Fuzzy Neural Network

The suggested dynamic block-diagonal fuzzy neural network (DBD-FNN) comprises *r* Takagi-Sugeno-Kang rules [10] of the following form:

$$IF\ \boldsymbol{u}(k)\ is\ \mathrm{A}^{(l)}\ \ THEN\ \ g_l(k) = BDRNN_l\left(\boldsymbol{u}(k)\right),\ l = 1, \ldots, r \qquad (5)$$

where $\mathrm{A}^{(l)}$ is the fuzzy region in the premise part and the sub-model $BDRNN_l$ is a block-diagonal recurrent neural network that implements the consequent part of the *l-th* rule. For the sake of simplicity, a multiple–input – single–output model is used. Based on the structural characteristics of the TSK model, it can be divided into three major parts: the premise, the consequent and the defuzzification part.

At each time instant k, the premise part is fed with the process variables $u_1(k), \ldots, u_m(k)$, which are used for defining the fuzzy operating regions. The firing strengths of the rules are calculated by the following static function:

$$\mu_l(k) = \prod_{j=1}^{m} \exp\left[-\frac{1}{2} \cdot \frac{(u_j(k) - m_{lj}(k))^2}{(\sigma_{lj}(k))^2}\right], \quad l = 1, \ldots, r \qquad (2)$$

where $\boldsymbol{m}_l = \{m_{lj}, j = 1, \ldots, m\}$ and $\boldsymbol{\sigma}_l = \{\sigma_{lj}, j = 1, \ldots, m\}$, $l = 1, \ldots, r$, are the premise part parameters.

The consequent parts of the model are dynamic, including the *r* sub-models of the rules. Each sub-model $BDRNN_l$ is a block-diagonal recurrent neural network, which is a two-layer network, with the output layer being static and the hidden layer being dynamic. The hidden layer consists of pairs of neurons (blocks); there are feedback connections between the neurons of each pair, introducing dynamics to the network. Therefore, the overall model is a locally recurrent globally feedforward fuzzy neural network [11]. The general representation of a MISO dynamic block-diagonal fuzzy neural network is shown in Fig. 3:

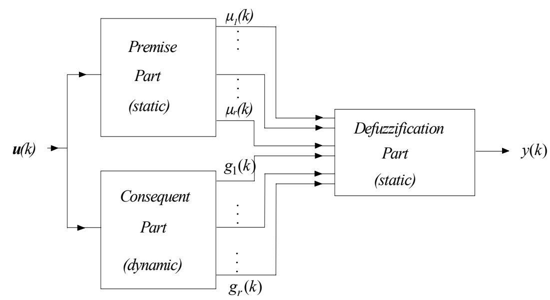

Fig. 3. General representation of the DBD-FNN

The operation of the BDRNN with *m* inputs, *r* outputs and *N* neurons at the hidden layer is described by the following set of state equations:

$$x_{2i-1}^{(l)}(k) = f_a\left[\sum_{j=1}^{m} b_{2i-1,j}^{(l)} \cdot u_j(k) + w_{1,i}^{(l)} \cdot x_{2i-1}^{(l)}(k-1) + w_{2,i}^{(l)} \cdot x_{2i}^{(l)}(k-1)\right] \qquad (6a)$$

$$x_{2i}^{(l)}(k) = f_a\left[\sum_{j=1}^{m} b_{2i,j}^{(l)} \cdot u_j(k) - w_{2,i}^{(l)} \cdot x_{2i-1}^{(l)}(k-1) + w_{1,i}^{(l)} \cdot x_{2i}^{(l)}(k-1)\right] \qquad (6b)$$

$$g_j(k) = f_b\left[\sum_{j=1}^{N} c_{lj} \cdot x_j^{(l)}(k)\right] = f_b\left[\sum_{j=1}^{N/2} c_{l,2j-1} \cdot x_{2j-1}^{(l)}(k) + \sum_{j=1}^{N/2} c_{l,2j} \cdot x_{2j}^{(l)}(k)\right] \qquad (7)$$

$l = 1, \ldots, r$, $i = 1, \ldots, N/2$

where

- f_a and f_b are the neuron activation functions of the hidden and the output layer, respectively. In the following, the activation functions are both chosen to be the sigmoid function $f(z) = \dfrac{1-e^{-a \cdot z}}{1+e^{-a \cdot z}}$.
- $\boldsymbol{x}^{(l)}(k) = [x_i^{(l)}(k)]$ is a N-element vector, comprising the outputs of the hidden layer of the *l-th* fuzzy rule. In particular, $x_i^{(l)}(k)$ is the output of the *i-th* hidden neuron at time *k*.
- $B = \left[b_{ij}^{(l)}\right]$ and $C = \left[c_{lj}\right]$ are $r \times N \times m$ and $r \times N$ input and output weight matrices, respectively.
- $w_{1,i}^{(l)}, w_{2,i}^{(l)}$ are the rules' feedback weights, that form the block diagonal feedback matrices, $W^{(l)}$. The *scaled orthogonal* form is employed in this work, [8], where the feedback matrices are described by the following formula:

$$W^{(l)} = \begin{bmatrix} w_{1,i}^{(l)} & w_{2,i}^{(l)} \\ -w_{2,i}^{(l)} & w_{1,i}^{(l)} \end{bmatrix} \quad i = 1, 2, \ldots, N/2 \tag{8}$$

The output of the model at time *k*, *y(k)*, is determined using the weighted average defuzzification method:

$$y(k) = \frac{\sum_{l=1}^{r} \mu_l(k) \cdot g_l(k)}{\sum_{l=1}^{r} \mu_l(k)} \tag{9}$$

The configuration of the proposed BDRNN consequent part is presented in Fig. 4, where, for the sake of simplicity, a single–input–single–output BDRNN with two blocks of neurons is shown.

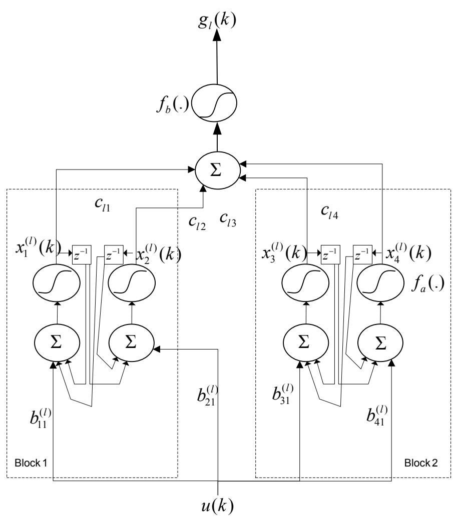

Fig. 4. Configuration of the consequent part of the fuzzy rules

The DBD-FNN is trained by use of the Dynamic Fuzzy-Neural constrained Optimization Method (D-FUNCOM) [6], a training algorithm for batch-wise learning of recurrent fuzzy systems with internal feedback. Only minor modifications are made, such that the method takes into consideration the special features of the DBD-FNN. The method is selected in order to provide a learning framework that is common to all cancellers, which will participate in the comparative analysis hosted in the following chapter. Since the scope of the paper is to highlight the model's operation as a canceller rather than the selected training algorithm, a detailed overview of D-FUNCOM can be found in [6].

IV. Results and Discussion

In this section, the aforementioned DBD-FNN filter is applied to a noise cancellation problem, where the noise $n(k)$ passes through a nonlinear dynamic channel, producing the reference noise $x(k)$. The passage's dynamics is simulated by a second order nonlinear auto-regressive model with exogenous inputs (NARX) [9]:

$$\begin{aligned} x(k) = {} & 0.25x(k-1) + 0.1x(k-2) + 0.5n(k-1) + 0.1n(k-2) \\ & -0.2n(k-3) + 0.1n^2(k-2) + 0.08n(k-2)x(k-1) \end{aligned} \tag{10}$$

The information signal *s(k)* is a saw-tooth signal of unit magnitude, 50 samples period long, as shown in Fig. 5. The signal *s(k)* is corrupted by a uniformly distributed white noise sequence varying in the range [-2,2] shown in Fig. 6, while the noise-corrupted signal *d(k)* is depicted in Fig. 7. The training data set consists of 12000 pairs [*x(k),d(k)*], while the testing set comprises 1000 data pairs. It should be noted that, due to the recurrent property of DBD-FNN, the model is trained in a parallel mode, therefore only the current value of the reference noise is directly used as the sole input to the adaptive filters. Accordingly, formulation of time-delay lines, necessary for static network structures, is avoided, with the channel mapping learned by means of the internal dynamics being introduced in the filter.

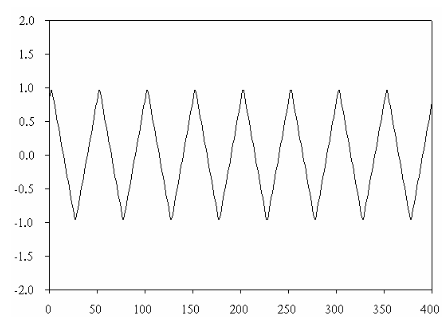

Fig. 5. Information signal *s*(*k*)

A time section of eight periods of the recovered signal is presented in Fig. 8. It is evident that, even though the information signal has half the amplitude of the additive noise, the former is accurately identified, with the exception of a few high frequency components.

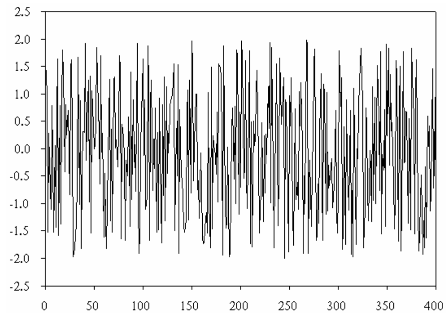

Fig. 6. Additive noise $s(k)$

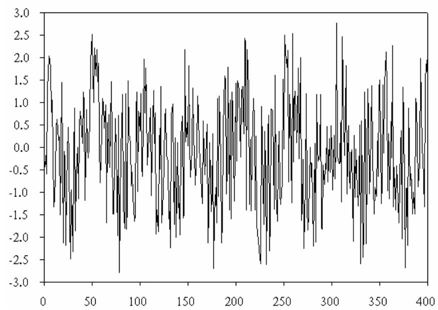

Fig. 7. Noise corrupted signal $d(k)$

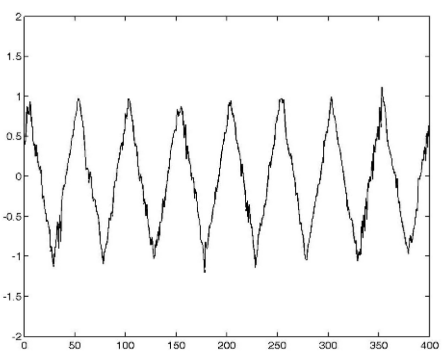

Fig. 8. Recovered signal

Several DBD-FNNs with different structural characteristics are examined and various combinations of the learning parameters are tested. Selection of the model and the parameter combination is based on the criteria of *(a)* effective identification of the plant's behavior, i.e. effective suppression of noise, and *(b)* moderate complexity of the resulting model. The selected structural and learning characteristics are given in Tables I and II, respectively. Training lasts for 1000 epochs. The input space is normalized to [-1,1] and is uniformly partitioned to fuzzy membership functions. The final membership functions are shown in Fig. 9. It can be noticed that the final model efficiently covers the input space, while the membership functions do not overlap considerably, thus preserving the local modeling approach of the Takagi-Sugeno-Kang fuzzy systems.

TABLE I
CHARACTERISTICS OF THE DBD-FNN STRUCTURE

Number of rules	3
Number of blocks per rule	4
Coefficient of the sigmoid function, *a*	2
Overlapping coefficient between initial membership functions, [0,1]	0.6

TABLE II
D-FUNCOM LEARNING PARAMETERS

Premise part				
n^+	n^+	Δ_{min}	Δ_0	ξ
1.05	0.6	1E-4	0.03	0.9
Consequent part				
n^+	n^+	Δ_{min}	Δ_0	ξ
1.1	0.6	1E-4	0.2	0.9

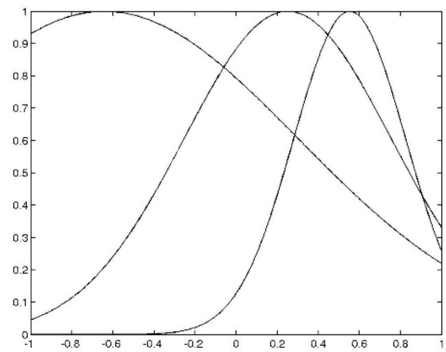

Fig. 9. Final membership functions

In the sequel, a comparative analysis is attempted between the suggested DBD-FNN filter and a representative class of competing noise cancellation filters, including:

(i) the Dynamic Fuzzy Neural Network (DFNN), taken from [6]. It is a recurrent fuzzy model with internal feedback in the consequent parts of the fuzzy rules, resembling the proposed model but with a significantly more complicated structure.

(ii) the Block-Diagonal Recurrent Neural Network, taken from [7]. It can be considered as the equivalent neural version of DBD-FNN, that performs global modeling.

(iii) A three-layer finite impulse response neural network (FIRN), in the form of 1-*H*-1, having a linear input layer, and FIR synapses at the hidden and output layers, as described in [6]. The outputs of the hidden and output layers are given by the following formulas:

$$O_i^1(k) = \tanh\left(\sum_{q=0}^{Ou}\left[w_{iq}^1\, u(k-q)\right] + w_i^3\right),\; i = 1,\dots,H \tag{11}$$

$$y(k) = \tanh\left(\sum_{j=1}^{H}\sum_{q=0}^{Oy}\left[w_{jq}^4\, O_j^1(k-q)\right] + w^6\right) \tag{12}$$

(iv) A three-layer infinite impulse response neural network (IIRN), in the form of 1-*H*-1, having a linear input layer, and Frasconi-Gori-Soda [12] neurons in the hidden and output layers. The outputs of these neurons are determined by the following formulas:

$$O_i^1(k) = \tanh\left(\sum_{q=0}^{O_u} w_{iq}^1 u(k-q) + \sum_{j=1}^{Oy_1} w_{ij}^2 O_i^1(k-j) + w_i^3\right),\quad i = 1,\dots,H \tag{13}$$

$$y(k) = \tanh\left(\sum_{j=1}^{H}\sum_{q=0}^{Oy_3} w_{jq}^4 O_j^1(k-q) + \sum_{j=1}^{Oy_2} w_j^5 y(k-j) + w^6\right) \tag{14}$$

The performance analysis for the DFNN and the BDRNN is taken from [6] and [7], respectively. For the rest of the competing models exhaustive experimentation has been carried out in order to extract the most efficient structure, which is going to participate in the comparative analysis. Since the FIRN can be regarded as an IIRN without feedback, the D-FUNCOM [6] algorithm is chosen to be the training method for these neural models. Moreover, BDRNN is trained using the RENNCOM algorithm, a modification of D-FUNCOM for neural schemes that is based on the same notion of constrained optimization. Thus, the structures of the suggested filter and those of the fuzzy and the neural networks are evaluated using a similar learning method, since our concern in this paper is to investigate the performance of different models rather than focusing on the learning attributes of the training algorithm.

The structural and learning characteristics of the competing filters are given in Table 3, while Table 4 hosts the simulation results. Training lasts for 1000 epochs for all the comparing filters and the Mean Squared Error is selected as the performance criterion.

Based on the results cited in Table IV, it becomes evident that the dynamic nature of the channel is clearly reflected to the results, since the FIRN neural network fails to sufficiently track the passage dynamics, $T^{-1}(\cdot)$. Moreover, the proposed filter exhibits superior performance compared to IIRN and FIRN, and a slightly ameliorated performance compared to DFNN and BDRNN, requiring less parameters than its competing rivals. Therefore, it can be argued that the suggested dynamic model constitutes an effective noise cancellation tool, having a simpler structure than the DFNN, while maintaining the advantage of interpretability of the TSK fuzzy models over its neural contestants.

V. Conclusion

A novel recurrent fuzzy filter, with a relatively simple structure, is proposed for the task of adaptive noise cancellation. The filter model is based on the classic TSK fuzzy model, with the consequent parts of the fuzzy rules consisting of small block-diagonal recurrent neural networks. This results to the dynamic block-diagonal fuzzy neural network (DBD-FNN).

The proposed filter has been applied to a noise cancellation problem, where the noise passes though a nonlinear dynamic channel. For the analysis purpose, the cancellation problem has been transformed to a system identification problem. The DBD-FNN's performance on the problem is compared with a series of competing recurrent neural and fuzzy filters, underlining the effectiveness of the proposed noise canceller. It is concluded that the suggested dynamic model constitutes an effective noise cancellation tool with a structure simpler than that of its contestants.

TABLE III
CHARACTERISTICS OF THE COMPARING FILTERS

Model	*Learning method*	*Models' characteristics*				
BDRNN	RENNCOM	Blocks=8	Coefficient of sigmoid function, *a*=2			
DFNN	D-FUNCOM	*H*=4	*Ou*=2	Oy_1=1	Oy_2=2	Oy_3=1
FIRN	D-FUNCOM	*H*=12	*Ou*=2		*Oy*=2	
IIRN	D-FUNCOM	*H*=12	*Ou*=2	Oy_1=1	Oy_2=2	Oy_3=1

TABLE IV
COMPARATIVE ANALYSIS

Model	*MSE training data set*	*MSE testing data set*	*Parameters*
DBD-FNN	0.01279	0.01226	60
BDRNN	0.01290	0.01240	64
DFNN	0.01360	0.01310	62
IIRN	0.01570	0.01690	87
FIRN	0.06170	0.06180	85

ACKNOWLEDGMENT

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GRASP: A Greedy Reconfigurable Approach for Shortest Path

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***Abstract*—This paper presents GRASP ("Greedy Reconfigurable Approach for Shortest Path"), a new shortest path algorithm using reconfigurable logic. It has time complexity O(P), where 'P' is maximum of number of edges along the shortest paths from source to other nodes. It is a modification of Bellman-Ford algorithm and is highly parallel and scalable. Unlike most other shortest path algorithms, GRASP does not need to find the minimum of nodes/adjacent nodes. Hence its FPGA implementation is faster compared to other FPGA implementations. Preliminary experimental results show that a 17-node GRASP runs about 4.7 times faster compared to parallel Bellman-Ford algorithm on Xilinx Virtex II.**

I. INTRODUCTION

Shortest path (SP) problem in graphs is still an active area of research[4], due to the demands for faster SP algorithms by applications like CAD for VLSI[8], robotics[6] and computer networks[3][5]. SP algorithms which run on instruction set based processors like Dijkstra's[1] algorithm and others[4][7], iterate hundreds of instructions and are sequential in nature, and hence have high computation time. Reconfigurable logic based approaches have been used in the past [3][9] to accelerate SP algorithms. But they are slowed down by the process of finding minimum of nodes/adjacent nodes.

Reconfigurable computing[2] achieves high performance by spatially spreading computation on hardware instead of iterating hundreds of instructions on a processor. Reconfigurable computing has execution time close to ASICs with flexibility to reconfigure. It can be used to efficiently and effectively mimic "natural" solutions: an implementation that replicates the way nature tackles analogous problems.

This paper presents GRASP ("Greedy Reconfigurable Approach for Shortest Path"), a new SP algorithm using reconfigurable logic. It has time complexity O(P), where P is the maximum of number of edges along the shortest paths from source to other nodes. It avoids finding minimum of nodes/adjacent nodes and hence is faster compared to other approaches. It is a modification of Bellman-Ford algorithm. In Bellman-Ford algorithm, for every node 'i', distance X_i from source is set as $Min(X_j + D_{ij})$, where X_j is adjacent node's distance from source and D_{ij} is distance between 'i' and 'j'. It is possible to update X_i of all nodes (i =1 to N) in parallel and we call this as parallel Bellman-Ford(PBF) algorithm. In GRASP, a daisy chain is used to to select first value of $X_j + D_{ij}$, which is less than X_i, in a greedy way and thus avoids finding $Min(X_j + D_{ij})$. Use of daisy chain makes it to operate at a much higher clock frequency compared to PBF and hence is much faster compared to PBF. GRASP has more iterations compared to PBF, still it is faster than PBF. GRASP assumes undirected graphs and positive integer edge weights.

The rest of the paper is organized as follows. Section 2 explains related work for finding SP. GRASP and its implementation details are given in Sections 3 and 4 respectively. Section 5 presents experimental results and compares GRASP with other approaches. Section 6 suggests future extensions and the paper concludes with Section 7.

II. RELATED WORK

A. Shortest path algorithms

Dijkstra's algorithm [1] is a popular SP algorithm and is $O(N^2)$. Let x_i be the current distance of node 'i' from source and D be the adjacency matrix. When there is no edge between nodes 'i' and 'j', D_{ij} is set to large value to indicate infinity. Let 's' be the source and 'd' be the destination. Dijkstra's algorithm is as shown in Figure 1.

1. Initialize x_j to D_{sj}, where j =1 to N, and $j \neq s$, set x_s as 0.
   Add 's' to set of labeled nodes F={s}
2. Find minimum among x_j
   x_i=Min(x_j) for j= 1 to N and node 'j' not in F
3. Add node 'i' to F and update each node's distance using
   x_j=Min(x_j , $x_i + D_{ij}$) where j=1 to N
4. Repeat steps 2 and 3 until destination is reached

Figure 1: Dijkstra's Algorithm.

Dijkstra's algorithm has been improved using efficient data structures like radix heap and two level radix heap[7], and have time complexities O(M+NlogC) and O(M+NlogC/loglogC), where C is edge weight, M is number of edges and N is number of nodes. A recent improvement[4] has time complexity $O(M+D_{max}\log(N!))$, where D_{max} is maximal number of edges incident at a vertex. Implementation of Dijkstra's algorithm on reconfigurable logic is presented in [3] and this uses a comparator tree to find minimum instead of using a loop. But the algorithm has to repeat steps 2 and 3(Figure 1) until destination is reached and hence its time complexity is O(N). Ralf Moller[6] has reformulated Dijkstra's algorithm and implemented that using the concept of signal propagation, and has time complexity O(L).

Bellman-Ford algorithm is as shown in Figure 2 and has time complexity $O(N^3)$. Here, distance of a node is updated using the relation $X_i = Min(X_j + D_{ij})$ and this is continued till there are no changes in X_i. The parallel implementation of Bellman-Ford algorithm updates all node values in parallel and has time complexity O(P). For the example graph in Figure 3 trace is as shown in Figures

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 393–398.

4 to 7. Nodes which are at infinity are not shown in Figures.

1. Initialize x_j to D_{sj}, where j =1 to N, and $j \neq s$, set x_s as 0.
2. Update each node position by the relation $x_i = \text{Min}(x_j + D_{ij})$ for j = 1 to N and $j \neq i$
3. Repeat step 2 till there are no changes.

Figure 2: Bellman-Ford Algorithm.

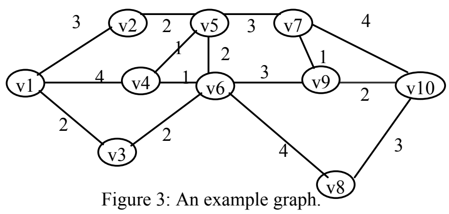

Figure 3: An example graph.

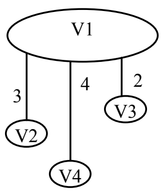

Figure 4: After first clock cycle.

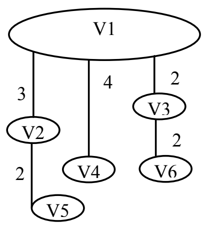

Figure 5: After second clock cycle.

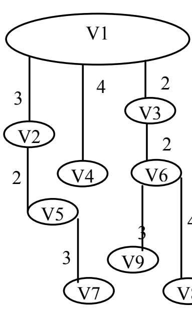

Figure 6: After third clock cycle.

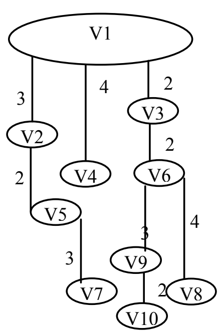

Figure 7: At the end.

B. *Problems and opportunities*

Most of the existing algorithms find minimum of nodes/adjacent nodes and are sequential in nature (select nodes one by one), and hence have high computation time.

Finding minimum can be avoided by using a daisy chain to select a better value for a node. This approach is faster compared to the approach which finds minimum. In Parallel Bellman-Ford(PBF) algorithm, nodes that are having same number of edges along the shortest path from source get fixed in parallel overcoming the sequential selection of nodes. In GRASP also nodes get fixed in parallel fashion.

III. GRASP

In GRASP, N nodes synchronously update their X_i using a daisy chain. Each node 'i', finds $X_j + D_{ij}$ for all adjacent nodes 'j', in parallel. It scans adjacent nodes from 1 to N to find a X_j+D_{ij} that is less than its current X_i using daisy chain and the first value thus found is set as new X_i and corresponding 'j' is set as previous node of 'i'. We associate with each node a flag O_i and is set to 0 in the beginning of the clock cycle, and is later set to 1 whenever there is a change in X_i in that clock cycle. If 'i' does not find any $X_j + D_{ij}$ as less than current X_i, then old position is retained and O_i remains as 0. This process of updating X_i needs adjacent node distances X_j and D_{ij}'s from adjacency matrix. Given this information, a node can update its value independently and this makes GRASP scalable. The algorithm stops when there are no changes in any X_i or, in other words when all O_i are 0.

For graph in Figure 3, let v1 be the source and v10 be the destination. The trace of GRASP is as shown in Figures 8 to 13 and the algorithm is as shown in Figure 14. Initially X_i of v1 is 0 and all others will have 99999(indicating infinity and nodes which are at infinity are not shown in Figures). At first clock cycle, v2 finds $X_1 + D_{21}$ (i.e. 0+3) as less than X_2 (i.e.99999) and hence sets X_2 to 3. Similarly v3 and v4 set their X_i to 2 and 4 respectively. In next clock cycle, v5 and v6 set their X_i to 5 and 4 respectively. In third clock cycle v7, v8 and v9 set their X_i to 8, 8 and 7 respectively. In fourth clock cycle, v10 sets its X_i to 12 through v7, as through v7 it finds X_i as 12 lesser than 99999. At this time though through v9 it is possible to set v10's X_i to 9, it selects v7, as it is seen first. In next clock it sets X_i to 11 through v8 and at the end it sets X_i to 9 through v9 and stops.

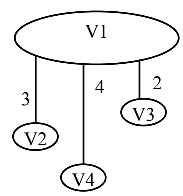

Figure 8: After first clock cycle.

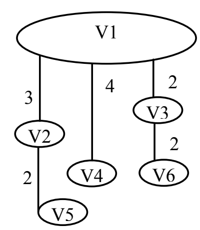

Figure 9: After second clock cycle.

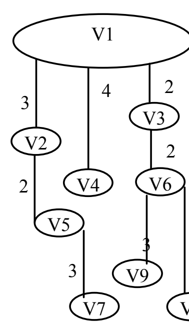

Figure 10: After third clock cycle.

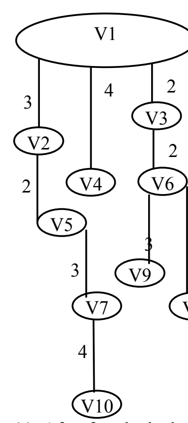

Figure 11: After fourth clock cycle.

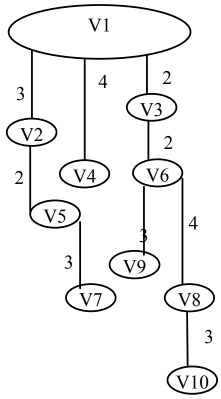

Figure 12: After fifth clock cycle.

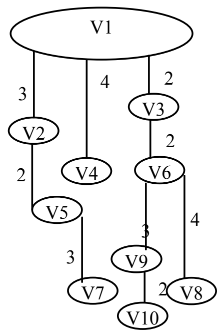

Figure 13: At the end.

```
1. For all non source nodes set Xj as 99999 (indicates
   infinite) and for source set this as 0. Set all flags Oi to 1

2. while any Oi is 1 loop
       for i=1 to N do parallely
         Oi =0
         for j=1 to N do parallely
            temp(j)=Xj + Dij
         end for loop
         flag=1
         j=1
         while(flag=1 and j<= N) loop
           if(temp(j) < Xi )then
              Xi =temp(j)
              previous_node(i)=j
              flag=0
              Oi =1
           end if
           j=j+1
         end while loop
       end for loop
   end while loop
```

Figure 14: GRASP algorithm.

A. *Trace Back*

Trace back is done to get nodes along the shortest path. The pseudo code is as shown in Figure 15 and in that, nodes along the shortest path are stored in sp_nodes. Trace back starts from destination and moves to wards source using previous_node values, and stops on reaching source. For the example graph in Figure 3, let v1 be the source and v10 be the destination. Starting from v10, trace back is done as v10 to v9, v9 to v6, v6 to v3 and v3 to v1, as shown in Figures 16 to19.

```
temp_node=destination
i=1
while(temp_node ≠ source) loop
   sp_nodes(i)=temp_node
   i=i+1
   temp_node=previous_node(temp_node)
end while loop
sp_nodes(i)=temp_node
```

Figure 15: Trace back pseudo code.

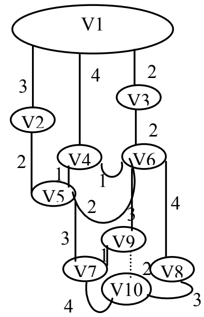

Figure 16: Tracing from 10 to 9.

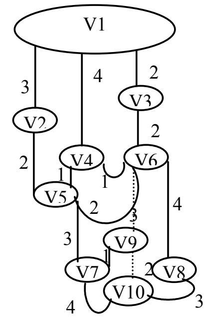

Figure 17: Tracing from 9 to 6.

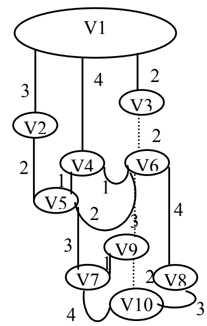

Figure 18: Tracing from 6 to 3.

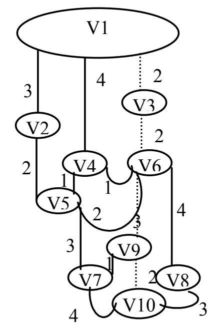

Figure 19: Tracing from 3 to 1.

IV. IMPLEMENTATION

GRASP is implemented using Xilinx[10] Virtex II development kit with XC2VP30 FPGA and is coded in VHDL. Chipscope pro 8.2i is used to check the results. As shown in Figure 20, GRASP consists of 'N' nodes moving synchronously with clock. Node takes information about adjacent nodes as input, and updates its position(X) and previous_node(PV) to new values. X and PV are declared as signals(integer array) and O is declared as std logic vector of size N. D is declared as a two dimensional constant array. A signal by name "load" is used for initialization. Initially load is 0 and initialization is done as said in the algorithm, and after initialization load is set to '1'. In the subsequent clocks(when load='1') nodes update their X_i as described in the algorithm and comes to halt when there are no more changes in X_i. Flag O_i at the beginning of clock is set to 0 and if X_i changes, flag O_i is set to 1. If all O_i are 0, it implies there is no change in the system and system can halt.

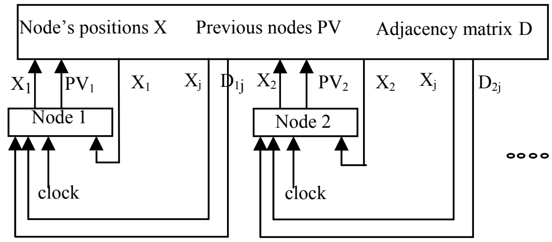

Figure 20: GRASP, as a collection of nodes.

A. *Node functionality*

Node structure is as shown in Figure 21. Adjacent node distance X_j is added to D_{ij} to get $X_j + D_{ij}$, which is then compared with X_i and flag C_j is set to 1 if $X_i<X_j+D_{ij}$ else C_j is set to 0. Similar adder and comparator pairs are kept in parallel for other adjacent nodes. Select signal(SS) is ANDed with C_1 to get F_1. F_1 is used to check whether $X_i<X_1+D_{i1}$ is true, if so X_i is set as X_1+D_{i1} and 1 is set as previous node of 'i'. F_1 is inverted and ANDed with SS, which is forwarded as SS for second adjacent node. If F_1 is 1, it deasserts SS to 0 for all other nodes. If F_1 is 0, then SS to node 2 will be 1. Thus first 'j' which satisfies $X_i<X_j+D_{ij}$ becomes previous node of 'i' and X_i is set to X_j+D_{ij}. O_i is the inverted value of select signal SS propagated through all adjacent nodes, taken at last adjacent node (not shown in Figure 21 due to less space).

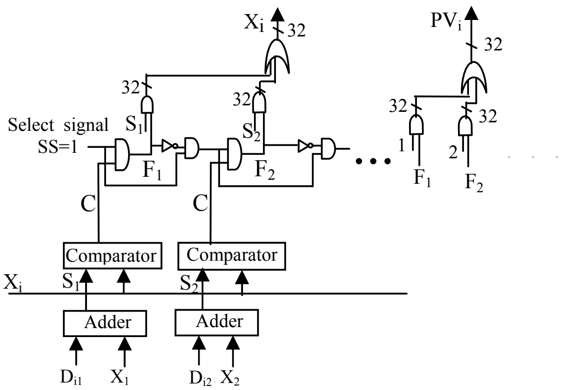

Figure 21: Node structure of GRASP.

B. *Scalability*

GRASP handles graphs that are larger than an FPGA's capacity. This section explains how the computation to be done in a clock cycle is divided into smaller steps and how these steps are executed sequentially on the FPGA.

Figure 22 shows the logical interface between host computer and FPGA. If FPGA can accommodate only 'k' nodes(where 1<=k<= N), then computation to be done in an iteration (clock cycle) is divided into N/k steps and in each step, computation of 'k' nodes is done. To start computation for an iteration, all X_i are loaded into FPGA memory, as for computation of a node, adjacent node's information is needed. X_i of first 'k' nodes and first 'k' rows of D are sent to FPGA and computation is done. Though in the beginning X_i are loaded in to FPGA, for 'k' nodes' computation, these values are sent again. This is because node structure is same for all nodes, except current node's X_i and D_{ij} s'. So these values are loaded each time computation is done. After the computation, results from FPGA are stored in temporary storage in host computer, as previous iteration values(X_i) should be used. Computation of all nodes is to be completed by taking 'k' nodes at a time. Once computation for an iteration is over, values held in temporary storage are updated as new values. Before starting next iteration once again X_i of all nodes are loaded to FPGA. With this execution time becomes O(PN/k).

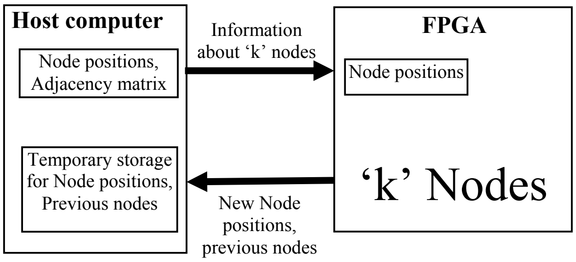

Figure 22: Interface between FPGA and host computer for 'k nodes computation.

Suppose we want to find shortest path for graph in Figure 3 and FPGA has resources to accommodate only 4 nodes, in that case computation to be done in an iteration is to be divided in to 3 steps with nodes 1,2,3,4 in step1, nodes 5,6,7,8 in step2 and nodes 9, 10 in step3 as shown in Figure 23. Before starting computation for an iteration, all X_i are to be loaded to FPGA, as they are needed for computation of all nodes. Load time is not proportional to value shown in Figure 23 and there will be small load delay before each step to load 'k' node's information.

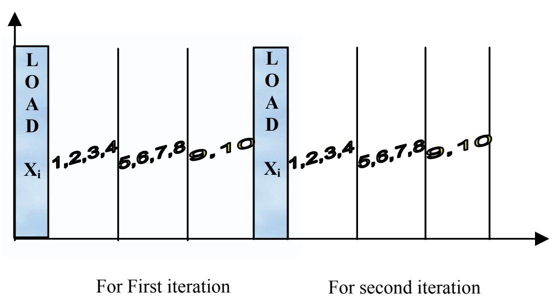

Figure 23: Computation of '4' nodes in a clock cycle.

Results at each step should be stored in temporary storage in host and after computation is done for all 10 nodes in the iteration, results stored at temporary storage should be updated as new values.

V. RESULTS

Experiment is conducted for graphs of 5, 10 and 17 nodes by placing all nodes in FPGA. Results for different values of N are as shown in Table I. It clearly shows that though GRASP takes a few extra clock cycles, it is faster compared to parallel Bellman-Ford algorithm due to high operating clock frequency. In parallel Bellman-Ford algorithm operating clock frequency is low compared to GRASP, due to finding minimum. A graph is plotted for execution time of GRASP and parallel Bellman-Ford algorithm as shown in Figure 24.

GRASP is also implemented using C(on Linux Fedora Core 4) on Pentium IV processor running at 2.66GHz and having 1 GB RAM. Results are given in Table II and a graph is also plotted for execution time(Figure 25) which clearly shows FPGA implementation is good.

TABLE I. RESULTS OF GRASP AND PARALLEL BELLMAN-FORD ALGORITHM

	GRASP			**Parallel Bellman-Ford**		
Number of nodes	5	10	17	5	10	17
Max. clock frequency in MHz	123.07	102.42	89.43	55.28	30.81	18.99
LUTs used %	8	31	82	7	28	75
Gate Count	18,841	65,071	1,71,424	17,621	60,254	1,59,043
Clock cycles	2	6	12	2	4	8
Execution time in ns	16.25	58.58	234.82	36.17	129.82	421.27
Actual execution results on Xilinx Virtex II Pro kit.						
Clock Frequency	100	100	50	50	25	16.66
Execution time in ns	20	60	420	40	160	480

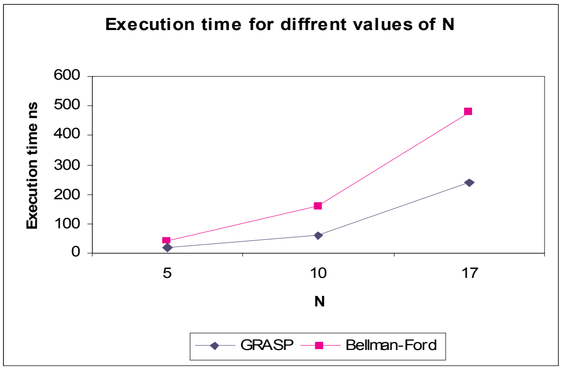

Figure 24: Execution time for GRASP and parallel Bellman-Ford algorithm.

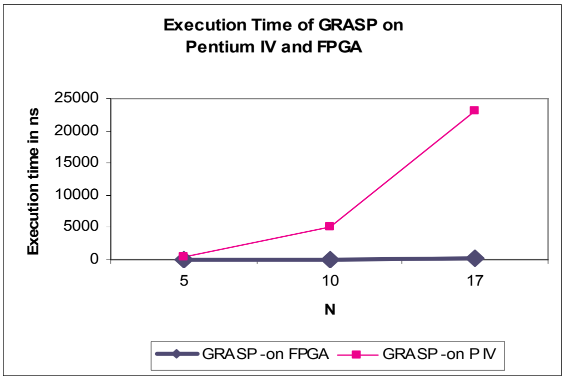

Figure 25: Execution time for Software(C) and FPGA implementation of GRASP.

TABLE II. EXECUTION TIMES IN NS FOR SOFTWARE (C) AND FPGA IMPLEMENTATIONS OF GRASP

N	5	10	17
GRASP in FPGA	20	60	240
GRASP in C	500	5000	23000
Speedup C/FPGA	25	83	95

A simulation of GRASP is done to see increase in number of clock cycles when compared to parallel Bellman-Ford algorithm. The results are as shown in Table III, which shows that worst case is 10 times the Bellman-Ford algorithm. For 17 nodes, GRASP has clock frequency 4.7 times that of parallel Bellman-Ford approach and this ratio increases with increase in N. Hence GRASP is efficient.

Among the existing approaches to find shortest paths, the best approach is parallel Bellman-Ford algorithm, as its time complexity is O(P). But the parallel Bellman-Ford algorithm has the disadvantage of finding minimum among the adjacent nodes(Min[X_j+D_{ij}]). Dijkstra's algorithm has a time complexity (N^2) and has disadvantages of finding minimum among the nodes (i.e. Min[x_j, x_i+D_{ij}]) and sequential selection of nodes, which makes it slower. Reconfigurable implementation of Dijkstra's algorithm[3] has time complexity O(N) and uses finding minimum, and hence is slower. Ralf Mollers's[6] approach always takes L clock cycles, where L is shortest path length and is not scalable, as all nodes need to be in the hardware for computation. GRASP has time complexity O(P), does not find minimum of nodes/adjacent nodes and is implemented on reconfigurable logic. Hence GRASP runs at higher clock speed and has low execution time, and is better than existing algorithms.

TABLE III. NUMBER OF ITERATIONS IN BELLMAN-FORD AND GRASP

Data range	**1-10**		**1-100**		**1-1000**	
N	Bellman -Ford	GRASP	Bellman -Ford	GRASP	Bellman-Ford	GRASP
10	4	7	5	9	7	7
25	4	9	8	13	6	17
50	4	11	8	16	9	17
100	3	13	8	24	15	34
200	3	14	7	25	11	33
500	3	15	6	28	14	39
1000	3	11	5	34	11	52
2000	3	12	4	36	9	58
5000	3	14	4	33	7	59
10000	3	17	4	40	6	65

Each node consists of N adders and comparators, and for N nodes, N^2 adders and comparators are needed. Hence resources needed is $O(N^2)$. If number of adjacent nodes(maximum is say Z) is smaller compared to N, then we can allocate resources for Z adjacent nodes at each node and hence resources needed gets reduced to O(N). This applies for both GRASP and Bellman-Ford algorithm.

Memory required is $O(N^2)$ because of storing D. By storing only required Z D_{ij}'s for each node, memory needed is 2ZN (Z for D_{ij} s + Z for 'j'). For node position and previous node memory needed is 2N(N for each). So total storage is 2NZ + 2N, which is O(N).

VI. FUTURE EXTENSIONS

GRASP actually gives single source all pair shortest paths. By having N such systems in parallel with each system having one node as source, we can find all pair shortest paths. This has time complexity O(P). Another approach is to use single GRASP system sequentially N times, with each time one node as source. With this we can get all pair shortest paths in O(NP) time. A similar approach can be used for finding minimum spanning tree(i.e. replace finding minimum by daisy chain).

VII. CONCLUSION

GRASP is a new shortest path algorithm using reconfigurable logic and has time complexity O(P), where 'P' is maximum of number of edges along the shortest paths from source to other nodes. It is based on Bellman-Ford algorithm and is highly parallel and scalable. It avoids finding minimum by using daisy chain and hence it runs at much higher operating clock frequency compared to other reconfigurable based approaches, and hence is faster. GRASP is 4.7 times faster compared to parallel Bellman-Ford algorithm for a 17 node graph. FPGA implementation of GRASP is 95 times faster than its C implementation.

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NATR: A New Algorithm for Tracing Routes

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***Abstract*— This paper presents NATR ("New Algorithm for Tracing Routes"), a new shortest path algorithm using reconfigurable logic and has time complexity O(L), where L is shortest path length. It uses ball and string model and is highly parallel and scalable. Unlike most other shortest path algorithms, NATR does not need to find the minimum of nodes/adjacent nodes. Hence its FPGA implementation is faster compared to other FPGA implementations. Preliminary experimental results show that a 17-node NATR runs about 6.3 times faster compared to parallel Bellman-Ford algorithm on Xilinx Virtex II.**

I. Introduction

Shortest path (SP) problem in graphs is still an active area of research[4], due to the demands for faster SP algorithms by applications like CAD for VLSI[8], robotics[6] and computer networks[3][5]. SP algorithms which run on instruction set based processors, like Dijkstra's[1] algorithm and others[4][7], iterate hundreds of instructions and are sequential in nature, and hence have high computation time. Reconfigurable logic based approaches have been used in the past [3][9] to accelerate SP algorithms. But they are slowed down by the process of finding minimum of nodes/adjacent nodes.

Reconfigurable computing[2] achieves high performance by spatially spreading computation on hardware instead of iterating hundreds of instructions on a processor. Reconfigurable computing has execution time close to ASICs with flexibility to reconfigure. It can be used to efficiently and effectively mimic "natural" solutions: an implementation that replicates the way nature tackles analogous problems.

This paper presents NATR ("New Algorithm for Tracing Routes"), a new SP algorithm using reconfigurable logic and has time complexity O(L), where L is shortest path length. It avoids finding minimum of nodes/adjacent nodes and hence is faster compared to other approaches. It mimics the formation of ball and string model[5]. In NATR, nodes fall down synchronously from the source by comparing their position with positions of adjacent nodes, and stop at shortest distance from source. Given adjacent node's information, a node can move independently and this makes NATR scalable. NATR is intended for large graphs in which L < N, where N is number of nodes. NATR assumes undirected graphs and positive integer edge weights.

The rest of the paper is organized as follows. Section 2 explains related work for finding SP, the ball and string model and its formation. NATR and its implementation details are given in Sections 3 and 4 respectively. In Section 5, we implement parallel Bellman-Ford algorithm, as it takes 'P' clock cycles to find shortest path, where 'P' is maximum of number of edges along the shortest paths from source to other nodes. Section 6 presents experimental results and compares NATR with other approaches. Section 7 suggests future extensions and the paper concludes with Section 8.

II. Related work

A. *Shortest path algorithms*

Dijkstra's algorithm[1] is a popular SP algorithm and has time complexity $O(N^2)$. Let x_i be the current distance of node 'i' from source and D be the adjacency matrix. When there is no edge between nodes 'i' and 'j', D_{ij} is set to large value to indicate infinity. Let 's' the be source and 'd' be the destination. Dijkstra's algorithm is as shown in Figure 1.

1. Initialize x_j to D_{sj}, where j =1 to N, and $j \neq s$, set x_s as 0.
   Add 's' to set of labeled nodes F={s}
2. Find minimum among x_j
   x_i=Min(x_j) for j= 1 to N and node 'j' not in F
3. Add node 'i' to F and update each node's distance using
   x_j=Min(x_j , x_i + D_{ij}) where j=1 to N
4. Repeat steps 2 and 3 until destination is reached

Figure 1: Dijkstra's algorithm.

Dijkstra's algorithm has been improved using efficient data structures like radix heap and two level radix heap[7], and have time complexities O(M+NlogC) and O(M+NlogC/loglogC), where C is edge weight, M is number of edges and N is number of nodes. A recent improvement[4] has time complexity $O(M+D_{max}\log(N!))$, where D_{max} is maximal number of edges incident at a vertex. Implementation of Dijkstra's algorithm on reconfigurable logic is presented in [3] and this uses a comparator tree to find minimum instead of using a loop. But the algorithm has to repeat steps 2 and 3(Figure 1) until destination is reached and hence its time complexity is O(N). Ralf Moller[6] has reformulated Dijkstra's algorithm and implemented that using the concept of signal propagation, and has time complexity O(L).

B. *Ball and String model(BSM)*

Ball and string model[5] of a graph is a network of balls connected by strings, where balls and strings represent nodes and edges respectively. For graph in Figure 2, Figure 3 shows equivalent BSM. In BSM, a straight line is a fully stretched string and a curve is a string with slack.

To illustrate formation of BSM from the graph, let v1 be the source. Assume all balls(nodes) are together as shown in Figure 4 and are at a distance of 0 from source. Source is fixed, which is shown by hatching. A fixed ball cannot move down. Keeping source fixed, when other balls are released, they fall down as shown in Figures 5 to 7. Figure 5 shows positions of balls after they fall down by unit distance and at this point no string is stretched to full. Figure 6 shows positions of balls after they fall down by distance of 2. Now, the string between v1 and v3 is stretched to full. Hence v3 cannot fall beyond 2 and gets fixed at 2. After falling by a distance of 3, balls are as

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 399–405.

shown in Figure 7 and at the end we get BSM as shown in Figure 3. In BSM, all strings along the shortest path are stretched to full and other paths will have one or more slacks.

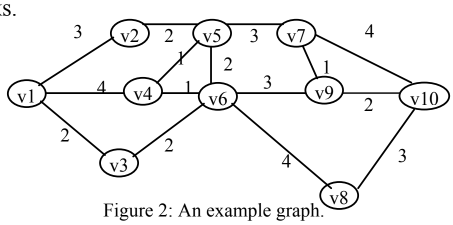

Figure 2: An example graph.

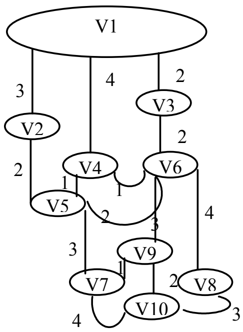

Figure 3: Ball and String model for the graph of Figure 2.

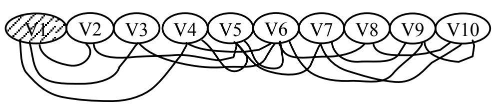

Figure 4: Initially all balls are together and v1 is fixed.

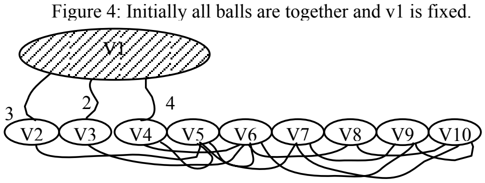

Figure 5: Balls after moving by distance of 1.

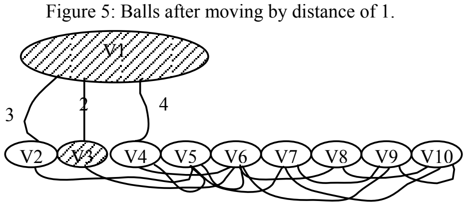

Figure 6: Balls after moving by distance of 2, v3 is fixed.

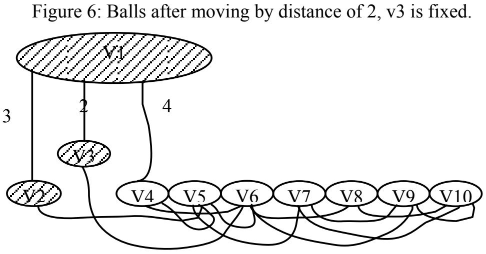

Figure 7: Balls after moving by distance of 3, v2 is fixed.

In [5], BSM is used to rebuild shortest path tree(SPT), whenever SPT gets disturbed due to changes in edge weights. This problem is represented as a linear programming problem and is solved to get new SPT. NATR is a simple approach to find shortest path and it uses reconfigurable logic, and hence is much faster compared to [5].

C. Problems and opportunities

Most of the existing algorithms are sequential in nature(select nodes one by one) and find minimum of nodes/adjacent nodes, and hence have high computation time.

In BSM, strings will have lengths equal to corresponding edge weights in graph and hence a closer node will have shorter length and gets fixed first, thus eliminating the need for finding minimum. In addition, during the formation of BSM, nodes that are at the same distance from source get fixed in parallel(like, v4 and v6), which overcomes sequential selection of nodes.

III. NATR

NATR mimics the formation of BSM to find shortest path. In NATR all nodes fall down from source synchronously. They fall under the constraint of not breaking any strings and stop at shortest distance from source. To fall down, a node needs information about adjacent nodes and this consists of adjacent node position, its status(whether node is fixed or movable) and weights on edges connecting the adjacent nodes(from adjacency matrix, D). Given this information, a node can fall down independently and this makes NATR scalable. Each node consists as its information; node position, status flag, step_size and previous_node. Initially for all nodes, position is set to 0 and step_size is set to 1. For all non source nodes flag is set to 0 and for source it is set to 1. Logic behind a node's move is as said below.

- Find next position(new X_i) of node 'i' by adding its position value and step_size.
- Find the actual distance($Dist_j$) from each of its adjacent nodes using $Dist_j = X_i - X_j$ where $1 <= j <= N$.
- Compare D_{ij} with $Dist_j$ set flags C_j and E_j indicating less and equal respectively.
- If any of C_j is 1, then move is failure(as string connecting node 'i' and 'j' breaks) and old position is retained.
- If none of the C_j's are 1 then move is successful and position is set to new position found in step1.
- On a successful move, if any E_j is set to 1(string is stretched to full) with corresponding O_j is set to 1, then node 'i' gets fixed through 'j' and 'j' is set as previous_node of 'i' and status flag O_i is set to 1.

Here initially step_size is set to 1. Later at each clock cycle, if move is successful step_size is multiplied by 'k', where 'k' is acceleration factor. Whenever a move is failure, if step_size >=k then it is divided by 'k' else step_size is set to 1. Step_size is varied to accelerate node's move. NATR with k=1, takes exactly L clock cycles and hence its time complexity is O(L). For k=1, the moves are as shown in Figures 4 to 7, and at the end it looks as shown in Figure 3. With k > 1, NATR takes less than L clock cycles to find shortest path, for large edge weights. But for small edge weights it takes more than L clock cycles due to excessive failed attempts. So, for small edge weights (<5) k=1 will be efficient. Figures 8 to 10 show the moves in NATR for k=2. In first clock cycle all nodes move by distance of 1 as shown in Figure 9 and double their step_size to 2. Now all nodes try to move by a distance of 2, but only v2, v8 and v10 are successful and other nodes fail(as their threads break). Successful nodes double their step_size and others halve their step_size. At the same time v2 gets fixed as shown in Figure 10. Moves will continue till destination node is fixed. Algorithm is given in Figure 11.

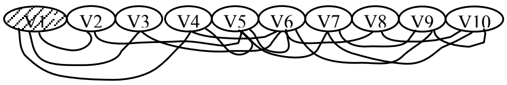

Figure 8: Initially all balls are together and v1 is fixed.

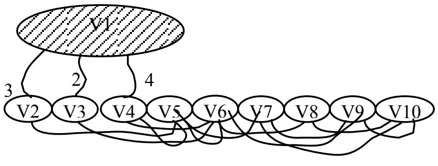

Figure 9: Balls after first move.

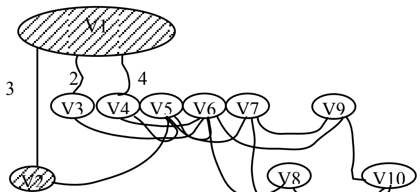

Figure 10: Balls after second move.

```
i. Initialize
   For all nodes initialize step_size to 1, position X_i to 0.
   Set status flag O_i of all nodes to 0 except for source, for
   which it is set to 1.

ii. Move Nodes
while destination is not fixed loop
  for all nodes i=1 to N do parallely
    If node ' i' is not fixed then
      Find next position of 'i',  X_i =X_i + step_size_i
      for all adjacent nodes j= 1 to N do parallely
          find distance from each adjacent node ' j ',
          Dist_j = (X_i -X_j )
          If any ( D_ij < Dist_j ) then
              X_i=X_i-step_size_i
              If (step_size_i >= k) then
                  step_size_i =step_size_i /k
              else
                  step_size_i =1
              end if
          else
              If any (( D_ij = Dist_j ) and (O_j = 1) )then
                  Fix 'i' at present location by setting
                  status flag O_i to 1
                  Set  node ' j' as the previous_node
              else
                  step_size_i =step_size_i *k
              end if
          end if
      end for loop
    end if
  end for loop
end while loop
```

Figure 11: Algorithm NATR.

For graph in Figure 12, though the shortest path length is 90 from v1 to v10, it is found in 27 clock cycles with k=2. But with same k=2, for graph in Figure 2, though the shortest path length is 9 from v1 to v10, it is found in 15 clock cycles. For k>1, NATR has a drawback of excessive moves for small edge values and this can be avoided by limiting step_size to minimum of adjacent edge values. Minimum of N adjacent edge values of a given node can be found using a comparator tree and has time complexity O(logN). With a single comparator tree, time complexity of the algorithm is O(L + NlogN), as finding minimum is to be done for all N nodes. If N comparator trees are used in parallel, then algorithm will have time complexity O(L + logN).

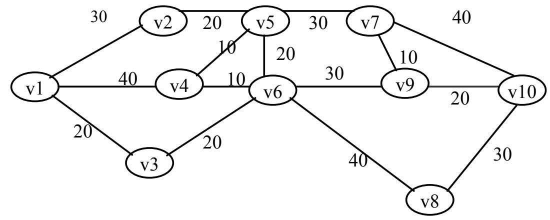

Figure 12: Another example graph

Many adjacent nodes may satisfy the condition(string stretched to full)to become previous_node for a node, giving alternate paths. In Figure 3, when v5 reaches a distance of 5, both v2 and v4 satisfy condition to become previous_node. But in hardware, handling multiple nodes to get alternate paths is difficult and hence one node is set as previous_node using a daisy chain (Section IV B).

A. Trace Back

Trace back is done to get nodes along the shortest path and its pseudo code is given in Figure 13. Nodes along the shortest path are stored in sp_nodes. Trace back starts from destination, moves towards source using previous_node and stops at source. For graph in Figure 2, trace back from v10 to v1 is shown in Figures 14 to 17.

```
temp_node=destination
i=1
while(temp_node ≠ source) loop
   sp_nodes(i)=temp_node
   i=i+1
   temp_node=previous_node(temp_node)
end while loop
sp  nodes(i)=temp  node
```

Figure 13: Trace back pseudo code.

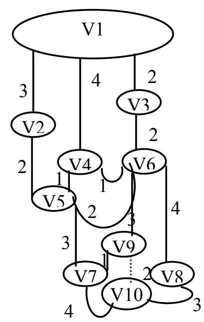

Figure 14: Tracing from 10 to 9.

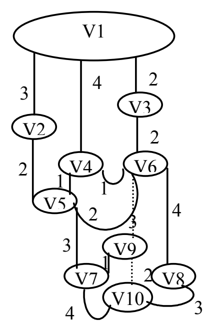

Figure 15: Tracing from 9 to 6.

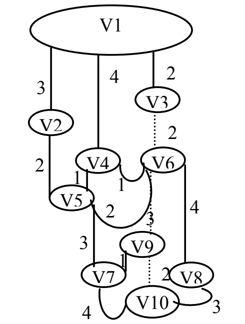

Figure 16: Tracing from 6 to 3.

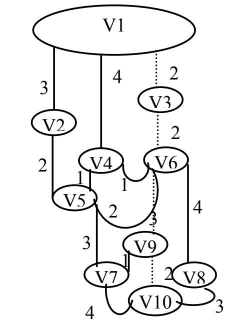

Figure 17: Tracing from 3 to 1.

IV. IMPLEMENTATION

NATR is implemented using Xilinx[10] Virtex II Pro development kit with XC2VP30 FPGA and is coded in VHDL. Chipscope pro 8.2i is used to check the results. As shown in Figure 18, NATR consists of 'N' nodes moving synchronously with clock by taking adjacent node's information. X, step_size and PV are declared as signals (integer array), and O is declared as a std_logic_vector of size N. D is declared as a two dimensional constant array. A signal by name "load" is used for initialization. Initially load is 0 and initialization is done. After initialization load is set to '1'. In the subsequent clock cycles(when load='1') nodes move as described in the algorithm and system halts when destination is fixed.

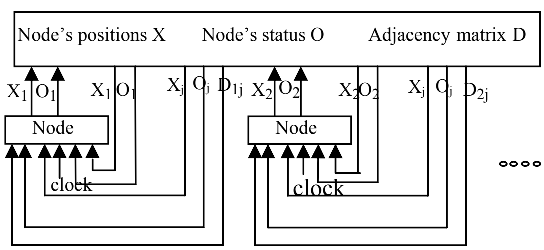

Figure 18: NATR as a collection of nodes.

A. *Node functionality*

Implementation has two versions: (1)NATR-B, a basic version with k=1 and (2) NATR-A, an adaptive version with k=2. In NATR-A, step_size is altered by shifting left or right to get step_size*2 and step_size/2 respectively.

1) *NATR-B*

In NATR-B, k=1 and hence step_size will always be 1. Node structure is as shown in Figure 19. In this, new position X_i of node 'i' is found by adding 1 to its current position. Actual distance between node 'i' and an adjacent node 'j' is found using X_i-X_j. This actual distance is compared with D_{ij} and flag E_j is set to 1 if $D_{ij} = X_i$-X_j else E_j is set to 0. Similar subtracter and comparator pairs are kept in parallel for other adjacent nodes. A node gets fixed when a E_j is 1 with corresponding O_j is 1. So all E_j values are ANDed with corresponding O_j to get F_j and F_j's are ORed to get FO, which is used to set O_i. An OR gate is used to retain old O_i, once it is set to 1. Similarly old position is retained once a node gets fixed. A daisy chain is used to select previous_node.

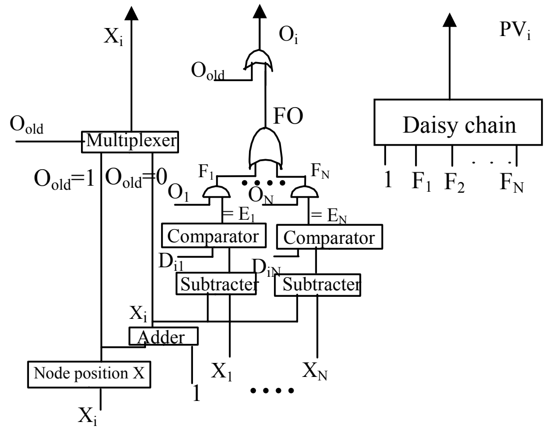

Figure19: Node structure for basic version.

2) *NATR-A*

Node structure for NATR-A is as shown in Figure 20. New position X_i of node 'i' is found by adding current X_i and step_size$_i$. Actual distance between node 'i' and an adjacent node 'j' is found using X_i-X_j and is compared with D_{ij} to set flags C_j and E_j, which indicate D_{ij}<(X_i-X_j) and D_{ij}=(X_i-X_j) respectively. Similar subtracter and comparator pairs are kept in parallel for other adjacent nodes.

If any C_j value is set to 1, it means string between 'i' and 'j' is broken. All C_j 's are ORed to get CO. If CO is 1, it implies one or more strings have broken and move is a failure. On failure, old position is retained using CO and CO is used as SHR, which is used to halve the step_size by shifting it right.

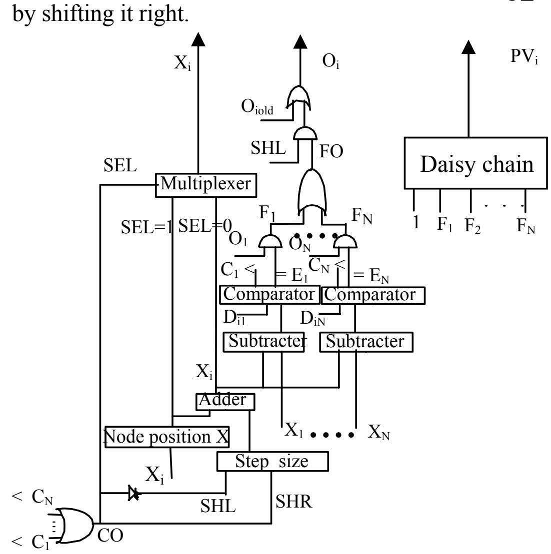

Figure 20: Node structure for NATR-A.

If CO is 0, move is successful and hence new X_i is selected as node position. Negated value of CO is used as SHL, which is used to double the step_size by shifting it left. A node gets fixed when a E_j is 1 with corresponding O_j is 1. So all E_j values are ANDed with corresponding O_j to get F_j and all F_j's are ORed to get FO. If FO is 1 and CO is 0(or SHL=1), node is fixed by setting O_i to 1. An OR gate is also used to retain old O_i, once O_i is set to 1 in earlier clock cycle. A daisy chain is used to select previous_node.

B. *Daisy chain*

Figure 21 shows the daisy chain and in that 32 is integer size of node numbers. F_j (Figures 19 & 20) is ANDed with select signal(SS) and the result is used for sending node number 'j' to OR gate. Whenever some 'j' is selected as previous_node of 'i', F_j deasserts SS to 0 and prevents high numbered nodes from becoming previous_node. If only shortest distance is needed, daisy chain can be skipped and node position is the shortest distance from source.

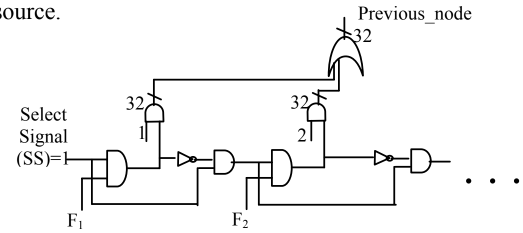

Figure 21:Daisy chain for selecting a node as previous node.

C. *Scalability*

When FPGA can't accommodate all nodes, NATR divides the computation to be done in a clock cycle into smaller steps and executes them sequentially on the FPGA.

Interface between host computer and FPGA is as shown in Figure 22. If FPGA can accommodate only 'k' nodes(where 1<=k<= N), then computation to be done in an iteration(clock cycle) has to be divided into N/k steps and in each step, computation of 'k' nodes should be done. To start computation for an iteration, all X_i and O_i are loaded to FPGA memory, as for computation of any node, adjacent node's information is needed. Positions, status flags, step_sizes of first 'k' nodes and first 'k' rows of adjacency matrix are sent to FPGA and computation is done. Though in the beginning X_i and O_i are loaded in to FPGA, for 'k' nodes' computation, these values are sent again. This is because node structure is same for all nodes, except current node's information(position, status flag, step_size) and D_{ij}'s. So these values are loaded each time computation is done. After the computation, results from FPGA are stored in temporary storage in host computer, as previous iteration values(X_i and O_i) should be used. Computation of all nodes is to be completed by taking 'k' nodes at a time. Once computation for an iteration is over, values held in temporary storage are updated as new values. Before starting next iteration, once again X_i and O_i of all nodes are loaded to FPGA. Because of dividing function to be done in an iteration to N/k steps, the time complexity will be O(L*N/k).

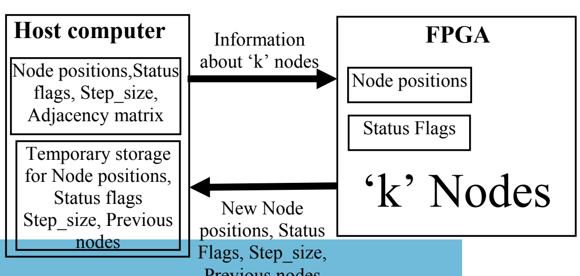

Figure 22: Interface between FPGA and host computer for 'k nodes computation.

Suppose we want to find shortest path in a graph of 10 nodes, with an FPGA that can accommodate only 4 nodes, then computation to be done in an iteration is divided as shown in Figure 23.

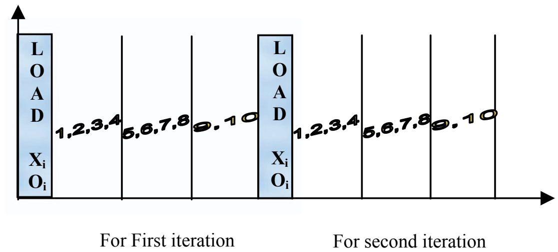

Figure 23: Computation of '4' nodes in a clock cycle.

V. Bellman-Ford Algorithm on FPGA

Parallel Bellman-Ford(PBF) algorithm[1] is as shown in Figure 24. Let 's' and 'd' be source and destination respectively. Let x_i be the distance of node 'i' from source and D be the adjacency matrix. When there is no edge between 'i' and 'j', D_{ij} is set to large value like 99999 to indicate infinity. Trace of PBF is as shown in Figures 25 to 27.

1. For all non source nodes set x_j as D_{sj}, j=1 to N, and for source set x_s as 0.
2. Update each node position in parallel using $x_i = \mathrm{Min}(x_j + D_{ij})$ for all j from 1 to N and $j \neq i$
3. Repeat step 2 till there are no changes in x_i.

Figure 24: Bellman-Ford Algorithm.

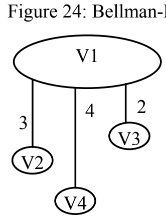

Figure 25: After first clock cycle.

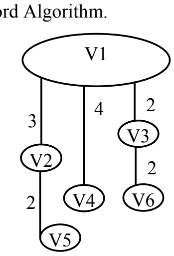

Figure 26: After second clock cycle.

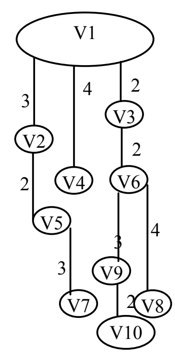

Figure 27: At the end.

Implementation involves updating X_i synchronously as per step 2 of algorithm and is implemented on Xilinx Virtex II kit. Let p1, p2 ...pn be the number of edges along the shortest paths from source to other nodes and $P=\mathrm{Max}(p_i)$. Algorithm needs P clock cycles to stop and hence is O(P).

VI. Results

Experiment is conducted for graphs of 5, 10 and 17 nodes by placing all nodes in FPGA and results are given in Table I. As shown in Table I, NATR has high operating clock frequency and it gets reduced by small values, with increase in N. As all subtracter and comparator pairs are kept in parallel, increase in N does not contribute to additional delay at this point. Small reduction in clock frequency is due to increased propagation delay in daisy chain and in N input OR gates used for ORing C_j and F_j.

TABLE I. RESULTS OF PBF, NATR-B AND NATR-A.

	Parallel Bellman-Ford		
Number of nodes	5	10	17
Max. clock frequency in MHz	55.28	30.81	18.99
LUTs used %	7	28	75
Gate Count	17,621	60,254	1,59,043
Clock cycles	2	4	8
Execution Time in ns	36.17	129.82	421.27
	NATR-B		
Max. clock frequency in MHz	145.57	131.34	120.63
LUTs used %	4	12	30
Gate Count	9,556	28,995	70,170
Clock cycles	5	9	15
Execution Time in ns	34.34	68.52	124.34
	NATR-A		
Max. clock frequency in MHz	94.70	92.75	87.28
LUTs used %	10	35	80
Gate Count	32,711	1,05,011	2,39,366
Clock cycles	9	15	27
Execution Time in ns	95.03	161.72	309.34
Execution time on Virtex II development kit			
	Parallel Bellman_Ford		
Clock frequency in MHz	50	25	16.66
Execution Time in ns	40	160	480
	NATR-B		
Clock frequency in MHz	100	100	100
Execution Time in ns	50	90	150
	NATR-A		
Clock frequency in MHz	50	50	50
Execution Time in ns	180	300	540

PBF has low operating clock frequencies and it drops down significantly with increase in N, which is due to finding minimum. Figures 28 to 30 show graphs of operating clock frequencies, execution times (with Max. operating clock frequency) and execution times on Virtex kit respectively. Though for N=5, PBF is efficient, but with increase in N, NATR is efficient.

In Table I, NATR-A has high execution time compared to PBF. But operating clock frequency falls down significantly in PBF and hence for large value of N NATR-A will be efficient. It can be seen in Figure 29. For N=17, NATR-A is better compared to PBF.

NATR is also implemented using C on Pentium IV (2.66GHz). Execution times are given in Table II and a graph is also plotted (Figure 31), which show FPGA implementation is good.

TABLE II. EXECUTION TIMES FOR C AND FPGA IMPLEMENTATIONS OF NATR.

	NATR-Basic version			**NATR-Adaptive version**		
N	**C**	**FPGA**	**Speed up C/FPGA**	**C**	**FPGA**	**Speed up C/FPGA**
5	2500	34.34	72.80	5000	95.03	52.61
10	20,000	68.52	291.88	30,000	161.72	185.50
17	1,10,000	124.34	884.67	2,00,000	309.74	645.70

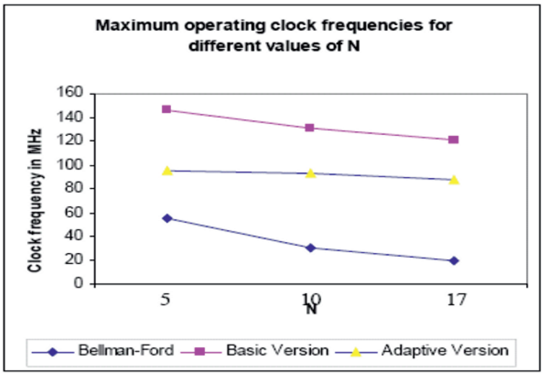

Figure 28: Operating clock frequency for different values of N.

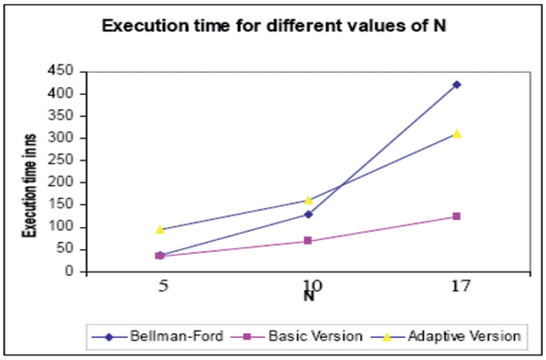

Figure 29: Execution times with maximum operating clock frequency.

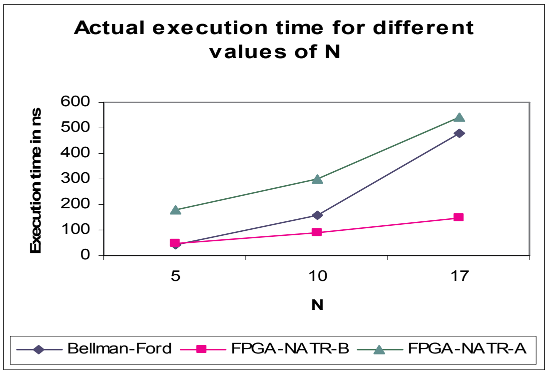

Figure 30: Execution time for different values of N on Virtex II.

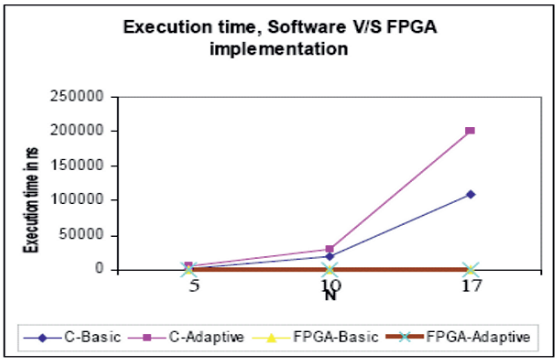

Figure 31: Execution time for software(C) and FPGA implementations.

Each node consists of N subtracters and comparators, and hence for N nodes, resources needed is $O(N^2)$. If maximum number of adjacent nodes is Z<<N, then we can allocate resources for Z adjacent nodes at each node and hence total resources needed is O(N).

Memory required is $O(N^2)$ because of storing D. By storing only required Z D_{ij}'s for each node, memory needed is 2ZN (Z for D_{ij} s + Z for 'j') and is O(N).

In [3] Dijkstra's algorithm is implemented on FPGA and it uses a comparator tree to find minimum instead of a loop, and hence has time complexity O(N). An experiment is conducted for 100 graphs of 1000 nodes with random edge weights. Shortest path length and number of outer loops in Dijkstra's algorithm are found. Results are sorted on number of outer loops and a graph is plotted (Figure 32). Graph shows that shortest path length is very small compared to Dijkstra's outer loop and hence NATR is efficient for large graphs. [3] finds minimum and hence operating frequency will fall down with increase in N, and makes it slower.

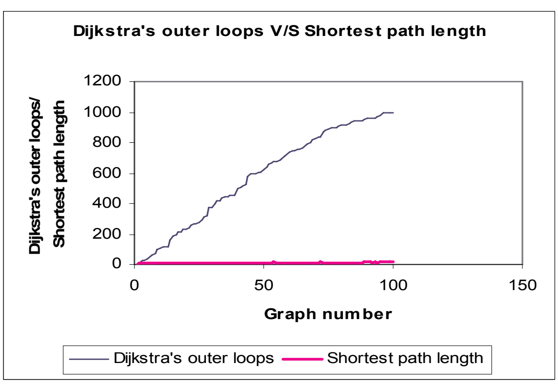

Figure 32: Dijkstra's outer loops V/S shortest path length.

In [6] signal propagation concept is used for finding shortest path and has time complexity O(L). [6] is not scalable, as all nodes need to be in FPGA for execution. It needs 'L' clock cycles for finding shortest path. NATR is scalable and NATR-A takes less than L clock cycles to find shortest path.

VII. Future Extensions

A. *Finding Minimum Spanning Tree*

NATR can be extended to find minimum spanning tree by resetting all node positions to 0 and making nodes to fall again, whenever a node gets fixed.

B. *Finding All Pair Shortest Paths*

In NATR, if clocks are applied till all nodes are fixed; it gives shortest paths from source to other nodes. If this is repeated for each node as source, we get all pair shortest paths in O(NL) time. Using N such systems in parallel and each system having one node as source, we can find it in O(L) time.

VIII. Conclusions

NATR is a new shortest path algorithm using reconfigurable logic and has time complexity O(L). It is based on BSM and is highly parallel and scalable. The adaptive version of NATR takes less than L clocks, for large edge weights. It runs at much higher operating clock frequency compared to other reconfigurable based approaches, as it avoids finding minimum and hence is faster. Both versions of NATR is implemented on Xilinx Virtex II and is 6.3 times faster compared to parallel Bellman-Ford algorithm for 17 node graph. FPGA implementation of NATR is 884 times faster than its C implementation.

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Basic Concepts and Advantages of the On-line Voting Solution

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***Abstract*-This contribution focuses on the opportunity of using web-based electronic voting as part of e-government services. Many countries have been researching the benefits of e-voting solutions. Every country uses individual ideas to solve e-voting problems. Only few are thinking about web-based electronic voting. This contribution describes the basic proposals of web-based electronic voting solution research and the web-based application advantages for the electronic voting.**

I. Introduction

Many countries have been researching the benefits of e-voting solutions. Every country uses individual ideas to solve e-voting problems. The range nowadays is from electronic ballot reading devices to ballot boxes installed in polling stations. The second electronic voting system is internet voting. The internet voting solution allows voting through the internet network. Very few countries have been working on the internet based voting system.

E-Voting has been attracting considerable attention during the last few years. There is nowadays a great deal of interest in e-solutions. There are two more practical reasons for the interest in e-voting solutions. The first an interest in an e-voting, system which can help to solve problems with domestic election systems e.g. lacking flexibility with respect to time frames (not in Europe, but e.g. in the USA) Also the physical accessibility of polling stations, which may possibly prevent citizens from casting their votes. The second reason is connected to the number of elections during the year. People do not usually want to visit polling stations very often. In a time when referendum is becoming increasingly popular, it is the right time to research other possibilities of practicing democracy.

II. THE ELECTRONIC VOTING DESCRIPTION

Electronic voting is similar to classic "paper-form" voting. In classical "paper-form" voting voters entering the polling station have to be identified. If identification is passed, they are able to vote. The whole scenario of classical voting can be seen in figure 1

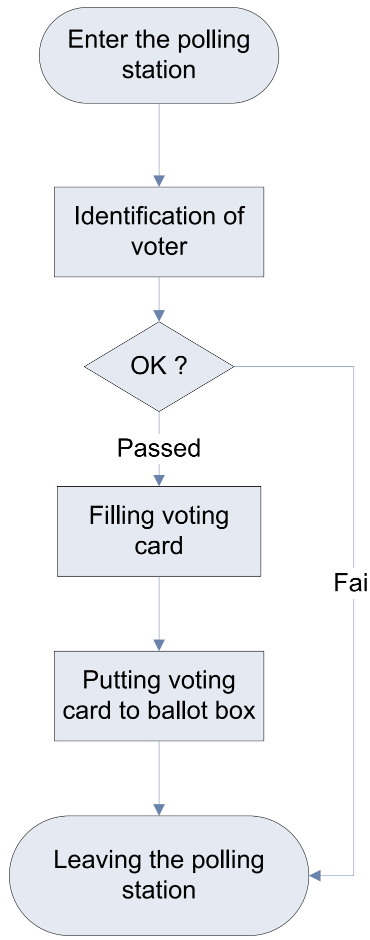

Fig. 1. The classical "paper form" voting process

There are two recognised types of electronic voting systems. The first one is based on visiting a polling station. In this case voters are still identified by using identification cards. Voters

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 406–409.

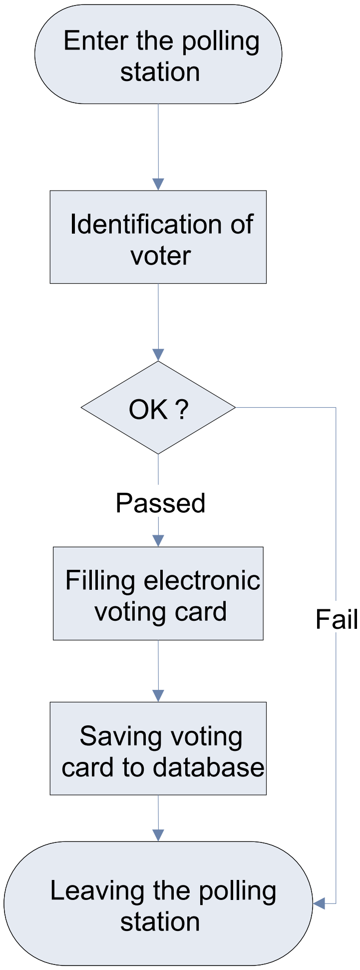

Fig. 2. The in-site electronic voting system

do not fill voting cards in the paper form but push buttons on various electronic devices. Then voters use the electronic device to vote

The second type of electronic voting system is based on remote technology. Usually voters have the chance to vote by using computers at remote locations or at polling stations. They use computer and internet networks for voting. Voters can vote out with the normal interval for voting (usually office hours). They can also vote from abroad. These constitute the most important advantages of the remote-based voting system. This idea is usually called internet voting.

III. THE IDEA OF WEB-BASED VOTING SOLUTIONS

The main point of this contribution is the utilisation of a web-based application solution for remote electronic voting,

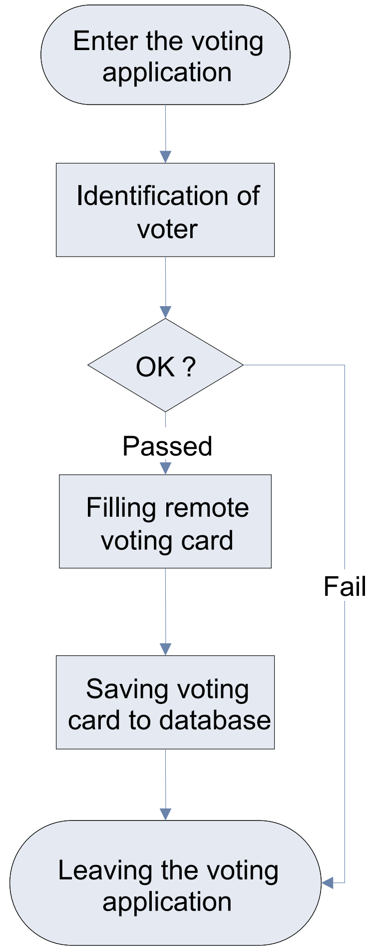

Fig. 3. The remote voting process

because it is a solution, which allows the connection of electronic voting and internet voting.

By using a web-based voting system you are not merely limited to internet voting. Web-based voting is useful in electronic voting too. The web platform can be useful and helpful for every electronic voting system. If the e-voting system uses a web-based solution, it can be used by individual voters through the internet network or it can be used in polling stations. In the polling station touch-screen terminals can be utilised.

The web-based voting solution can be divided into four parts:

1. The voting application for individual voters
2. The voting application for polling stations
3. The database of votes
4. The results of elections

The voting application for individual voters would be used in internet explorer (web browser). The voter has to be identified and will be able to access the voting application. The voter identification has to be done by using the list of voters. The list of voters has to be in electronic form. When access is granted the process of vote casting can start. Each voter is permitted to access a list of candidates in his election district only.

The voter uses web-based applications for the voting process. The first step is choosing and editing a ballot. Then the ballot is secured by cryptophytic method (e. g. public key). The secured ballot is saved into a database. Each ballot has an individual transaction ID. Each voter receives a ticket with the transaction ID and results of their voting. This is necessary for voting control and maintaining anonymity.

At the polling station the application is used without electronic identification of voters. Voters are identified by the election committee and process of voting continues as for electronic voting (Figure No. 2). The election committee has to control the voter if the voter does not vote by the internet channel.

The third part - database of votes - can be registered and monitored by various database technologies. Every vote is saved only once. During the process of identification, the system has to check voter status. It is means that the system will recognise if voter already log in or not. If the voter had previously logged-in the voter is not allowed to log-in again.

The result of elections can be prepared automatically. The system has to support the algorithm(s), which are used in the country. Results are controlled by an election committee or directly by voters.

IV. CONDITIONS FOR THE ELECTRONIC VOTING SYSTEM

There are several conditions for electronic voting systems. The law in the country has to support the electronic voting systems. The web-based voting solution has to follow the technical and process conditions listed below:

1. Participation in the voting process is granted only for registered voters.
2. Each voter has to vote only once.
3. Each voter has to vote personally.
4. Security and anonymity of voters and voting.
5. Security for the electronic ballot box.

V. EXAMPLES OF USING WEB-BASED VOTING

There are two examples of using web-based voting in Europe. The first one describes a project prepared in Switzerland. The situation in Geneva - Switzerland is a good example for using electronic voting systems. Voters are called 4 to 6 times to the elections.

Lots of voters in Geneva used to vote by postal message and 2/3 of citizens have internet access. The electronic ballot box is locked by two independent digital keys. In Geneva a "paper-form" voting card is used. Each card has a transaction number. Transaction numbers are stored in the database. Voters have to insert a number into the system as an authorisation code. Then they can vote. Each voting card has a PIN code too, which is used for confirmation of vote.

The second example is taken from Estonia. Identification cards are used in electronic voting system in Estonia. This smart card has a chip. The chip contains data about the holder and two certificates, two private keys. Smart cards can be used for various applications, not only for eGovernment services. Smart cards were important for starting the project of electronic voting in Estonia. These cards are used for identification of voters and for authorization of votes. Each voter can change their vote several times.

VI. WEB TECHNOLOGY FOR ELECTRONIC VOTING

The web based technology is useful for electronic voting systems. This technology is based on a client-server. The client-server technology has advantages in the field of support and installation. There is only one central server, which is the main part of the solution. Application development can use several technologies. The web based application can use AJAX user-interface, which produces a more comfortable and easy

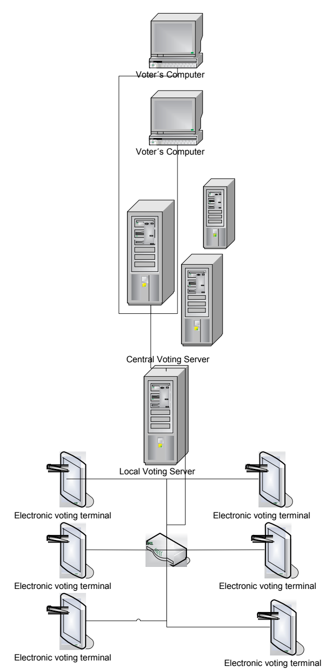

Fig. 4. Basic architectural concept of a web-based voting system

environment. Web services can be used for communication between various areas of the solution. Web services are very useful especially in the decentralisation of the client-server system. A demonstration of the solution can be found on figure No. 4

VII. CONCLUSION

The main task of this contribution was to introduce the idea of a web-based electronic voting solution. The author in his research has been working on the concept and the architecture of the web-based electronic voting solution for the Czech Republic. The contribution focused on web-based systems because these systems are more flexible and comfortable for voters.

The next obvious advantage of electronic voting systems and especially of web-based systems is related to the cost of elections.

Many countries have researched the benefits of e-voting solutions. Every country uses individual ways to solve e-voting problems. Only a few are seriously studying web-based electronic voting. Web-based electronic voting solutions represent the future of electronic voting in Europe.

Increasing internet access supports the e-Government and e-Democracy. People are used to communicating through the Internet. In the future people will be used to electronic elections too.

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Towards a Generic Autonomic Architecture for Legacy Resource Management

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Abstract **Half a decade has passed since the objectives and benefits of autonomic computing were stated, yet even the latest system designs and deployments exhibit only limited and isolated elements of autonomic functionality. From an autonomic computing standpoint, all computing systems – old, new or under development – are legacy systems, and will continue to be so for some time to come. In this paper, we propose a generic architecture for developing fully-fledged autonomic systems out of legacy, non-autonomic components, and we investigate how existing technologies can be used to implement this architecture.**

I. Introduction

The vision of autonomic computing [1] set an unprecedented number of engineering and scientific challenges [2] directed at a single goal: the development of self-managing computing systems. Since the launch of the autonomic computing manifesto over five years ago, tremendous resources have been dedicated to solving these challenges, yet systems that "manage themselves according to an administrator's goals" [3] are far from ubiquitous.

The limited adoption of autonomic solutions is largely due to the fact that existing IT system components exhibit only restricted and isolated elements of autonomic functionality. Based on insights from the development of a commercial framework for the autonomic management of data centre resources [4] and on best practices presented in a separate paper [5], we propose a generic autonomic architecture for the development of autonomic systems out of non-autonomic components. This approach differs significantly from other autonomic frameworks that target the management of autonomic-enabled components, e.g., applications implemented to use the API of the autonomic framework explicitly [6].

The core component of our architecture is a *policy engine* that enforces a set of user-specified business policies. The policy engine can be configured to manage resources whose types are unknown at implementation time. This configuration is achieved by means of a model of the managed system, and allows the integration of the policy engine into systems comprising a heterogeneous mix of legacy resources. Again, this represents a major improvement over existing frameworks that are dedicated to the management of a specific type of resource [6, 7, 8]. Another novel feature of our generic autonomic architecture is the policy engine's ability to expose the collection of IT resources it manages as an atomic, higher-level resource. This enables the integration of an autonomic system as an individual resource into another instance of the same architecture, thus supporting the development of hierarchical systems-of-systems [9].

The remainder of the paper is organised as follows. Section II introduces our generic autonomic architecture and contrasts it with existing autonomic computing frameworks. Sections III through VII describe in detail the components of the architecture, and investigate the extent to which existing technologies can be used for their implementation. A preliminary specification of the managed system model used to configure the policy engine is proposed, and sample policies based on this model are presented in these sections. We then describe the requirements for the policy engine, and the criteria used to distinguish between its different realisations. Section VIII summarises our findings and the next steps of the project.

II. Autonomic Architecture for Legacy Resource Management

A large number of projects have investigated isolated aspects related to the development of autonomic computing systems out of non-autonomic components. Some of these projects addressed the standardisation of the policy information model, with the Policy Core Information Model (PCIM) [7, 8] representing the most prominent outcome of this work. Recent efforts such as Oasis' Web Services Distributed Management (WSDM) project were directed at the standardisation of the interfaces through which the manageability of a resource is made available to manageability consumers [9]. An integrated development environment for the implementation of WSDM-compliant interfaces is currently available from IBM [10].

In a different area, expression languages were proposed for the specification of policy conditions and actions, and used to implement a range of policies [11, 5, 12, 3]. In addition to the development of standards and technologies, complete autonomic computing solutions have been produced recently [4, 5, 6], typically for the management of specific systems, and with limited ability to function in different scenarios from those they were originally intended for.

The generic autonomic computing architecture depicted in Fig. 1 builds on all these recent developments, and generalises the author's previous work on policy-based resource management [5]. The policy engine at the core of the architecture is a generic module that can be configured to manage heterogeneous systems comprising components that vary from traditional computing resources such as servers and software applications to application servers, virtual machines and devices including load balancers, switches and PDAs.

The use of a generic policy engine across such a broad variety of domains requires that the engine configuration is done by means of a model of the system to be managed. This model has to specify the relevant system resources, alongside with their characteristics and relationships. A rich and expressive meta-model of managed systems is required to ensure that the manageability capabilities of all types of systems, whether small and simple or large and complex, can be specified as a model that the policy engine understands.

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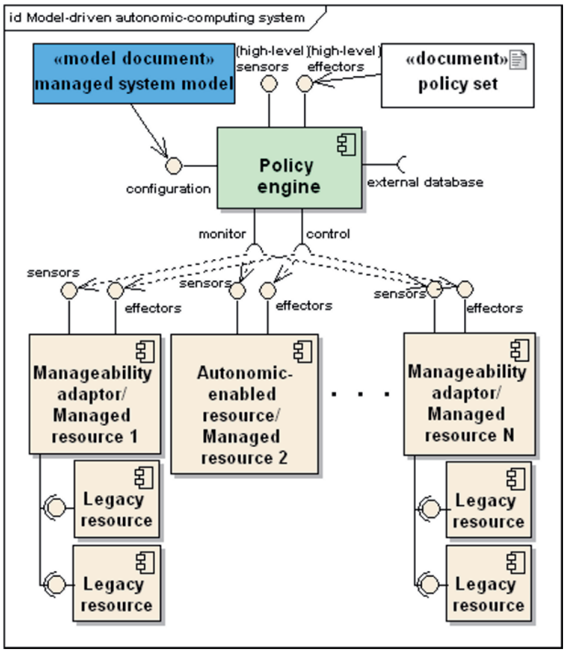

Fig. 1: The generic autonomic architecture for legacy resource management

A configured instance of the policy engine is capable of reading in and applying the set of policies on which to base its decisions in the management of the system. These policies are presented to the policy engine in a declarative language that makes references to the resources defined in the managed system model. They specify how the engine is required to monitor the resources, aggregate them into resource collections, report their state and act upon them to enforce higher-level business policies.

The managed system can include both legacy resources with no or limited autonomic capabilities, as well as autonomic-enabled resources. The legacy resources are exposed to the engine through a management-enabling layer with standard *sensors* (i.e., interfaces for gathering state information about resources) and *effectors* (i.e., resource control interfaces). The autonomic-enabled resources can be accessed directly by the engine. In particular, an implementation of the generic autonomic architecture can become a resource in a larger instance of the same architecture.

The next sections describe each part of the architecture in detail, specifying its required and desirable characteristics. Existing standards and technologies that can contribute to the realisation of the architecture are overviewed, together with their benefits and limitations.

III. LEGACY RESOURCES

The legacy resources that the generic autonomic architecture should support include a heterogeneous mix of "traditional" IT resources, recent types of IT resources, and devices. Typical traditional IT resources include:

- Physical servers and clusters of servers with their CPU, memory and disk resources, and the applications running on them. Starting/stopping, monitoring, reporting, allocating CPU, memory and disk service-level agreements to applications, and powering servers on and off in response to variations in load, failures, time of day/week and other business policies are among the activities covered by the autonomic management of a traditional IT system [5].
- Networks, collections of networks and the consumers that make use of them. Quality-of-service management, dynamic admission and provisioning control in the presence of variable demand, failures and changing business policies are the typical targets of a self-managed network [13].

Analogous but less traditional IT resources that will benefit from being part of a self-managed system include:

- Application servers with their web applications. Setting application service levels and access to the resources of the underlying hardware, monitoring, reporting and all functionality that is normally expected from a classical IT system will eventually be extended to these platforms.
- Virtualisation environments and the virtual machines they provide.

Some of the devices that will become increasingly present in autonomic systems are:

- Devices that are typical components of a standard IT system—printers, backup systems, switches, load balancers and power supplies. The latest models of all these devices exhibit interfaces that provide an ever increasing scope for automation.
- Common household devices—televisions, home cinemas, telephones, home security devices.

IV. MANAGEABILITY ADAPTORS

Despite an increasing trend to add management interfaces to new computing components and devices, and to make existing ones public, achieving self-management in even small computing systems is hindered by the broad diversity of architectures and technologies these interfaces are based upon.

The generic autonomic architecture requires that a standard interface is used to expose the manageability of all types of resources presented in the previous section in a uniform way. The manageability interface comprises:

- Sensors for accessing the state of the managed resources. The sensors should support both explicit reading of specific state information, and a notification mechanism that the policy engine can use to subscribe and receive notifications of certain state changes.
- Effectors for configuring the resource parameters in line with the policies supplied to the policy engine.

The interface is solely responsible for the interoperability of a diverse spectrum of resources with the universal policy engine. To achieve this, the interface needs to be simple and flexible, and to associate only limited semantics to the state information and configuration parameters it exposes. Resource properties such as state variables and configuration parameters are uniquely labelled and strongly typed, but their roles and relationships are specified instead in the system model, as described in the next section.

A very good approach at defining a manageability interface standard that satisfies these requirements is represented by the Web Services Distributed Management (WSDM) standard. The Management Using Web Services (MUWS) component of WSDM [9] leverages web service technology benefits such as platform independence, loose coupling and security support to define a web service architecture enabling the management of generic distributed resources. The MUWS specification describes a standard way in which manageable resources can expose their capabilities, and defines a number of built-in capabilities that resources should provide (e.g., *ResourceId*, *Description* and *Version*). Resource-specific capabilities can be provided and listed as elements of the *Manageability-Characteristics* built-in capability. The WSDM/MUWS standard specifies ways for accessing resource capabilities by means of web services, and requires that a "resource properties document" XML schema is provided as a basic model of the managed resources. As a result, an implementation of the standard [10] provides a superset of the functionality required for a managed resource from our architecture.

V. Managed System Model

The system model used to configure the policy engine must specify all resources to be managed and all their relevant properties. As the policy engine can always be reconfigured using new versions of the model, resources and resource properties not referred to in the policies need not be specified. The model should also provide details about the characteristics of the resource properties, thus allowing the use of adequate operators in the policies and reducing the amount of work by the policy engine. Finally, to enable the reuse of model components and policies, standardised terminology and resource (property) definitions must be used in the model.

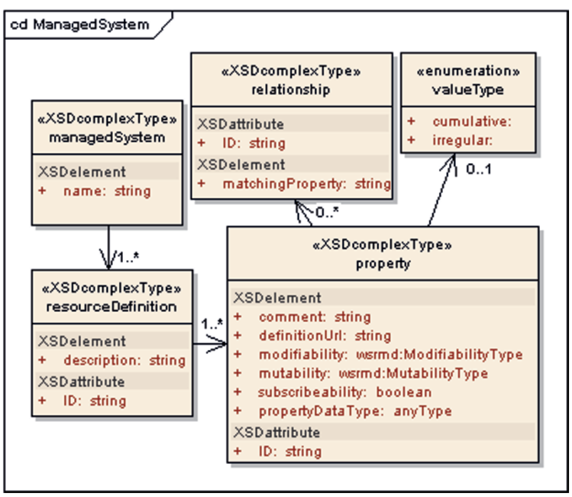

Fig. 2: Prototype meta-model of a managed system

The prototype *meta-model of a managed system* in Fig. 2 satisfies these requirements and is the preliminary result of a project to generalise the author's previous work on policy-based resource management [5]. The meta-model specifies a managed system as a named sequence of one or several resource definitions. Each resource definition (i.e., *resourceDefinition* in the UML diagram) comprises a unique identifier, a description and a set of resource properties with their characteristics. These properties should be drawn from a controlled metadata repository for the IT area of interest. Each property has a data type (*propertyDataType*), and is associated a unique ID and the URL within the metadata repository where its definition is located. The following property characteristics are exposed by the current version of the meta-model:

- *modifiability* – the *ModifiabilityType* from the *WSResourceMetadataDescriptor* (WS-RMD) 1.0 specification [14] is used to state if the value associated with this property is "read-only" or "read-write".
- *mutability* – the WS-RMD MutabilityType used specifies if the property is read-only or can be set. The possible values for this characteristic are "constant", "mutable" and "appendable".
- *subscribeability* – this element specifies if a client/agent such as the policy engine can subscribe to receive notifications when the value of this property changes.
- *valueType* – optionally, the model can specify for numerical properties if their value is cumulative (such the CPU utilization of a process over the process lifespan) or the property values follow no pattern.
- *relationship* – relationships between instances of a resource can optionally be specified as pairs comprising a unique ID and the ID of a "matching" property. Two resource instances are in the relationship if the current property of the first and the matching property of the second have the same value.

The sample model in Fig. 3 defines the processes and servers of an IT system. A policy engine configured to use this model can handle policies that refer to these two types of resources and their properties.

Microsoft's System Definition Model (SDM) is a meta-model used to create models of distributed systems [15] with a high degree of detail. The ongoing Dynamic Systems Initiative programme [16] intends to use these complex models as enabling elements in the development of manageable systems that exhibit elements of autonomic behaviour. Given its complexity, the SDM meta-model is less suited for use in conjunction with the generic policy engine employed by our generic architecture. The WSDM/MUWS standard [9] uses the WS-Resource Metadata Descriptor framework to describe the metadata for a resource manageability endpoint. This allows the specification of the properties of specific resource state variables and parameters, and the definition of resource relationships and operable collections. The Managed Resource Document used by version 1.1 of IBM's Policy Management for Autonomic Computing (PMAC) framework, and the combination of web services and autonomic computing standard specifications that version 1.2 of PMAC uses are further examples of managed system models [3].

```
<managedSystem xmlns="http://www.softeng.ox.ac.uk/system" xmlns:xsi="
               xsi:schemaLocation="http://www.softeng.ox.ac.uk/system
  <name>IT system</name>
  <description>A set of servers running user applications.</description>
  <resourceDefinition ID="process">
    <description>A process run by the operating system.</description>
    <property ID="serverId"> [11 lines]
    <property ID="pid">
      <comment>The process identifier.</comment>
      <definitionUrl>http://www.it-metadata.org/process/pid</definitionUrl>
      <propertyDataType>
        <xsd:simpleType>
          <xsd:restriction base="xsd:positiveInteger" />
        </xsd:simpleType>
      </propertyDataType>
      <mutability>constant</mutability>
      <modifiability>read-only</modifiability>
      <subscribeability>false</subscribeability>
      <relationship ID="child">
        <matchingPropertyID>ppid</matchingPropertyID>
      </relationship>
    </property>
    <property ID="ppid"> [11 lines]
    <property ID="name"> [11 lines]
    <property ID="uid"> [11 lines]
    <property ID="groupId"> [11 lines]
    <property ID="cmdline"> [11 lines]
    <property ID="cpuUtilisation"> [12 lines]
    <property ID="memoryUtilisation"> [12 lines]
    <property ID="cpuAllocation"> [11 lines]
    <property ID="memoryAllocation"> [11 lines]
  </resourceDefinition>
  <resourceDefinition ID="server">
    <description>A physical server is a data centre.</description>
    <property ID="serverId"> [11 lines]
    <property ID="numProcessors"> [11 lines]
    <property ID="processorCpu"> [11 lines]
    <property ID="memory"> [11 lines]
    <property ID="status">
      <comment>The status of the server: 'active' or 'standby'.</comment>
      <definitionUrl>http://www.it-metadata.org/server/status</definitionUrl>
      <propertyDataType>
        <xsd:simpleType>
          <xsd:restriction base="xsd:string">
            <xsd:enumeration value="active"/>
            <xsd:enumeration value="standby"/>
          </xsd:restriction>
        </xsd:simpleType>
      </propertyDataType>
      <mutability>mutable</mutability>
      <modifiability>read-write</modifiability>
      <subscribeability>true</subscribeability>
    </property>
   <property ID="command"> [11 lines]
  </resourceDefinition>
</managedSystem>
```

Fig. 3: Basic model of an IT system

VI. Policy Set

A. Overview

Policies tell the policy engine how to manage the underlying system, and how to expose it to the outside world. Note that although the former role of policies is the only one considered by most autonomic computing frameworks, the latter role is equally important as it allows the architecture as a whole to become a managed component of a larger managed system. The policies employed by the autonomic architecture in Fig. 1 achieve these roles by specifying:

- How the modifiable properties of the resources (i.e., the resource configuration parameters) need to evolve as a function of the system state and of time.
- The exposed resources of the system, and their properties. As an example, consider the traditional IT system introduced in the previous section, whose resources are a set of servers and the processes running on them. A set of policies can specify that the policy engine exposes as high-level resources the applications running on the system, one property of this "application" resource being the number of servers on which the application is running.

Note that in a particular instance of the architecture, one or the other of these roles (but not both) can be missing.

The language used to express policies needs to be sufficiently flexible to support the use cases below.

B. Resource group specification

Policies are about resources of the managed system and their properties. Therefore, the policy language needs to allow the specification of the set of resources to which the policies apply. Specifying the scope of policies typically organises system resources into groups that are regarded as a single entity from the standpoint of a policy or set of policies. Resources grouped together for this purpose can be exposed as a higher-level resource by the policy engine. To illustrate this with an example, consider the IT system defined in Section V. The XML fragment below shows how the transitive closure of the child process relationship applied to all processes whose name is 'httpd' can be used to group the processes of an Apache web server and all their descendents:

```
<resourceGroup ID="Apache">
  <includes resource="process">
    child*(name=="httpd")
  </includes>
<resourceGroup>
```

C. Higher-level resource definition

Policies can specify higher-level resources that the policy engine exposes to the outside world, e.g., to present system administrators with a summary of the state of the managed system or to enable its integration into a larger managed system. The example below instructs the policy engine to expose an 'application' as a higher-level resource:

```
<resourceDefinition ID="application">
  <description>
    A software application.
  </description>
  <property ID="name"> [...]
  <property ID="numServers"> [...]
</resourceDefinition>
<exportedResourcePolicy type="application">
  <policyScope>
    <resourceGroup ID="Apache"/>
  </policyScope>
  <policyCondition>TRUE</policyCondition>
  <policyAction>
    <property>
      <name>name</name>
      <value>Apache web server</value>
    </property>
    <property>
      <name>numServers</name>
      <value>COUNT(p:process|p.serverId)</value>
    </property>
  </policyAction>
</exportedResourcePolicy>
```

D. Resource configuration

Policies specify the desired value of modifiable resource properties as a function of the state of the managed system and of time. The following sample policy illustrates how between 8:00 and 18:00 the processes in a resource group are allocated 80% of the CPU power of their servers:

```
<resourceConfigurationPolicy>
  <policyScope>
    <resourceGroup ID="Apache"/>
  </policyScope>
  <policyValue>100</policyValue>
  <policyCondition>Hour IN 8..18</policyCondition>
  <policyAction applyTo="EACH(process.pid)">
    <property>
      <name>groupId</name>
      <value>1</value>
    </property>
  </policyAction>
  <policyAction applyTo="EACH(process.serverId)">
    <property>
      <name>cpuAllocation</name>
      <value>80%</value>
    </property>
  </policyAction>
</resourceConfigurationPolicy>
```

Other policies can be used to define how these resources should be managed outside this time interval, or policies with a higher policy value can enforce different actions between 8:00 and 18:00 on certain week days.

E. Resource scheduling

In resource scheduling, system capacity specified by resource properties are allocated to resource groups. Similar to other policies, this involves setting the value of specific resource properties. For instance, in our basic IT system scheduling policies could be used to specify how the server CPU and memory is to be partitioned among software applications. This may involve setting the "cpuAllocation" property of processes to allocate CPU to running groups of processes, and/or using the "command" and "state" properties of servers to start/stop applications and power on/off servers, respectively [5].

F. Workflow

Each configuration policy is a simple, one-step workflow. More complex workflows are often needed in which a sequence of actions is performed, with well-defined delays and state validations between successive actions in the sequence. Although this behaviour could potentially be simulated using a number of configuration policies and supporting additional resource properties, this approach would unnecessarily complicate the implementation of the manageability layer, the system model and the policies themselves. The use of BPEL workflows [17] represents a significantly more effective approach to expressing and handling workflow policies.

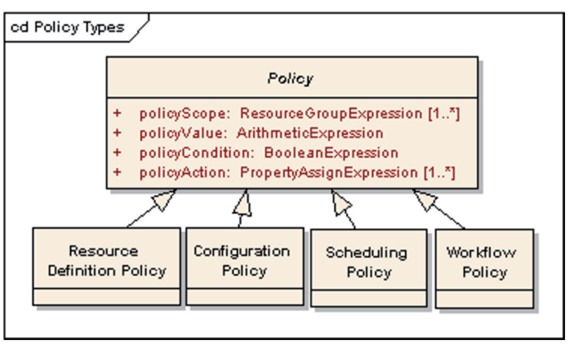

Fig. 4: Policies encountered in a generic autonomic architecture

G. Summary

Fig. 4 summarises the types of policies described in this section, illustrating how policy components are formulated in terms of expressions that depend on the resource properties of the managed system. These expressions vary in complexity from the very simple to the sophisticated, and the effectiveness of the policies supported by a realisation of the architecture is dependent on the power of its underlying expression language. Several autonomic computing expression languages have been proposed in the recent years. The language used by the policy-based resource allocation framework in [5] enables the specification of policies for resource monitoring and management in a data centre through the use of combinations of arithmetic and logic operators, pre-defined functions that can be applied to resource properties and built-in variables. While this works well for the system that the framework is targeting, the use of system-specific pre-defined variables such as PercCpuUtilServer (i.e., the percentage of CPU that an application is using on a given server) and AbsCpuHeadroomServer (i.e., the amount of CPU unused on a given server) is not generic enough for our system. However, the built-in variables used by the system-specific approach in [5] suggest the type of operators that would be needed in a realisation of the generic autonomic architecture.

The Windows System Resource Manager [4] uses regular expressions, logical and string operators, and built-in time variables to specify the process-matching criteria that define the WSRM policy scope, as well as the policy conditions and actions. The Autonomic Computing Expression Language (ACEL) [12] used by IBM's PMAC framework [3] supports a wide variety of primitive types (e.g., Boolean, several integer and float types, and String), and a selection of complex data types—Calendar, Composite and Collection. The standard operators are employed to combine resource properties and constants of these types into expressions. The extensive operator set in ACEL covers most of the use cases envisaged by the architecture described in this paper, although some very useful (albeit more complex) operators such as set comprehension and transitive closure are not supported.

VII. Policy Engine

The core component of the autonomic architecture implements a set of policies by monitoring and controlling the sensors and effectors of the managed resources, respectively. The "high level" resources of the managed system are exposed through the (high-level) sensors interface, enabling the inclusion of the system into another instance of the same architecture, a key requirement for the design of manageable systems of systems [1]. As indicated in Fig. 1, the engine is expected to make use of an external database for storing its internal state, e.g., the managed system model, the active policies and historical resource property values. To keep the architecture generic, we do not propose any particular way in which the policy engine should learn about the actual set of resources it is responsible for. Possible options include direct configuration, the use of a discovery technique [25] or a combination of the two.

Internally, the engine comprises modules for evaluating the expressions in the four policy components, an internal clock for time-based expressions, and an implementation-dependent set of schedulers, linear programming solvers and other optimisers, workflow engines, etc. An internal cache can optionally be used in addition to the external database for the rapid retrieval of state information. To keep the architecture generic, we are not going to propose a particular way in which the policy engine should be informed about the actual set of resources it is responsible for. Possible options include a static configuration by means of the policy set itself, the use of a discovery technique [19] or a combination of the two.

Given the generality of its specification, the engine can be implemented using a number of very different technologies, including standalone software applications/agents, a web services, or hardware appliances. As the field progresses and agreement is reached on a standard specification for the universal policy engine, its largely interchangeable implementations will differ in:

- The presence or absence of certain areas of functionality. The management of certain legacy resources may not require the use of scheduling and/or workflow policies. In this case, the use of a fast, off-the-shelf hardware appliance that does not support these parts of the specification could be ideal.
- The "quality" (i.e., the complexity and effectiveness) of the algorithms and heuristics involved. For instance, some implementations may use suboptimal, fast scheduling heuristics, while others may provide optimal decision making but a longer response time. Each of these implementations may be suitable for use in some systems but not in others.
- The total cost of ownership (TCO). Open-source and proprietary implementations of the engine will inevitably come with different TCO and TCO breakdowns. An open-source solution may involve no initial expenditure but significant effort to integrate and configure. Conversely, commercial implementations will require a major initial investment but offer the guarantee of a high-quality documentation and support over a long period of time.

VIII. Conclusions

Starting from a policy-based management framework targeted at data-centre resources [4, 5] and building on recent advances in autonomic computing [2, 3, 13, 16, 19, 23], we proposed a generic autonomic architecture and a universal policy engine for autonomic solution development. Our policy engine can be configured to monitor and control a wide variety of systems comprising heterogeneous mixes of legacy resources. The policy engine is configured by presenting it with a model of the system to be managed, i.e., a formal specification of the legacy resources in the system and of their relevant properties.

The components of the generic autonomic architecture were defined, and their requirements were discussed in the paper. Existing technologies that could be used to build these components were briefly analysed, and possible approaches to implementing the architecture were outlined.

Work is underway to validate the proposed system meta-model and the types of policies supported by the universal policy engine in data-centre resource management scenarios similar to those addressed by the commercial framework in [4]. The project is currently investigating the best way to use policies to define the high-level resources exposed by the policy engine so that an instance of the architecture can be integrated as a managed resource into a system of systems [18]. In the future, this work will continue in conjunction with the development of an IT metadata repository from which the models used to configure the policy engine will draw their resource property definitions. In the longer term, this should allow the definition of reusable policies and policy templates that will ease the adoption of the architecture.

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Constrained Linear Spectral Unmixing Technique for Regional Land Cover Mapping Using MODIS Data

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***Abstract*—Over the last few decades, there has been a significant land cover (LC) change across the globe due to the increasing demand of the burgeoning population and urban sprawl. In order to take account of the change, there is a need for accurate and up-to-date LC maps. Mapping and monitoring of LC in India is being carried out at national level using multi-temporal IRS AWiFS data. Multispectral data such as IKONOS, Landsat-TM/ETM+, IRS-1C/D LISS-III/IV, AWiFS and SPOT-5, etc. have adequate spatial resolution (~ 1m to 56m) for LC mapping to generate 1:50,000 maps. However, for developing countries and those with large geographical extent, seasonal LC mapping is prohibitive with data from commercial sensors of limited spatial coverage. Superspectral data from the MODIS sensor are freely available, have better temporal (8 day composites) and spectral information. MODIS pixels typically contain a mixture of various LC types (due to coarse spatial resolution of 250, 500 and 1000 m), especially in more fragmented landscapes. In this context, linear spectral unmixing would be useful for mapping patchy land covers, such as those that characterise much of the Indian subcontinent. This work evaluates the existing unmixing technique for LC mapping using MODIS data, using end-members that are extracted through Pixel Purity Index (PPI), Scatter plot and N-dimensional visualisation. The abundance maps were generated for agriculture, built up, forest, plantations, waste land/others and water bodies. The assessment of the results using ground truth and a LISS-III classified map shows 86% overall accuracy, suggesting the potential for broad-scale applicability of the technique with superspectral data for natural resource planning and inventory applications.**

***Index Terms*—Remote sensing, digital image processing, superspectral, Geographic Information System.**

I. Introduction

Land cover (LC) relates to the discernible Earth surface expressions, such as vegetation, geology, and hydrologic or anthropogenic features, and thus describes the Earth's physical state in terms of the natural environment and the man-made structures. Essentially, LC can have only one class or category at a given time and location, and can be mapped using suitable image data with spectral signatures. Land use is an expression of human uses of the landscape, e.g. for residential, commercial, or agricultural purposes, and has no spectral basis for its unique identification. Thus it can not be explicitly derived from image data, but only inferred by visual interpretation or assessed in the framework of object-based contextual analysis. It involves both the manner in which the biophysical attributes of the land are altered and the intent underlying that alteration, and the purpose for which the land is used. LC changes induced by human and natural processes play a major role in global as well as at regional scale patterns of the climate and biogeochemistry of the Earth system. Variations in topography, vegetation cover, and other physical characteristics of the land surface influence surface-atmosphere fluxes of sensible heat, latent heat, and momentum of heated air particulates caused by conduction, convection and radiation, which in turn influence weather and climate [1].

Many remote areas of the world are now being opened to exploration and development, generating a growing demand for up-to-date knowledge of topography, LC and other geo-spatial information. By using as many data sources as possible, a more complete and accurate knowledge of a landscape can be obtained. Therefore, users are seeking to integrate a multitude of spatially referenced information into their management and decision-making systems, a step that is facilitated by the standardisation of digital formats and the rapidly expanding market of GIS (Geographic Information System). There is a growing need for a global standardised LC and land use mapping system, similar to the CORINE approach. Many classification systems and innumerable map legends exist, but in most cases even from the same country, they are incompatible with each other. This system would enable a variety of end-users to use the results for their specific application (e.g. from rural planning to energy planning) and would enable intercomparison of existing data and a harmonised approach of data collection in areas where this information is not available or obsolete, whilst minimising the data and processing cost. Therefore, LC mapping remains an important research field, one what has grown more sophisticated with more recent technical developments in object oriented analysis or ontology [2], [3], [4].

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LC features such as vegetation, water, and soil are important components in regional planning and management for monitoring the dynamics associated with the Earth. LC changes induced by human and natural processes play a major role in the climate, hydrology and biogeochemistry at global as well as regional scales. LC features can be classified using remotely sensed satellite imagery of different spatial, spectral and temporal resolutions. Recently, LC mapping of the entire territory of China was done at a 1:1 million scale to understand the LC change, based on existing LC maps, field surveys, NOAA's (National Oceanic and Atmospheric Administration) AVHRR (Advanced Very High Resolution Radiometer) imagery and aerial photos. The CORINE LC map of the European Union includes 44 LC classes divided into 5 main categories (agricultural areas, artificial surfaces, forests & semi natural areas, wetlands and water bodies). It is based primarily on Landsat TM (Thematic Mapper bands 4, 5 and 7) data of different vegetation periods with additional information in the form of topographic maps and orthogonal photos [5], [6].

In India, spatial accounting and monitoring of LC have been carried out at a national level at 1:250,000 scale, using multi-temporal IRS AWiFS (Indian Remote Sensing Satellite Advanced Wide Field Sensor) with 4 bands (Green, Red, NIR and SWIR) at 56 m resolution to address the spatial and temporal variability in cropping patterns and other LC classes. A decision tree classifier method was adopted to account for the variability of temporal datasets [7].

The above attempts are based on monotemporal remote sensing (RS) data with the analysis being done on an annual basis. Monitoring LC dynamics with time series satellite data would not be economical for regional or national level mapping with commercial data such as IRS LISS (Linear Imaging Self Scanner)-III, LISS-IV, SPOT or Landsat TM/ETM+ (Enhanced Thematic Mapper plus). This imposes a major limitation on the use of such data despite their high spatial resolution. RS data such as ASTER (Advanced Spaceborne Thermal Emission and Reflection Radiometer) are inexpensive and have a better spatial resolution, but are not regularly available for whole regions. MODIS (Moderate Resolution Imaging Spectroradiometer) data, with a spatial resolution of 250m to 1 km, have better spectro-temporal resolution (7 bands, and composite-data with Level 3 processing and 36 bands every 8 days, or every 1-2 day availability with Level 1B processing) can be downloaded freely and may be suitable for regional mapping and planning activities in many developing countries. Their frequent availability is especially useful to account for seasonal variations and changes in LC pattern.

In order to obtain these LC types, remotely sensed data are classified by identifying the pixels according to user-specified categories, by allocating a pixel to the spectrally maximally "similar" class, which is expected to be the class of maximum occupancy within the pixel [8]. MODIS based LC mapping addresses large area coverage but is limited to classification of whole pixels [9], [10]. A variety of other classification methods exist, including spectral matched filter [11], mixture tune matched filtering and spectral angle mapper [12], which are appropriate when pixels do not contain mixtures of materials with correlated spectra, especially in higher spatial resolution data. However, MODIS pixels generally have the problem of spectral mixing. The mixed pixel problem is normally found at boundaries between two or more mapping units, or along gradients, etc. when the occurrence of any linear or small subpixel object takes place, and is usually dealt with by subpixel classifiers that assume either linear or non-linear mixtures [8] and [13]-[16]. It is often the case in RS that one wants to deal with identification, detection and quantification of fractions of the target materials for each pixel for diverse coverages in a region using unmixing approaches to discern the proportion of heterogeneity. A suitable way of extracting information is to estimate the composition of each pixel (proportion of the category contents) by spectral unmixing, i.e. soft classification techniques. The concepts of spectral unmixing emerged in the early 1970's [17], [18] and gained more prominence in 80's and 90's [8], [13]-[16] and [19]. During the last two decades, methods have been proposed ranging from modelling the component mixtures to solving the linear combinations to obtain abundances. Later, a number of techniques were developed to estimate and extract endmembers from the scenes or use spectral libraries.

In the present study, constrained linear spectral unmixing (CLSU) technique is applied on MODIS data to assess the suitability of the method for regional LC mapping with six LC classes. Linear Mixture Model (LMM), also known as a macro spectral mixture model, assumes no interaction between materials, and a pixel is treated as a linear combination of signatures resident in the pixel with relative concentrations [14]. This approach is different from the Spectral Angle Mapper (SAM) technique, where for the identification of pixel signature spectra only the angular information is used, which is based on the idea that an observed reflectance spectrum can be considered as a vector in a multidimensional space, and where the number of dimensions equals the number of spectral bands.

II. OBJECTIVE

Objectives of this study are: (i) LC mapping at regional level using superspectral MODIS data and (ii) to evaluate the suitability of CLSU to identify endmembers and generate abundance maps with MODIS (250 m spatial resolution).

III. ENDMEMBER EXTRACTION

Before modelling the linear mixture for unmixing, endmembers for the given study area have to be extracted. Various techniques, such as the Pixel Purity Index (PPI), Orasis (Optical real-time Adaptative Spectral Identification System), N-FINDR, Iterative Error Analysis (IEA), Convex Cone Analysis (CCA), Automated Morphological Endmember Extraction (AMEE) and Simulated Annealing Algorithm (SAA) have been developed to extract endmember spectra automatically from remotely sensed data [16], [20]-[23]. In general, each algorithm finds appropriate spectra for end-members. Orasis is capable of rapidly determining endmembers and unmixing large scenes. The N-FINDR method finds the set of pixels that define the simplex with the maximum volume potentially inscribed within the dataset. In the IEA algorithm, a series of constrained unmixing operation is performed, each time selecting as endmembers the pixels that minimize the remaining error in the unmixed image. CCA is based on the fact that some physical quantities, such as radiance and reflectance, are nonnegative. AMEE uses a morphological approach where spatial and spectral information are equally employed to derive endmembers. SAA is a method for constructing a simplex from a partition of the facets of the convex hull of a data cloud. The endmember extraction techniques used in this study are:

A. Pixel Purity Index (PPI)

This involves a dimensionality reduction using the Minimum Noise Fraction (MNF) transformation and the calculation of the PPI for each point in the image cube. This is accomplished by randomly generating lines in the N-dimensional space comprising a scatter plot of the MNF transformed data. All points in the space are then projected onto a line. After many repeated projections to different lines, those pixels above a certain threshold are declared "pure". There can be many redundant spectra in the pure pixel list. The actual endmember spectra are selected by a combination of review of the spectra themselves and through N-dimensional visualization. This provides an intuitive means to understand the spectral characteristics of materials [20].

B. Scatter Plot

A scatter plot with the set of scene spectra shows the endmember spectra occurring at the extremities (at the corners of the plot) [20]. In two dimensions, pure endmembers fall at the two ends of the mixing line, while in the case of a three endmember mix, mixed pixels fall inside a triangle, four endmembers fall inside a tetrahedron, etc.

C. N-Dimensional Visualization

Pixels from the spectral bands are loaded into an n-dimensional scatter plot and rotated on the visualization tool until points or extremities on the scatter plot are exposed. These projections are marked using a region of interest (ROI) tool and are repeatedly rotated in lesser dimensions to determine if their signatures are unique. Mean spectra are then extracted for each ROI to act as endmembers for spectral unmixing. These endmembers are then used for subsequent classification and other processing.

IV. Linear Spectral Unmixing

Models designed to estimate class proportions (rather than a single class label) for individual pixels, while addressing the problem of mixed pixels and considering the spectral response from a mixture of classes, are referred to as mixture models [24], [25]. Such spectral mixture (SM) modelling was used on resampled 25 m Landsat-7 ETM+ data for a subpixel classification that achieved 87% accuracy for DN (Digital Number) values and 93% for radiance values [26]. The technique is useful for discerning information from data with low spatial resolution, and would thus be ideal for free MODIS data with large ground coverage [27]. The spectral radiance measured by the MODIS sensor consists of the radiances reflected by all materials present, thus the radiance can be summed in proportion to the sub-pixel area covered by each material, given that the endmembers are the reference spectra of each of the individual pure materials, and under the condition that these spectra are linearly independent. The sites of pure LC for each class (or component) of interest are identified, and their spectra are used to define endmember signatures as discussed in section III. The position of the spectral signature of an input pixel along this continuum indicates directly the percentage cover for each component [14]. Constrained Least-Squares method (CLSM) aids in computing $n-1$ variables with n simultaneous equations [19]. In the case of MODIS, seven bands were designed for LC mapping [28] and hence the maximum number of LC categories that can be obtained is only six.

Linear unmixing (and its variants - Multiple Endmember Spectral Mixture Analysis) has been used earlier for LC mapping using MODIS surface reflectance data of 250 m and 500 m spatial resolutions of Northern Africa [29] and the unmixed results were compared with high resolution classified maps that gave an overall classification accuracy of 54% with significant confusion between alluvial surfaces and regs (surface covering of coarse gravel/pebbles or boulders from which all sand and dust have been removed by wind and water; a stony desert), and between sandy and clayey surfaces and dunes. A second validation using 20 Landsat images in a stratified sampling scheme gave a classification accuracy of 70%, with confusion between dunes and sand sheets.

LC fractions derived from MODIS 1 km resolution data and MISR (Multi-angle Imaging SpectroRadiometer) using SM have been compared to results of a Bayesian-regularized artificial neural network (ANN), as well as with 30 m reflectance data, yielding a quantitative improvement over spectral unmixing of single-angle, multispectral data (Landsat-7/ETM+) [30].

V. MATERIALS AND METHODS

A. Data

MODIS data with atmospheric corrections, also known as the "MOD 09 Surface Reflectance 8-day L3 global" product at 250 m (band 1 and 2) and 500 m (bands 3 to 7) in Hierarchical Data Format (HDF), were downloaded from the Earth Observing System Data Gateway [31]. They are radiometrically corrected, fully calibrated and geolocated radiances at-aperture for all MODIS spectral bands and are processed to Level 3G. The spectral range is from 0.45 to 2.15 µm [28]. IRS - 1C/D LISS-III MSS (Multi Spectral Scanner) data in 3 bands (G, R and NIR, 0.52 to 0.86 µm) with a spatial resolution of 23.5 m were purchased from NRSA (National Remote Sensing Agency), Hyderabad.

B. Study area

The Kolar district in Karnataka State, India, located in the southern plain regions (semi arid agro-climatic zone) and extending over an area of 8238 km^2 between 77°21' to 78°35' E and 12°46' to 13°58' N, was chosen for this study (Fig. 1). Kolar is divided into 11 taluks for administration purposes. Rainfall occurs mainly during southwest and northeast monsoon seasons. The average population density of the district is about 209 persons/km^2 [32]. The study area is mainly dominated by agricultural land, built up (urban/rural), evergreen/semi-evergreen forest, plantations/orchards, waste lands and water bodies. There are a few other LC classes (barren/rock/stone/others) that have very limited ground area proportions and are unevenly scattered among the major six classes, and were grouped under the waste land category.

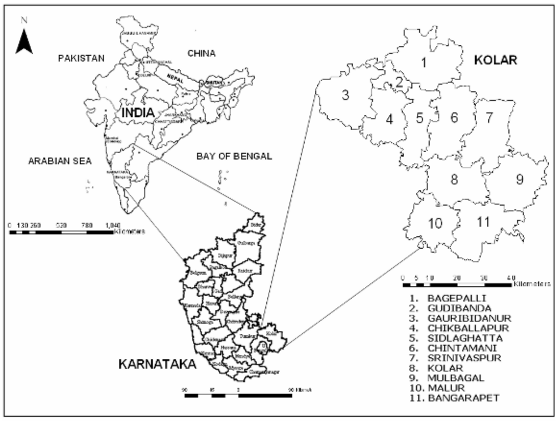

Fig. 1. Study area – Kolar district, Karnataka State, India.

C. Methods

The study involved creation of base layers, such as district, taluk and village boundaries, road network, etc. from the Survey of India (SOI) topographic maps of scale 1:250000 and 1:50000. The LISS-III bands were geo-registered using ground control points (GCPs). LISS-III bands were resampled to 25 m, which helped in pixel level comparison of abundance maps obtained by unmixing MODIS data with LISS-III classified map. This was followed by cropping and mosaicing of data corresponding to the study area from the image scenes. Supervised classification was performed on LISS-III MSS data using a Gaussian Maximum Likelihood Classifier (GMLC) followed by accuracy assessment. The MODIS data were geo-corrected with an error of 7 meters with respect to LISS-III images. The 500 m resolution bands 3 to 7 were resampled to 250 m using nearest neighbourhood technique (with Polyconic projection and Evrst 1956 as the datum). Minimum Noise Fraction (MNF) components were derived from the 7 bands to reduce noise and computational requirements for subsequent processing. Endmembers were extracted directly from the data without using existing spectral libraries through: (a) Pixel Purity Index (PPI) with the MNF components, (b) Scatter Plot, (c) N-Dimensional Visualization and (d) Collected training data - training polygons (≥ 250 x 250 m) of homogenous patches corresponding to MODIS pure pixels were collected in the study area, thus enabling direct selection of assumed pure pixels from the images. The spectral characteristics of the endmembers were analysed by plotting them and analysing their separability using a Transformed Divergence matrix. It shows that the endmembers selected for the analysis are separable and can be distinguished from each other. The abundance maps were generated via constrained linear unmixing of MODIS data. Accuracy assessment was done for the abundance maps: LC percentages were compared at boundary level and at pixel level with a LISS-III classified map and also with ground truth data which is discussed later (Accuracy assessment).

VI. RESULTS

A. LC analysis using LISS-III MSS data

The class spectral characteristics for six LC categories using LISS-III MSS bands 2, 3 and 4 were obtained from the training pixels spectra to assess their inter-class separability and the images were classified with training data uniformly distributed over the study area collected with pre calibrated GPS (Fig. 2). This was validated with the representative field data (training sets collected covering the entire district and a detailed validation in Chikballapur Taluk covering ~ 15% of the study area) and the LC statistics are given in Table I. Producer's, user's, and overall accuracy computed are listed in Table II. A

TABLE I
LAND COVER DETAILS USING LISS -III MSS

Class	Area (%)
agriculture	19.03
built up land (urban/rural)	17.13
evergreen /semi-evergreen forest	11.41
plantations/orchards	10.96
wasteland/others: barren/rocky/stony	40.39
water bodies	01.08
Total (%)	100.00

TABLE II
ACCURACY OF LC CLASSIFICATION USING LISS-III MSS FOR CHIKBALLAPUR TALUK

Category	Producer's Accuracy (%)	User's Accuracy (%)	Overall accuracy
agriculture	94.21	84.54	95.63%
built up	96.47	83.11	
forest	94.73	96.20	
plantation	92.27	91.73	
waste land/others	97.49	89.88	
water bodies	96.13	98.33	

kappa (k) statistics of 0.95 was obtained indicating that the classified outputs are in good agreement with the ground conditions to the extent of 95%.

Classification errors can occur when the signal of a pixel is ambiguous, perhaps as a result of spectral mixing, or due to overlap of spectral reflectance (as in the case of certain agriculture and horticulture crops) or when the signal is produced by a cover type that is not accounted for in the training process. Another possible source of error may be due to the temporal difference in training data collection and image acquisition.

B. Spectral unmixing of MODIS data

Here each pixel spectrum of an image is modeled as a linear combination of a finite set of known components (or endmembers) given by (1)

$$r_i = \sum_{j=1}^{n} (a_{ij} x_{ij}) + e_i \qquad (1)$$

where, r_i = Spectral reflectance of a pixel in i^{th} spectral band containing one or more components; a_{ij} = Spectral reflectance of the j^{th} component in the pixel for i^{th} spectral band; x_j = Proportion value of the j^{th} component in the pixel; e_i = Error term for the i^{th} spectral band; j = 1, 2, 3, ... n (Number of components assumed; in this study n=6); i = 1, 2, 3, ... m (Number of spectral bands for the sensor system; in case of MODIS m=7). In principle, according to constrained assumptions, the proportions of abundance for each class range between 0 and 1, are non-negative and add to one as give by (2).

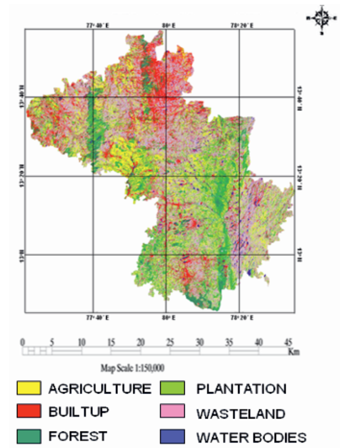

Fig. 2. Supervised classification of high resolution LISS-III image.

$$\sum_{j=1}^{n} (x_j) = 1 \qquad (2)$$

Visual inspection as well as accuracy assessment of abundance images corresponding to each endmember showed that the CLSU algorithm maps LC categories spatially and proportionally similar to supervised classified image of LISS-III. The proportions of the endmembers in these images (Fig. 3) range from 0 to 1, with 0 indicating absence of the endmember and increasing value showing higher abundance.

Bright pixels represent higher abundance of 50% or more stretched from black to white. Errors were found due to confusion between agriculture and horticulture (plantation) in the central regions and built up and wasteland in northern regions of the district. Overall, the distribution and abundance values of other classes are comparable within ±6 % to the classified outputs of LISS-III. There are many areas where proportions of agriculture are properly identified, mainly in the western central portion of the study area. However, there is also underestimation of agriculture in the south-central region due to errors of commission and omission.

C. Accuracy assessment

1) Comparison with LISS-III MSS: Table III provides the difference in LC percentage between the abundance maps generated taluk wise using CLSU and classified LISS-III data. Hence, further analysis was carried out at pixel level to understand the sources of these differences.

2) Pixel level assessment: A pixel of MODIS (250 m) corresponds to a kernel of 10 x 10 pixels of LISS-III spatially (Fig. 4). Endmembers generated along with abundance maps were validated pixel by pixel with the

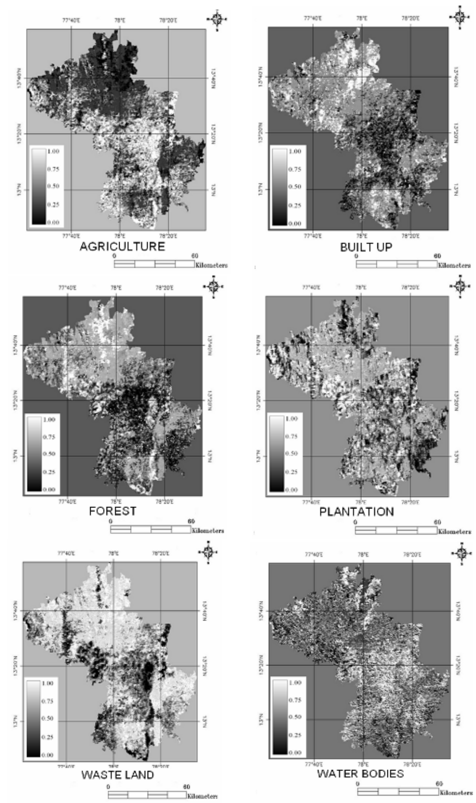

Fig. 3. Gray scale abundance maps for agriculture, built up, evergreen/semi-evergreen forest, plantations/orchards, wasteland and water bodies.

TABLE III
LC AREA DIFFERENCE (IN %) BETWEEN FRACTION IMAGES OBTAINED FROM MODIS DATA AND LISS-III DATA

Taluk	agri-culture	built up	forest	plantation	waste land	water bodies
1	+0.4	-4.3	+0.2	+1.4	-1.5	+3.7
2	+0.2	0.0	-2.4	-0.3	+0.2	+2.1
3	-2.8	+0.4	-0.7	+0.5	+0.8	+1.5
4	-4.4	-4.2	+3.3	0.0	+4.2	+1.2
5	-0.9	-0.7	+0.2	+2.0	-3.6	+3.0
6	-3.5	+1.9	+3.7	+2.4	-5.9	+1.4
7	+3.4	-3.8	+4.1	-2.4	-4.1	+2.8
8	-0.4	-2.5	+0.8	+0.8	0.0	+1.1
9	-1.0	+1.4	+1.9	+0.6	-3.4	+0.5
10	-2.9	-0.7	+1.5	+1.5	-1.9	+2.3
11	+1.2	-0.4	-2.7	+2.9	+3.4	+3.2
District	-2.0	-1.3	-1.8	-1.7	+4.8	+2.0

1. Bagepalli, 2. Bangarpet, 3. Chikballapur, 4. Chintamani, 5. Gauribidanur, 6. Gudibanda, 7. Kolar, 8. Malur, 9. Mulbagal, 10. Sidlaghatta, 11. Srinivaspur

classified LISS-III image for Chikballapur taluk. Endmember for a particular category was considered as pure when its abundance value was greater than 90%.

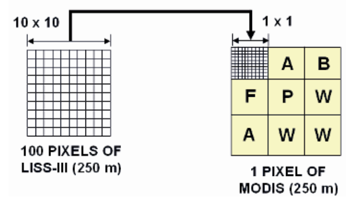

Fig. 4. 10 x 10 pixels of LISS-III equals to 1 x 1 pixel of MODIS spatially.

A total of 51 pixels with respect to the six LC classes were verified, and the result is tabulated in Table IV. The validation indicates that 35 pure and 14 mixed were correctly classified. Two pixels were misclassified; belong to water bodies which constitute < 1% of the total area.

3) Assessment based on the field data of proportion LC: The abundance maps were also verified through field investigations using calibrated (with known benchmarks and digital SOI topographic maps) GPS. In the case of mixed pixels, proportional LC were mapped and overlaid on the abundance map, and producer's and user's accuracy (Tables V and VI), overall accuracy and the kappa statistics were computed.

VII. Discussion

Sustainable management of natural resources entails spatio-temporal information related to various surface features such as vegetation, water, and soil. In this regard, LC mapping using temporal RS data helps in capturing the dynamics associated with these LC components [33]. In principle, LC mapping can be done with any multispectral dataset, but currently available data have limitations such as poor spatial and temporal resolutions, or high spatial but low spectral resolution (IKONOS PAN, CARTOSAT-II) or moderate spatial and spectral resolutions (IRS LISS-III/IV, SPOT, Landsat TM/ETM+) as well as limited coverage, reduce the suitability of the available data types. Most of these RS data are commercial and many scenes would be required if mapping is to be carried out for larger regions. Even though some of the data (such as ASTER) are free, they are not available for all parts of the world and at variable temporal resolution. In this context, high temporal resolution MODIS data are freely available and are useful in providing insights into LC dynamics.

The present study evaluated the scope of CLSU technique for LC mapping using low spatial resolution MODIS data. Endmembers and abundance maps for the six LC categories were generated using MODIS data through linear unmixing. The proportion of the classes in each pixel of MODIS was computed and compared with the classified image of better spatial resolution and also through field data.

TABLE IV
VALIDATION OF LC CLASSES IN CHIKBALLAPUR

Class	Number of pixels identified	Pure pixels	Mixed pixels	Wrongly classified
agriculture	23	18	5	0
built up	6	4	2	0
forest	2	1	1	0
plantation	11	7	4	0
waste land	6	4	2	0
Water	3	1	0	2
Total	51	35	14	2

TABLE V
PRODUCER'S ACCURACY (IN PERCENTAGE)

Percentage→ Classes ↓	20 %	40 %	60 %	80 %	100 %
agriculture	50	83	100	100	100
built up	100	50	-	-	100
forest	100	100	100	-	100
plantation	63	88	100	100	100
waste land	33	100	33	100	100
water bodies	100	100	-	-	100

TABLE VI
USER'S ACCURACY (IN PERCENTAGE)

Percentage→ Classes ↓	20 %	40 %	60 %	80 %	100 %
agriculture	66	100	100	100	100
built up	50	100	-	-	100
forest	60	75	100	-	100
plantation	63	88	100	100	100
waste land	50	75	100	100	100
water bodies	100	100	-	-	100

Overall Accuracy = 86 %
Kappa (KHAT value) = 85 %

The results show that five of the six identified classes exhibit high interclass variability, allowing linear spectral unmixing. The variability of the pixels that represent the local pure classes is also responsible for the uncertainity of the mixture proportions. Probability density functions and parametric representation of the constituent pure classes did represent local conditions evident from the post classification validation. Assigning the categories to endmembers in consultation with the field data proved to be critical as it helped to discern the adjacency effect that exists between contrasting features such as forest and plantation, agriculture and plantation, etc. [34], [35]. This highlights that LMM, with improved estimates of proportional abundance values, has greater scope compared to other hard classification techniques in handling coarse spatial resolution data. The spectra of the different endmembers were modelled by linear equations. This introduces another form of quality assurance, as these equations can be inverted, taking the endmember spectra to reconstruct the spectra of each pixel from the original image. Divergences between the original and reconstructed image then give idea of the goodness of fit of the model, and provide insights to which bands add more to the errors. When no more recognizable patterns are found and the error is overall small, it can be deduced that a near perfect model has been reached.

Nevertheless, even though the number of spectral bands in MODIS is higher compared to LISS-III, with the CLSM [36], only six categories can be discerned (number of bands available minus one). The condition of identifiability (the number of classes should be one less than the number of bands) can be solved by a two step process provided that many spectral endmembers are available. A subset with a prefixed number of endmembers that optimally decompose the candidate pixel is first selected by a Gram-Schmidt orthogonalisation process [37]. This restricted subset is then used in conventional LMM. The final result is the decomposition of the scence into all end-members considered while reducing the residual errors. Also, this approach fails in heavily fragmented landscapes or small isolated areas of high contrasting nature. This can be addressed through image fusion techniques [38] of low spatial resolution (MODIS) with high resolution (LISS-III/Landsat-TM/ETM+). To account for contributions from the neighbourhood pixels at the same time, non-linear mixture models are suitable for unmixing of coarse spatial resolution data.

VIII. Conclusion

Spectral linear mixture analysis provides an efficient mechanism for the classification of superspectral RS data (e.g. from MODIS). It aims to identify a set of reference signatures (endmembers) that can be used to model the reflectance spectrum at each pixel of the original image. These endmembers are extracted from the images using techniques such as PPI or scatter plots. Thus, the modelling is carried out as a linear combination of a finite number of ground components and their reflectance spectra. An abundance map helps in estimating the proportion of each endmember in a pixel. The performance of linear unmixing technique for identification of endmembers, interclass variability and presence of adjacency effect using 250-500 m MODIS data was evaluated. MODIS LC outputs are comparable to LISS-III classification results, evident from validation, as most endmembers were correctly classified while mixed pixels were within 10-15% of the values obtained in LISS-III classification. The MODIS data tested here have an advantage over alternative high or medium resolution datasets, given their coverage, high temporal resolution, cost-free availability, and utility for LC mapping as shown in this study. The challenges associated with this data type are (i) georegistration of the pixels when no or limited identifiable GCPs exist, (ii) misclassification of pixels due to LC mixtures, and (iii) mapping of LC in heavily fragmented areas with highly contrasting nature. One way of overcoming problems (ii) and (iii) is to use MODIS data with high spatial resolution images. Even, in case MODIS ceases to function, the technique would still be applicable for datasets with low spatial resolution.

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Extending GridNexus to Support the Visual Assembly of Web Services

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Abstract - ***This paper describes a set of extensions (primitives, bindings and graphical modules) and their implementation to facilitate the visual assembly of web services from scientific workflows within the GridNexus computing environment. The implementation approach for this work involved employing a strategy for defining WSDL message types and procedures such that each will be bound to a defined workflow for evaluation. Because of its Axis2 foundation, web services produced by GridNexus can run under any J2EE application container by providing the service as a deployable archive file. The paper concludes with a discussion of various design alternatives including the advantages and disadvantages of each.***

I. Background and Motivation

An identified need in the user community provided the motivation for this project. GridNexus[1] users expressed a need to visually assemble web services as this would greatly facilitate the creation and sharing of their scientific workflows. An evaluation of existing web service toolkits highlighted the need to provide researchers with the ability to create services themselves. It was determined that GridNexus was an ideal and extensible platform to provide this functionality.

GridNexus was created at the University of North Carolina Wilmington as an extension of Berkeley's Ptolemy project [1]. GridNexus supports the creation of scientific workflows in an easy to use graphical environment. GridNexus enables users to orchestrate grid services, web services, and more using simple configurable drag and drop boxes [2]. Using these boxes, users can develop and run complex processes, called workflows, without having to concern themselves with the syntactical details of code implementation. The GridNexus GUI acts as a graphical front end that creates script in an underlying XML based scripting language known as JXPL which is responsible for the workflow execution [3, 4].

In instances where remote JXPL execution is required, GridNexus generates a JXPL script that is sent to a remotely bound JXPL engine. These engines are pre-bound using RMI, HTTP, or the JXPL grid service. This solution makes

[1]The GridNexus Project web site: http://www.gridnexus.org/

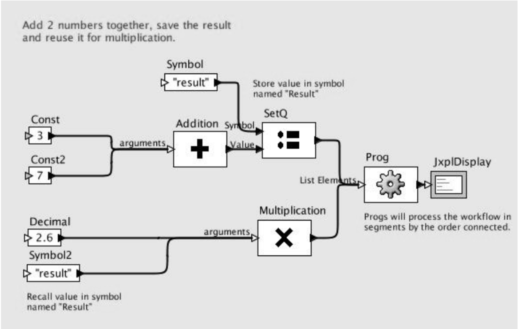

Fig. 1. Sample GridNexus Workflow

distributing a JXPL process easy, but is less than ideal when making the workflow available beyond GridNexus itself. In this paradigm, the client must be capable of dictating the script that the remote engine is to execute. Clearly, providing this level of strict control over the remote engine may not always be in the best interest of an organizations security practices as it may be possible to execute any arbitrary script or command on the remote system if not configured properly.

While working with the GridNexus user community, it became clear that being able to visually define and create a web service from within the GridNexus graphical editor would greatly facilitate the development of scientific workflows. Although, GridNexus has the capability of producing workflows that can do a variety of different tasks, before a client can use a workflow it must also run within the GridNexus environment. In many instances, users expressed interest in publishing the workflow as a stand-along fully functioning web service. This would better facilitate workflows by adding three key benefits: security, accessibility and maintainability.

II. Related Work

A. JOpera

JOpera is a similar workflow project developed by a team at the Swiss Institute of Technology in Zurich [5]. Like GridNexus, JOpera is a tool for rapid development of workflow processes incorporating a runtime environment and

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 424–428.

Table 1. Benefits of Publishing Workflows as Web Services

Benefit	Explanation
Security	Hide details such as paths. Prevent users from changing workflow details.
Accessibility	Make available to any Web service enabled client, .Net, Java, C++, etc.
Maintainability	If sharing workflow, create one endpoint which would impact all users.

a graphical editor (JOpera).

JOpera is designed as a plug-in for Eclipse and relies on existing frameworks such as the GMF (Graphical Modeling Framework) plug-in and many custom written modules. At runtime JOpera compiles Java byte code incrementally during execution [5].

B. OMII BPEL

Open Middleware Infrastructure Institute (OMII) Business Process Execution Language (BPEL) is built upon two open source projects that have been developed for hosting BPEL processes. It utilizes both the Eclipse BPEL Project[2], and the open source ActiveBPEL[3] engine. Because these tools were being used to target business processes, the OMII group decided to extend and tie these two together in an attempt to use BPEL for scientific uses.

C. Problems and Weakness

JOpera may be robust and flexible but it is less than ideal for a non-programmer end user. In order to include a web service, for example, the user must go through a myriad of menus to create and configure the web service as a program. First the user has to go and import a WSDL file found on the Internet. Next the user must return to JOpera and create a new process in the process menu from which the service will actually be defined. The user must then select "edit" process to arrive at a screen displaying options for naming and defining parameters for the "process." After this is complete, the user will be taken to a "data flow" graph containing components corresponding to the process and any programs that may have been defined. The purpose of which is to specify the path in which data will travel throughout the system. The user must expand each input/output property from each of the components and connect them accordingly. If the user wishes to establish conditional control this must be accomplished in yet another graph called the "control flow."

With OMII BPEL, the BPEL Editor and ActiveBPEL together have key weaknesses. First, the BPEL editor is a plug-in designed for Eclipse 3.2 or greater. Although this is a great strength it also is a weakness. For new users and non-programmers, using Eclipse involves a bit of a learning curve getting used to "perspectives" and figuring out which features they need and which to ignore. Aside from the learning curve associated with Eclipse, the BPEL editor is designed to be abstract and generate BPEL in general and as of June 2007 does not provide any explicit ties with any BPEL Engines. This is good in that it is loosely coupled and universal. However, this is bad when you want to make it work with a specific engine such as ActiveBPEL [6]. In order to make it work with ActiveBPEL, the user must provide additional XML configuration files to run the service. The OMII BPEL solution has made some strides at overcoming this deficiency by providing better linkage between the two. However, the combination does have another weakness – BPEL can only consume web services. This may be a limiting factor for researchers who frequently need to incorporate specialized applications or utilities within their service.

III. Extending GridNexus

A. Methodologies

Publishing workflows require three key components. First, a means of describing workflows as procedure calls needs to be defined. Second, a means of representing these calls with a WSDL interface needs to be described. Last, a method in which the web service container can connect the web service invocation to the JXPL Script needs to be implemented.

During analysis, two general approaches were identified for publishing JXPL workflows as a web service. The methods that were identified are: (a) the Wrapper Approach and (b) the Dynamic Approach. Although each of these methods has its own strengths and weaknesses, the Wrapper Approach was ultimately chosen for implementation due to its ease of implementation and because it required no modification of the Axis2 API.

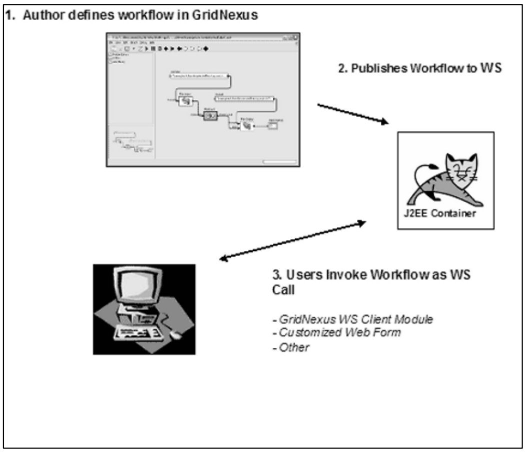

Fig. 2. Workflow Publication

[2]BPEL for Eclipse project web site: http://www.eclipse.org/bpel/
[3]Active Endpoints web site: http://www.active-endpoints.com/

Wrapper Approach

In the wrapper approach, a special class would be generated that would wrap all necessary functionality into a source file and then need to be compiled before it could be deployed. Although simple, this approach requires the presence of a compiler and involves auto generating and compiling new source code. This approach would place most of the effort of creating a mapping up front. The weakness of this approach is the need for dynamically compiling code. The WSDL would be generated dynamically from the wrappers template.

Dynamic Approach

In the dynamic approach, no new source code needs to be generated and everything can be done with precompiled code. However what needs to be generated with this approach are the WSDL documents for each prospective service. Binding in this scenario would be accomplished by custom Axis2 message receivers that in turn map the incoming SOAP messages into JXPL requests. The custom receiver would sever the typical ImplClass / Message Receiver relationship to create a JXPL Processor / Message Receiver relationship.

This approach is advantageous in that the JXPL Engine need only be concerned with generating XML (as it itself is an XML language) instead of building new wrapper classes, the Axis receiver itself can then negotiate the mapping.

However, because this approach lacks an ImplClass there is no way of utilizing the auto WSDL generator included with the Axis2 API. At time of this writing the WSDL generator can only generate a WSDL if provided a compiled Java Class. This means that a JXPL to WSDL generator would have to be created. The WSDL is a very complicated document and would add unneeded complexity to the implementation.

B. JXPL Extensions

To implement the Web services infrastructure, a set of primitives have been included to describe the service, as well as the operations and parameter types that are associated with the service.

SOAPService

The SOAP Service actor is the primary actor that describes the SOAP Service. Properties associated with the SOAP Service include “Name”, “Namespace”, and a toggle determining whether or not the service is going to be packaged for deployment or run locally. At runtime, if the service is to be run locally, it will block further execution of the thread until the thread is shutdown by the user. On remote JXPL processors, this mode of operation will not permitted unless specified in the policy file. This primitive will return the JxplSymbol “true”. The ultimate task of this primitive is to ensure that the service is defined and generated properly.

SOAPOperation

The SOAPOperation is the primitive that provides the executable portions of the SOAP Service. JXPL already has a function called “Defun” that enables the user to define functions. SOAPOperation elaborates on the same idea as “Defun.” The difference with SOAPOperation is an expanded set of parameters that enable it to specify the “type” of parameters the operation bindings will accept and return. To make it easier to use, the service will specify parameters based upon an example. A type map will convert the JXPL types to the appropriate interface types at runtime.

Figure 3 shows an example of a workflow published as a simple web service. The workflow will accept a single integer and double its value. The Soap Service module accepts two inputs in the editor, a “SoapOperation” (defining business logic, parameters and return type), and a location in which to store the file. Upon execution of this script the archive will be written to the file “DoublerService.aar” which may then be deployed in a J2EE application container.

IV. Evaluation and Discussion

A. Runtime Differences

GridNexus and JOpera are alike in goal but dissimilar in how they approach it. JOpera contains a modeling language but for the executable runtime code relies on compiling Java byte code through the framework provided by Eclipse. In comparison, GridNexus borrows its modeling language from the Ptolemy front end. At runtime, the GUI will generate an XML runtime JXPL script to facilitate the execution of the workflow, as opposed to byte code [7].

At runtime the BPEL script is deployed on the server and compiled as a web service. Evaluation is performed by a special engine that is designed to parse the BPEL XML script from Eclipse and bind it to executable byte code. Afterwards, the script can be invoked as a typical web service. GridNexus is very similar in that JXPL gets bound and interpreted as a function call. However, one difference with JXPL is that after being deployed as a web service, it is not bound to byte code until first invoked, when the JXPL Processor is invoked. In comparison, the BPEL engine is bound to byte code during deployment.

B. Creation Tool Differences

JOpera has many robust options for creating workflows. Many of which are based around constructs found in the Eclipse Development environment. As such, many of these options require going through a set of configuration screens to get there.

By contrast, GridNexus, while not based on Eclipse, contains a set of modules that are already available to perform trivial tasks. Instead of having to go through a variety of menus, GridNexus is driven by a dialogue box accessed from double clicking on an actor. Thus eliminating the need to go through a sequence of views to establish the configuration the user wants. The only time a user needs to concern themselves with multiple views of a workflow is if the user opts to compartmentalize the visual workflow into composites.

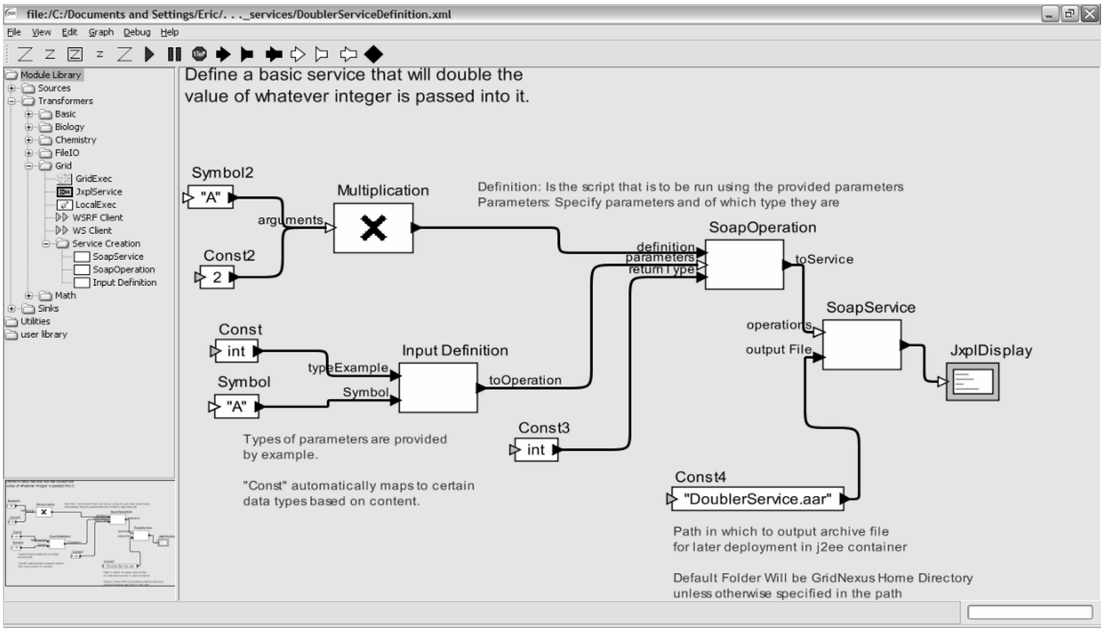

Fig. 3. Example of a GridNexus Workflow Published as a Web Service

In JOpera, to create web service modules, the user must go under the "New" menu to add a "WSDL" for each of the Web Services, each time running through a series of steps required to import them as program modules to be used in the workflow. After the program modules have been created, the user must then switch to the processes Dataflow graph. The user will first import each of the modules required. Second the user must expand each module to reveal its input and output components and connect them accordingly.

Because the Dataflow graph doesn't include any conditional support, JOpera utilizes a separate graph which manages Control Flow. In the Control Flow graph each module is either going to be true or false (on or off). If true (or switched to "true" by an event) the workflow will be required to execute the module upon receiving data. Otherwise, the module will be skipped.

GridNexus does not separate the Dataflow from the Control flow. Instead control is determined by a set of primitives designed for handling conditionals (behaving much like a switch statement). Another key difference is that, while JOpera creates modules by going through a set of menus, GridNexus creates them by including a pre-created library of adjustable modules for a variety of different tasks.

Like JOpera, OMII also takes advantage of Eclipse for its front end. However being based on the BPEL Project, it contains a set of predefined modules that correspond to specific functions found within the BPEL language. Creating connections to web services involves using an import wizard to grab the WSDL's and binding them to "Partner Links" before they can be used in the web service. "Partner Links" are then used to define which web service to invoke from within the BPEL script (defined in the modules' properties).

The idea of the BPEL Project to include predefined modules is similar to that of GridNexus. For conditional support, the BPEL Project even includes a switch statement from which to make decisions and include splits in the path of the workflow. However, the BPEL Project utilizes XPath statements to evaluate logic, whereas GridNexus includes special modules for logical comparisons.

V. Summary and Conclusions

Adding the ability to publish workflows as web services helps make GridNexus a more powerful and valuable tool. Now GridNexus can not only be used to orchestrate, but also to share web services easily and securely. Additionally, building GridNexus' WS support on the Axis2 ensures that GridNexus can easily be adapted to support many of the latest features and expansions developed for the Axis2 web services stack. Table 2 summarizes the results of the comparisons between the different software toolkits discussed throughout the paper.

Upon completion of the project, a variety of extensions were introduced into JXPL for creating Web Services directly from an XML script. To accompany these changes three GUI modules were created to enable a simplistic approach for defining a service. The *Soap Service*, *Soap Operation* and *Input Definition* components provide a framework to make service creation almost trivial for the user. Furthermore, the framework is extensible as it can quickly take advantage of any add-on provided for Axis2 with little to no modification necessary.

Table 2. Comparison of GridNexus, OMII BPEL and JOpera

Issue	GridNexus	OMII BPEL	JOpera
Extensibility	*Very Good* New primitive and add to classpath. Invoke using qualified name.	*Bad* Active Endpoints Engine does not yet implement the extension tags specified in BPEL 2.0.	*Good* Java Snippets offer limited extensibility. Ability to add new frameworks.
Portability	*Very Good* Deploy in any axis2 enabled J2EE container.	*Good* Deploy wherever Active Endpoints is installed.	*Limited* Generates its own modified web service environment. Based on older Axis stack.
Installation	*Very Good* Only requires Java SDK 1.5 or greater.	*Average* Requires Eclipse with proper dependencies.	*Moderate* Requires Eclipse with proper dependencies.
Graphical Interface Usability	*Very Good* Pre-built Drag and Drop components.	*Moderate* Pre-built drag and drop components. Eclipse can be challenging for new users and non-programmers.	*Bad* Not a true drag and drop system. Eclipse can be challenging for new users and non-programmers.
Flexibility (Multitude of Tasks)	*Very Good* Supports: Local Apps, Web Services, Grid Services, Remote Apps Logical Operations	*Bad* Only supports Web Services.	*Very Good* Supports: Local Apps, Web Services, Grid Services, Remote Apps
Robustness	*Limited* Needs support for complex types.	*Good* Uses framework from industry	*Good*

In addition to these changes, it was necessary to also overhaul the original Generic WS Client in GridNexus to support the newer Axis2 web services stack, where it had originally supported Axis 1.x. Finally, all the necessary documentation and another version of GridNexus (version 2.01) was produced and published as Open Source[4].

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Controlling Electrical Appliances Using an Open Source Technology

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Abstract— **This paper discusses the development of a universal remote control system in a house. It utilizes the use of Bluetooth and Java technology on mobile phones in controlling electrical appliances. The aim of this universal remote control is to eliminate the need for having many remote controls at home in view of the fact that almost all electrical devices and appliances come with their own proprietary remote control. This system also expands the capability of the IrDA remote control technology since typical IrDA remote controls does not confine to the physical barriers. The developed universal remote control consists of two categories, which are client application that involves a mobile user interface and mobile phone for Bluetooth communication that interacts with the circuit where the electronic components reside. It applies immediate direct manipulation of the circuit to implement a real time communication and update the states of appliances as soon as it is been modified. In a nutshell, the end product gives flexibility to the users by promoting the usage of their mobile phones as a universal remote control for their electrical appliances.**

Keywords- Universal remote control, J2ME

I. Introduction

Home automation can be defined as automating the requirements of private homes. It can be as simple as remote or automatic controlling of a few lights [1] or as intelligent as turning a home air conditioner to an energy saving device when the house is empty and putting it back to the desired temperature when the house is occupied [2]. Remote controls have been used widely in controlling these home devices and throughout the years, the usage of remote control is vastly expanded as well as the technology behind it. However, almost all electrical appliances come with their own set of remote controls. Thus, the consumers are bombarded with remote controls for their televisions, audio systems, air-conditioners, security systems and even toys.

Therefore, instead of having many modular remote controls that only control one device, having a universal remote control would be a great help in reducing the amount of remote controls.

This project is about turning a Bluetooth-enabled mobile phone into a universal remote control that is able to control all electrical appliances in a single house. Bluetooth is chosen primarily because this technology can be found easily in mobile phones and mobile phones themselves are widely being used [9]. One of the great advantages in Bluetooth lies in the huge support in all kind of devices. It is supported in USB dongles to ordinary PCs, PDAs, mobile phones and other embedded devices. Also, it provides advantage over other technologies such as IrDA, Home Radio Frequency (RF) and Wireless LAN [4].

The developed application requires the users to connect to a Bluetooth server using their mobile phone, and manipulate the logic switch from on state to off state, and vice-versa by using piconet topology. The server and client are assigned with a unique UUID that allow communication between them.

By having this system implemented in households, mobile phone owners are able to expand their phone capabilities by having to control as many devices and electrical appliances as possible. This will then reduce the need of having one remote control for every electrical appliance available in the house

II. Background and Related Work

There has been various significant home automation system using remote controls being developed. An example of a project in [5] is a prototype of such new idea of home automation. The project focuses on replacing the legacy technology of a remote control which is the Infrared technology with the Bluetooth technology. The mobile application utilizes the use of Java 2 Micro Edition (J2ME) on a Nokia Series 60 phone. The project was successful with such native interfaces for both server and client applications. However, the project was conducted by manipulating only a single lamp. The assumption made is by controlling a single electrical appliance will prove that other appliances are able to be controlled by expanding the circuitry. Besides that, the mobile application is not user-friendly enough for the users. It was mainly because of the project's initial objective is to prove the concept of Bluetooth remote control. As for the operating

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 429–434.

system, the author was using Windows XP SP2 to host the Bluetooth server. Communicating directly to the I/O port is not fully recommended by Microsoft in that release. Thus, the author needed to run a couple of other applications prior to running the Bluetooth server. This is to allow the server to send I/O command and receive status message from the parallel port.

The project also indirectly suggests the use of built-in parallel port available on CPUs and laptops instead of using expansion slots such as PCI cards. This is due to the port base address constraint found in UserPort software, an application that is needed to be run prior to the Bluetooth server. This reduces the convenience towards users who do have any built-in parallel port on their system.

The other related project in [6] introduces another way of remotely controlling devices with the use of Bluetooth and web page. The web page was design using Java language and contains the list of appliances that can be controlled. The web page is also able to provide the current state of those appliances to the users. The project also suggested the usage of Bluetooth chips on separate devices for them to be able to be linked in Scatternet form. Since the project uses web page to control the appliances, it reduces the mobility of the users in controlling the appliances because it means that the users are expected to control their appliances from in front of the computer. Other than that, the usage of Bluetooth chip on every device is not cost effective from the users' point of view. Users have to bear with the high cost of implementation if they decide to adopt this system.

In [7], the home automation system is designed to be controlled by the users when they are not at home. By using mobile phones, users can control the system using SMS, WAP or other mobile development technologies. These technologies are the intermediary between the users and the PC which in turn controls the devices attached to X10 modules. The PC will then return the status of every remote request that the users have made. Having this kind of approach is not really efficient and user-friendly as it will burden the mobile phone users through the SMS charges besides having to spend more in buying the X10 product adapters.

III. System Architecture

The architecture of the home automation system is shown in Figure 1. A J2ME application is installed in the mobile phone as the interface for the remote control. The mobile phone then communicates with the server using Bluetooth. The server, which has a Bluetooth dongle, receives the signal from the mobile phone and then control the parallel port pins via System interface. Before that, it returns the status of pins back to the mobile phone in order to let the users know the state of the appliances, whether it is in ON or OFF mode. When the users decide to change the mode, the mobile phone will send the command signal back to the server. Then, the server updates the parallel port pin with the new data bit and send it to the circuit interface. When the operation is done, the server then updates the status and notifies the users of the status of the particular appliance.

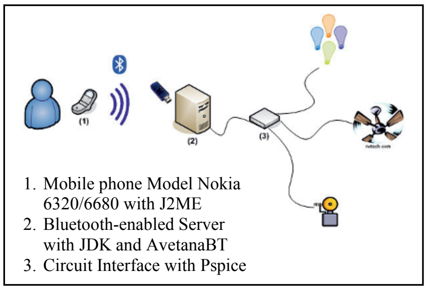

Figure 1. The proposed system architecture

IV. Curcuit Development

The circuit design consists of a parallel port, and six LEDs, six resistors, a 5V fan and a buzzer. The parallel port has two roles: (1) acting as a communication interface between the PC and the circuit, and (2) controlling the voltage for all the electronic components. Ideally it is assigned 0V when it is in low logic level (0) and +5V when it is in high logic level (1). When a value of 1 is sent out to the data pin where the components are connected, that respective component will be turned on. When a value of 0 is sent to that same pin, the component will be switched off. Figure 2 shows the schematics diagram of the circuit.

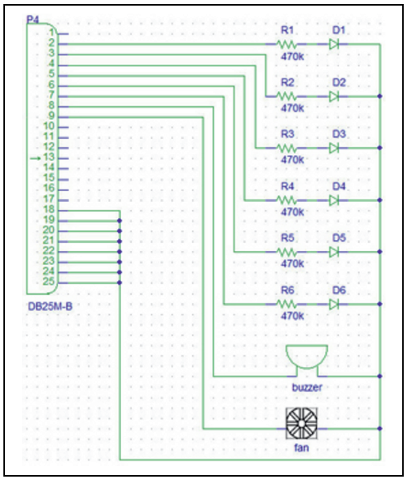

Figure 2. Circuit Schematics

The system utilizes all eight data pins on the parallel port. 470_ resistors are used to limit the current that flows into the LED, fan and buzzer. The fan and buzzer do not need a resistor because they can handle the 5V supplied to them.

V. MOBILE PHONE SYSTEM INTERFACE

To enable the use of the program to control appliances in a particular room domain inside the house, the program needs to locate the Bluetooth room server first. To enable this, start up screen is created to locate the room by providing a list of room server to choose from. Users will then select which room to control. Once the room is chosen, the mobile phone will attempt to establish connection with the computer using the Bluetooth connection of the mobile phone. The current state of the appliances is then shown in the next screen. The next option is to send the command to the server to change the state of the appliances. When the 'Switch' option is selected, a message will appear asking whether to allow the connection to the server and a simple confirmation needs to be selected. However, the message sent may vary on different type of mobile phones in which the program is installed.

The system logic flow shown in Figure 3 is the communication process between the server and the client from the start. The server needs to open the Bluetooth connection so that up to maximum number of 34 phone clients will be able to locate and acquire the server to establish the connection.

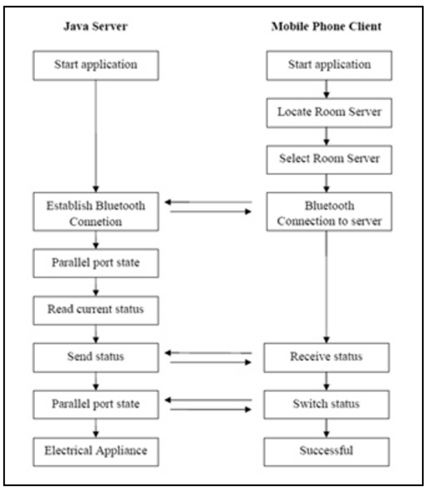

Figure 3. System Logic Flow

J2ME is the chosen language to deploy this part of the project because MIDP (Mobile Information Device Profile) applications written in J2ME are cross platform programs and work on most modern mobile phones. J2ME is minimal version of Java for mobile devices such as mobile phones or PDAs. Because a program written in J2ME can be used on most mobile phones, the library is more limited to enable 28 compatibilities. In order to smooth things up, Netbeans 5.0 and Wireless Toolkit are used to manage application packages and emulators.

VI. BLUETOOH SERVER

The server program opens the Bluetooth connection from the server program and waits for the client connection to connect. Once connected, the server will read the current state of the parallel port and send it to the mobile phone. After each time a command is received from the mobile phone program, the server will change the current state of the appliances to ON or OFF. The Bluetooth server is developed using J2SE. This reduces the possibility of incompatibility issues between the PC and the mobile phone. There are also other open-source applications that are being used for the server such as BlueZ stack, Parport IO Component, AvetanaBT and Blue Tips Server.

A. BlueZ Stack

BlueZ Stack is the official Bluetooth stack for Linux. The stack was initially developed by Max Krasnyansky at Qualcomm in 2001 [8]. Later Qualcomm decided to release it under the GPL open source license. It was then included as the official stack in the Linux kernel from the 2.4.6 release. Any recent kernels have the BlueZ stack build in, either as loadable modules or compiled into the kernel. BlueZ provides vast support for the entire core Bluetooth layers and protocols, like: L2CAP, RFCOMM, SDP, SCO, BNEP etc. It also ships with a large amount of tools and sample programs to test the Bluetooth equipment, which makes it easier for a new developer to make out new code from the small code fragments of the samples.

B. Parport IO Component

This is a piece of coding that was written by Juan Gabriel Del Cid Portillo in 2005 [9]. It is a Java class that enables applications to read and write bytes to and from the parallel ports on the computer. By using this, developer does not have to implement Java Comm API that is far more complex and tedious.

The following function is the sample of the written program using J2SE. Before we can use the function, we will first need to copy the file libparport.so file into the system directory.

```
public class ParallelPort {
private int portBase;
public ParallelPort (int portBase)
{
this.portBase = portBase;
} …
}
```

In order to write commands to the parallel port data pins, this function will be executed by accepting base 2 numbers.

```
public void write (int oneByte)
{ParallelPort.writeOneByte(this.portBase, oneByte);}
```

This command will be used load the libparport.so library to the system.

```
{System.loadLibrary("parport");}
```

C. *AvetanaBT*

This software is based on the most widely spread Bluetooth protocol stacks and does not use special Bluetooth hardware or software. It allows programmers to easily use and offer Bluetooth services. Avetana Bluetooth is a Java JNI implementation of JSR-82 for J2SE and different stack implementations. The implementation is actually quite platform independent and supports different stacks on various platforms such as Widcomm (Windows), Apple System stack (Macintosh) and BlueZ (Linux)

AvetanaBT needed to be patched in order to be used in any Linux distributions that uses kernel 2.6 and later. For those distributions that come with new kernels, all distribution related packaged have been migrated to the new library. However, some 3rd party software still refers to the old library such as this AvetanaBT. Upon compilation, the compiler will halt with an error caused by missing symbol "sdp_cstate_t".

In order to get AvetanaBT compiled, the following statement is needed to be inserted in the structs declaration of BlueZ.cpp source file which is located in the "c" directory of the distribution package.

```
typedef struct {
uint8_t length;
unsigned char data[16];
} __attribute__ ((packed)) sdp_cstate_t;
```

D. *Blue Tips Server*

This J2SE application handles both Bluetooth communication between the mobile phone application and the server; and the data transfer towards circuitry via parallel port. Both tasks must be declared and initialized as stated below.

```
//0xa400 is the printer port base address
private static ParallelPort lpt1 = new
ParallelPort(0xa400);
// 11111 is the UUID being set in both
// server and client mobile application
// to allow Bluetooth communication
UUID uuid = new UUID("11111", true);
```

The next statements allow the server to send the appliances current state by the mean of base 2 numbers to the mobile phone application.

```
DataOutputStream out =
btConn.openDataOutputStream();
out.writeInt(x);
out.flush();
```

processConnection() function is to handle data stream or actions to be performed summoned by the mobile phone application.

```
void processConnection(StreamConnection conn) {
try {
DataInputStream in = conn.openDataInputStream();
try{
x = in.readInt();
} catch (IOException ex) {
40
System.out.println("Unable to handle incoming
data");
ex.printStackTrace();
}
in.close();
…
}}
```

VII. THE SYSTEM

To allow the communication of this application, the Bluetooth connection is set to open so that the mobile phone client is able to locate and acquire the server to establish the connection. Figure 4 shows the server interface waiting for a connection.

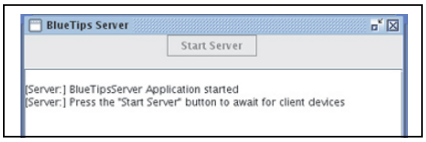

Figure 4. Server Interface - Waiting for Connection

Once the server is up and running, the mobile phone client starts searching for the server to establish a connection with a particular server. The client can then establish a connection with the server. The server is then acknowledges the connection and starts retrieving the parallel port status. Figure 5 shows the mobile phone interface for room server and appliance status retrieval.

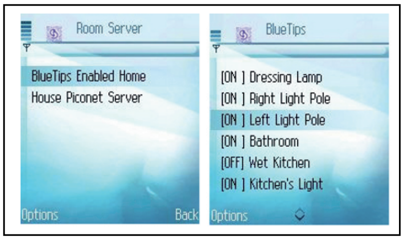

Figure 5. Mobile Interface for Room Server and Appliance Status

BlueTips server will receive the command to switch the state of the appliance from the mobile phone as in Figure 6.

Figure 6. Mobile Interface – Appliance Status Switch

Figure 7 shows a server receiving a command and the updated room status view from the mobile interface. The server will then send it to the control unit via parallel port and switch the particular appliance. The state will remain until the next command is received to change the state.

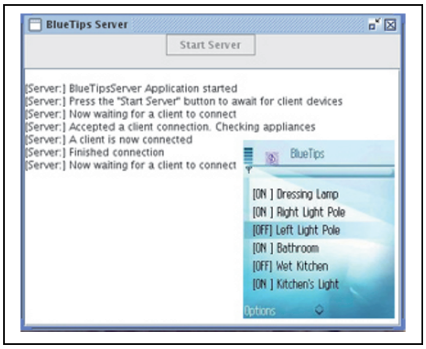

Figure 7. Server and Mobile Interface – Execution of Status Update

The Bluetooth server and client need to have the same set of UUID to allow communication between them. The UUID assignment is being done in SPP_Server class

```
public final static UUID uuid = new UUID("11111", true);
```

`read_SPP_message()` function is in `SPP_Client` class to retrieve the appliance state sent from the server.

```
public void read_SPP_message (ServiceRecord r)
{
…
con = (StreamConnection) Connector.open( url);
//read command (current status) using input stream
in = con.openDataInputStream();
x=in.readInt();
…
}
```

Below is the portion of the function that converts command from the mobile application to the base 2 number before sending it to the server.

```
public int pow(int a, int b)
{
…
for (int x=1;x<=power;x++)
{
value = value*2;
}//end of for.
…
}
```

VIII. CURCUIT REALIZATION

Currently, the circuit that is developed is a simple circuit which consists of a 25-pin D-shaped male connector, six LEDs, six 470_ resistors, a 5V miniature fan and a buzzer as shown in Figure 8. The circuit board is connected to the house model as shown in Fihure 9. The resistors are assigned to limit the current that flow through the LEDs. Bread board is used instead of a PCB to ease up the swap of components and debugging process.

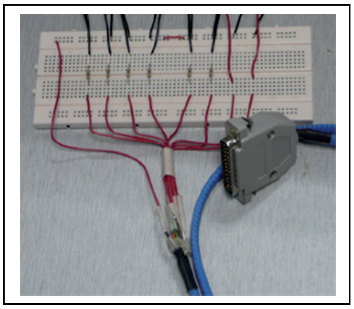

Figure 8. Product Image - Circuit View

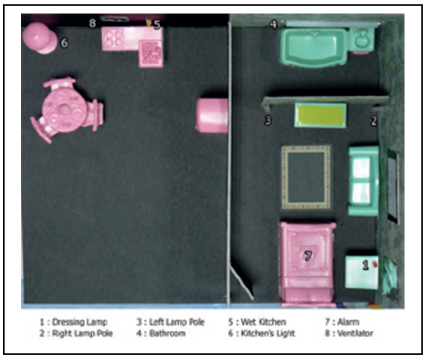

Figure 9. Product Image – Home Model

IX. Conclusion

The outcome of the project has proven that Bluetooth technology can be used to control electrical appliances in a restricted domain. Theoretically, the circuitry part needs to be expanded to change from electronic components to electrical appliances. LEDs, a miniature fan and a buzzer are used to resemble the real life appliances such as lamps, fans and security alarm respectively. The introduction of J2ME into this project enables the client application to be easily installed in various mobile phones. The client application for this project has been tested on Nokia 6230 (Series 40) and Nokia 6680 (Series 60) mobile phones. J2ME enables Java application to run on small, resource-constrained computing devices. It has become a standard in current mobile phone developments that most mobile phone manufacturers bundle their mobile phones with J2ME support. In a nutshell, the end product gives flexibility to the users by promoting the usage of their mobile phones as a universal remote control for their electrical appliances.

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Optimal Allocation of Load Using Optimization Technique

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Abstract: **An optimized system for distribution planning is proposed in this paper. By using mathematical algorithm the proposed paper is capable of optimizing the conductor layout. The method takes into consideration the capital expenditure and the cost associated with power losses.**

The paper discusses the distribution of load demands from various substations using the concept of Simplex Method and Branch and Bound technique. This paper gives us a method for meting the load demands from various available substations. The results of the discussed techniques will lead to a configuration of substations that will minimize substation construction cost. It will further lower long range distribution expenses as it will lead to optimum feeder path. The result is verified using MATLAB.

I. Introduction

A power distribution network consists of a number of substations connected to each other via feeders, an electric conductor carrying power from a substation to meet load demands along its route. Distribution planners must ensure that there is adequate substation capacity (transformer capacity) and feeder capacity (distribution capacity) to meet the load forecasts within the planning horizon. Alternatives such as procuring transformers, building new feeders and new substations need to be evaluated carefully. In general, the decisions in the planning of power distribution system include:

- Optimal location of substations
- Optimal allocation of load
- Optimal allocation of substation capacity

The paper discusses a method for the optimal allocation of load and hence determines the optimal conductor layout, the algorithm that is proposed aims at minimizing both the capital expenditure and the cost associated with power losses. The principle of selecting the substation sites is to minimize the total cost of the capital expenditure and the cost associated with power losses.

Presently work has been carried out with two substations and ten load points.

Let us assume 'm' substations denoted by Sub_i where i= 1,2,…., m and 'n' load points denoted by L_j where j=1,2,….., n let the distances of the load point 'j' with respect to the substation 'i' be D_{ij}.

So the total cost of supply involved in meeting the demand of a load is given as[9]

$$C = \sum_{i=1}^{m}\sum_{j=1}^{n} I_{ij} \times D_{ij} \times w_{ij} \times t_{ij} \quad (1)$$

where C= cost of supply involved in meeting the demand

I_{ij} = is the load demand supplied from substation 'i' to load point 'j'.

w_{ij} = the unit product cost of load and distance.

t_{ij} = 1 when load point 'j' is supplied by substation 'i'.

t_{ij} = 0 when load point 'j' is not supplied by substation 'i'.

D_{ij}= distances of the load point 'j' with respect to the substation 'i'

So we can frame the objective function as:

$$MinC = \sum_{i=1}^{m}\sum_{j=1}^{n} I_{ij} \times D_{ij} \times w_{ij} \times t_{ij} \quad (2)$$

Constrain:

$$\sum_{j=1}^{n} I_{ij}t_{ij} \le S_i \text{ where i=1,2,…., m} \quad (3)$$

S_i = capacity of substation i.

$$\sum_{i=1}^{m} I_{ij}t_{ij} = L_j \text{ where j=1,2…., n} \quad (4)$$

L_j = demand of load point j.

Solving the above using simplex method and then applying branch and bound for having an integer solution of the above problem have resulted in a satisfactory output.

II. PROBLEM

The problem is tested for ten load points with given load demands. Suppose there are two substations to be built to serve these load points. The maximum allowable supply capacities for the substations are given. By the use of the proposed technique we get the maximum supplies by the substation to meet the demands of the various load points minimizing the total cost of the capital expenditure and the cost associated with power losses. While meeting the load demands the distance of the load points with respect to the substation is taken into account, the load points are feed from the substation having minimum distance among them.

Let us assume a problem having two substations which are suppose to feed 10 load points. The supply and the demand

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of the load points are given below. The representation of the problem in fig form is shown below:

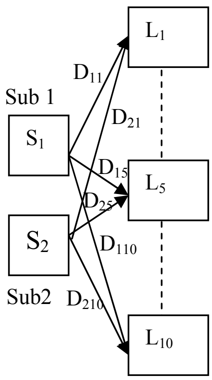

Fig 1

S_1 and S_2 are capacity of substation 1 and 2.
L_1, L_2 ……, L_j j=10 are the demand of load point L_j.
D_{ij} = Distance of substation i from load point j.

Let us assume:

S_1 = 100KVA ; S_2 = 300KVA

L_1 = 45KVA ; L_2 = 80KVA ; L_3 = 25KVA
L_4 = 50KVA ; L_5 = 100KVA ; L_6 = 30KVA
L_7 = 15KVA; L_8 = 5KVA ; L_9 = 30KVA
L_{10} = 20KVA

D_{11} = 10 units; D_{12} = 5 units; D_{13} = 50 units
D_{14} = 5 units; D_{15} = 15 units; D_{16} = 10 units
D_{17} = 28 units; D_{18} = 45 units; D_{19} = 5 units
D_{110} = 45 units; D_{21} = 24 units; D_{22} = 80 units
D_{23} = 10 units; D_{24} = 15 units; D_{25} = 10 units
D_{26} = 35 units; D_{27} = 5 units; D_{28} = 27 units
D_{29} = 45 units; D_{210} = 25 units

III. RESULTS

After applying the simplex and branch and bound technique:

Sub 1 – load 1= 0 KVA Sub 2 – load 1= 45 KVA
Sub 1 – load 2= 80 KVA Sub 2 – load 2= 0 KVA
Sub 1 – load 3=0 KVA Sub 2 – load 3= 25 KVA
Sub 1 – load 4= 0 KVA Sub 2 – load 4= 50 KVA
Sub 1 – load 5= 0KVA Sub 2 – load 5= 100 KVA
Sub 1 – load 6= 0KVA Sub 2 – load 6= 30 KVA
Sub 1 – load 7= 0KVA Sub 2 – load 7= 15 KVA
Sub 1 – load 8= 0KVA Sub 2 – load 8= 5 KVA
Sub 1 – load 9= 20 KVA Sub 2 – load 9= 10 KVA
Sub 1 - load 10= 0 KVA Sub 2 – load 10= 20 KVA

Load point	1	2	3	4	5	6	7	8	9	10
S_1 100	0	80	0	0	0	0	0	0	20	0
S_2 300	45	0	25	50	100	30	15	5	10	20

The result obtained from the work of Dale M.Crawford and Stewart B.Holt [1]

Load point	1	2	3	4	5	6	7	8	9	10
S_1 100	0	80	0	0	0	0	0	0	20	0
S_2 300	45	0	25	50	100	30	15	5	10	20

The result is validated with the MATLAB simulation and with the work of Dale M.Crawford and Stewart B.Holt [1].It bears a close resemblance with the result obtained by Dale M.Crawford and Stewart B.Holt [1].Hence this is the optimum division of load among the substations to minimize the cost incurred.

IV. CONCLUSION

The techniques discussed will lead to optimum distribution of loads among sub stations and further it can be extended for configuration of substations that will minimize substation construction cost. It will further lower long range distribution expenses as it will lead to optimum feeder path.

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Object-Oriented Analysis Using Event Patterns

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Abstract- Object Oriented Analysis (OOA) of requirements has always been the most difficult, critical and an important step in developing applications using object-oriented paradigm. Identification of types of objects from the requirements is the primary goal of every OOA. Most commonly recommended approach to identify objects is to search for nouns directly from the requirements or indirectly from the Use Cases derived from requirements. Experience has shown that both these approaches are inefficient in their own ways. In this paper, we have proposed a new approach of capturing Object Oriented requirements based on analysis of events and actions occurring in the system and then identify all static and dynamic components of the system from it. Our approach captures requirements in the form of Event Patterns that are templates in our Event-oriented approach analogous to Use Cases in the conventional Object-orientation. These templates are used in finding out object oriented components of the system during the process of OOA. We have also proposed an Event Meta-model that forms the basis of our event based class identification process.

I. Introduction

Object-Oriented Analysis and Design has now become a major approach in the design of software systems. In this methodology classes and objects play a vital role in system's goals and requirements. Booch [1] was one of the first authors to formalize the object oriented approach. Now there are several popular object oriented methods such as Object-Oriented Analysis (OOA) by Coad and Yourdon, Object Oriented Design (OOD) by Booch and Object Modeling Technique (OMT) by Rumbaugh et al. All these OO approaches examine requirements from the perspectives of classes and objects found in the vocabulary of the problem domain.

Identifying classes of objects and then building OO software model for a problem is the most difficult and important part of Object Oriented Analysis and Design (OOAD). It is important because not only other steps of software development life cycle will use the identified classes and objects but also future extensibility and maintainability of the software depends on it. It is difficult as every object can play different role and hence can be categorized in more than one ways. So, there are no crisp guidelines and no theoretical or pragmatic methods that can help to tackle this difficulty. Reference [1] has stated that "There is no such thing as perfect class structure, nor the right set of objects. As in any engineering discipline, our design choice is compromisingly shaped by many competing factors." He also stated "It's a holy grail. There is no panacea".

A commonly recommended technique for identifying classes and objects from the requirements was proposed by Abbott J in [2]. It uses proper nouns, common nouns and noun phrases to identify classes. The benefit of this technique is that nouns are easily mapped to objects and classes. Analysts need not to have the experience of domain entity classification, so no learning curve is involved but they must have understanding of parts of speech. The disadvantage is that it often results in too many nouns being identified which are to be refined by removing nouns that represent redundant classes or those which are actually adjectives or attributes. In order to deal with this problem Jacobson in [3] proposed a way to break the entire scope of system functionality into a number of Use Cases. Each Use Case describes a sequence of behaviorally related events that flow through a system. Since Use Cases are smaller statements of system functionality so identifying nouns becomes simpler and efficient. However experience has shown some problems with Use Case such as lack of precise definition. The concluding statement given by Ian Graham was "There is a need for another approach with sound theoretical basis and a precise definition" [4].

In this paper we have given a new approach where Events are the primary entity that can help in OO analysis of requirements. Unlike the two above approaches we have followed a direct way of capturing and representing requirements as Event patterns that are templates in our Event-oriented approach analogous to Use Cases in the conventional Object-orientation. These templates help us to find out object oriented components of the system. We have also critically analyzed how concept of events has been used in different domains in general and for OO class identification in particular. We have also devised a Basic Event Meta-model that forms the basis of our event based class identification process. The remainder of the paper is structured as follows: Section 2 presents a critical review on Application of Event Concepts. Section 3 details our proposed methodology of class identification on basis of events. Section 4 illustrates our approach on a case study and

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Section 5 concludes our work with future scope and directions.

II. REVIEW ON APPLICATION OF EVENT CONCEPT

A . Role of event in multiple domains

The concept of events is not new to the research community. In this section, we try to explore how event concept has been widely used in multiple domains like Real Time Systems, Control Systems, Software Engineering , Simulation, Modeling, Active Databases etc to name a few. X Zheng [5] et.al has used Events and Actions to derive object-oriented model of real time control systems. According to them, Events are like interrupts in system and stimulate action procedures. They have treated events and actions as abstract data type on which operations like create, destroy, enable, disable, enqueue and dequeue can be performed. In their approach, every event is derived from a general event class that adds on its own specific operations as per the requirements. Jingzhou Li [6] et.al has used concept of events to realize the distribution as well as the control mechanism in their object-oriented workflow model. To illustrate this they have made a meta-model consisting of event classes and relationships. To support dynamic modification of workflow definitions, their meta-model have separated the event definition and notification mechanisms from the event reaction objects and also separated the definition of functions from the triggering mechanism of the functions. Monique Snoeck and Geert Poes [7] used events to demonstrate how a generic domain model can be analogically reused using event-based, object oriented domain model. They have modeled domain objects in terms of their participation in real-world events. According to their approach, information and objects come into existence, get modified and disappear from Universe of Discourse through events. Events affect more than one domain object and classes. They have used an Object-Event Table (OET) to show relationship between domain class and its related events. Alan S Abhramans [8] et.al have illustrated that their event centric paradigm (covering notion of events and types of events) can better specify and implement E-Commerce application requirements in comparison to Object-oriented framework. They have given event based architecture to facilitate the executable specification of E-Commerce application. Specifications are decomposed into events and stored in event store. A complete guidance on how events and their temporal relationship can be expressed in English specification is also given. N H Gehani [9] et.al have used events in the context of Object Oriented Databases. They have focused on different types of events that can occur in OO Databases and have given a model with a language for specifying the basic and composite trigger events in the databases. Their language is equivalent to regular expression but focuses on sequence and subsequences. Expression of their language can be easily translated into finite state automata. They have simplified Event-Condition-Action model by making condition part of the event specification. Ping-peng Yuan [10] et.al have used concept of events to build a collaborative support software architecture and also given a new Knowledge Interchange Format (KIF) based event specification language to facilitate representation and exchange of event information in their architecture.

B. Role of Event in Class Identification

From all the above approaches analyzed so far, one can clearly see that usage of events have not been made purposely for class identification problem. Use of Event-based class identification in the early phase of Object-oriented Analysis and Design has been done in the work by Danny C.C.Poo [11]. In his approach, Events in Use Cases have been used to identify classes and business rules. For each event identified in the Use Case, an Event Script is written. Information in Source, Participants Sets, Pre-Event Conditions and changes from the Events script help the analysts to identify classes and objects, their attributes, operations and business rules. These identified components form a class specification. The disadvantage of his approach is that to identify classes one has to adopt a tedious and lengthy process. This is due to the fact that from the requirements given by the users, Use Cases are written first followed by writing event scripts for each event in every Use Case. If we start directly from events captured from requirements we would make the analysis simpler and shorter. In his approach class relationship identification has not been dealt properly but is carried out intuitively in a conventional manner. Unlike our approach where events have been categorized into various generic classes (data flow, control and temporal) each of which have specific pattern, author has treated all events alike. In such case of lack of categorization there is a problem of having large number of events which have to be optimized for saving time involved in analysis. Due to lack of precise definition of Use Cases we start directly from capturing events in requirements. Events are our main entity that can collectively make up a Use Case so we target to capture events first and not the Use Case, after all use comes after events. The author has not identified any relationship between events and presented his approach as an extension to Use Case modeling. In contrast our event patterns take up the place of Use Case template which can be temporarily ordered to show relationships between them. Even the analysts can further use these templates to structure and understand the requirements and functionality of the system.

III. PROPOSED METHODOLOGY OF CLASS IDENTIFICATION

A. Principal of Our Methodology

Conventional approach identifies classes of objects using nouns in the domain description. This technique finds all static components and relationship in a system at a given point of time. According to our principal we use events as the base for identifying classes. We would then be able to record the changes that have occurred over a period of time. We would be able to identify which object(s) have stimulated events, which object(s) have been affected by events, what operations have made the changes and in which state members. Thus we propose that not only Objects but the relationships between objects can also be identified by analyzing the participating objects in an event and values that have changed in an event gives the idea of the attributes of an object. Our proposed Event Meta-model justify the principle of our methodology to take events as the base in identifying classes from the requirements.

B. Basic Event Meta-model

According to our Event Meta-model given in Fig-1(Refer to Last Page), Events are anything that happens in a system and causes a system to change state. Creation and destruction of objects, call to method as well as response from it are all events. System is composed of various Objects. These objects interact and collaborate through events to render the functionality of the System. Every Object has a State which is composed of various attributes possessed by them and their relationships with other Objects in or outside the System. Objects play a role of actors in an event. These objects stimulate each other and that stimulus is an event. In the meta-model the Object that initiates the event is a stimulator and the one who is affected by event becomes an affecter.

Events have a Response associated with it which can be an Activity or an Action. Activity is a continuous or sequential operation. Activity takes time to complete whereas actions are instantaneous operations associated with an event. Events and Response have a cyclic relationship in the sense that events trigger Responses and Response in turn cause other events. Response also generates an information that changes or modifies the State of Objects and hence their attributes and relationships with other Objects.

An event occurring in a system may carry or generate any information. Such events are called data-oriented events. This information is the information required either for processing the event or a result of an outcome of an event. Data flow can be from source to destination objects or vice-versa. Optionally events may also carry a timestamp denoting the time of their occurrence. Such events are called time-oriented events. These events are assumed to be triggered by an internal clock of the system. This clock determines the passage of time. Some events occur at an unpredictable time and are called control-oriented events. Most of the events in a general application domain are data-oriented and time-oriented. Control-oriented events are more common in the domain of real time systems. Events in a system can be formally expressed as Event-patterns. Irrespective of the type of event, every event has a set of attributes which are collectively modeled as an Event-Attribute class shown in the meta-model.

C. Methodology for Event based Class Identification

As a first step towards event based class identification, we present an outline of the methodology in Fig-2. It shows the process of deriving classes from the domain requirement statements using events.

More precisely the steps we followed in deriving candidate classes were the following.

1. End user provides the requirements which are gathered by the analysts by adopting various modes of information gathering techniques. These requirements could be unstructured text or could be structured into domain description documents.
2. Analysts understand the problem domain and requirements statements in order to extract all the events occurring in the system and make a list of events. Event list is an exhaustive list of all possible events occurring in the system.
3. Events from the event list are categorized into data-oriented, time-oriented and control-oriented events.
4. Each event in the list is formalized and documented as an event pattern.
5. Using our mapping rules we can identify the candidate classes their methods, attributes and relationships from these patterns and generate class diagram.

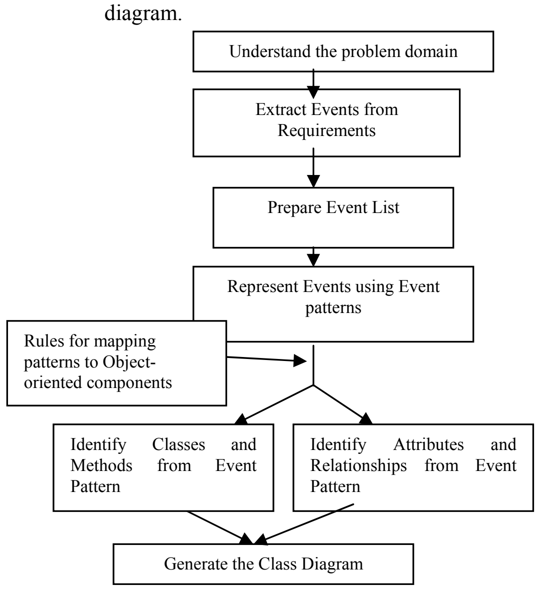

Fig. 2. Event Based Class Identification process

D. An Example of Event Pattern

In our process of class identification use of Event Patterns in the Object-oriented analysis plays a vital role. In this section we are proposing an outline template for Event Pattern that strongly supports the whole class identification process from the requirements analysis to generation of class diagram. We also propose that using Event patterns in the analysis phase has benefit in reducing development time by introducing reusability. Event Pattern will capture events occurring in a system or conceptual models in an application domain in order to allow reuse across applications. Following is template of Event pattern that we are working on and will refine it further in future work:

Event Identifier: This section gives the unique identifier of the event identified in the list.

Event Name: This section states the name of the event described in the list.

Initiator of Event: The originator of the event.

Facilitator of Event (Optional): The Entity who helps in carrying out the event.

Affecter of Event: The Entity who is affected by the event.

Pre-Event Conditions: This section states the conditions that must hold before the Event in consideration can take place.

Site of Event Occurrence (Optional): This is a location specified of event occurrence if any.

Guard Condition (Optional): This is a condition to test whether event will occur or not.

Activities/Actions Triggered: These are actions or activities that get initiated as a response of Events.

Facts Changed: These could be some information exchanged. It could give us idea of attribute or candidate class.

Post-event Condition (Optional): This section states the conditions for triggering an operation after the event is successfully completed.

Timestamp Assigned (Optional): This is a system time assigned for event occurrence.

Event Type: This could be Data, Control or Time oriented

IV. Reservations Online Case Study

We shall illustrate our proposed approach described above using a case study named "Reservations Online" on object-oriented analysis. Following is the Tours Online user requirements of the case: "A software for a travel agency provides reservation facilities for the people who wish to travel on tours by accessing a built in network at the agency bureau. The application software keeps information on tours. Users can access the system to make a reservation on a tour and to view the information about the tours available without having to go through the trouble of asking the employees at the agency. The third option is to cancel a reservation that he/she has made. Any complaints or suggestions that a client may have could be sent by email to the agency or stored in the complaint database. Finally, the employees of the corresponding agency could use the application to administrate the system's operations. Employees could add, delete or update the information on the customers and the tours. For security purposes, the employee should be provided a login ID and password by the manager to be able to access the database of the travel agency."

Following our methodology from the case study we make a list of all events as shown in Fig .3

Customer view information about the tours.

Customer makes reservation for a tour.

Customer cancels a reservation.

Customers send complaints by email to agency.

Complaints stored in a complaint database.

Manager provides login ID and password to Employees.

Employees add customer information

Employees delete customer information

Employees update customer information

Employees add tour information

Employees delete tour information

Employees update tour information

Fig. 3. Events from Case Study

If we try to map the above template on an one of the events like "**Customer make reservation on tour**", following will be the resulting event pattern in Fig.4.

Event Identifier: E01

Event Name: **Customer makes reservation on tour**

Initiator of Event: **Customer**

Affecter of Event: **Tour**

Pre-event Condition (Optional): **Customer has viewed information on tour.**

Activities/Actions Triggered: **Reservation made**

Facts Changed: **Number of Seat decremented**

Post-event Condition (Optional): **Compute Ticket and Print Reservation slip.**

Event Type: **Data-oriented**

Fig. 4. Example used as Event Pattern /Template

Using the above Event pattern, our potential candidate classes can be the initiator, facilitator or affecter of the event. Facts changed can also give us the idea of either candidate classes or their attributes. Name of the event or actions and activities triggered in the pattern can give us the name of the class methods. They can also provide the class name e.g. Reservation. The relationship between the initiator, facilitator or affecter of the event can give us the class relationships.

V Conclusion and Future Work

All Object-Oriented Analysis and Design (OOAD) methods start from the process of class identification. Identifying objects and classes and then building OO software model for a problem is the most difficult and important part of OOAD. Unfortunately however, none of the current approaches of OO analysis have used events occurring in the system as the base to identify classes. In this paper we have made an attempt to look at requirements from event-oriented perspective and tried to break the domain requirements in terms of events. We have shown that these events derived from requirements can be documented as Event Patterns. From these patterns object-oriented components of the system can be derived. We have illustrated an outline of our process we are working on to derive classes from domain requirements using events as basis. A Basic Event-Meta Model is also made to focus on the concept of event as basis of class and object identification. Our Meta-model addresses certain issues like what is an event in the context of OOAD, how objects are identified from events and why events are basis of class identification. In future we will be presenting a detailed framework to show how our basic Event Meta-model and patterns can help us systematically derive the classes and their attributes, methods and relationships. We are also working on refining our basic event patterns to make it complete so that it can cover all types of events. We are also defining rules to transform the event patterns to class specifications automatically.

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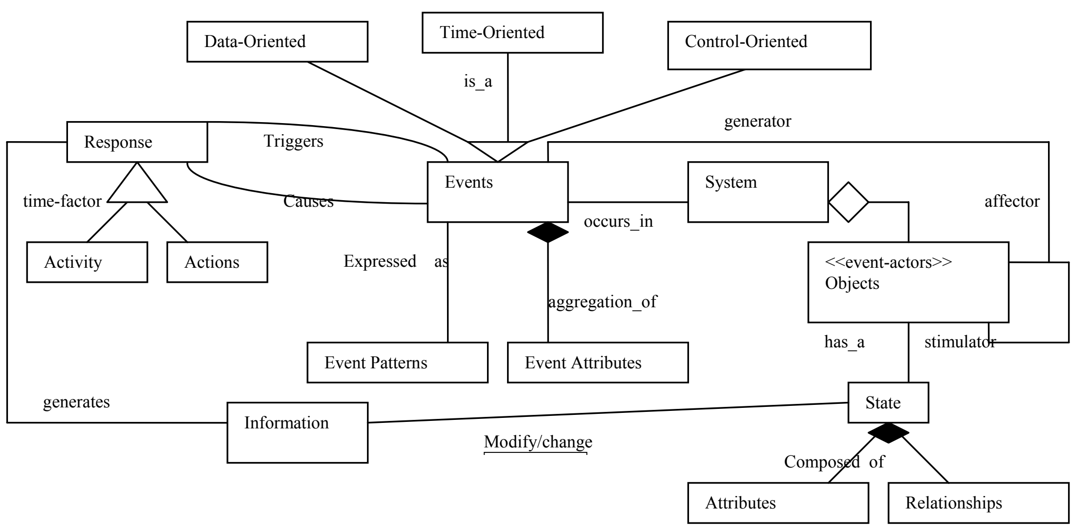

Figure 1 –Basic Event Meta-Model

Unit Tests Construction Based on Business Rules

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***Abstract*-Software unit testing involves generating or creating test data and specifying a test oracle. The test oracle verifies if software under testing behave correctly when it is given test data as an input.**

We present a technique for creating a test oracle in this paper. The test oracle is created by converting (or specifying directly) Business Rules to Object Constraint Language (OCL) statements. The created OCL statements are used as test oracles. These OCL statements in conjunction with any test data generation technique acts as test cases. We present a method how to convert Business Rules to OCL statements and use them for software unit testing.

I. Introduction

Software development process consists of several phases, such as requirements gathering, design, implementation, testing, deployment and maintenance. Software testing consumes 50 or more percent of all software project costs [1, 2]. In order to reduce these costs, software developers employ various testing automation techniques. Automation is usually regarded as automatic tests execution [1] (such as record-playback tools for user interface testing or unit testing frameworks [3] for executing already prepared tests as a regression suite repeatedly). In this case software tests have to be created manually. Another way of automation is to generate tests automatically. Tests generation is based on software models and/or code. Test generators (such as a random test generator) usually automate only the test data preparation process and do not provide any means to recognize if software under testing performs correctly with given test data. A tester has to act as a test oracle and to verify manually if software has produced the correct output when it was executed with generated test data. This process is labour intensive and costly. Only several test generation techniques provide means to evaluate if software behaves correctly when it is executed with test data as input. These techniques usually are model based. They evaluate if software under testing behaves the same way as its model. These techniques rely on artefacts produced in the software design phase.

We present the test generation technique for generating tests from artefacts of software requirements phase. Requirements can be expressed as natural language texts or as models. Our approach uses requirements expressed as Business Rules [4] for generating unit tests.

A. Business Rules

Business Rules can be used to describe software requirements. Business Rule is a statement, which defines or constraints an aspect of a business [5]. The purpose of Business Rules is to show business structure, to control or influence its behaviour. Business Rules are atomic; they can not be subdivided further.

Business Rules can be used for software testing. They can be used as a testing oracle, which decides when software behaves correctly with given test data and when not.

B. Business Rules classification

Not all types of Business Rules can be used for software testing. Business Rules can be groped by their type (syntax, semantic). Rules of some types can be more useful for software testing.

There are several Business Rules classification schemes. One is presented in Guide project [4], the other is proposed by M. Weiden, L. Hermans, G. Schreiber and S. van der Zee [6].

The guide project classifies Business Rules based on their syntax: business terms definition, facts joining the terms, constraints and derivations [4, 5].

The main drawback of Guide business rules classification is that it is based on rules syntax differences (for example, type verification and function evaluation). The classification scheme proposed by M. Weiden, L. Hermans, G. Schreiber and S. van der Zee groups rules by their semantic characteristics (roles in processes). Their types are grouped in three categories: Structural, Behavioural, and Managerial. The structural category defines the static structure of business, the behavioural one defines tasks workflow in business and the managerial one defines higher level business constraints. The rule types and their categories are presented in Table I [6].

TABLE I
Business-Rule Classification Scheme

Category	Rule Type
Structural	Concept Structure
	Persistency
	History
Behavioural	Information Flow
	Control Flow
	Pre-condition
	Post-condition
	Frequency
	Duration
	Task knowledge
Managerial	Organization
	Goal & Value

II. Testing Framework

Our testing method uses Business Rules as a test oracle in unit testing. They can be used as an oracle, because they define how software has to behave and provide some constrains on its behaviour. The software testing framework is presented in Fig. 1.

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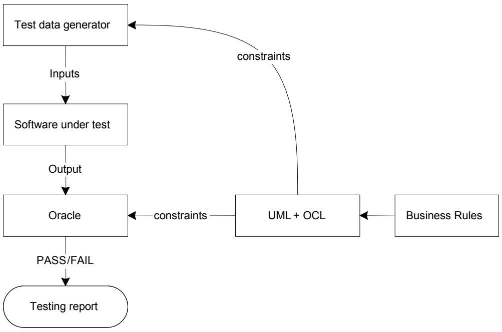

Fig. 1. Testing framework.

In order for this method to work we have to get OCL statements. These statements are created from Business Rules or Business Rules are described as OCL statements itself. When we have OCL statements we can generate some test data (using any test data generation technique). After test data generation we can feed them to a software under test directly, or we can filter some of them out, by checking if they satisfy constrains for data input (in OCL it is called pre conditions) and feed them to a software under test only filtered test data. The test data are passed to software under testing and the software is executed. Software under testing produces some output. The oracle takes the produced output and evaluates its correctness against the UML+OCL model. It evaluates if the produced output satisfies the provided OCL statements (post conditions and invariants) and produces the testing report. If the output satisfies OCL constraints the PASS record is added to the testing report and the probability of the likelihood that the test has passed is calculated. If the output does not satisfy the OCL constraint, the FAIL record is added to the testing report.

The testing procedure:

1. Converting or describing Business Rules as OCL statements.
2. Generating test data
3. Filtering test data according to OCL statements (pre conditions)
4. Executed software with passed test data
5. Verifying if the produced output satisfies OCL statements (post conditions, invariants)
6. Generating the testing report: if OCL statements are satisfied then the test has passed, else the test has failed.

A. Defining Business Rules as OCL Statements

We have selected Business Rules classification scheme proposed by Weiden, L. Hermans, G. Schreiber and S. van der Zee [6], as it is based on rules semantics (and not as in Guide project by rules syntax). The authors presented a way how to express each rule type in their classification as UML diagrams [6]. We have extended their classification by changing UML diagrams and extending them with OCL statements. B. Demuth, H. Hussmann and S. Loecher have presented another way for expressing business rules as OCL statements, but their work only defines how to specify business Rules of Structural type as OCL statements [7]. Our scheme for presenting Business Rules as OCL statements is presented in Table II.

TABLE II
BUSINESS-RULE REPRESENTATION AS OCL STATEMENTS

Rule Type	Representation UML + OCL
Concept Structure	UML class diagram. The <<concept>> stereotype is used for displaying business concepts. OCL statements are presented as invariants. Invariants are used for defining class properties and relations between classes.
Persistency	UML class diagram. An attribute of type DateTime is added in classes for which we want to control their lifetime. And OCL invariants are added for constraining the values for that attribute. For example: - inv: [self.atributeName] > 2005-01-1 and [self.atributeName] < 2006-01-01 (Object can only exist during the year 2005) - inv: Date::now -[self.attributeName] < 5-0-0; (The object can be no older than 5 days.)
History	-
Information Flow	In the class method some system function is added for performing. Preconditions are added as OCL statements.
Control Flow	-
Pre-condition	The method is added to UML class which represents a task. The OCL preconditions for this method are added; such preconditions specify limitations of input values passed to this method and limitations for object attributes.
Post-condition	The method is added to UML class which represents a task. The OCL post conditions for this method are added; such post conditions specify limitations of input values passed to this method and limitations for object attributes.
Frequency	The method is added to UML class which represents a task. The OCL preconditions for this method are added; such preconditions specify limitations of input values passed to this method and limitations for object attributes.
Duration	The method is added to UML class which represents a task. OCL statements are added for attributes as post conditions that have to be valid after executing the methods. For example: - post: [self.productionDate] == [self.serviceDate]
Task knowledge	The method is added to UML class that represents a task. OCL statements are added for attributes as post conditions that have to be valid after executing the methods. For example: - post: [self.productionDate] == [self.serviceDate]
Organization	-
Goal & Value	-
Actor Competences	-
Actor Responsibilities	-
Resources	-

Not all types of business rules can be expressed as OCL statements. For example, rules in the managerial category can not be expressed as OCL statements because they are too generic. For example, they define organization's goals, its vision. Business Rule "Software's purpose is to increase company productivity by 50%" can not be easily expressed as an OCL statement if any at all.

For some types of rules we have presented similar expressions in OCL. So these statements are presented in the same way, but they just cover different aspects of software.

B. OCL and Tests Generation

OCL standard stands for Object Constraint Language [8]. It was proposed by OMG organization. OCL allows UML [9] models to be extended with constraints. An UML diagram can not reflect all relevant aspects and constraints of a model. The

OCL standard has been developed to extend UML models by defining constraints.

For example, an UML class diagram can have an attribute of the integer type. It is not possible to define limits of that attribute's values or define relations with other attributes in the UML diagram. Let's suppose we want that this attribute could have its value within the range of 0 and 100. Using OCL, this constraint could be implemented. Such constraints are called invariants. They have to be satisfied during the whole lifetime of an object.

OCL can also provide constraints for class methods, such as preconditions for input values and post conditions for output values. Let's consider a class Rectangle:

```
class Rectangle
{
        public int calcArea(int width, int height); };
```

Let's define constraints for this method:

- The width has to be greater than zero
- The height has to be greater than zero

Let's define constraints for the result:

- The result is the width multiplied by the height

Constraints for a method calcArea such as the width and height have to be greater than zero (precondition) and the result is The width multiplied by The height (post condition) can be expressed in the UML model by using OCL as follows:

```
context Rectangle::calcArea
 pre: width > 0
 pre: height > 0
 post: result = width * height
```

The post condition can perfectly serve as an oracle. The test generator can easily select two input values which fall in the range of preconditions, execute the unit under testing and evaluate if software has calculated the result which matches OCL post statement. If the result matches, then the test has passed otherwise it has failed, thus revealing a bug in software.

Unfortunately, OCL statements are not always available for all software units. Some constraints are not as precise as the one given in the example. In the given example, the post statement completely reassembles the implementation of the method. When a post statement reassembles the method implementation (provides a full implementation) the statement can be automatically transformed into software implementation [10]. In this case no testing could be needed. C. Philippe and R. Roger [11] proposed the idea how to convert OCL statements directly to the code implementation in this way removing the need for testing of implementation, because implementation represents the model correctly. Statements have to be precise in order to perform such automatic implementation generation. But it is not so common situation.

B.K. Aichernig and P.A.P. Salas [12] have proposed a method for tests generation using OCL statements. Their method relies on OCL specification mutation. In this method, inputs are selected that could detect changes in OCL specification. Such inputs passed to software under testing could detect bugs in the implementation, too.

B. Korel and A.M. Al-Yami [13] presented a similar method, which injects constraints into software code. These constraints serve as an oracle inside the code and allow verifying if some method implementation variables are within the defined ranges.

N. Tracey, J. Clark, K. Mander and J. McDermid [14] presented a similar method for detecting bugs. They have analyzed the "exceptions" feature of programming languages. Their method tried to calculate the required input, which would force to raise an exception, when the input is passed to software under testing. If it is not possible to calculate such an input, software under testing is bug free.

C. Test Case Generation Using Imprecise OCL Statements

OCL [8] statements are usually imprecise, because it is easier to define imprecise statements or even derive them from other model elements. We propose a test case generation technique using UML models and imprecise OCL statements.

Imprecise OCL statement is a constraint which defines the post or pre conditions not by an exact formula, but by some boundaries or by a formula defining an approximate result value. For example, in the previous example we could replace the post condition with a new imprecise one:

```
 post: result > 0
```

This modification makes the OCL statement imprecise. Despite the fact that such type of statement is easier to define or derive, this statement does not allow to verify if the computation was performed correctly. For example, we are testing an implementation of the method calcArea:

```
class Rectangle
{
        public int calcArea(int width, int height)
        {
            return width/height;
```

There is usual mistyping in the given above implementation. The division '/' is used instead of multiplication '*'. Let's say the data set (2, 3) was generated for the width and height of input parameters, respectively. The unit under testing returned a value equal to 0.666. Using a precise constraint would reveal an error, but the imprecise constraint does not.

We have another implementation with a bug, for example

```
class Rectangle
{
        public int calcArea(int width, int height)
        {
            return width-height;
```

The error is a misplaced subtraction operator in the code given above. Instead of the multiplication '*' symbol, the subtraction '-'symbol was typed. When the test data (2, 3) are passed to the method, it returns -1. This value does not satisfy the model constraint, and we have found a bug.

Imprecise OCL statements can be deduced from other model elements. For example, we have the class Rectangle which represents a geometrical shape and extends the class Shape. The Shape is a generic class and serves as an interface for all geometric figures. The class Shape could define a method calcArea and have an OCL statement associated with it. The OCL statement defines that an area is always greater than zero or is equal zero. So this constraint has to be valid for all classes which extend the base Shape class. The constraint for the Rectangle class is derived from the base Shape class in this example.

D. Testing Procedure Using Generated/Created OCL Statements

During the software testing using imprecise OCL statements, we generate test data for each class method using some selected test data generation technique. The generated test data are filtered if they match the input OCL statements, thus reducing the test execution time. The test data which satisfy OCL input statements are passed to software under testing and a software output is received. The received output is verified against OCL statements. If the output does not satisfy OCL statements, then we have detected a bug and the test has failed. If the output satisfies OCL statements, then the test has passed.

The testing procedure is represented in Figure 2. The procedure generates the testing result as a set TR of records. Each record specifies the class C, the method M, the Boolean flag R (which defines if it contains bugs or not R), and the probability P of an inaccuracy of the flag R.

Test generation can be performed for the selected method until a bug is found or until we have reached some defined coverage criteria. For example, if we selected all path criteria then the testing continues by generating test input data until the input which forces the method to produce an invalid output is found (the input does not satisfy the OCL statement) (lines 11, 12). The testing can also be finished when all paths in the method are executed and no inputs that have invalidated constraints have been found or no more new test data can be generated (line 14). If the test passes, the EVALUATEPROBABILITY function is called that has to calculate the correctness of the testing result. If, using some test data, software under testing produced an output, which does not satisfy OCL statements, then we can say definitely that the test has failed. If the output satisfies OCL statements then we can only predict that test has passed. The EVALUATEPROBABILTY function calculates the probability that the test has passed. The probability depends on the number of test data executed.

```
Input. A set of classes, a set of methods in each class, a set of
OCL input constraints associated with each method, a set of OCL
output constraints associated with each method.
Output. The set TR of records <C, M, R, P>

1.  While there are untested classes
2.    Do select the class C1
3.      While there are untested methods in the class C1
4.        Do select the method M1
5.          Select the OCL constraint OCLI for
            the method M1 input
6.          While can generate test data (the
            required coverage criteria has not
            been yet met)
7.            Generate test data TD
8.            If TD matches the OCL input
              constraint OCLI then
9.              Execute M1 with test data TD
                and get the result MER
10.              Select the OCL constraint OCLR for
               the method M1 output
11.             If MER does not satisfy OCLR
                then
12.               add <C1, M1, FAIL, 100%> to TR
13.               go to the step 3
14.           Else
15.             Discard test data TD
16.         add <C1, M1, PASS,
            EVALUATEPROBABILITY()> to TR
```

Fig. 2. Testing procedure.

The probability is required because the OCL statements are imprecise. Using imprecise constraints we can not be sure completely if the value calculated by software is correct despite the fact that the constraints are satisfied.

E. Test data generation

In order to generate test data for software under testing any test data generation technique can be used. For test data generation random generation technique, genetic algorithms, symbolic execution, chaining approach, simulated annealing, variable dependence analysis, evolutionary algorithms and other white box testing techniques could be used [15]. The techniques could target to some code coverage criteria. For example, all paths criteria could be used as a mean for determining the test success probability.

OCL statements can filter some generated test data and reduce testing time by doing that. For example, when we are generating test inputs for method calcArea, the random test generator chooses values from the range -32K and +32K. Using the OCL statements values ranging from -32K to 0 can be removed because they do not satisfy preconditions. By removing them the testing time is reduced.

F. Oracle

OCL post conditions serve as test oracles. The methods are executed using the generated test data, and the output is checked against post conditions. If the function output does not satisfy the post conditions, the test has failed and a bug is found. But if the result value satisfies the post condition, we can only predict with some likelihood that the test actually has passed and there are no errors in the implementation. The probability that the test has passed could be calculated using some of coverage techniques. For example, we have calculated that software under testing has n paths in the code and using one of test data generation techniques, we have generated test data which covers m paths ($0 < m <= n$) of the code. After the execution we could get that all generated outputs satisfy OCL statements and we have to calculate the success probability. Using all-paths coverage criteria, we can assume that $p = m / n * a$. Here p is the probability of the accuracy of the testing result, m – the number of the covered paths with test data, n – the number of total paths in software under testing, a – a constant ($0 <= a <=1$) which defines the imprecision level of OCL statement. Here 1 could mean that OCL statements are precise, and 0 could mean that we have no OCL statements at all.

III. EXAMPLE

We have selected a cars rental software. It is presented in Business Rules Guide project as business rules example [4]. This software purpose is to support a fictional company EU-Rent which works in cars rental business. The software has to allow its users to manage the car park, support business functions such as renting a car, managing clients, managing customer's loyalty programs.

In this example we will use two following business rules:

- "Each car must be serviced every three months or 10,000 kilometers, whichever occurs first"
- "The car has to be in good condition after servicing".

These two rules deal with a business entity "car" and its function "service". The first rule falls under the "frequency" Business Rule type and the second under the "post-condition" type. These two rules can be expressed as an UML class diagram and OCL statement. The class diagram is presented in Fig. 3.

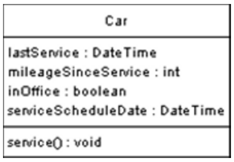

Fig. 3. Software's Implementation.

OCL statements for the first rule can be expressed as:

context Car::service()
pre: (Date::now – self.lastService >= 3) or
(self.mileageSincService >= 10000)

OCL statements for the second rule can be expressed as:

context Car::service()
post: self.inOffice = true
post: self.lastService = Date::now
post: self.mileageSinceService = 0

These constrains now can be used for testing class Car method service. The preconditions filter generated test data and the post conditions verify if the method performed correctly.
Suppose we have a buggy implementation of this method:

```
public void service()
{
  // performing maintenenace task..
  // ..
   This.inOffice = true;
   This.lastServiceDate = Date:Now + serviceDurationTime;
};
```

The developer missed to reset the milleageSinceService variable and incorrectly calculated the lastServiceDate variable (there can be a time gap between servicing scheduling and service start).

We used a random test data generator for generating test data. The testing and tests generation results are presented in Table III. The Date::now function returns 2010-12-10 during the testing and serviceDurationtime is 1 day.

TABLE III
TESTING RESULTS

Test No.	Initial state, inputs:	Test Status	Result	Testing Result
1.	Self.lastService = 2010-10-10 Self.mileageSinceService = 5000	Skipped	-	Skipped
2.	Self.lastService = 2010-7-10 Self.mileageSinceService = 11000	Executed	Self.inOffice = true Self.lastServiceDate = 2010-12-11	FAIL

The first test was skipped, because the generated test data did not satisfy the preconditions. The second test revealed an error in the implementation. With a given test data the attribute lastServiceDate was calculated incorrectly, it did not satisfy OCL statement “post: self.lastService = Date::now”.

IV. Conclusions and Future Work

We have presented a method for performing unit testing by using business rules as a test oracle. The testing method involves converting business rules to OCL statements and using them in test generation and testing results evaluation later. In order to detect an incorrect implementation a tester generates a sufficient amount of test data. By executing tests, he or she evaluates an output calculated by software against OCL statements. If constraints are not satisfied, the tester has established the fact of the existence of a bug in the implementation. The test data generation can be stopped at this point.

Our future work is targeting on developing method for automatically converting business rules from natural language texts to OCL statements (now this conversion has to be done manually). Also we want to develop a method for measuring the probability of the passed tests more precisely. We are willing to determine what kinds of faults our method can detect. We are also thinking about extending this method in order to define new testing criteria. The criteria should define when to stop generating test cases and finish testing when we have not found any errors.

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Global Positioning System – Working and its Applications

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Abstract- **The most promising and fully operational GPS (Global Positioning System) is a navigation system based on a group of 24 satellites developed by U.S. Department of Defence. Every satellite sends data to the receiver in the form of signals, having some information about satellite and orbital information. This paper aims to discuss the potential of GPS as constantly global communication medium and gives insights in terms of its business and military applications. It focuses on the architecture of GPS, its working, changing signal conditions and discusses its technical applicability and uses current advanced applications of GPS like vehicle tracking, location-based services as a case study. Even in worst weather conditions this GPS system is used to find out exact location with respect to timing information anywhere on the surface of globe. The paper will also consider different weather and geographical conditions to prove the suitability and compatibility of the GPS. The paper will also compare GPS with other navigation systems and will discuss its advantages such as high precision and continuous coverage.**

Keywords: Global Positioning System, satellite, signal, suitability, compatibility, location, navigation.

I. INTRODUCTION

Tracking and locating is an important part in every field of life whether it is in commercial sector or is in industrial. For finding location of anyone in anywhere on the surface of Earth, a modernized system was to be designed with its most sophisticated and modernized functions. Global Positioning System provides all the tracking and locating facilities through its satellite based network in any weather condition in any signal level.

First of all, this system was designed specifically for military usage for security reasons. Today, this system's applicability is everywhere in daily life applications. Due to continuous moving of satellites in orbit using GPS, every entity is to be identified on the surface of Earth with its more precise and accurate level. As compared to other tracking systems, this system provides the better services of locating and positioning with its real-time tracking [2].

This paper focuses on the current implications of Global Positioning System, its architecture and also this manuscripts defines its operations and working with clear aim. The paper concludes that GPS played a pivotal role in every part of modern life and is a revolution in the positioning and navigation systems.

Fig1.0. Satellites constellation orbiting the Earth [15]
Source: Bureau of Oceans and International Environmental and Scientific Affairs (2006)

II. GENERAL OVERVIEW

GPS, a well known Global Positioning System, is a collection of total 27 satellites moving around the Earth in an elliptical path [1]. This system was designed by American Department of Defense for the sake of security in different applications of military in 1970s [3]. After that, modifications like accurate timing, velocity related information are to be done in this system for getting better precision and accuracy level. GPS is the fully compatible to provide signals for locality, movement measurement and tracking of vehicles and other type of transport anywhere on the surface of Earth to satellites [1].

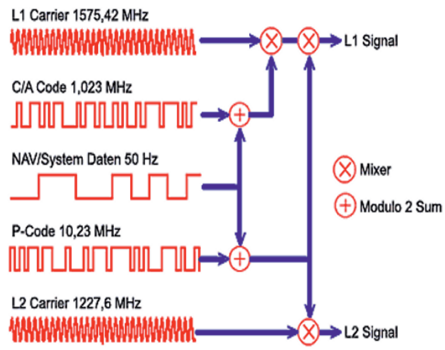

Fig 2.0. GPS Signal structure [16]
Source: Kowoma (2005)

As compared to other navigation systems, GPS has fully operational capabilities to define location and

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direction of receiver with full accuracy. Currently this system is used in many scientific as well as in civil applications like Surveying, GIS (Geographic Information System), Geology and Mobile Satellite Communications in modern way. Location can be determined through the time difference for each satellite signal through satellite receiver [5].

A. GPS Background & Evaluation

GPS is a system having full capability of tracking and locating services with the help of its satellites, receivers and built in antennas on each receive [1].Initially, American Navy planned to develop navigation system in the name of "TRANSIT" to pilot their ships for military operations in early 1960s. There are different drawbacks in this system like inefficiency and slow speed. After the TRANSIT, another navigation system "TIMATION" consisting of seven low altitude satellites having additional feature of atomic clock inside each satellite. The big hole in this system was its inefficiency, poor accuracy and two-dimensional navigation system.

In the meanwhile, U.S. Air Force decided to develop 612B system having three dimensional capabilities in the form of altitude, latitude and longitude coordinates. The other developed navigation systems were not full operational worldwide. In 1973, American Air force and Navy decided to make a system in the name entitled NAVSTAR Global Positioning System with full operational capabilities. This system was premeditated by American Defence Authority for its military safety particularly in Persian Gulf War. This satellites constellation was launched into orbit in 1980 and aftermath of this was in the form of better precision and exactness. Onwards, this fully operational designed navigation system is to be used in many civil applications worldwide.

B. Characteristics of GPS

GPS has been renowned due to its prominent and advanced features in civil applications and some business point of view. The first and foremost feature of GPS is its accurate description of location with the respect of 3 dimensional co-ordinates up to 100m but this accuracy can be more refined further up to 3cm with the help of DGPS (Differential Global Positioning System). This modernized system not only uses the satellites but also use the base stations on Earth [5].

GPS uses the triangulation method for finding more accurate position and precise timing with the help of timing clocks built on GPS satellites on the surface of Earth between GPS receiver and GPS satellite.

According to Theiss et al (2005), *"The atomic clocks on the satellites are accurate to within a nanosecond. The receivers contain only a quartz clock, but are very accurate due to the constant synchronization with the satellite' atomic clocks".*

Fig 3.0. Satellite sending the navigation messages [14]
Source: Kintner 2004

GPS receivers have latest features and compatible with its satellites through different techniques. The atomic clock used in GPS satellites are much costly as compared to quartz clock used in receiver but the clocks are much accurate and synchronized with GPS satellites due to its advanced built up features. GPS receivers can show the exact direction of satellites orbiting the Earth and also depicts the signal strength coming from GPS satellites. Another built up advanced feature in GPS receiver is to add graphical latitude and altitude coordinated in the shape of map datum [8] [5].

III. GPS STRUCTURAL DESIGN

Fundamentally, GPS comprises of basic three segments like space segment, user segment and control segment having their own working independently [3]. System time and orbital location is to be known very precisely through each satellite on-board computer and navigation message generator. The information coming from GPS receivers is broadcasted with the speed of light in the navigation messages to satellite for calculation of approximate ranges from each satellite. The messages used in satellite system travel in two code sequence like C/A- code with the bit length of 1023 and other encrypted code with the bit length of 6 trillion.

The system time and position of each individual satellite can be measured on the basis of these types of codes. The information coming from satellites are transported to control system through L-band frequency and the signals are uploaded to space segment through S-band frequency. Satellites use L-band frequencies to minimize the level of interference, ionospheric effect on the transmission of GPS signals. Now, all the segments contained by GPS will be discussed separately.

A. Space Segment

The space segment plays the pivotal role in this whole system and all the major functionalities are dependant on this part of GPS. This segment is a constellation of 27 satellites moving around the Earth in an elliptical path at an height of 11,000 nautical miles having the total no of six orbital planes with the inclination angle of 55• each other. Currently, this segment is further divided into four blocks and a 5^{th} block is to still under development. These blocks are to be shown diagrammatically here separately:

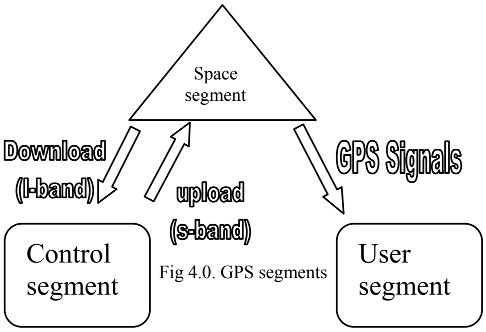

Fig 4.0. GPS segments

1) Block I / II

The initial version of GPS testing was in the shape of Block I satellites comprising of 10 satellites with the capability of storing navigation messages for 14 days. In this system, there was no capacity of having on-board momentum management [9].

The system having the average life capacity was five years and based on the Cesium and Rubidium frequency standards. After the working on the Block I, the need is to avoid satellite system from the cosmic rays effects. The Block II system with average life of 7.5years was designed for high security of military in support of extra features like selective availability and antispoof.

2) Block IIA / IIR

After the successful launching of above mentioned system, more modifications are to be done in the result of Block I IA and I IR with upgraded results. The promising feature in case of Block I IA was the enhancement of on-board storage capability in the range of 14-180 days. The drawback of this system was wholly dependable on ground control system. Block I IR series satellites were launched in 1998 after the careful consideration of more 20 satellites in Earth orbit with full accuracy having the average design life of 10 years [9].

3) Block IIF

The final version in the name of Block IIF of GPS constellation is under consideration and the existing positioning system will be replaced with the support of 33 I IF satellites. This enhanced version of the system with the support of second coded civil signal and enhanced cross link capability is less dependant on the ground control system [10][9]. Block IIF satellites have the average design life of 15 years.

B. Operational Control Segment

This is most important part of the GPS and plays a pivotal role in this whole scenario. All the activities supported by the satellites orbiting the Earth are to be managed through this control segment. It comprises of Master Control Station, Monitor stations and collectively different antenna system. The important atmospheric data, location of satellite etc. is uploaded via control segment to satellites through S-band frequency level.

This part of GPS is ground based and monitors all the movement of satellites moving around the Earth through intelligent built in software packages and hardware. The software used in Master Control Station is responsible for handling Block IIR satellites' functioning and is capable of handling 20 monitor stations at a time. The growing monitoring network results in the better precision and accuracy level.

C. User Segment

The third part of this system consists of user equipments like GPS receivers which are directly interact with the satellites. The signals transmitted by satellites are to be processed by receivers in L-band frequency range to evaluate user's direction, time etc.

GPS signals are to be received and manipulated through its receiver and predict the position of the user anywhere on the surface of Earth with the help of maximum four satellites. Now, more and more modern applications like air traffic management surveying and tracking are to be facilitated through this astonishing technology.

IV. GPS OPERATION AND ITS WORKING

As described in the above mentioned discussion, GPS is a navigation system which is used for tracking and locating of objects. Location can be identified anywhere on the surface of Earth through the concept of resection with the help of at least three satellites and a point from a GPS receiver [3]. Signals generated by each satellite continuously are composed of two carriers, encrypted or decrypted codes and a navigation message. GPS receiver has a built up antenna and it receives the signals directly coming from satellites through this antenna. The antenna' receivers process the signals automatically coming directly from satellites.

A. Errors sources in GPS

There are various sources of errors faced by GPS measurement. These errors may be caused by satellites, GPS receiver or signal propagation level. Different factors like signal strength, atmospheric changes, multi-path and sudden climate changes, system clocks error built in satellites play role in originating the errors. These sources of errors are to be eliminated through mathematical modelling, different algorithms and intelligent software etc. Next section will discuss the different levels of errors and the ways to eradicate these errors as follows, *1) Multi-path errors*

The prominent feature in signal propagation through free source is multi-path and sometime this results in the deterioration of the signal strength and other variations. The signals coming from satellites to GPS receiver follow direct LOS and also in NLOS (Non-Line-of-Sight). Multi-path signals coming from various ways distort the original signal strength and results in errors. These types of errors are to be overcome through advanced built up features in GPS receiver by Strobe correlator and the MEDLL technologies [19]. These types of errors can be eliminated by these technologies even in the reflective environment [13].

2) Ionospheric Errors

Ionospheric effect happens due to the presence of free negative and positive charges in atmosphere at the height of above 50km from Earth. The Ionosphere medium directly effects on the GPS radio signals and deteriorate the signal strength coming from different Ionospheric layers. The medium slows down the speed of signals before reaching to GPS receivers coming from satellites and results in range error [3].

3) Satellite and receiver clock errors

Each individual Satellite and GPS receiver uses their own clocks separately and these clocks are to be managed for getting better precision and accuracy. Satellite uses atomic clock and GPS receiver uses Quartz clock which is less accurate as compared to atomic clock but is cheaper than atomic clocks. The error range in satellite atomic clock is to be measured through multiplication of error clock with the speed of light. The core part of the satellite, "Ground Control System" monitors and checks all the errors occurred in the system.

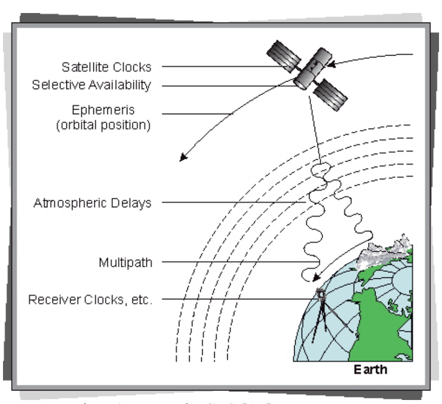

Fig 5.0. Errors in GPS [17]

4) Human errors

There are some types of errors caused by satellites Engineers. According to recent news in BBC, Engineers lost contact with the satellites. NASA (National Aeronautics Space Administration) found that faulty change made to the space craft's computer memory caused the battery to overheat and resulted in much loss [18].

V. GPS APPLICATIONS

In the past, GPS (Global Positioning System) was limited for the military purposes with the use of different satellites. Now, there are many applications in industrial as well in commercial sectors. The system having full potential of precision and accuracy gives cost effective solution of tracking and locating. The most exciting feature of this system is to be monitored and controlled automatically through "Operational Control System". This section describes the various applications from the areas of Transporting, Engineering, Tracking, Weather forecasting etc.

A. Accurate and Precise Timing

GPS has been applied to determine exact timing in civil application whether it is dynamic or static. The most precise and accurate timing is really important in any electric company or electrical system. The most prominent American electric company has applied the GPS technology for minimising the extra consumption of power and electricity.

Accurate and precise timing is required for generation and transfer of power with the help of modern Global Positioning System. For getting and analysis of appropriate information regarding electric company, GPS time tags are required. In contrast of GPS satellites, its receivers are useful for determining exact location where power goes down.

B. Surveying and Mapping

GPS is also a good tool for practical surveying and mapping of different utilities like water, gas etc. With the help of this system, distance between two locations can be determined. GPS can provide solution of water mapping and better location can be traced and verified for utility mapping through this attractive positioning system.

This provides cost effective solution for exact location of utility installation and mapping with the help of GIS (Geography Information System) and no extra ground marking is required. For mining industry in oil drilling purposes, this system provides the precise place for this and this eradicates the extra cost.

C. Traffic management and logistics usage

Another advanced feature of this navigation system is in the area of traffic management and logistics. In logistics, GPS plays also its own role with the efficient usage in big cargo companies. Space for shipping containers is to be carefully monitored with accurate and precise space to a few centimetres. Real time tracking is also possible with the help of this emerging technology and acknowledges back to the base station with accurate position.

The system provides strategic advantages to the companies having support of this technology with the cost effective solution. In transportation era, Global Positioning System also helps for management of traffic in the efficient way. Currently, an efficient named as traffic manager consisting GPS receiver provides updated information in case of jamming of traffic occurs and traces alternative routes for this case. Owing to this tremendous achievement in transportation management, better and cost effective solutions are available [2].

D. Security Measurement

In today's modern technology world, security plays a core role in any sort of application. GPS provides the solution for different companies and businesses to enhance security. Security is highly concerned in case of transporting and big vehicular companies.

All dangerous and toxic materials like weapons are uploaded through big cargo companies and highly monitored and keep in touch continuously with GPS base station. There are two cargo companies like PAR Logistics and Savi have the built in Global Positioning System for providing high security in case of hijacking and kidnapping [15][5].

E. GPS Vehicle Navigation and tracking

Most interesting and upcoming application done by GPS (Global Positioning System) is in the form of vehicular navigation or vehicular tracking and this technique provides an efficient solution for drivers having unfamiliarity with routes or roads. Before this technology, road drivers use paper road maps for identification of proper routes. Having inefficiency of this system in busy areas, this technique does not provide better solution for route tracking.

With the advancement of new GPS technology in this area having digital road map and a computer navigated system results in easily manipulation of road tracking electronically with the touch of button[16]. The digital road map having built in information regarding street names, road direction, airports gives the solution of easy and safe tracking continuously.

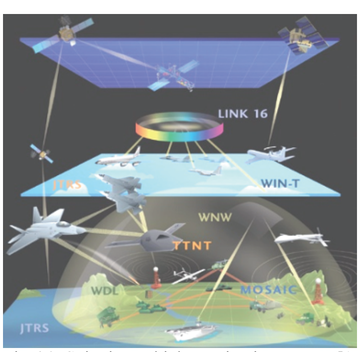

Fig 6.0. Spinning vehicle Navigation system [15]
Source: Rockwell Collins

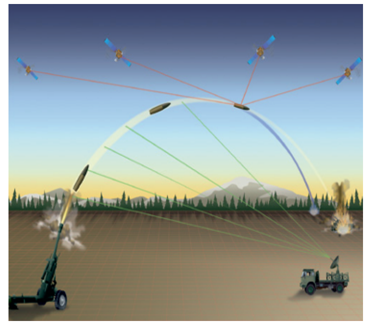

Fig 7.0. Vehicle navigation structure [15]
Source: Rockwell Collins

F. GPS Weather forecasting

Weather forecasting is an important application predicted by the modern Global Positioning System. It predicts the accurate weather forecast through its intelligent system. Accurate weather forecast and improvement of weather data is possible through the establishment of GPS [18]. Meteorologists predict the accurate weather forecast on the basis of signals change coming directly from satellites. This application is also an important factor for prediction of future threats in case of storm, tornadoes, hurricanes etc.

G. GPS for precision farming

The more precise and modernized positioning system in the name DGPS (Differential Global Positioning System) revolutionized the farming industry. Different attributes like soil sample collection, harvesting machinery can be controlled this technique. Vehicular steering system used in this application is directly interlinked with the satellite.

For spraying purposes, aerial guided system integrated with GPS can be used for right spots and location in the farms with less overlap and with accurate rate. The steering system built on automatic system is used for much time and controlled directly through satellites. Mapping of crop yields is to be done through this integrated positioning system [12].Yield rates are to be calculated through Differential Global Positioning System through mapping and now this system is available in existing market.

VI. COMPARISON OF DIFFERENT POSITIONING SYSTEM

Like GPS, there are other navigations systems like Galileo, Beidou and GLONASS developed by Europe, China and Russia respectively. All these navigation systems play their own role with less coverage as compared to GPS developed by U.S.A. Basically, GLONASS works like GPS and each satellite transmits signals having L-band carriers and a navigation message. Beidou is the navigation system and the satellites are placed in geostationary orbits with the maximum altitude of 36,000km above Earth surface. Galileo satellite based positioning system proposed by European standard bodies.

In this future based system, all of constellation types like LEO (low Earth orbit), MEO (medium Earth orbit) and IGSO (inclined geosynchronous orbits) are to be analysed. Galileo has an edge over other two GLONASS and Beidou due to its high security measures and two levels of services: free and chargeable.

VII. CONCLUSION AND ITS IMPLICATION

In the consequences of all above mentioned discussion, GPS played a pivotal role in every part of modern life and revolutionized in the positioning and navigation systems. After careful considerations, one should have a good knowledge about its working, its applicabilities in civil as well as in military side. This up and coming technology benefited the businessmen and they flourished their businesses due to this astonishing and cost effective navigation system.

As compared to other positioning systems like Galileo, Beidou and GLONASS etc, Global Positioning System provides more accurate and precise measurements with the support of more than two dozens satellites.

For future perspective, Global Positioning System has indulged in every walk of life and more and more applications are under considerations in this context. The problem faced by GPS is line of sight and this can be overcome through the use of special types of antenna on its receivers. Future implications and betterments in this system are possible through modern techniques like DGPS (Differential Global Positioning System), Precise monitoring. Another latest achievement in the form of GPS modernization is expected in coming few years in 2013 with the help of new civil signals and military code.

In this paper, I described the general overview of the Global Positioning System, its different applications, its working. Finally, this manuscript discussed the future implications and the ways of its betterment and enhancement. In a nutshell, this growing technology will be adopted by everyone in every part of life.

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Building Anti-Virus Email and File Storing Service Based on Grid Computing

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Abstract - Grid Computing is a new trend being developed in Information Technology. It helps us take full advantage of processing and storing capacity and other underused resources to provide an environment of high processing capacity and plentiful storing ability and using it to solve complicated problems with very low cost.

The growth and popularity of computer science and Internet in both work and daily life have been taken place in the world. Network security is a serious problem and now, users have been facing many problems such as computer virus infection as transmitting files, virus emails, spam emails, or attacking DoS/DDoS/ … on Internet.

This theme which is analysising and establishing a system based on Grid Computing technology first solves the most important problems protecting system from virus infection as transmitting/storing files and emails with large quantity.

I. INTRODUCTION

Grid computing technology is a new trend in Information Technology. The appearance of Grid Computing has marked a great achievement in the development of high performance computing. It helps us take full advantage of processing and storing capacity and other underused resources to provide an environment of high processing capacity and plentiful storing ability, and uses it to solve complicated problems which the current technology has difficulty dealing with or which can only be achieved at high financial cost [1], [2],[3],[4]. In the past years, a number of large IT organisations and corporations have chosen grid computing as their developing strategy and have invested much time in researching its practical applications[6],[18], [26].

II. THE NETWOKK SECURITY ISSUES

The growth and the popularity of computer science and the Internet in both work and daily life have been worldwide and nowhere less than in Vietnam. Network security is an urgent need and so has become one of great interests of Vietnamese IT companies.

Taking full advantages of the ability of a network environment to share data easily is wider and deeper. In some occasions and some situations, this requirement is crucial and it may lead the network system to easily pick up a computer virus infection.

At work Internet users often receive many emails daily, any of which may introduce a virus to their computer system. This problem is always inconvenient, and potentially disastrous.

When setting up a system to which users connect and to satisfy the user's desire for a safe network, the importance of avoiding unexpected security problems and inconveniences cannot be understated, especially if the user is part of a large company, organization or ISPs. This is of concern to people with limited IT experiences as in reality most of people who use IT don't know much about it.

This theme which is analysing and establishing a system based on Grid Computing technology first solves the most important problems protecting the system from virus infection whilst transmitting/storing files and emails with large quantity.

III. PROPOSED SOLUTION

As mentioned above, the system, which supports anti-virus scanning (in the files and emails) must scan an average of hundreds to thousands of emails/files every day, which can be equivalent to some tens of GBs..

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 454–459.

Nowadays, most of models in business as well as in different organizations are based on the conventional models (using strong configuration as mail/FTP servers). However, these systems have some weaknesses:

- Scale and range of operation of companies and corporations are more and more extended, which leads to their network systems facing more and more difficult processing or storage problems.
- Other weaknesses of these models are bottleneck and low reliability situations.

There are a lot of ways to solve the above problems such as buying expensive specialised hardware devices, using Cluster/Grid Computing technology, …

Using Grid Computing technology will have many benefits as follows:

- Allowing active managing of load balancing on servers, to schedule jobs in order to get the most benefits and also avoiding server overloading in grid computing.
- Replacing or updating some servers during operation does not delay the system
- The cost to employ the system is low, and the system can process very large quantities of data (emails/files)

IV. ANTI-VIRUS EMAIL SERVICE BASED ON GRID COMPUTING

A. Anti-virus problem in emails

Email systems in large companies and organizations, when totalled up, receive and process from hundred of thousands to millions of emails each day. These systems often integrate more anti-virus functions but scanning through such an enormous quantity of emails takes time.

Using Grid Computing technology takes full advantage of computers' processing ability in the Grids Computing system in order to decrease time in processing emails.

This scheme does not build the whole system but makes use of freely available open source tools/software existing on the Internet. The following open source soft wares are chosen to establish anti-virus system in emails based on Grid Computing:

1. QMail

Nowadays, there is a lot of Mail Transfer Agents (MTA) – often accompanied with Sendmail - in the UNIX environment; however, Sendmail is insecure and the software is very bulky. Qmail, which is programmed by Dan Bernstein, repairs many security errors. Qmail is MTA open source, which is used popularly and highly valued because of its ease of operation and high performance standards.

2. Clam AntiVirus

Clam AntiVirus (ClamAV) is an anti-virus tool. It is designed especially to scan emails on the email gateway/server. Clam AV provides a lot of flexible, extendable tools, dealing with multi-processes, scanning virus via command line and also providing a convenient tool to update data frequently.

B. The current email service model

The following figure describes the basic components and operation principles of the current email service

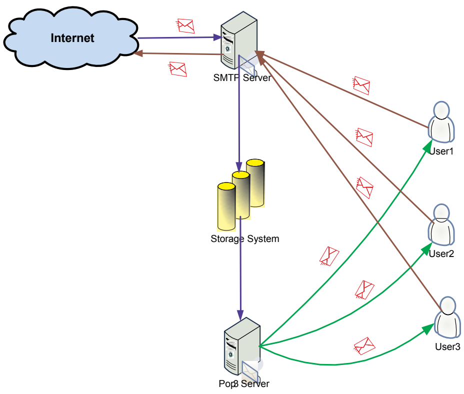

Figure 1: Current email service model

The above model is used by most large companies/organisations. Mail system consists of SMTP Server (receive/send mails), Storage System (store mails), POP3 Server (users access to receive mails by POP3). POP3 Server and SMTP Server can be in the same computer.

Anti-virus or anti-spam checking of emails is carried out at SMTP Server.

C. Email service model based on Grid Computing technology

With the email model in Figure 1, it's easy to recognize that more numbers of mails, more time to process emails. The following figure describes elements and operative principles of email service that improve to solve the above problem:

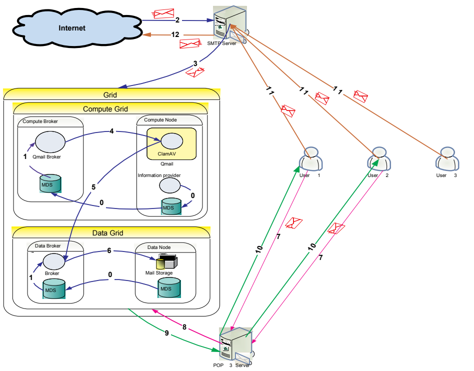

Figure 2: Email service model based on Grid Computing technology

The system is based on Grid Computing Technology by using open source code softwares:

- Middleware Globus Toolkits [7],[8],[9],[11]supports the possibility of managing system information and services in network by MDS service (Monitoring and Discovery System) [5],[13].
- Ganglia's function is to provide some information about local host such as CPU speed, the numbers of CPU, free RAM capacity, … to MDS (0) and Broker to schedule network processes (1).
- Qmail and ClamAV.

From the model in figure (1), when emails from Internet come into the system (2), SMTP server will get these emails and send them to Grid System (3), via Computer Broker at which the emails will be scanned before being stored. Broker program in Computer Broker System undertakes to monitor emails coming from SMTP server and then send to suitable Computer Nodes to be virus-scanned (4). This distribution is based on the information from MDS about these Computer Nodes.

After finishing the scanning process, emails that have virus will be deleted; virus-free emails will be sent to System Data Grid (5). Data Broker gets these emails and selects the suitable Data Node to store them.

When users want to receive mails, through Mail Client, they send requests to POP3 Server (7).After authenticating user account successfully, POP3 Server asks System Store email Data Grid (8) and gets emails from this system, then sends them back to users (10).

When users want to send mails, emails will be sent to SMTP Server (11). The SMTP Server will analyse the destination email addresses, if those addresses are members of an internal network then emails will be sent to Grid System (3) to be processed. Conversely, emails will be sent to more suitable other SMTP Server on Internet (12).

D. *Experiment Model*

The following figure describes real system deployed in the general model:

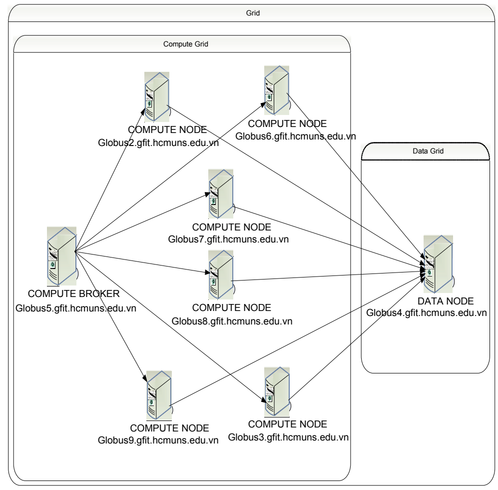

Figure 3: Experiment model for antivirus email service

The system is deployed at the lab of Network Computer and Telecommunication Department, Faculty of IT, The University of Natural Sciences, which consists of 9 PCs, three of which are Pentium IV 2.4 GHz, RAM 256MB-512MB, six of which are Pentium II 400MHz, RAM 192MB-256MB.

Compute Grid system is built with all functions that are analysed in the general model.

Data Grid system now can only store data at one assigned host.

V. STORING ANTI-VIRUS FILE SERVICE BASED ON GRID COMPUTING

A. *An anti-virus problem in file storing system*

Nowadays, there are many online file-storing services on Internet, which give top priority to the security and confidentiality of information passing into servers. So, it is necessary to have a method to control these data files and to make sure that each file is clear of virus infection before it is accepted; and can then be stored in the system.

With the basic choices and the technology mentioned above, the main purpose of this project is to build the system to provide online storing and virus scanning functions for files uploaded by users.

B. *General compositions-model*

The following figure describes basic elements and operative principles of anti-virus file storing service and relations among them:

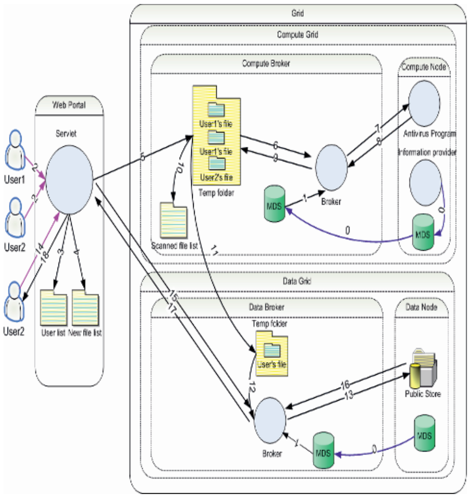

Figure 4 : Genaral compositions model in antivirus file storing service

System is built on Grid Computing Technology by using open source code softwares:

- Middleware Globus Toolkits [7],[8],[9],[11] supports the possibility of managing system information and services in network by MDS service (Monitoring and Discovery System) [5],[13].
- Gangila has function as providing some information about local host such as CPU speed, the numbers of CPU, free RAM capacity, … to MDS (0) and Broker to schedule network processes (1).
- Middleware Gridbus [15],[19],[21],[22],[23] supports the possibility of scheduling tasks to hosts in grid network environment.

Using this system, user can send requests about storing files through web interface to system by HTTP protocol (2). After user signs in successfully (3), system will save file name (4) and then starts receiving data. The data flow will be sent to Grid system by FTP protocol (5) and is virus scanned before it is accepted in this system

Broker program in Compute Broker system undertakes to observe files coming and distribute them to suitable Compute Nodes for virus scan (6, 7) which bases on information received from MDS. Result from the scanning will be written in a managed file (9, 10) and is used to decide if system should save that file. If detecting virus in file, system tries to delete virus (or delete infected file if it can't get rid of the virus). If there is no virus or the system completely deletes virus, it will transmit file to Data Grid system (11). In the Data Grid, there is also Broker monitoring new files to distribute them to suitable Data Nodes then to store them.

When users log in the system wanting to download a file (14), this request will be sent to Data Grid system (15). Broker will find a file having the most suitable version sought from the storage devices (16) and then creates a data flow to Web Portal by FTP (17). The data flow will be transmitted to users by HTTP (18).

Grid system's manipulations such as sending files to virus scanning system, receiving virus scanning results, storing files on distributed system using GridFTP [12],[14] and be transparent to users, and users only receive notice whether file is accepted to store on system or not.

C. *Experiment model*

The following figure describes real system employed in the general model:

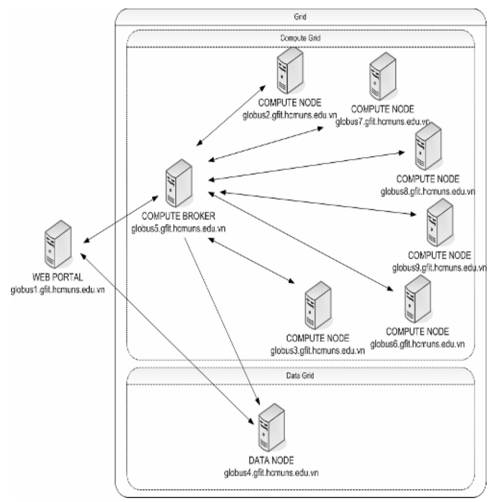

Figure 5: Experiment model for antivirus file storing service

The system is deployed at the lab of Network Computer and Telecommunication Department, Faculty of IT, The University of Natural Sciences, which consists of 9 PCs, three of which are Pentium IV 2.4 GHz, RAM 256MB-512MB, six of which are Pentium II 400MHz, RAM 192MB-256MB.

Compute Grid system is built with all functions, which are analysed in the general model.

Data Grid system now can only store data at one assigned host. Broker is used to manage distributed storage and has not been developed completely.

D. *Experiment results*

- The main goal is to test the ability of whole system operation based on the above design. Experiment proves that design is logical and appropriate.

- The second goal is testing the performance system with a gradual increase of file quantity. So in the experiments, it is supposed that users upload files to the system in advance, and only wait for the system to scan files for virus before storing them.
- Each file which is virus scanned has a capacity of 1MB and it is assumed that anti-virus programs are saved in advance at Compute Nodes, so it will not be necessary to download them to Compute Nodes whenever scanning virus.
- Experiment result is described in the following scheme:

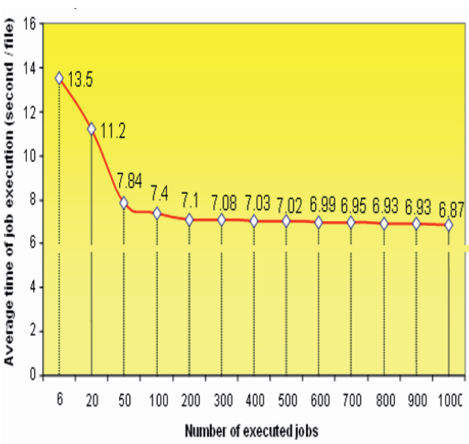

Figure 6: Average implement time graph per a scanned file

The above figure shows that the number of tasks are more and more, while time for implementing every task is less and less. So, system is more effective with the large quantity of tasks.

VI. CONCLUSTION AND TREND OF DEVELOPMENT

- Succecssfully designing detailed models of two email services such as anti-virus and file storage based on Grid Computing.
- Implementing successfully restricted system (without Data Grid) with two above services. System operates well with designed functions.
- Anti-virus file storing subsystem is tested about its performance and the result is suitable to the official experiments of Gridbus Broker group for applications used to store data by using Gridbus Broker [20]. However, performance of anti-virus mail subsystem hasn't been tested yet.
- Finishing researching, choosing and using open-source softwares suitable to implement the above system, but there is not enough time to correct and to modify them for optimising the system's performance.
- The above services can expand the ability of storing on Data Grid in order to increase flexibility, ability of large storing, security and reduce time consumed during processing.
- We can make more functions such as anti-spam based on Grid Computing technology for email service.

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Mathematical Model and Simulation of a Plate Heat Exchanger Operating as Steam Generator

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***Abstract*—This paper shows a one-dimensional mathematical model for a plate heat exchanger operating as steam generator; the generator is one of the main components of an absorption heat transformer. The model shows the global heat and mass balances, the heat and mass balances in the interface, the model of heat transfer, the heat transfer coefficients and the friction factor for the generation process. This is a complete model for the heat and mass transfer to determine the concentration and temperature profiles. Commercial software Agilent HP Vee Pro 7.5 was used for the simulation.**

Index terms: mathematical model, steam generator, plate heat exchanger, water – Carrol™ mixture.

I. INTRODUCTION

High energy consumptions are part of the industrialized world, the energy demand, principally, fossil fuels, has increased in the last years, with the consequent decrease in the disposition of these fuels, therefore, it is important the application of new technologies that allow the use of alternative energies. An Absorption Heat Transformer (AHT) allow to utilise alternative energy (like solar energy), increasing the temperature from this source to a higher level, which can be applied in different processes, like industrial, in effluents purification systems or in water purification systems. The AHT use a working mixture for the absorption process, one of these mixtures is *Carrol*, developed by the company Carrier [1], this mixture has demonstrated to have better performance in comparison with other mixtures as Water – Lithium bromide [2]. One of the main components of an AHT is the steam generator, its efficiency in the heat and mass transfer has an impact in the performance of the system. In this work it was developed a one-dimension model for heat and mass transfer in this component, the simulation will predict temperature and concentration profiles. As far as we know, there are not previous works, where a complete model is presented in a steam generator in an AHT with the mixture Carrol.

II. HEAT TRANSFORMER

An AHT consists of an evaporator, a condenser, a generator, an absorber and an economizer. Fig. 1 shows an AHT in a plot diagram pressure against temperature.

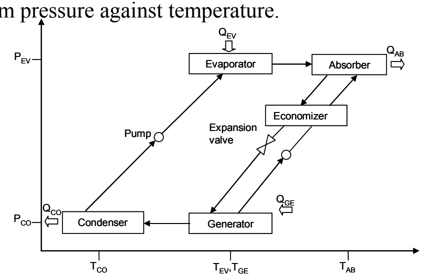

Fig. 1. Eschematic diagram of AHT

The function of the generator is to evaporate a quantity of water from a working mixture, and in this case the mixture Carrol™ (lithium bromide + ethylene glycol) - water is considered.

A one-dimension mathematical model for the steam generation from Carrol – water mixture is proposed. The steam generator is a horizontal plate heat exchanger, made of Stainless Steel 316; designed with a thermal capacity of 3000 W.

III. ONE-DIMENSION MATHEMATICAL MODEL

The model was established based on the control volume show in Fig. 2. It was considered a horizontal tube and it corresponds to a channel of the heat exchanger. The heat flux is transferred to the working solution that flows inside the tube, no heat losses are considered, water is the heating fluid. In the tube the flow is characterized by a complete separation of liquid and vapor phases (stratified flow), pool boiling process is not a good correspondence with this model. The liquid film thickness is assumed constant. The generator is in steady state.

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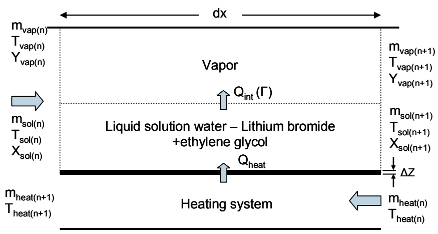

Fig. 2. Volume control

Mass balance:

$$m_{vap(n)} + m_{sol(n)} = m_{vap(n+1)} + m_{sol(n+1)} \tag{1}$$

$$m_{vap(n)} * Y_{vap(n)} + m_{sol(n)} * X_{sol(n)} = m_{vap(n+1)} * Y_{vap(n+1)} + m_{sol(n+1)} * X_{sol(n+1)} \tag{2}$$

Energy balance:

$$Q_{heat} = m_{heat}(H_{heat(n)} - H_{heat(n+1)}) \tag{3}$$

Heat transfer equation:

$$Q_{heat} = UA\Delta T_{GE} \tag{4}$$

Where:

$$U = \frac{1}{\frac{1}{h_{heat}} + \frac{\Delta Z}{k_{SS}} + \frac{1}{h_{sol}}} \tag{5}$$

As far as where it has been revised in the literature, there are not available correlations to predict the heat transfer coefficient for Carrol solution, reference [3] proposed a correlation for obtaining heat transfer coefficients for working mixture water – lithium bromide. In this work is proposed utilize the same correlation for the working mixture water - Carrol:

$$Nu = C_3 \mathrm{Re}^{0.7} \mathrm{Pr}^{1/3} (C_4 Bo^{0.5}) \tag{6}$$

Where C_3=0.2 and C_4=88 are constants to calculate the Nusselt number in the generator. The heat transfer coefficient of the heating fluid (water) the following correlation can be used:

$$Nu = C_1 \mathrm{Re}^{a} \mathrm{Pr}^{1/3} \tag{7}$$

Where C_1 and a depend on the geometry of the plates. For plates with an angle of 60°, $C_1 = 0.2$ and $a = 0.7$ [1].

The log mean temperature differences in counter-current and co-current flow respectively are:

$$\Delta T_{gencf} = \frac{(T_{heat(n)} - T_{sol(n+1)}) - (T_{heat(n+1)} - T_{sol(n)})}{\ln \dfrac{T_{heat(n)} - T_{sol(n+1)}}{T_{heat(n+1)} - T_{sol(n)}}} \tag{8}$$

$$\Delta T_{genfp} = \frac{(T_{heat(n+1)} - T_{sol(n+1)}) - (T_{heat(n)} - T_{sol(n)})}{\ln \dfrac{T_{heat(n+1)} - T_{sol(n+1)}}{T_{heat(n)} - T_{sol(n)}}} \tag{9}$$

Energy balance at the interface

The model for the total heat transferred through the vapor – liquid interface was proposed in based in reference [4]; it is obtained by the addition of sensible heat and latent heat:

$$q_{int} = q_{sen} + q_{lt} \tag{10}$$

Sensitive heat occurs due to the temperature difference of liquid and vapor

$$q_{sen} = h_{vap}(T_{liq} - T_{vap}) \tag{11}$$

The heat transfer coefficient proposed by Prandtl

$$h_{vap} = \frac{(f_{int}/2)\mathrm{Re}_{vap}\mathrm{Pr}_{vap}}{1 + (8.7(f_{int}/2)^{0.5}(\mathrm{Pr}_{vap} - 1)} \left(\frac{k_{vap}}{D_h}\right) \tag{12}$$

The vapor core diameter

$$D_h = D_i - \delta \tag{13}$$

The liquid film thickness is given by

$$\delta = \frac{D_i}{2}(1 - \alpha^{0.5}) \tag{14}$$

α is a function of density and flow quality:

$$\alpha = \left(1 + \left(\frac{1 - x_f}{x_f}\right)\left(\frac{\rho_{vap}}{\rho_{sol}}\right)^{2/3}\right)^{-1} \tag{15}$$

The interfacial friction factor is calculated from:

$$f_{int} = f_{vap}\left[1 + 12\left(\frac{\rho_{sol}}{\rho_{vap}}\right)^{1/3}(1 - \alpha^{0.5})\right] \tag{16}$$

The friction factor f_{vap}

$$f_{vap} = f_{Dc}/4 \tag{17}$$

And the Darcy friction factor f_{Dc}

$$\frac{1}{f_{Dc}^{0.5}} = -1.8\log\left[\frac{6.9}{\mathrm{Re}} + \left(\frac{\varepsilon / Di}{3.7}\right)^{1.11}\right] \tag{18}$$

ε/Di is relative roughness.

Latent heat flux is

$$q_{lt} = \frac{D_i}{4D_h}\Gamma H_{fg} \tag{19}$$

The evaporation rate is calculated from

$$\Gamma = \frac{4}{D_i}\frac{q_w - \sqrt{\alpha} q_{sen}}{H_{fg}} \tag{20}$$

The heat flux at the wall

$$q_w = h_{sol}(T_w - T_{liq}) \tag{21}$$

The heat transfer coefficient for the working solution can be calculated from the the Dittus – Boelter equation:

$$h_L = 0.453\frac{k_{sol}}{D_i}\mathrm{Re}_{Lo}^{\ 0.5}\,\mathrm{Pr}_{sol}^{\ 0.33} \tag{22}$$

The frictional pressure gradient can be written as follows:

$$\left(\frac{dP}{dx}\right)_F = 2 f_{lo}\frac{m_{sol}^{\ 2}}{D_i\rho_{sol}}\phi_{lo}^2 \tag{23}$$

The two phase multiplier can be determined from:

$$\phi_{lo}^2 = (1 - x_f)^{1.8}\left(1 + \frac{20}{X_{tt}} + \frac{1}{X_{tt}^2}\right) \tag{24}$$

Mass balance at the interface

The equation that defines the mass transfer at the interface is [5]:

$$\nabla N_{sol} + \frac{\partial C_{sol}}{\partial t} - R_{sol} = 0 \tag{25}$$

Considering steady state, no chemical reaction and without mass accumulation at the interface the above equation reduces to:

$$\frac{dN_{sol}}{dx} = 0 \tag{26}$$

Where the molar flux of solution

$$N_{sol,x} = -cD_{sv}\frac{dy_{sol}}{dx} + y_{sol}(N_{sol,x} + N_{vap,x}) \tag{27}$$

Where c is molar density of the working mixture, D_{sv} is the diffusion coefficient, $\frac{dy_{sol}}{dx}$ is the gradient of concentration. The first term of the equation corresponds to the contribution for the concentration gradient (diffusion), while the second corresponds to the contribution for the mass convection. We considered that the vapor is not absorbed in the mixture; therefore the previous equation reduces to:

$$N_{sol,x} = -\frac{cD_{sv}}{1 - y_{sol}}\frac{dy_{sol}}{dx} \tag{28}$$

Integrating this equation with limit conditions:

$$N_{sol,x}\int_{x1}^{x2} dx = cD_{sv}\int_{ysol1}^{ysol2} -\frac{dy_{sol}}{1 - y_{sol}} \tag{29}$$

Solving, we obtain the molar flux for the solution:

$$N_{sol,x} = \frac{cD_{sv}}{(x_1 - x_2)}\ln\frac{(1 - y_{sol2})}{(1 - y_{sol1})} \tag{30}$$

IV. RESULTS AND DISCUSSION

Fig. 3 shows a plot of heat transfer coefficients against heat flux for three different concentrations (50, 52 y 55%). Those results has been compared with similar systems previously reported [6], which operate with the mixture water – lithium bromide. The values of the heat transfer coefficient for the water – Carrol mixture are slightly lower that for the mixture water - lithium bromide. Heat transfer coefficients decrease with an increase in the concentration. The presented model is an initial approach, since the system will be model in two dimensions, which will allow a bigger approach to the plate heat exchanger geometry. Fig. 4 shows a part of the program for the one – dimension model. HP VEE Pro 7.5 software was selected for programming. This software allows data acquisition and to call MATLAB functions, bringing more options for, in the future, to simulate the steam generator in two dimensions.

V. CONCLUSIONS

A one – dimension model for a steam generator in an absorption heat transformer coupled was presented and programmed. The model considers the heat and mass transfer for two phase flow of the working mixture water –Carrol™. This model is a first approach for a future two - dimensions model, which will allow, more accuracy in the geometry of the PHE. The model was programming in HP VEE Pro software with satisfactory results.

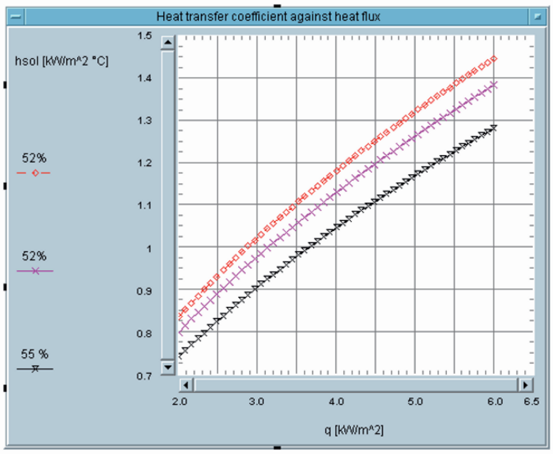

Fig. 3. Plot heat transfer coefficient against Boiling number for three different concentrations

VI. ACKNOWLEDGMENTS

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VII. NOMENCLATURE

Nomenclature
A *Area; [m²]*
Bo *Boiling number; [dimensionless]*
D_h *Diameter of vapor core; [m]*
D_{sv} *Diffusivity; [m²/s]*
D_i *Inner diameter; [m]*
f *friction factor; [dimensionless]*
f_{Dc} *Darcy friction factor; [dimensionless]*
h *Heat transfer coefficient; [W/m² K]*
H *Enthalpy; [J/kg °C]*
H_{fg} *Enthalpy of vaporization; [J/kg °C]*
k *Thermal conductivity; [W/m K]*
L *Total tube length; [m]*
m *Mass flow rate; [kg/s]*
Nu *Nusselt number; [dimensionless]*
P *Pressure; [Pa]*
Pr *Prandtl number; [dimensionless]*
Q *Heat flux; [W]*
q *Heat flux; [W/m²]*
Re *Reynolds number; [dimensionless]*
R *Rate of production [mol/m³ s]*
T *Temperature; [K]*
U *Global heat transfer coefficient; [W/m² K]*
x *molar fraction; [dimensionless]*
X *Solution concentration in liquid phase; [%W]*
X_f *Flow quality; [dimensionless]*
X_{tt} *Lockhart – Martinelly parameter; [dimensionless]*
Y *Solution concentration in vapor phase; [%W]*

Greek symbols
α *void fraction; [dimensionless]*
δ *Liquid film thickness; [m]*
Γ *Evaporation rate; [kg/m³ s]*
ρ *density; [kg/m³]*
μ *Viscosity; [Pa s]*
ΔZ *Plate thickness; [m]*
ΔT *Log mean temperature difference; [°C]*

Subscripts
AB *Absorber*
CO *Condenser*
EV *Evaporator*
GE *Generator*
gencf *Generator in counter-current flow*
genfp *Generator in co-current flow*
heat *Heating system*
int *interface*
liq *Liquid*
lo *The total mass flux flowing with the liquid properties*
lt *Latent*
sen *Sensible*
sol *Solution (working mixture)*
ss *Stainless Steel*
vap *Vapor*
w *wall*

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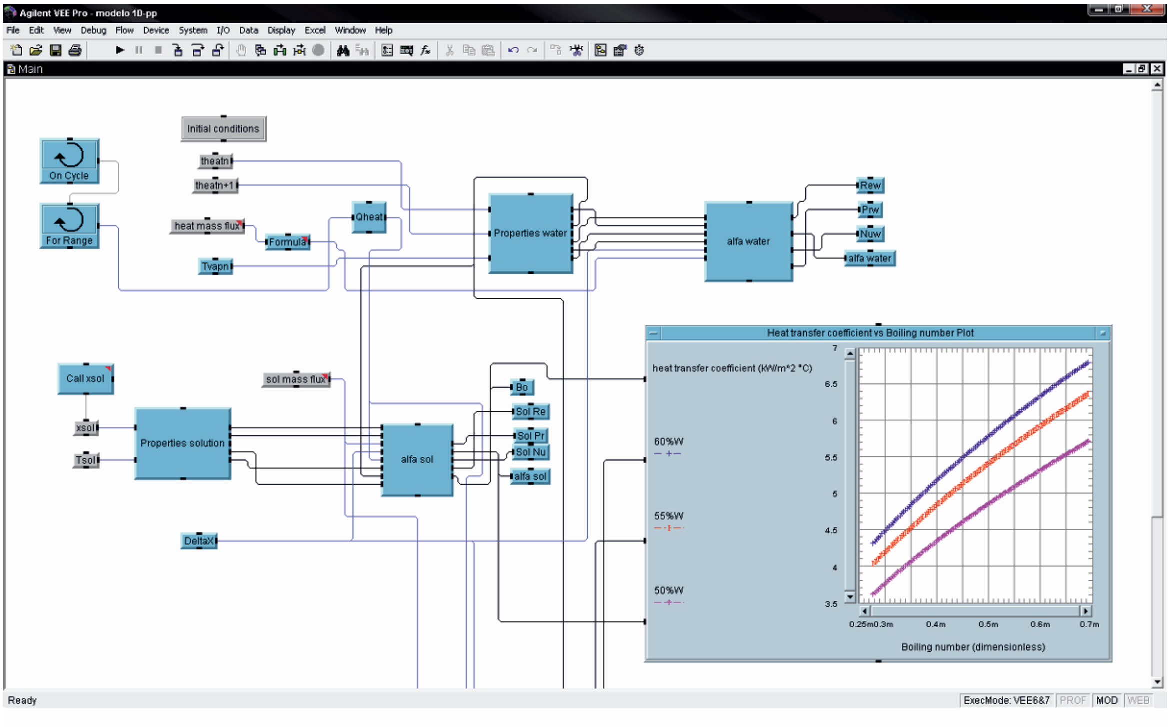

Fig. 4. Part of the one – dimension program for the steam generation model.

Architecture for Belief Revision in Multi-Agent Intelligent Systems

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Abstract: **The Semantic Web provides a framework for agents to easily publish and consume structured knowledge suitable for automatic reasoning. However, the open and distributed nature of the Web causes information of dubious quality to be published by sinister or incompetent agents. At the same time, Web accessible knowledge can change often. As a result, agents are forced to function in an environment of unreliable, incomplete, and contradictory information. They require mechanisms to change their beliefs dynamically and remove contradictions while maintaining the quality of the knowledge base. Computational agents might rely on social networks to judge information quality. Relationships of trust help determine the degree of belief an agent may ascribe to a given proposition. A previous work has proposed a formal logic for agent-oriented reasoning about belief and trust. This logic allows agents to deduce their confidence levels on both received and inferred knowledge. The present paper shows how to employ this logic in a collaborative document repository for scientific literature. Such a system relies on user-supplied classification data to construct comprehensive, personalized taxonomies for document browsing and search. It is based on a system of multiple agents representing users, constantly involved in knowledge acquisition and belief revision to provide the users with the best data available. The paper describes the theoretical foundations and briefly outlines a software implementation.**

1 INTRODUCTION

The Semantic Web [3] is a vision of a new generation of the WWW that will enable computational agents to process data that have well-defined formal semantics. Agents can use reasoning to infer new information from statements provided by users. As a model example, consider an archive of publicly available resources updated by a community of knowledge workers (e.g., a preprint archive of scientific papers). Indexing information in a digital archive requires extensive metadata schemas, enabling accurate categorization for navigation and search. Such a metadata system would include concepts organized in a taxonomic hierarchy, thesaurus, or more complex knowledge structure utilizing ontology semantics. One obstacle for utilizing such schemas for large and/or evolving archives, such as digital repositories of research papers, is that the users often have conflicting perspectives on overlapping areas of inquiry. This makes it difficult to difficult to define a taxonomy or ontology of concepts that will satisfy the needs of the majority of users. Ideally, the metadata should reflect the perspectives of different groups while spanning the entire content of the library, as well as recognize links between alternative conceptualizations. At the same time the system should maximize the quality of the ontology. The Swoogle search engine [11] is a model application of these ideas.

It is our position that this task is best left to the evolving community of users to carry out. This paper outlines a Web-based system that allows users, individually and through collaboration, to define ontologies for classifying documents in a collection. Users will form online communities around domains of interest, contributing to ontology engineering through discussions and collaborative editing. The key idea is that different ontologies will be created by different individuals or groups, and the resulting ontologies can then be combined on overlapping concepts and used to improve search in the repository. Each user's or group's association with their ontologies will provide the context necessary for personalizing their search and browsing

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experience, identifying items of their particular interest. The collaborative environment can also be used to create communities of practice, supporting informal knowledge exchange.

Sufficiently expressive concept taxonomy embodies a knowledge structure that reflects the real world. We treat the taxonomy as a knowledge base expressed in a formal language that allows inference of explicit facts. The knowledge base can be used by computational agents that act on behalf of a user to assist in browsing, query, and search. We require that agent's knowledge base be locally consistent, meaning that it does not contain any obvious contradictions (such as saying that a particular document is both in and not in a given classification category). We also require that the agent use as much information as possible, meaning that it must merge knowledge bases provided by different users. If the merging is to be performed non-critically, the consistency requirement would probably be violated due to the unrestricted distributed nature of the Semantic Web.

2 TAXONOMIC KNOWLEDGE

The overall system is comprised of automated agents that act on behalf of the human users. These agents act as both information suppliers and consumers. As a consumer, the agent's goal is to find documents relevant to the user's interests. As an information supplier, agents make their own documents available for other agents. We assume that it is in the best interests of the human users to make their documents easily searchable, and that they will provide rich metadata to make it so.

We use first-order logic as a basis for our bibliographic metadata scheme. Documents will be represented as individual constants (names) denoted by alphanumeric identifiers starting with a lowercase letter. For the purposes of this treatment we call them simply *document1*, *document2*, etc. Document categories, or topics, are denoted by a unary predicates, e.g., $Logic(x)$, $Philosophy(x)$, $Mathematics(x)$, $ArtificialIntelligence(x)$. Thus, the formula $ArtificialIntelligence(document1)$ postulates that document1 belongs to the topic Artificial Intelligence.

Users will define taxonomic relationships using first-order logic. The four allowable types of relationships are illustrated by the following examples:

Subtype: $\forall x(Logic(x) \rightarrow Math(x))$

Disjointedness:

$$\forall x(Mathematics(x) \rightarrow \neg Philosophy(x))$$

Or-Definition:

$$\forall(ClassicalLanguages(x) \leftrightarrow Latin(x) \vee Greek(x))$$

And-Definition:

$$\forall x(Biochem(x) \leftrightarrow Biology(x) \wedge Chemistry(x))$$

The system also allows for negative classifications, e.g., $\neg\ Logic(document1)$. Agents acquire these expressions either from its human user or from other mechanical agents. Whenever such a statement is entered into a agent's knowledge base, various reasoning tasks are triggered automatically. These are aimed at deducing any classification statements that may arise as logical consequences of the new statement. For example, if the knowledge base already contains the above subtype statement, and the agent receives the statement $Logic(document1)$, then the system will automatically deduce $Mathematics(document1)$ and add this to the knowledge base. As another example, if the agent's knowledge base contains the statement $Logic(document1)$, but does not yet contain the above subtype statement, then adding the latter would again cause the agent to infer $Mathematics(document1)$. Similar routines can be invoked for the other types of statements. The objective is to derive as many such classification statements as is feasible, as these in turn provide the information necessary to support search. When the user queries the system for documents related to some topic, all the necessary reasoning tasks will have already been performed.

3 MULTI-AGENT ENVIRONMENT

Each agent in our world maintains two data structures: its belief set, used for requesting documents that are classified within its knowledge base, and public metadata of documents that are intended to be exposed to the public by the user. We assume that our agents are sincere: they share all their beliefs concerning the topic taxonomy with the entire agent community. This helps other agents to classify documents the agent publishes. Thus, all published document descriptions are available from their owners and are globally public. We can represent this as either the existence of a global knowledge repository, which pools the knowledge bases of the participating agents, or ability of agents to provide knowledge upon query. We can assume, without loss of expressivity, than individual and predicate names are globally unique.

There are no assumptions concerning global consistency---we expect that formulas that are attributed to different agents can and will contain contradictions. Moreover, we do not require that, at any given time, an agent's view of a taxonomy be consistent, although this can be seen as a goal of each agent's belief revision process.

An agent's actions include the following:

- Responding to queries for documents.
- Exposing a new document created or discovered by the user. This includes classifying the document into one or more of the taxonomic categories. It introduces a new named individual into the domain description, effectively expanding the language.
- Accepting a previously unknown document into the agent's belief set. This may come from the human user or another agent.
- Inferring new beliefs from previously acquired beliefs in the manner described in the foregoing Section 2.
- Modifying currently held beliefs to remove a contradiction. This action is triggered when a reasoning step results in deriving falsity. In this event, a belief revision process is initiated, removing beliefs that lead to contradiction.

Along with the model of the bibliographic domain, agents maintain beliefs regarding other agents. We assume that a human user provides his or her agent with an initial list of "known" agents for the agent to communicate with, along with a level of trust bestowed onto statements those agents have. These statements form an oriented weighted graph of familiarity relationships between agents, or a *social network*. The level of belief that an agent may place on a statement acquired from another agent accordingly depends on the degree of trust the first agent has in the second agent. The belief level may be computed according to a rule for combining belief and trust. Such a rule that is based on numeric values for belief and trust has appeared in [8]. A logic for reasoning about linguistic values of knowledge and belief was published in [10], and this has been extended to incorporate reasoning about linguistic trust values in [13]. The latter is employed in the present work.

4 BELIEF REVISION

Agents use their belief sets to satisfy user requests for documents. Regardless of the specific query mechanism (information retrieval or taxonomy browsing), the agent relies on the quality of the taxonomy and the completeness of document information in its belief set. Therefore, an agent must continuously acquire document descriptions from other agents. This process can introduce contradictions into belief set.

When dealing with contradictions, an agent may have to abandon some beliefs. The well-known AGM framework [1, 6] defines *rationality criteria* or "integrity constraints" that propose criteria for abandoning previously-held beliefs. In informal terms, those postulates include:

- Belief change should keep the epistemic state (belief set) consistent.
- Any statement logically entailed by statements in the belief set should be included in the belief set.
- Loss of information should be kept to a minimum. This principle is referred to as a principle of informational economy or, more narrowly, the principle of conservation.
- Beliefs that are in some sense "more important" should be retained in favor of those "less important".

The AGM framework then defines, for an epistemic state *K* (here the agent's knowledge base) and epistemic input ά (some statement), belief change operators transitioning one epistemic state into another (i.e., modifying the knowledge base). These are Belief Expansion (denoted $K \rightarrow K^+_\alpha$), Belief Contraction (denoted $K \rightarrow K^-_\alpha$), and Belief Revision (denoted $K \rightarrow K^*_\alpha$). From the rationality criteria described above, formal rationality postulates are derived, constraining the classes of possible change operators (i.e., of the three kinds just mentioned). For example, for Belief Revision AGM provides 8 postulates are as follows (where $K_\perp$ refers to an absurd belief set that contains all possible formulas):

- **Closure:** For any input α and any belief set K, K^*_α is a belief set.
- **Success:** $\alpha \in K^*_\alpha$
- **Inclusion:** $K^*_\alpha \subseteq K^*_\alpha$
- **Preservation:** If $\neg\alpha \notin K$, then $K^+_\alpha \subseteq K^*_\alpha$
- **Vacuity:** $K^*_\alpha = K_\perp$ if and only if $\neg\alpha$ is derivable from K
- **Extensionality:** If $\alpha \leftrightarrow \beta$ (α and β are semantically equivalent), then $K^*_\alpha = K^*_\beta$
- **Superexpansion:** $K^*_{\alpha\vee\beta} \subseteq (K^*_\alpha)^+_\beta$
- **Subexpansion:** If $\neg\beta \notin K^*_\alpha$, then $(K^*_\alpha)^+_b \subseteq K^*_{\alpha\wedge\beta}$

Similar postulates are formulated for Belief Expansion and Belief Contraction. It turns out that the expansion operator is uniquely determined by the corresponding rationality postulates: to wit, the expanded set is the closure of the union of the initial set and the new formula, or, in symbols, $K^+_\alpha = Cl(K \cup \{\alpha\})$. Revision and contraction, moreover, are connected by Levi's identity:

$$K^*_\alpha = (K^-_\alpha)^+_\alpha$$

and also by Harper's identity:

$$K^-_\alpha = K \cap K^*_\alpha$$

Therefore, only one of contraction or revision need be defined. Abstract ways to construct such operators are proposed, with representation theorems proving that the resulting sets of operators are equivalent to ones defined by postulates. One of the constructions introduces the notion of epistemic entrenchment, an ordering over the set of all formulas (or, equivalently, over the set of possible worlds). Intuitively, this ordering represents a preference ordering over formulas in the belief set.

Our system adopts this AGM framework, but with two important modifications. As mentioned, AGM requires that a belief set be closed under implication, i.e., that it contain all logical consequences of the agent's beliefs. While theoretically appealing, this requirement does not lend itself to efficient computational implementation. In our treatment we replace this potentially infinite knowledge structure with a derivation path, a finite sequence of statements that the agent is aware of at a given moment in time. This makes use of the notion of a "path logic", first published in [9]. This is accomplished by viewing inference rule applications as occurring in discrete time steps, and taking the agent's knowledge base as consists of only those statements that have been derived (or received) as of the present time.

Thus we drop the foregoing AGM requirement that any statement logically entailed by statements in the belief set should be included in the belief set. But we do require that all inference processes be logically sound. We do not require completeness, however, although this too would be a desirable property. It is required only that the inference mechanisms be adequate to allow the agents to achieve their practical goals.

A second modification to the AGM framework is to replace the consistency requirements with a capability for the agent to eventually reach a consistent state, given enough time. Thus, at any given time, and assuming the absence of further inputs, an agent should be able reach a consistent state in a finite number of steps. Actually executing those steps, and ultimately reaching the consistent state, however, is not required. A rationale for this modification is that this more closely resembles actual human reasoning. Peoples' views of the world often harbor inconsistencies without their being aware of it. It is only when a contradiction appears in their consciousness (here represented by the agent's knowledge base) are they motivated to modify their view so that the inconsistency can be removed. In this case, a contraction operation must be performed to remove one or more assumptions that led to the contradiction. It may be noted also that in the particular application studied here, namely, managing indexes for digital libraries, inconsistencies in the metadata (the knowledge base) have the character of being nuisances but typically will not seriously damage the system's overall usefulness as a tool for search.

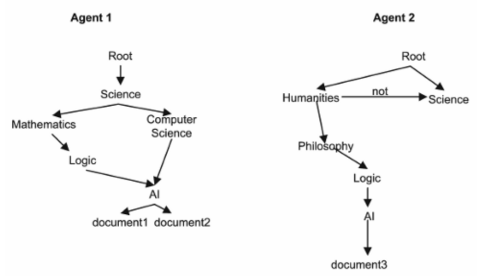

Figure 1. Example knowledge bases.

Let's consider an example. Assume that two agents have beliefs as shown in Figure 1. Using first-order logic, the beliefs of the agents are:

Agent 1

$\forall x(Science(x) \rightarrow Root(x))$

$\forall x(Mathematics(x) \rightarrow Science(x))$

$\forall x(CompScience(x) \rightarrow Science(x))$

$\forall x(Logic(x) \rightarrow Mathematics(x))$

$\forall x(ArtificialIntelligence(x) \rightarrow Logic(x))$

$\forall x(ArtificialIntelligence(x) \rightarrow CompScience(x))$

$ArtificialIntelligence(document1)$

$ArtificialIntelligence(document2)$

Agent 2

$\forall x(Humanities(x) \rightarrow Root(x))$

$\forall x(Science(x) \rightarrow Root(x))$

$\forall x(Humanities(x) \rightarrow \neg Science(x))$

$\forall x(Philosophy(x) \rightarrow Humanities(x))$

$\forall x(Logic(x) \rightarrow Philosophy(x))$

$\forall x(ArtificialIntelligence(x) \rightarrow Logic(x))$

$ArtificialIntelligence(document3)$

Further assume that Agent 1 acquires the description of document3 from Agent 2. This description naturally contains all the taxonomic knowledge presented.

The resulting knowledge base will lead to a contradiction: document3 (in fact, all three documents) belongs both to Science and Humanities, two categories that are declared disjoint. Thus, the agent must disbelieve at least one sentence in his knowledge base. The "culprit" formulas that might be removed include:

- $\forall x(Humanities(x) \rightarrow \neg Science(x))$
- $\forall x(Philosophy(x) \leftarrow Humanities(x))$
- $\forall x(Logic(x) \rightarrow Philosophy(x))$
- $\forall x(ArtificialIntelligence(x) \rightarrow Logic(x))$

After carefully examining the results of such revisions, we can see that, in each case, we lose some potentially useful knowledge. This includes not only the statement in question, but also all statements whose derivations have depended on it. Thus in making our decision regarding which statements to disbelieve, we adopt a principle of informational economy, choosing beliefs that are the "least valuable", where a statement's value as measured simply as the number of subsequent statements whose derivations depend on it. In this case, applying contraction to the statement that is least valuable will remove the smallest number of statements from the knowledge base.

5 IMPLEMENTATION CONSIDERATIONS

We next present a model for the architecture of a software system supporting our collaborative digital library. The proposed architecture demonstrates the practical applicability of the concepts and techniques discussed. The system design will be created using a combination of agent-oriented and object-oriented approaches.

The simplest system architecture for a collaborative document repository consists of: (i) a central database to store knowledge, (ii) business logic implementing multiagent system behavior, and (iii) a Web-based user interface. Alternatively, one can design a distributed, peer-to-peer software system to implement the same conceptual model. A distributed architecture will additionally contain protocol specifications for information exchange.

At the center of the software specification lies the data model for the application domain. Section 2 presented a conceptual vocabulary of taxonomic knowledge, represented using first-order logic. Our goal is to create a schema that supports this vocabulary, allows for a straightforward physical implementation using mature technology (e.g., relational database or XML). From the previous discussion we can derive two major requirements for the model of the bibliographic domain:

- **Personification:** Knowledge elements must be explicitly associated with agents. Our data model contains all the knowledge of the multi-agent system, so that every agent has access to knowledge of other agents. At the other hand, each individual knowledge base (or belief set) is clearly identified.
- **Dynamics:** The belief revision mechanisms mentioned above rely on explicit representations of theagent's belief evolution in time.

Our domain model implements these requirements by explicitly representing users (agents) and by adding a timestamp to each taxonomy element. Thus, all historic data is always present in the system. All knowledge elements are identified both by the timestamp and the user. At any given time, different users might have different views to the same model. The view is affected by the user's group membership and relationships between groups. This allows for great flexibility and is a great asset in a collaborative environment where global consensus is unlikely or undesirable. (We assume this is true for the bibliographic system we describe).

The object-oriented design community has adopted a concept of design patterns, borrowed from the architectural design [2]. Design patterns are the semi-formal, systematic descriptions of the solutions for common design problems. They show a particular commonly occurring problem in a specific context and then describe a proven method of solving such a problem. The goal of patterns is to increase the quality of design, namely maintainability and extensibility, by communicating best industry practices. Among commonly used patterns in object-oriented design are "Model-View-Controller" for flexible user interface, and "Factory", which suggests encapsulating object creation to a special class to avoid relying on the type information when creating objects during runtime.

Metadata schemas, like software in general, have two facets. First, they are meant to naturally represent those aspects of a specific domain that are important for users. Second, they are dictated by the architecture of the software system they are part of and constrained by various technological requirements. Design patterns are meant to deal with the second facet, addressing nonfunctional requirements like extensibility, maintainability, and performance.

In [12], we discuss the use of object-oriented design patterns in defining metadata models, using a variation of the application domain described here. We showed how two patterns, Composite and History, can be used in improving the characteristic of a data model. The two patterns are combined, resulting in the data model depicted in Figure 2. The two design patterns complement each other: Composite allows treating thesaurus elements uniformly, while History solves change management issues for unified metadata pieces. Each pattern facilitates the other in achieving its advantages.

This schema supports the dynamic metadata management system. Metadata is stored in the relational database. Classes are implemented as tables. Inheritance is implemented as a one-to-one relationship. Stored procedures are used for querying Taxonomy classes, allowing the merging of the Entity table with the appropriate subclass table.

Using object-oriented approaches and patterns had shown to be a useful metaphor for metadata construction. Along with improving metadata architecture maintainability and extensibility, using patterns for metadata facilitates software design. The envisioned system can be implemented using an object-oriented platform, e.g. Enterprise Java technologies or Microsoft .NET and would include a Web-based user interface. Its main components should include a taxonomy browsing interface, a full text search facility for bibliographic data, collaboration facilities, and an administrative module.

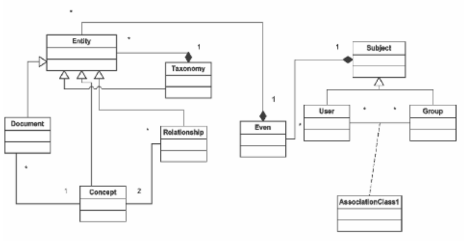

Figure 2. Draft data schema.

Business logic components mapped to the taxonomy follow the classic Composite architecture, adding behavior to the data model. The History pattern supports multi-user collaboration logic. Software components for data access can be designed using the Model-View-Controller architecture, with versioning support along the lines of the "Gang of Four" [5] Command pattern, structurally similar to History.

To function as a component of the wider Semantic Web infrastructure, the system should support well-defined metadata vocabularies for interoperability. To achieve this, the data must be exportable to the de-facto standard RDF/S and OWL formats. Web services are created to export taxonomy data using the SKOS RDF schema [7]. Additional bibliographic data will conform to Dublin Core vocabulary. Social network data is exposed as FOAF graphs [4]. Internal pattern based metadata schema is sufficiently rich in semantics to yield useful mappings to the vocabularies mentioned, while providing adequate flexibility for supporting the information system's object-oriented architecture and reasoning services.

After structural components of the system, including data structures and static software architecture, are defined, dynamic aspects must be addressed. The basic behavior of the agents will be captured in our software design. In addition to macro behavior specification, inference and belief revision algorithms discussed earlier will be expressed in terms of the specific data model, and responsibilities will be assigned to specific classes and methods. The appropriate tools to express these design decisions are UML sequence diagrams and state charts. A detailed design specification will serve as evidence of practicality and applicability of concepts we introduce, as well as a starting point for further research and development. As a secondary goal, our design effort will serve as an example of applying established software and data engineering methodologies to knowledge-based systems development. In future work, ties between object-oriented paradigm and formal semantics for metadata, e.g., as envisioned in the ongoing Semantic Web project, will be explored.

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Feature Selection Based on Semantics

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***Abstract* – The need for an automated text categorization system is spurred on by the extensive increase of digital documents. This paper looks into feature selection, one of the main processes in text categorization. The feature selection approach is based on semantics by employing WordNet [1]. The proposed WordNet-based feature selection approach makes use of synonymous nouns and dominant senses in selecting terms that are reflective of a category's content. Experiments are carried out using the top ten most populated categories of the Reuters-21578 dataset. Results have shown that statistical feature selection approaches, Chi-Square and Information Gain, are able to produce better results when used with the WordNet-based feature selection approach. The use of the WordNet-based feature selection approach with statistical weighting results in a set of terms that is more meaningful compared to the terms chosen by the statistical approaches. In addition, there is also an effective dimensionality reduction of the feature space when the WordNet-based feature selection method is used.**

I. Introduction

The task of document categorization is being carried out everyday. In today's computerized environment, many categorization tasks are still being done manually. This is due to the fact that most digitized documents are in the natural language. Therefore, those documents need to be preprocessed beforehand in order to make it understandable by the computers. The task of preprocessing the documents involves many processes, among which, one of the most significant process is the feature selection process. The feature selection process involves selecting a subset of keywords in a category to represent the category in the categorization task. Feature selection based on statistical approaches is commonly used. However, these approaches do not take into consideration the semantics of the natural language. Almost all our everyday documents are in the natural language. Therefore, in order to categorize them more effectively, there is a need to have the semantics component in the feature selection process.

In this research, we explore the hypothesis that incorporating semantics knowledge into feature selection can improve categorization accuracy and identify keywords that best describe a particular category. In the works that are carried out in this research, we attempt to explain how text categorization can be made more effective by incorporating WordNet as the semantics database in the feature selection process.

In Section II, we give an overview of text categorization. Section III will discuss statistical and semantics feature selection. An introduction to WordNet will be given in Section IV. Section V will briefly describe categorical sense disambiguation. Section VI will give an overview of the approach used for feature selection based on WordNet. Section VII will emphasize the dimensionality reduction achieved in this research. In Section VIII, the experiments are described and the results and analysis are presented in Section IX. Finally Section X concludes the paper with a summary.

II. Text Categorization

Text categorization is defined as assigning new documents to a set of pre-defined categories based on the classification patterns suggested by a training set of categorized documents. Automated text categorization is a field that has been around since the early 1960s [2]. In those days, categorization of text was done manually by constructing classifiers using some knowledge engineering techniques. In other words, it was done by gathering the knowledge of domain experts and then defining a set of rules that incorporate the experts' knowledge in categorizing the documents into a given set of categories. No doubt this technique is time consuming especially if the amount of documents is abundant. With a steep increase in digital documents over the years, manual categorization proves to be inefficient.

As the paradigms shifted in this computer age, the machine learning approach to text categorization starts to gain popularity. Many machine-learning schemes have been applied to text categorization and among them are Naïve Bayes [3], support vector machines (SVM) [4], decision trees [5] and so on. The application of machine learning in the field of text categorization only emerged in the 1990s. The concept behind machine learning in the task of categorization is generally described as a learner that automatically builds a classifier by learning from a set of documents that has already been classified by experts [2].

In this research, we apply the machine learning approach to text categorization. Fig. 1 shows the framework of the text categorization process. Our research focus is on the feature selection process to improve the effectiveness of the text categorization process.

III. Feature Selection: Statistical Vs Semantics

Feature selection is performed in text categorization to tackle the problem of the large dimensionality of the feature space. This process involves selecting a subset of features from the

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 471–476.

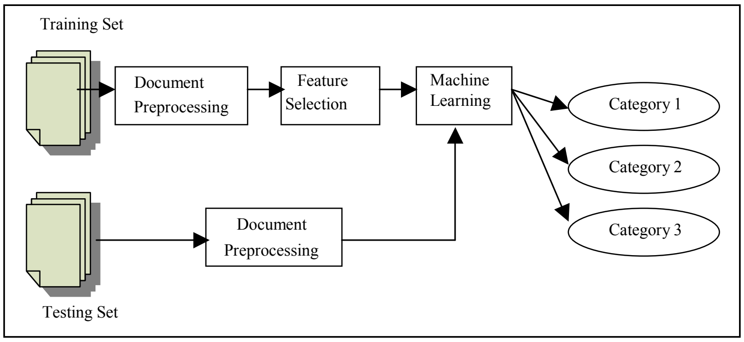

Fig. 1. The text categorization framework

feature space to represent the category. A feature space can contain thousands of features; however, it is not computationally efficient to process a large feature space. A good feature selection approach needs to be employed to select the most suitable features for category representation. There are a number of feature selection approaches, which over the years, are used in a wide range of text categorization tasks. The more widely used ones are statistical-based approaches, which will be discussed in the next section.

A. Statistical Feature Selection

Among the more widely used statistical-based feature selection approaches are Information Gain (IG) [2], [6], [7], [8] and Chi-square (Chi2) [2], [6], [8], [9]. Both IG and Chi2 can reduce the dimension of the vector space by a factor of 100 with no loss of categorization effectiveness [10]. It is thus desirable to develop feature selection approaches with a performance comparable to both IG and Chi2.

1. Information Gain (IG)

Information Gain (IG) or more popularly known as InfoGain, is a feature selection approach that makes use of the presence and absence of a term in a document to select its features. It is frequently used as a term-goodness criterion in the field of machine learning. The number of bits of information is measured for category prediction by using the knowledge of the presence and absence of a term. The amount of information term t_k contains about category c_i is measured and terms that are more indicative of a category based on their presence or absence are chosen.

For each unique term in the training set, information gain is computed and those terms that are above a predetermined threshold are selected as features. A term with a high information gain indicates that it is a good feature for category prediction. The formula of IG is shown in (1).

$$\sum_{c\in\{c_i,\bar{c}_i\}} \sum_{t\in\{t_k,\bar{t}_k\}} P(t,c)\cdot \log 2\frac{P(t,c)}{P(c).P(t)} \qquad (1)$$

2. Chi-Square (Chi2)

This method measures the degree of dependence between a term and a category. If a term is independent of a category, it will have a value of zero. A low value of Chi2 signifies a high degree of independence of term t_k and category c_i while a high value shows otherwise. A term with a high value of Chi2 shows that it is more dependant on a category and is therefore, more likely to be added to the feature space. These highly dependent features are selected because of their discriminating power. For each category, the Chi2 value for each unique term is computed and those terms that are below a predefined threshold are removed from the feature space. The formula of Chi2 is shown in (2).

$$\frac{\left[P(t_k,c_i)\cdot P(\bar{t}_k,\bar{c}_i) - P(t_k,\bar{c}_i)\cdot P(\bar{t}_k,c_i)\right]^2}{P(t_k)\cdot P(\bar{t}_k)\cdot P(c_i)\cdot P(\bar{c}_i)} \qquad (2)$$

B. Semantics Feature Selection

Semantics feature selection is not as widely explored as the statistical approaches. It refers to the selection of features based on its semantics value. Semantics is the study of word meanings. Digital dictionaries and word databases are commonly used to handle the linguistics aspects of text documents. Works by [11] investigated the usage of cascaded feature selection (CFS) in SVM text categorization. Their work highlights the potential of making use of synonyms in feature selection. Their approach shows promising results. Further to that, they also explore the use of parts-of-speech (POS) in a variable CFS [12]. Here, they use a two-step POS selection for SVM based text categorization.

In this research, WordNet, a lexical database, will be used to add in semantics information in the feature selection process. Unlike statistical feature selection, in semantics feature selection, a feature is chosen based on its semantics content, rather than based on its statistical value.

IV. Introduction To Wordnet

WordNet is an online thesaurus and an online dictionary. It can be considered as a dictionary based on psycholinguistics principles. WordNet contains nouns, verbs, adjectives and adverbs as parts-of-speech (POS). Function words are omitted based on the notion that they are stored separately as part of the syntactic component of language [1].

WordNet is organized by relations such as synonym, antonym, hyponym/hypernym and holonym/meronym. While synonym and antonym are lexical relations between word forms, both hyponym/hypernym and holonym/meronym are semantics relations between word meanings. Generally, synonyms are words having the same meaning and antonyms are words having opposite meanings. On the other hand, semantics pointers include “IS-A”, “PART-OF/HAS-PART”, “MEMBER-OF/HAS-MEMBER” and “SUBSTANCE-OF/HAS-SUBSTANCE” [1], [13]. The “IS-A” relationship is also known as the hyponym/hypernym relationship, where hyponym is the subset and hypernym is the superset. “PART-OF/HAS-PART”, “MEMBER-OF/HAS-MEMBER” and “SUBSTANCE-OF/HAS-SUBSTANCE” is also known as the holonym/meronym relationship. Holonym is the inverse of meronym where, if x is a holonym of y, then y is a meronym of x.

The information in WordNet is organized into sets of words called synsets. Each synset in WordNet has a unique signature that differentiates it from other synsets. Each of the synset contains a list of synonymous words and semantics pointers that illustrate the relationships between it and other synsets.

In this research, WordNet is chosen over other alternatives, as there are a few advantages of WordNet that can be exploited. First and foremost, it links related words in a structure defined as a synset. Words are ordered hyponymically, that is, they are grouped and sorted in a hierarchy based on their meanings. Different concepts are represented by different synsets [1]. Besides that, it is able to provide semantics information, consistently structured and electronically available.

V. Categorical Sense Disambiguation

WordNet contains a list of senses for each of the words in its dictionary. Therefore, WordNet is able to provide each word with a list of senses that it has. By looking at the context of a word in a category, WordNet can be used to provide the sense of the word. When two synsets overlap, the sense of each of the corresponding term is identified.

Although categorical word senses are identified in this research, the sense information is not used. We merely use the sense information to identify terms with dominant senses by finding the overlapping synset sense signatures. Further processing is required to incorporate the actual word sense of each noun. Therefore, categorical sense disambiguation is applied only to determine the sense of the synonymous terms for a category.

VI. Wordnet-Based Feature Selection

In the WordNet-based feature selection approach, only nouns are considered. Preliminary experiments indicate that the use of other parts-of-speech (POS) does not significantly enhance performance. Therefore, the nouns are first identified based on the nouns in the WordNet’s dictionary. Synonyms that co-exist in a category are cross-referenced with the help of WordNet’s dictionary. The terms obtained from cross-referencing will be the features that will be used to represent a category. This approach is illustrated in Fig. 2.

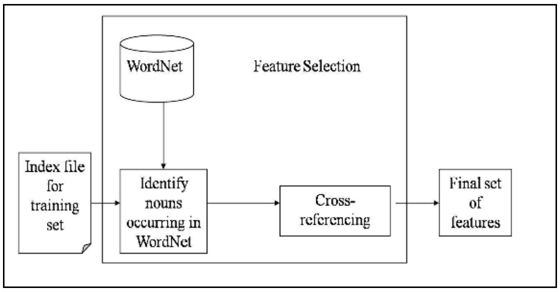

Fig. 2. The WordNet-based feature selection approach

The use of the WordNet-based approach allows us to determine whether semantics feature selection can enhance the quality of features for automated text categorization.

The difference between the semantics approach and the statistical approach is that, in the semantics approach, synonyms are chosen as features and are then weighted using Chi2 and IG. Our research makes use of the Chi2 and IG formulae from the works of [6].

With the WordNet-based approach for feature selection, insignificant words and noise can be filtered. These insignificant words consist of non-English words, wrongly spelt words, insignificant abbreviations and names. Terms like “govodi” and “pik” are actually meaningless in representing a category. Statistical approaches like Chi2 and IG do not take into consideration whether a term is misspelt or is reflective of a category. By using the WordNet-based approach for feature selection, we can actually tackle this problem by filtering these terms and at the same time, make use of the available synonym relationship and word senses in WordNet to identify semantics features in a category. The WordNet-based feature selection approach is able to choose a set of terms that is more reflective of a category’s content. This is in line with inducing a classifier that will act more like a human expert rather than having a classifier rely only on statistical findings.

The example below illustrates the approach used. We will look at all the senses for four nouns; “corn”, “maize”, “acquisition” and “ship”. Each sense has a signature, which is referred to as a synset. Every synset contains synonyms to reflect a sense.

TABLE I
LIST OF TERMS AND THEIR SYNSETS FOR EACH SENSE

Terms	Synsets for all senses
Corn	**Sense 1: {corn, maize, Indian corn, Zea mays}**
	Sense 2: {corn}
	Sense 3: {corn, edible corn}
	Sense 4: {corn, clavus}
	Sense 5: {wheat, corn}
	Sense 6: {corn whiskey, corn whisky, corn}
Maize	**Sense 1: {corn, maize, Indian corn, Zea mays}**
	Sense 2: {gamboge, lemon, lemon yellow, maize}
Acquisition	Sense 1: {acquisition}
	Sense 2: {acquisition}
	Sense 3: {learning, acquisition}
	Sense 4: {skill, accomplishment, acquirement, acquisition, attainment}
Ship	Sense 1: {ship}

From Table I, two identical synsets are identified (indicated in bold). They are sense 1 of "corn" and sense 1 of "maize", which have the same synset signatures. Thus, the terms "corn" and "maize" will be selected as terms to represent a category in feature selection. The use of categorical sense disambiguation is employed here to automatically disambiguate semantically related terms. The dominant senses for terms in each category can be determined by cross-referencing to find identical synsets with the same signatures.

VII. DIMENSIONALITY REDUCTION

The WordNet-based feature selection approach is also effective in reducing the dimensionality of the feature space. Table II shows the percentage of terms reduction when the approach is used compared to the number of unique terms in each category. Generally, the WordNet-based approach for feature selection is able to reduce the number of terms by more than 67%.

TABLE II
PERCENTAGE OF TERMS REDUCTION FOR THE REUTERS-21578 TOP TEN CATEGORIES

Category	No. of unique terms in each category	No. of terms selected by the WordNet-based approach	Percentage of terms reduction (%)
Acq	10760	2793	74.0
Corn	3089	955	69.1
Crude	5890	1834	68.9
Earn	10152	2342	76.9
Grain	5231	1716	67.2
Interest	3679	1147	68.8
Money-fx	5323	1667	68.7
Ship	3902	1258	67.8
Trade	5569	1823	67.3
Wheat	3358	1109	67.0

The effective reduction displays the ability of the WordNet-based approach to reduce noise while preserving the original contextual information of the documents.

VIII. EXPERIMENTS

The experiments that are carried out are to test and compare the effectiveness of semantics feature selection and statistical feature selection. The dataset used is the Reuters-21578 top ten most populated categories. The experiments are carried out using the Waikato Environment for Knowledge Analysis (WEKA) [14] machine learning tool, applying the multinomial Naïve Bayes machine learning scheme.

Experiments are carried out to compare and contrast the following feature selection approaches; Information Gain (IG), Chi-square (Chi2), WordNet-based feature selection using IG weighting (W-IG) and WordNet-based feature selection using Chi2 weighting (W-Chi2). The formulae for IG and Chi2 are obtained from [6].

Chi2 and IG are chosen in this research to be used for comparison because previous experiments by other researchers have proven that these approaches are successfully implemented as statistical feature selection approaches. Therefore, by making use of these two approaches as benchmark for comparison, we will be able to see how well the proposed WordNet-based feature selection approach can perform.

There are two sets of experiments. Each set of experiments consists of 6 different term sizes: 10, 20, 50, 100, 200 and 500. These different term sizes were chosen to evaluate the effectiveness of the classifier to see which term size can give the optimal performance for the classifier. The two sets of experiments are:

1. Comparison between Chi2 and WordNet-based feature selection using Chi2 weighting (W-Chi2).
2. Comparison between IG and WordNet-based feature selection using IG weighting (W-IG).

The aim of these experiments is to determine the effectiveness of the WordNet-based feature selection approach.

IX. RESULTS AND ANALYSIS

The F_1 measure with micro-averaged scores across all the ten categories is used in measuring the results. F_1 measure combines both the value of precision (P) and recall (R) to give a more effective result to indicate the effectiveness of the classifier's performance. The formula of F_1 measure is given in (3).

$$F_1 = 2 \cdot P \cdot R / (P + R) \tag{3}$$

Fig. 3 and 4 shows the micro-averaged F_1 measure of Chi2 and W-Chi2, as well as, IG and W-IG, for the Reuters-21578 top ten categories.

From Fig. 3 and 4, it is noted that both W-Chi2 and W-IG is able to perform better than the statistical approach itself, with the exception at term size 10. At all other term size thresholds, there is a slight increase in the F_1 measure value when the WordNet-based approach is used with the statistical weighting. The reason for this is that the WordNet-based approach needs a larger number of features to capture adequate representative categorical features, as well as, semantics information. This is the reason why it did not improve on the results of the statistical approaches at term size 10, while across all other term sizes, the WordNet-based approach with the statistical weighting is able to produce some improvements over the statistical approach itself.

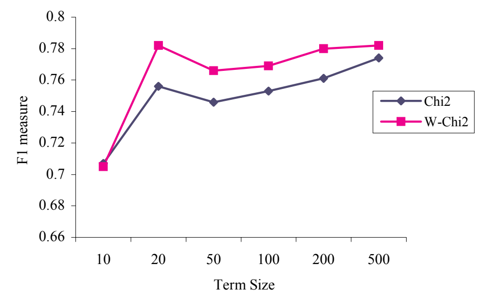

Fig 3. Comparison of micro-averaged F_1 measure between Chi2 and W-Chi2 for the Reuters-21578 top ten categories

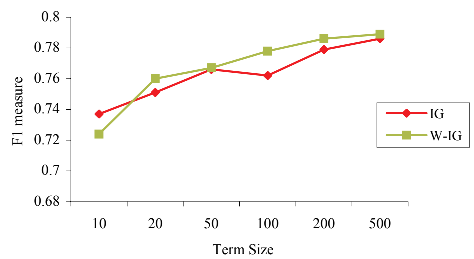

Fig. 4. Comparison of micro-averaged F_1 measure between IG and W-IG for the Reuters-21578 top ten categories

By using the WordNet-based approach for feature selection, a set of terms that is more meaningful and more reflective of a category's content can be obtained. We illustrate this by using two examples. The first example is the top 20 terms chosen by Chi2 and W-Chi2 for the category "acquisition" (acq), which is listed in Table III.

TABLE III
THE TOP 20 TERMS CHOSEN BY CHI2 AND W-CHI2 FOR THE CATEGORY "ACQ"

Feature selection approach	Top 20 terms for category "acq"
Chi2	Shares, Offer, Lt, Stake, Merger, Cts, Acquisition, Company, Inc, Acquire, Net, Loss, Corp, Usair, Common, Mln, Unit, Shr, Stock, Sell
W-Chi2	Shares, Offer, Stake, Merger, Cts, Acquisition, Company, Net, Loss, Corp, Common, Unit, Stock, Sell, Buy, Takeover, Shareholders, Trade, Transaction, Bid

When the list of terms chosen by Chi2 and W-Chi2 is compared to each other, it is noted that there are six terms that differ. These six terms differentiate the results between the two approaches. The six terms for each approach are listed in Table IV.

TABLE IV
THE SIX TERMS THAT DIFFERENTIATE BETWEEN CHI2 AND W-CHI2

Chi2	W-Chi2
Lt	Buy
Inc	Takeover
Acquire	Shareholders
Usair	Trade
Mln	Transaction
Shr	Bid

If human experts were asked to choose a set of terms to represent the category "acq", it is very likely that they would choose the terms listed under W-Chi2 in Table IV. All the terms under W-Chi2 clearly reflect the concept of acquisition. Under Chi2, there is only one term that strongly represents the concept of acquisition, which is "acquire". With W-Chi2, the term "acquire" is not chosen simply because the approach only consider nouns and not verbs.

The second example is the top 20 terms chosen by IG and W-IG for the category "wheat", which is listed in Table V.

TABLE V
THE TOP 20 TERMS CHOSEN BY IG AND W-IG FOR THE CATEGORY "WHEAT"

Feature selection approach	Top 20 terms for category "wheat"
IG	Wheat, Tonnes, Vs, Lt, Agriculture, Net, Export, Loss, Soviet, Grain, Crop, Winter, Usda, Department, Company, Bank, Barley, Lyng, Eep, Program
W-IG	Wheat, Tonnes, Agriculture, Net, Export, Loss, Grain, Crop, Department, Company, Program, Farm, Profit, Subsidy, Share, Farmers, Shares, Tonne, Commodity, Corn

From the comparison of the list of terms chosen by IG and W-IG, it is noted that there are nine terms that differ. These are the nine terms that differentiate the results between the two approaches. The nine terms for each approach are listed in Table VI.

TABLE VI
THE NINE TERMS THAT DIFFERENTIATE BETWEEN IG AND W-IG

IG	W-IG
Vs	Farm
Lt	Profit
Soviet	Subsidy
Winter	Share
Usda	Farmers
Bank	Shares
Barley	Tonne
Lyng	Commodity
Eep	Corn

Again, it is seen in Table VI that W-IG has terms that closely represent the category "wheat" as compared to the terms chosen by IG. Both Chi2 and IG will include terms that are statistically significant regardless of whether they are reflective of the category or not. For example, in Table IV and VI, we can see that both Chi2 and IG choose the word "Lt". As a human being would think, this word bears no connection to both categories "acq" and "wheat". It is not meaningful to both the categories. Therefore, with the use of the WordNet-based approach for feature selection, it is seen that a set of terms that is more meaningful and more reflective of a category's content can be obtained.

X. Conclusion

In this research, it has been shown that there is another approach for feature selection other than the statistical approach. The semantics feature selection is seen as a promising approach for feature selection, as it is able to select features that are more meaningful and more reflective of a category's content. It is also observed that when this approach is used with the statistical weighting, it can perform better than the statistical approach itself with improvements in categorization accuracy. Apart from that, this approach is also effective in reducing the dimensionality of the feature space. To summarize, this research has demonstrated the ability to extract meaningful terms from statistical features. It could thus be applied as a means to filter terms and potentially lead towards better text understanding. In conclusion, the incorporation of the semantics component using WordNet is capable of improving the effectiveness of tasks that involves the natural language.

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Using HPC and PGAs to Optimize Noisy Computational Models of Cognition

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***Abstract*-Cognitive modeling on high performance computing platforms is an emerging field. A preliminary analysis is presented on the use of parallel processing and genetic algorithms for optimizing the fit of non-linear, multivariable symbolic models of human cognition to experimental data. The effectiveness of this experimental optimization methodology is illustrated with a prototype model of a serial arithmetic task built in the ACT-R cognitive architecture. The results confirm that HPC-based optimization techniques could replace the manual optimization techniques used by cognitive modelers up until the present.**

I. INTRODUCTION

The number of available parameters for manipulating a cognitive model while running under the constraints of a cognitive architecture often outnumbers the experimental data points especially as the complexity of the task being modeled increases. For example, modifying model parameters to represent the effect of a threatening task appraisal in serial mental arithmetic performance, or the effects of 400 mg of caffeine on working memory capacity. The size of the search space grows combinatorially with the number of parameters used in the cognitive model.

Our research considers the role of genetic algorithms (GAs) in overcoming the combinative search spaces associated with cognitive models. GAs are a type of random search algorithm inspired by genetics and natural selection. They allow exploration of the space of potential cognitive theories, without preconceived notions of what the best parameters may be. GAs have disadvantages of being demanding in terms of computational load and memory. However, because the GA is an inherently parallel algorithm, parallel implementations of GAs (parallel genetic algorithms or PGAs) can provide considerable gains in terms of performance and scalability when studied and used on parallel machines.

The high-performance computing (HPC) platform utilized for the PGA component of the project is a Xeon Linux cluster with 1,450 dual-processor Dell PowerEdge 1750 servers located at the National Center for Supercomputing Applications.

The PGA runs the ACT-R cognitive architecture and the cognitive model. ACT-R is a theory for simulating and understanding human cognition. Researchers working with ACT-R are interested in understanding how individuals organize knowledge and produce intelligent behavior [1]. ACT-R provides a rigorous framework for cognitive modeling as well as an extensive set of parameters (over 100) and constraints on cognition to facilitate a priori predictions about a behavior of interest and psychologically plausible models in general.

In this case the behavior of interest is a serial mental subtraction task. Serial subtraction, repeatedly subtracting a 1- or 2- digit number from a 4-digit number is part of the Trier Social Stress Test used extensively to examine the physiological effects of stress in a laboratory setting [2]. Human performance data for the serial subtraction task was collected as part of a series of experiments investigating the effects of stress and caffeine on cognitive performance.

The human performance data is used to validate the cognitive model. A close correlation between the model's behavior and the human performance data is the goal. This fitting process is a key component in the Cognitive Science field, and in the end, determines success or failure of the research project.

Integrating HPC platforms, parallel processing, and optimization algorithms such as GAs into the modeling process points the way towards a more efficient and accurate model-to-data fitting process for the computational modeling community.

The paper briefly describes parallel implementations of GAs and the type of PGA used to optimize the prototype model. Cognitive models built in the ACT-R architecture are discussed next, as well as, the cognitive task being modeled, and the experimental data set used in the fitting process. Section IV compares the manual optimization used to date in the field to the parallel optimization methodology. The paper concludes with a discussion of the results from two PGAs.

II. PARALLEL GENETIC ALGORITHMS

Based on principles of natural selection and genetics, genetic algorithms (GAs) have been applied successfully to numerous problems in business, engineering, and science [3]. GAs are randomized, parallel search algorithms that search from a population of points [4]. The points (often referred to

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as genotypes) represent individuals in a population. The genotypes are evaluated for fitness, then propagated to later generations by means of probabilistic selection, crossover, and mutation. In the problem context of the project the genotypes are sets of ACT-R parameters applied to the cognitive task model. The population evolves to find better 'solutions' by selecting the most fit parameter sets (those that give the best match to the human data), and propagating these solutions to the next generation.

The stochastic search properties of genetic algorithms provide an efficient tool for solving problems with large, poorly understood search spaces, thus, allowing for exploration of the space of potential cognitive theories in which to apply to the problem. The search space can also be seeded (constrained) as knowledge about the context of the problem space becomes known.

In many practical applications, GAs find good solutions in reasonable amounts of time. However, in some cases, GAs can require hundreds of thousands of expensive function evaluations, and depending on the cost of each evaluation, the GA may take days, months, or even years to find an acceptable solution [5]. In this project, the function evaluation consists of running the model in the cognitive architecture, analyzing the model's performance output, and calculating a fitness value for the model's predictions. When considered over a generation of 200 genotypes, for example, the computational resources required on a single processor would be significant.

The parallel nature of genetic algorithms has been recognized for a long time, and many researchers have successfully used parallel GAs to reduce the time required to reach acceptable solutions to complex problems. GAs, working with a population of independent solutions, can easily distribute the computational load among multiple processors.

There are several classes of PGAs distinguished by their level of parallelization. This project utilizes a master-slave global parallelization PGA. This type of PGA is characterized by a high computation to communication ratio. In a master-slave PGA, one master-processing node (with rank 0) executes the GA-related functions (selection, crossover, mutation), while the fitness function evaluation is distributed among several slave processors. The slaves evaluate the fitness of the genotypes in the population that they receive from the master, and then return the fitness results back to the master node. Figure 1 is pseudo code for optimizing a cognitive model using a master-slave PGA with a message-passing interface (MPI).

In the project, the slaves each receive a different set of cognitive architecture parameters from the master, run the cognitive model in the architecture, collect the model output, and calculate the associated statistics and fitness value from the model's performance. Each slave then sends its fitness value from the model run back to the master.

```
MPI_Init . . .
if (rank is 0)  // master
     Initialize population
. . . . .
for (each generation)
{
     if (rank is 0)  // master
     {
          Selection
          Crossover
          Mutation
     }

     // find fitness of genotypes in population
     // master and slaves
     MPI_Scatter individuals out to processors
     Run cognitive model
     Calculate fitness of model predictions
     MPI_Gather up resulting fitness values

     if (rank is 0)  // master
          Print out generational statistics
}
Test best solutions found  // master and slaves
MPI_Finalize . . .
```

Fig. 1. Pseudo code for master-slave GA using MPI

III. Cognitive Models

A symbolic approach to cognitive science holds that cognition can be explained using operations on symbols, by means of explicit computational theories and models of mental (but not brain) processes analogous to the working of a computer.

A cognitive model, in the form of a working computer program, is intended to be an explanation of how some aspect of cognition is accomplished by a set of primitive computational processes. A cognitive model performs a specific cognitive task or class of tasks and produces behavior that constitutes a set of predictions that can be compared to data from human performance. *A cognitive model produces both a theory of human behavior on a task and a computational artifact that performs the task.*

To represent the intended level of abstraction, many programming languages designed for cognitive modeling are production systems. Production systems are used as a flexible model of the control structure of human cognition. The flow of processing is controlled by a set of production rules (condition-action pairs) that can be selected to fire when their conditions are satisfied. Therefore, the flow of control is at run time, and is a function of dynamically evolving memory contents triggering the productions. A cognitive model written in a production system makes theoretical commitments at the level of the production rules, and when built within a cognitive architecture, defines a computationally complete system. In this cognitive architecture approach to modeling, the model is a byproduct of three components: cognitive constraints offered by the architecture; background knowledge residing in memory; and the task to be performed.

A. ACT-R

Many instances of cognitive architectures exist, for example: ACT-R [6], Soar [7], and Epic [8]. ACT-R is the product of a community of researchers led by John Anderson at Carnegie Mellon University. ACT-R is a two-layer modular cognitive architecture on a production system framework. One layer contains symbolic representations and has a serial flow in that only one production can fire at a time. The second layer is a sub-symbolic layer whose representations are numeric quantities that are the result of computations preformed as if they were executed in parallel. In ACT-R cognition emerges through the interaction of a number of independent modules. Each of these modules is associated with specific brain regions and theories about the internal processes of these modules [9].

The modularity of ACT-R permits the parallel execution of the verbal system with the control and memory systems (specifically involved in the serial subtraction task). ACT-R has been used in models of working memory tasks and arithmetic processing tasks by other researchers.

B. Serial Subtraction Task

The cognitive model for this project simulates a human subject performing the serial subtraction task. Serial subtraction is the mental arithmetic stressor portion of the Trier Social Stressor Test (TSST). The TSST has been used to provide an acute physiological stress response in human subjects since the 1960's. The serial subtraction task consists of four 4-minute blocks of mentally subtracting by 7's and 13's from 4-digit starting numbers.

C. Experimental Data

The cognitive model of the serial subtraction task is validated with human subject data collected from a larger project to study the effects of stress, task appraisal, and caffeine on biomarkers of cardiovascular health [10].

In the serial subtraction task, subjects' answers were scored against a list of correct answers from the starting number. Task performance was voice recorded on a digital camera and laptop computer. For each subject the number of subtraction problem attempts were recorded and a percent correct score was calculated by dividing the total number of correct attempts by the total number of attempts for each block of the subtractions. The audio recordings were transcribed to obtain subtraction pace and details about error types.

Table I shows the subtraction rates for the subjects' performance on two 4-minute blocks of subtracting by 7s. There is a wide range of performance on this task suggesting a high degree of individual differences within the subject pool.

TABLE I
HUMAN SUBJECT (N=15) MEAN PERFORMANCE AND STANDARD DEVIATION FOR SERIAL SUBTRACTION ON 4-MINUTE BLOCKS OF SUBTRACTING BY 7S

	7s – 1st block	7s – 2nd block
Number of Attempts	47.3 (15.2)	47.8 (19.2)
Percent Correct	82.0 (10.0)	88.8 (7.0)

Subtraction performance was also analyzed by task appraisal. During the experiment, pre- and post-task appraisals were assessed immediately before and at the end of the serial subtraction stressor session. Based on the appraisal responses, subjects were categorized into one of two appraisal groups: challenge or threat. A challenge condition equates to a subject's perceived stress being less than or equal to their perceived ability to cope with the task. In a threat condition, the subject's perceived stress is greater than their perceived ability to cope with the task. Table II shows the subtraction rates for the subjects grouped by post-task appraisal condition. For the project the cognitive model was fit to the mean of each appraisal group.

TABLE II
MEAN PERFORMANCE AND STANDARD DEVIATION BY POST-TASK APPRAISAL GROUP

7s – 1st block	Threat (N=8)	Challenge (N=7)
Number of Attempts	40.3 (10.1)	55.3 (16.7)
Percent Correct	78.1 (8.2)	85.4 (10.8)
7s – 2nd block		
Number of Attempts	44.8 (10.2)	70.7 (23.7)
Percent Correct	84.2 (4.6)	92.5 (6.2)

IV. OPTIMIZATION PROCESS

A. Manual Optimization

Traditionally, cognitive modeling researchers use a manual optimization process to fit the model to the human data. This time consuming iterative process involves selecting a set of parameters, assigning a numeric value to each parameter, running the model in the cognitive architecture, and evaluating the resulting output against the human data. If the fit is unsatisfactory, the process is repeated.

The optimization process can be complicated by the stochasticity built into the cognitive architecture. With a static set of parameters and values, the combination of the model and architecture yield a distribution of performance scores, not a single value. When models include stochastic effects, the model may require 10, 20, or 100s of runs in order to compute stable predictions. Table III compares five example sets of ACT-R parameters used in the serial subtraction model. For each parameter set, the model was run 10 times and then 100 times. The number of attempts and percent correct are averaged over the number of model runs.

TABLE III

MEAN PERFORMANCE AND STANDARD DEVIATION ON SERIAL SUBTRACTION BY 7S FOR ONE 4-MINUTE BLOCK BY POST-TASK APPRAISAL GROUP

ACT-R Parameters			Mean Across 10 Model Runs		Mean Across 100 Model Runs	
ANS	BLC	SYL	Number of Attempts	Percent Correct	Number of Attempts	Percent Correct
0.463	1.693	0.526	42.30	82.56	42.88	85.00
0.399	1.839	0.555	40.30	78.05	40.03	89.74
0.251	1.588	0.529	42.50	80.49	41.57	82.93
0.766	2.619	0.531	40.90	90.00	41.11	78.57
0.654	2.078	0.588	39.60	78.95	38.77	89.47

When comparing model performance between 10 and 100 runs, percent correct shows more variance than number of attempts. This makes for difficult optimization especially if both performance statistics are used simultaneously in the fitting process. Previous attempts at fitting the serial subtraction model to data from other human subject experiments using manual optimization techniques have been unsuccessful [11].

B. *Parallel Optimization*

The ACT-R architecture and cognitive model are written in the Lisp. Generally, message-passing interfaces available on cluster computing resources are called from C or Fortran programs. To utilize parallel processing in the cognitive model optimization process, ACT-R and the cognitive model are packaged into an executable Lisp image or core file. This image file can be run by a system call from a C program on each processor in parallel while utilizing MPI to communicate genotypes and fitness values among the processors.

The population of genotypes (ACT-R parameter sets), in the form of a matrix, are 'scattered' row-wise to the processors. Each processor executes the Lisp image file that runs the model within the ACT-R architecture. Each processor then calculates a fitness based on the model's performance predictions and the human data statistics. In this case, sum of the squared error is calculated on both number of attempts and the percent correct from a block of subtracting by 7s. The fitness values calculated by the processors are 'gathered' up by the master process, which then applies genetic functions to the population based on the fitness of the genotypes (refer to Figure 1). This is repeated through any number of generations with the effect of evolving a set of candidate solutions.

C. *Serial Subtraction Optimization*

Two PGAs were set up to run 50 generations of 200 binary-encoded genotypes. One PGA optimized the model to the challenge appraisal group means (55.3 attempts, 85.4% correct), and the other to the threat appraisal group means (40.3 attempts, 78.1% correct).

A genotype consisting of one 36-bit chromosome is divided into three 12-bit substrings each representing the value of an ACT-R parameter. We investigated: activation noise representing variance in applying procedural knowledge (ANS), the base level constant affecting declarative memory retrieval (BLC), and syllable rate, seconds per syllable (SYL)—because the model verbalizes the answers as the human subjects do. One processor was allocated for each genotype.

The selection probability (selection of the fittest) was set to 0.5 meaning half the population is replaced each generation by offspring of the fittest genotypes. Random mutations alter a certain percentage of the bits in the list of chromosomes. This operation introduces traits in the original population and keeps the GA from converging too quickly before sampling the entire search space. The mutation rate was set at 0.15. The terminating condition was a specified number of generations (50), instead of proximity to the appraisal data means. The fitness function compared the sum of the squared error for the model's predicted number of attempts and percent correct to the corresponding human data.

V. RESULTS

Typically, GAs generate new points in the search space by applying operators to current points and statistically moving toward more optimal positions in that search space. In this optimization problem, the fitness is in terms of error (or cost) and is the discrepancy between the model's predictions and the actual human performance on the cognitive task. The PGA in this case is seeking a global minima in the ACT-R parameter space.

Figures 2 and 3 plot the progress of the PGA as it seeks a global minima across the 50 generations. Figure 2 shows the minimum and average fitness when optimizing to the challenge appraisal group means. Figure 3 is optimizing to the threat appraisal group means.

Normally, what would be expected for this type of plot is a smooth, maybe slightly bumpy, curvilinear downward sloping line as the GA converges on a solution. Figure 2, and especially Figure 3, show the PGA 'bouncing' around the search space; finding a fit solution in one generation, and then tossing it out in the next. Additionally, it appears that this pattern would continue for infinitely many generations.

Because of the previously discussed stochastic effects embedded in the model and architecture these results are not that surprising. Several modifications were built into the PGA to compensate for a single genotype returning a distribution of performance predictions instead of an exact value.

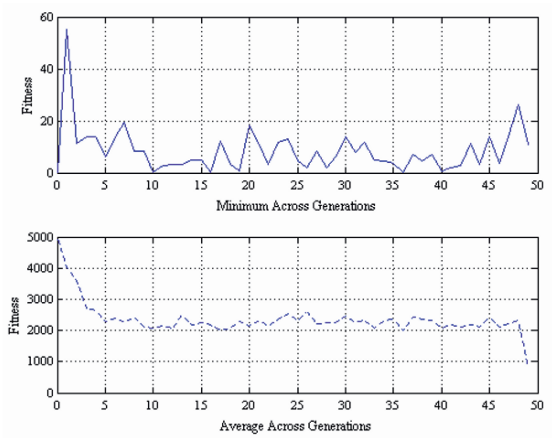

Fig. 2. PGA optimizing to challenge appraisal group

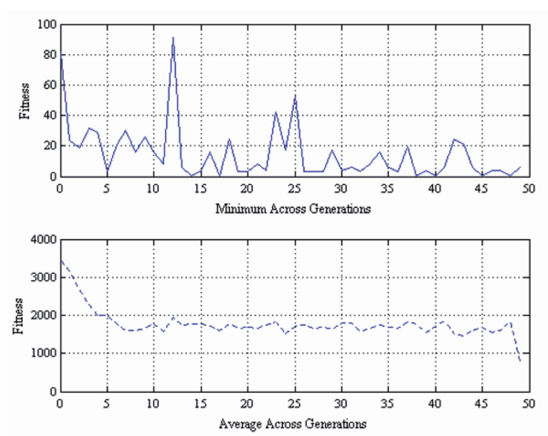

Fig. 3. PGA optimizing to threat appraisal group

During its generational journey, if the PGA finds a ‘good enough’ solution, as determined by a boundary fitness value, that particular genotype is remembered for a post-PGA testing phase. In essence, the PGA is gathering up good solutions across the generations, instead of converging on a so-called best set of solutions. Once the PGA terminates, each of the collected genotypes is run on all the processors with the fitness calculated from the mean number of attempts and percent correct across all runs (200 runs per genotype).

While optimizing to the challenge appraisal group the PGA collected 17 genotypes for additional testing. During threat appraisal group optimization, 9 genotypes were collected. After the final generation of the PGA, these genotypes were run in parallel on all the processors using a master-slave/MPI approach. Table IV lists the four best fitting genotypes (by post-PGA test) from each appraisal optimization collected by the PGA. The second column shows the genotypes’ original fitness value as reported by the PGA compared to their fitness values from post-PGA testing in the third column.

TABLE IV
GENOTYPE FITNESS COMPARISON BETWEEN PGA AND POST-PGA TESTING

Genotypes	Fitness	
Challenge Optimization	PGA Reported	Post –PGA Testing
0.500, 2.083, 0.365	0.093	0.133
0.271, 1.558, 0.360	0.093	1.244
0.561, 2.279, 0.366	0.093	1.696
0.500, 2.083, 0.365	3.597	3.610
Threat Optimization	PGA Reported	Post-PGA Testing
0.727, 2.538, 0.535	6.191	0.986
0.729, 2.524, 0.593	6.008	7.072
0.693, 2.446, 0.586	3.075	7.896
0.713, 2.446, 0.593	3.614	8.325

In the threat appraisal optimization, the genotype producing the best fitness value reported in the PGA (3.075) does not correspond to the genotype producing the best fitness value from the post-PGA testing phase (0.986). In the challenge appraisal optimization, there were three genotypes with a fitness of 0.093. One of those genotypes produced the best post-PGA test fitness value (0.133).

As a validation effort, the best fitting set of ACT-R parameters from each appraisal optimization was tested with three additional sets of 200 runs each. Table V shows the number of attempts and percent correct averaged over each of the 200 runs, and a comparison of the model’s mean performance to the subjects’ mean performance by appraisal group.

TABLE V
VALIDATION OF BEST FITTING PARAMETER SETS FROM APPRAISAL OPTIMIZATIONS

Challenge Appraisal Performance		
ACT-R Parameters	Number of Attempts	Percent Correct
0.500, 2.083, 0.365	55.0	83.5
	55.1	83.0
	55.0	84.7
Model Performance Means	55.0	83.7
Human Data Means	55.3	85.4

Threat Appraisal Performance		
ACT-R Parameters	Number of Attempts	Percent Correct
0.727, 2.538, 0.535	40.8	76.5
	40.9	78.5
	40.8	77.4
Model Performance Means	40.8	77.5
Human Data Means	40.3	78.1

For number of attempts the fit is nearly perfect; a difference of 0.3 for the challenged subjects, and 0.5 for the threatened subjects—half a subtraction problem or less. The fit is slightly less accurate for percent correct; a difference of 1.7 correct subtractions for challenged subjects, and half a correct subtraction (0.6) for threatened subjects.

In summary, this is a very good fit considering the complexity of model and the wide range of human performance on the serial subtraction task. The total run time on the cluster was minimal; 117 minutes for the two PGAs including the post-PGA testing phase, and 4 minutes for the additional best solution validation runs.

It would be important to consider past cognitive science research and potential theory development in the analysis and interpretation of the most promising of the PGA genotypes returned from the testing phase.

VI. Concluding Remarks

By integrating parallel processing on high-performance computing platforms with stochastic search algorithms, such as PGAs, cognitive models can be optimized to fit human subject data efficiently and more accurately then tradition manual optimization techniques.

The stochasticity built into the architecture requires cognitive models of tasks characterized by wide performance variance to be run 10, 20, or 100s of times to compute stable performance predictions. The serial subtraction task is one such task showing a wide range of human performance. Using manual optimization techniques to fit a wide distribution of performance is difficult.

Using 200 processors and approximately two hours of HPC run time the prototype model of the serial subtraction task produced over 25,000 predictions of human performance enabling the fitting of subject appraisal groups and, in the future, individual subjects. Additionally, the results from this exploratory optimization process introduce questions about the nature of the ACT-R parameter space and validity of the architecture in general. Visualization of the parameter space is needed to determine if rough terrain corresponding to noisy data or non-continuity exists.

Acknowledgments

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Personalized Web Search Using Information Scent

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Abstract

Web Search engines explore the existing structure of the web and try to find documents that match the user search criteria. The major challenge for the search engines these days is to determine and satisfy the information need of the user searching the web efficiently and effectively. The approach presented in this paper is to personalize the web search according to the information need of the user using Information Scent in Query log mining. Query log is preprocessed to find the query sessions. Information need associated with query sessions is modeled using Information scent and content of clicked documents. The work done in this paper uses clustering techniques to cluster query sessions with similar information need modeled using Information Scent and content of clicked URLs. The information need of the current user session is used to identify the cluster that has the information need similar to the information need of the current session of the user. The selected cluster is used to recommend useful link on top of next requested result page. Recommended URLs will help the user to find the relevant documents which are closed to his needs and direct the search in a fruitful direction. This approach personalizes the search process to the need of user. Performance of the proposed approach is evaluated with an experimental study of query sessions mining of the "Google" search engine web history data and the experimental results shows the improvement of the Information Retrieval precision.

Keywords
Information scent, clustering, search engine, personalized search.

1. Introduction

Search engines provide the interface to access to vast pool of information on the World Wide Web. Current search tools retrieve so many documents of which only a small fraction is relevant to a user query [10]. Now a days the web search engines provide the user friendly interface by which user can issue queries that are simply a list of keywords. From a study of the log of a popular search engine, Jansen [6] concludes that most queries are short and imprecise. It is known that most terms have different meaning in different context. Due to ambiguity of query terms and short length of query, keywords of query can not determine the information need associated to the query. As a result many documents are retrieved which are not relevant to the information need associated to the query and retrieval precision is degraded. Research is going on to improve the search results and work has been done in [3] where query sessions of query log are clustered using content of clicked URLs to recommend queries for the improvement of the precision of search results. The query log contains much more information implicitly than it is exploited. In [9] new concept of Information Scent is introduced in the field of information retrieval to improve the precision by improving the rank of those pages in result set which are relevant to the user information need. Information Scent is derived from Information foraging theory according to which user behavior in information environment is guided by information scent which is high if the information seen by the user matches the need of user. Information Scent is the perception of the value and cost of information with respect to the goal of the user [1]. The user when presented with search results tends to click those documents which have high Information Scent. Information scent of those documents is high as they match the information need of the user. In this

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paper web search is personalized according to the need of user as user browse the search results which is not considered in [9]. The proposed approach is based on the assumption that people with the similar information need tend to click similar documents. In the proposed approach web history data of "Google" search engine is used which stores the history of interaction of users for all query topics issued on search engine. The data is preprocessed to find the query sessions. A particular Query session shows the clicked documents for a particular query. The clicked documents in query sessions represent the information need associated with query sessions but all the clicked pages in a particular query session do not contribute equally to the information need of the session. Information Scent is used to give high weightage to those clicked URLs in the query sessions that uniquely identify the information need associated with the query sessions in which they are present. The information need associated with the query sessions is modeled using Information Scent and content of clicked URLs. The information scent is used to access the information need from the traversal path of clicked URLs in the sessions. Query sessions are obtained as weighted vector of clicked web pages in which high scent pages are weighted high in comparison to low scent pages. Query sessions are clustered to get the clusters of query sessions with similar information need. The input query is issued to search engine to get the search results. The user response to search results is tracked as the user browses the search results. Whenever the next result page is requested, the user partial search session is used to find the cluster that best matches the need of the user. The selected cluster is used to recommend useful link on top of next requested result page. Recommended URLs will help the user to find the relevant documents which are closed to his needs and which he could not get from initial query issued .The recommended URLs are those URLs which are clicked by those users whose information need is similar to information need of current user. This way the search process is personalized to the need of user using his current session with search engine. This search process will soon converge to the information need of user.

This paper is organized as follows: section 2 describes the concept of information scent, section 3 explains the Clustering query sessions with similar information need using information scent, section 4 gives the proposed algorithm for Personalized Web Search, section 5 present experimental study and section 6 concludes the paper.

2. Information Scent

On the web, users search for information by navigating from page to page along the web links. Their actions are guided by their information need. Information scent is the subjective sense of value and cost of accessing a page based on perceptual cues with respect to the information need of user. More the page is satisfying the information need of user, more will be the information scent associated to it. Information scent is used to derive the quantitative measure of the sense of value of the clicked page with respect to the information need of the user. The interaction between user needs, user action and content of web can be used to infer information need from a pattern of surfing [1] [7][8]. High Information Scent URLs are those clicked URLs in the query sessions that are close to the information need associated with the query sessions. High Scent pages are uniquely clicked for a given information need. For a given sequence of clicked documents in particular query session more unique is the page to the session relative to the entire set of query sessions of query log, more likely it is close to the information need of the current session and thus more is the information scent associated to it in determining the information need of the session. Another parameter that is taken in accessing the information scent of the clicked pages is the time spent on the clicked pages. The reason for considering the time factor is that the clicked page which consumes more user attention is more likely to satisfy his information need than the page which takes less time of user. Thus both the parameters decide the weightage of the pages in determining the information need associated to query sessions using Information Scent.

2.1 Information Scent Metric

The Inferring User Need by Information Scent (IUNIS) algorithm provides various combinations of parameters to quantify the Information Scent [1] [5].The factors that are taken and adapted according to the proposed approach are page access PF.IPF weight and TIME that are used to quantify the information scent associated with the clicked page in a query session. In page access PF.IPF the PF is the access

frequency of the clicked page in the given query session and the IPF is the ratio of total query sessions in the log to the number of query sessions in which this page is clicked. This helps to reduce the weight of those pages that are accessed in many query sessions and may not be very relevant to the information need associated with the current query session. The second factor that is taken is Time spent on a page in a given query session. By including the time more weightage is given to those pages that consume more user attention. The information scent s_{id} is calculated for each page P_{id} in a given session i as follows.

$$s_{id} = PF.IPF(P_{id}) * Time(P_{id}) \forall d \in 1..n \quad (1)$$

$$PF.IPF(P_{id}) = f_{P_{id}} / \max(f_{P_{id}}) * \log(M/m_{P_{id}}) \quad (2)$$
$$d \in 1..n$$

where PF.IPF(P_{id}) and Time(P_{id}) are defined as follows.

PF.IPF(P_{id}) : PF corresponds to the page P_{id} normalized frequency $f_{P_{id}}$ in a given query session Q_i and IPF correspond to the ratio of total number of query sessions M in the whole log to the number of query sessions $m_{P_{id}}$ that contain the given page $P_{id.}$

Time(P_{id}) : It is the ratio of time spent on the page P_{id} in a given session Q_i to the total duration of session Q_i.

3. Clustering using Information Scent

Clustering is the process of grouping the data into classes or clusters so that objects within a cluster have high similarity in comparison to one another, but are very dissimilar to objects in other clusters. Dissimilarities are accessed based on attribute values describing the objects. In this paper weighted content vector of the pages P_d in the query sessions is used.

P_d = Content$_d$ for each document d.

Content$_d$: The content vector of a page P_d is a keyword vector ($w_{1,d}, w_{2,d}, w_{3,d}, \ldots, w_{v,d}$) where v is the number of terms in the vocabulary set V describing the content of the page $P_{d.}$

Vector Model [2] is used for representing content feature of each page P_d in all query sessions. Each page P_d is represented by vector ($w_{1,d}, w_{2,d}, w_{3,d}, \ldots, w_{v,d}$) where v is the number of terms in the vocabulary set V.

TF.IDF (term frequency * inverse document frequency) term weighing scheme is used to represent the content vector for a given page P_d [2]. The importance of each item of V in a given page P_d is calculated using TF.IDF item weight. Vocabulary V is a set of distinct terms found in all distinct clicked pages in whole dataset relevant to a content feature. The TF.IDF term weight is calculated as number of times a term appears in the given page weighted by the ratio of the number of all pages to the number of the pages that contain the given item.

The information scent associated with the given clicked page is calculated by using two factors i.e. PF.IPF page access and TIME. Each query session is constructed as linear combination of vector of each page P_{id} scaled by the weight s_{id} which is the information scent associated with the page P_{id} in session i. That is

$$Q_i = \sum_{d=1}^{n} s_{id} P_{id} \quad (3)$$

In above formula n is the number of distinct clicked pages in the data set and s_{id} (information scent) is calculated for each page P_{id} in a given session i using (1) and (2).

Each query session Q_i is obtained as weighted vector using formula (3). This vector is used to model the information need associated with the query session $Q_{i.}$

The similarity of query sessions Q_i and Q_j in k-means clustering is calculated using cosine measure.

3.1 Clustering Queries

Query sessions are clustered using k-means algorithm because of its good performance for document clustering [4][11]. Query sessions in our approach are similar to the vectors of web pages. Thus clustering queries in our approach are similar to those for clustering pages.

A score or criterion function measures the quality of resulting clusters. This is used by common vector space implementation of k-means algorithm [12]. The function measures the average similarity between vectors and the centroid of clusters that are assigned to. Let C_p be a cluster found in a k-way clustering process ($p \in 1..k$) and let c_p be the centroid of p^{th} cluster. The criterion function I is defined as follows:

$$I = 1/M \sum_{p=1}^{k} \sum_{v_i \in C_p} sim(v_i, c_p) \quad (4)$$

where M is the total number of query sessions in all clusters and v_i is the vector representing some query session belonging to the cluster C_p and centroid c_p of the cluster C_p is defined as given below.

$$c_p = \left(\sum_{v_i \in C_p} v_i\right) / |C_p| \qquad (5)$$

where $|C_p|$ denotes the number of query sessions in cluster C_p . sim (v_i, c_p) is calculated using cosine measure.

4. Algorithm for Personalized Web Search

The algorithm is based on clustering process that defines neighborhood of query sessions driven by similar information need using information scent and content vector of clicked URLs in the query sessions. Each query session consist of a query along with the clicked URLs in its answer.

querysession=(query,(clicked URLs)$^+$)

where clicked URLs are those URLs which user clicked before submitting another query.

4.1 Algorithm

1. Offline Preprocessing phase at regular and periodical intervals
   - 1.1. Extract the queries and associated clicked URLs from the data set.
   - 1.2. Preprocess the Extracted Queries to find the query sessions.
   - 1.3. Model the Information need associated to each query session using information scent and content vector of pages in the session using (1)(2)(3).
   - 1.4. Cluster the query sessions using information need associated to each query session using cosine measure.
   - 1.5. For each cluster C_j maintain a list of high scent URLs P_j which are present in the cluster C_j .
2. Online searching:
   - 2.1. Model the information need associated with the partial session of input query q .

     $$q = \sum_{d=1}^{n} s'_d P_d$$

     $$s'_d = \mathrm{Time}(P_d)$$

     Time(P_d): It is the ratio of time spent on the page P_d in session q to the total duration of the session q.
   - 2.2. Find the C_j cluster which is most similar to the vector representation of input query as per the threshold value set for similarity measure.
   - 2.3 Recommend the URLs in set P_j associated with selected cluster C_j on the top of next requested result page for the input query q.

where the set P_j is defined as follows.

P_j = {URL_j: URL_j are ordered in descending order of the value of Information Scent s_j upto certain threshold value for Information Scent x}

5. Experimental Study

Experiments were performed on the data set containing the clicked documents associated with queries issued on the Google search engine. The data set was collected from the web history of Google search engine. The data set was generated by the users expertise in specific domain who were asked to issue the queries in some selected domain. The experiments were performed on the data set containing queries mainly in three domains that are entertainment, academics and sports. The web history data contains the following fields.

1. Time of the Day
2. Query terms
3. Clicked URLs

The data set generated from web history contains several thousand entries out of which we had extracted 5325 entries. The data set generated from the web history was loaded into the database to be processed further.

On the submission of input query, Google search engine returns result page consisting of URLs with information about URLs. In the experiment, only those queries in data set were selected which had at least one click in their answer. Query sessions considered consists of query terms along with clicked URLs. The clicked URLs were those URLs which user clicked before he submits another query. The distinct URLs in data set were found to be 4489.The data set was preprocessed to get 1450 query sessions. The query sessions were clustered using k-means algorithm. The k-means algorithm was executed several times for different values of k and criterion function was computed for each value of k. The criterion function was found to have maximum value at k=109. The threshold value set for similarity of two vectors was 0.5. The threshold value for information scent x for each set P_j associated with each cluster was 0.25 .The experiments were performed on randomly selected queries which were categorized into trained queries set and untrained queries set. The trained queries were those queries which had sessions associated with them in data set and untrained queries were those queries which did not have sessions associated with them in data set.

The experiment was performed on Pentium IV PC with 512 MB RAM under Windows XP using Java and Oracle database. WebSphinx Crawler was used to fetch the clicked documents of query sessions in the data set. Each query session was transformed into the vector representation using Information Scent and content of clicked URLs. Query sessions were clustered and each cluster of query sessions was represented by mean value of vector of terms.

The relevance of URLs in the selected cluster for a given input query was decided by some anonymous user having knowledge in domain to which input query belongs. The relevance was judged by analyzing the answer of the result set along with the recommended URLs and determined the URLs in answers which were relevant to the input query.

The experiment was performed on randomly selected 25 trained queries and 31 untrained queries. The average precision of queries of trained and untrained query set was calculated.

The Fig 1 and Fig 2 compare the average precision of google search results using without proposed approach and proposed approach presented in this paper. Average precision had been improved using proposed approach. It shows that recommended URLs brings more relevant documents early in search results. Recommended URLs help the user to satisfy his information need based on the fact that users with the information need similar to the current user had clicked documents in their sessions which were relevant to the information need of current user.

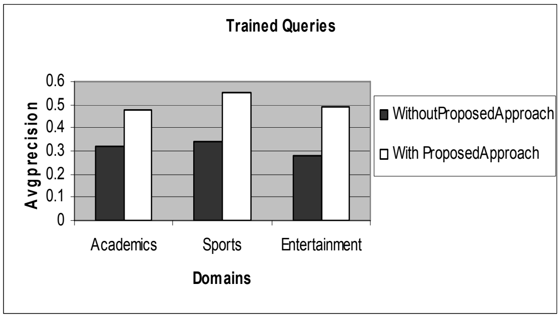

Fig 1 The average precision of without and with proposed approach on trained queries

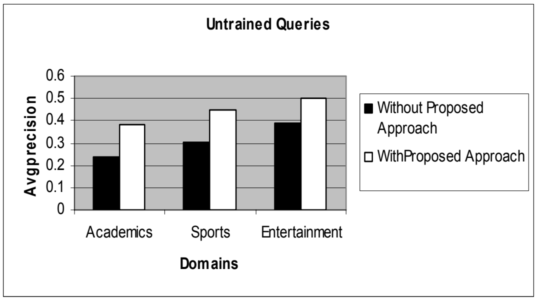

Fig 2 The average precision of without and with proposed approach on Untrained Queries

6. Conclusion

In this paper efforts have been made to satisfy the information need of the user and improve the information retrieval precision by recommending URLs associated with the queries which approximate the information need associated to the input query. The information need associated to the query is modeled using information scent and content feature of clicked pages in the session. The suggested URLs help the user to find the documents relevant to his information need which he could not get through his initial query as keywords of query are few and ambiguous. The personalized web search approach proposed in this paper personalizes the web search for a particular information need of the user. Experimental results show the improvement of the precision of information retrieval.

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Stabilization of a Numerical Model Through the Boundary Conditions for the Real-Time Simulation of Fuel Cells

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***Abstract*-The fuel cells are to convert chemical energy directly to electricity with high efficiency. From an environmental point of view, it is worth emphasizing that the fuel cells are not emitting any greenhouse gases or other pollutants. To increase the efficiency of the fuel cells their properties must be analyzed deeply. To this aim numerical modeling is a very useful tool. The fuel cell system models are usually solved by applying constant currents with long time step. For our purposes high amplitude sine perturbation and real-time simulation are needed, because during the control of a fuel cell the information needs to be available in real-time. In this article our method for approaching the real time simulation is formulated including the so-called scaling method. Using this method, the real-time simulation of fuel cells is getting easier and more accurate.**

I. Introduction

Fuel Cells (FC) are electrochemical devices that convert the chemical energy directly into electricity and are widely regarded as potential alternative stationary and mobile power sources. They are not emitting any greenhouse gases and the theoretical efficiency is not constrained by any theoretical limit like the Carnot limit. While the thermal efficiency of the spark-ignition (gasoline) engine is 15-25 %, efficiency of a H_2/Air fuel cell is 50-55% [1]. In practice the application of an electric motor diminishes less power then the mechanical drives, which improves the tank-to-wheel power. Application of fuel cells in automotive powertrains is emphasized in this publication, partly because ground vehicle propulsion conditions present the most challenging control problem, and partly due to their importance in global fuel consumption and emission generation.

At a glance, fuel cells seem to be simpler devices as internal combustions engines (ICE), i.e., they have no moving parts, but the non-linear and coupled phenomena among the different subsystems make it hard to understand and control them in real time. There are three different approaches to build an adequate model of an FC system and develop an efficient control strategy [2]:

1. Detailed FC models based on partial differential equations.

2. Steady-state FC system models based on experimental maps and look-up tables.

3. Dynamic FC system models that neglect spatial variations.

However these are not real-time models and the transient responses of FCs cannot be handled by using these approaches. Hence a control-oriented real time model is needed. Such a model can be used for real-time parameter estimation [3] for feedback control, or for more effective and energy-saving drive system.

In this paper different non-linear numerical models of FCs are investigated, determining the speed, stability, accuracy and the conservation of basic qualitative properties. A new approach, the parameter scaling method, is introduced to simulate FCs in real time.

II. Model Description

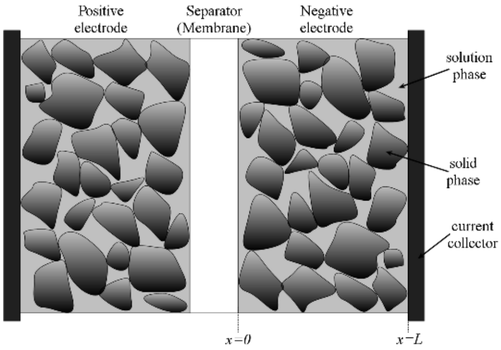

Fig. 1. Schematic figure of a fuel cell

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A. *The electrochemical model*

In Fig. 1, a schematic structure of a proton exchange membrane fuel cell (PEMFC) is shown. Fuel cells 'burn' hydrogen fuel (H_2) and oxygen (O_2) to water, but instead of heat electrical energy is produced. This can happen if the overall chemical reaction is separated by a membrane to the oxidation and to the reduction. The oxidation takes place at the anode (negative electrode) and the reduction takes place at the cathode (positive electrode). Electrons pass through the circuit and generate power while protons pass through the membrane.

$$H_2 + \tfrac{1}{2}O_2 \rightarrow H_2O : Overall$$
$$H_2 \rightarrow 2H^+ + 2e^- : Anode$$
$$\tfrac{1}{2}O_2 + 2e^- + 2H^+ \rightarrow H_2O : Cathode$$

The structure of the half cell is porous so that the electrolyte (conducts protons) from one side and the gas from the other side can penetrate it, while the pores itself conduct electrons to the current collector. The reaction takes place where the three phases are contacting.

Both electrodes have the same structure and the governing equations have the same form. Therefore, according to our purposes it is enough to perform the simulation on the half cell, only. The porous electrodes are usually modeled by macro homogeneous models [4] under the following assumptions:

- Only a one dimensional model is considered.
- The concentration distribution of gases is neglected.
- Double-layer charging as well as faradaic reaction occurs.
- The double-layer capacitance, the material and the kinetic properties are taken to be constants ($C_{dl}, a, \alpha, \sigma_{eff}, \kappa_{eff}, I_0$ [see Table I. [4]]).

B. *The mathematical model*

For the current in the two phases (solid phase [i_1] and solution phase [i_2]) we have [4]:

$$i_1 = -\sigma_{eff}\frac{\partial \varphi_1}{\partial x}; \qquad (1)$$

$$i_2 = -\kappa_{eff}\frac{\partial \varphi_2}{\partial x}. \qquad (2)$$

According to the electron neutrality the equality:

$$-\frac{\partial i_1}{\partial x} = \frac{\partial i_2}{\partial x} \qquad (3)$$

holds.

TABLE I
NOMENCLATURE

Symbol	Description
A	specific interfacial area (cm^{-1})
C_{dl}	double-layer capacitance (F/cm^2)
F	Faraday's constant (*96487 C/mol*)
i_0	exchange current density (A/cm^2)
i_1	solid phase current density (A/cm^2)
i_2	solution phase current density (A/cm^2)
I	total cell current density (A/cm^2)
j_n	interfacial current density (A/cm^2)
L	thickness of the porous electrode (cm)
R	universal gas constant (*8.313 J/mol K*)
t	time (s)
T	temperature (K)
x	distance (cm)
X	dimensionless distance x/L
α	transfer coefficient
η	overpotential (V)
κ_{eff}	effective solution phase conductivity (S/cm)
σ_{eff}	effective solid phase conductivity (S/cm)
ν^2	dimensionless exchange current density
τ	dimensionless time
φ_1	solid phase potential (V)
φ_2	solution phase potential (V)

The electrochemical reaction and the double layer charging take place at the interphase, and the interfacial current density is defined as [4]:

$$j_n = A\cdot C_{dl}\frac{\partial(\varphi_1-\varphi_2)}{\partial t} + A\cdot i_0\exp\left(\alpha\frac{F}{RT}(\varphi_1-\varphi_2)\right) - A\cdot i_0\exp\left(-\alpha\frac{F}{RT}(\varphi_1-\varphi_2)\right). \qquad (4)$$

On the other hand, the relation

$$j_n = -\frac{1}{A}\frac{\partial}{\partial x}\left(\kappa_{eff}\frac{\partial \varphi_2}{\partial x}\right) \qquad (5)$$

also holds. Introducing a new function for the difference of the potentials on the solid and solution phases – the so called overpotential function:

$$\eta := \varphi_1 - \varphi_2, \qquad (6)$$

from (4) and (5) we obtain

$$A\cdot C_{dl}\frac{\partial\eta}{\partial t}=-\frac{\partial}{\partial x}\left(\kappa_{eff}\frac{\partial\varphi_2}{\partial x}\right)-A\cdot i_0\exp\left(\alpha\frac{F}{RT}\eta\right)+A\cdot i_0\exp\left(-\alpha\frac{F}{RT}\eta\right). \qquad (7)$$

Our aim is to transform (7) to a simplified form which contains only the unknown overpotential function η.

C. Initial and boundary conditions

When there is no passage of current, for the sake of simplicity the equilibrium potential of the phases is taken equal to zero, so the initial conditions are

$$\varphi_1(x,0)=\varphi_2(x,0)=0, \qquad (8)$$

i.e.,

$$\eta(x,0)=0,\quad x\in[0,L], \qquad (9)$$

When current flows at the membrane | negative electrode interface only ionic charge transport, while at the current collector | negative electrode interface only electron transfer occurs, therefore the following boundary conditions prevail:

$$i_1(0,t)=I(t);\quad i_2(0,t)=0 \qquad (10)$$

and

$$i_1(L,t)=0;\quad i_2(L,t)=I(t). \qquad (11)$$

Based on (10), (1) and (2), we get

$$-\sigma_{eff}\frac{\partial\varphi_1}{\partial x}(0,t)=I(t),\quad -\kappa_{eff}\frac{\partial\varphi_2}{\partial x}(0,t)=0. \qquad (12)$$

Therefore, for the overpotential function we obtain

$$\frac{\partial\eta}{\partial x}(0,t)\equiv-\frac{1}{\sigma_{eff}}I(t). \qquad (13)$$

Similalry at $x=L$:

$$-\sigma_{eff}\frac{\partial\varphi_1}{\partial x}(L,t)=0,\quad -\kappa_{eff}\frac{\partial\varphi_2}{\partial x}(L,t)=I(t), \qquad (14)$$

which results in the boundary condition

$$\frac{\partial\eta}{\partial x}(L,t)\equiv\frac{1}{\kappa_{eff}}I(t). \qquad (15)$$

D. Transformations

Next we formulate the canonical form of the equation (7) and the boundary and initial conditions, as well. We introduce the new time variable

$$\tau:=\frac{t}{A\cdot C_{dl}\left(\frac{1}{k_{eff}}+\frac{1}{\sigma_{eff}}\right)L^2}=\frac{t}{p} \qquad (16)$$

and the new space variable

$$X\equiv\frac{x}{L}, \qquad (17)$$

respectively.

Easily can be verified that, for the new function

$$u(X,\tau):=\alpha\frac{F}{RT}\eta(X,p\tau) \qquad (18)$$

we get

$$\frac{\partial u}{\partial\tau}(X,\tau)=\frac{\partial^2 u}{\partial X^2}(X,\tau)-\nu^2\exp(u(X,\tau))+\nu^2\exp(-u(X,\tau)), \qquad (19)$$

Here we have used the notations

$$\nu^2\equiv\frac{\alpha F}{RT}A\cdot i_0\cdot L^2\left(\frac{1}{\kappa_{eff}}+\frac{1}{\sigma_{eff}}\right) \qquad (20)$$

and

$$p=A\cdot C_{dl}\left(\frac{1}{k_{eff}}+\frac{1}{\sigma_{eff}}\right)L^2. \qquad (21)$$

Then the initial condition for (19) is

$$u(X,0)=0, \qquad (22)$$

while the boundary conditions are transformed into

$$\frac{\partial u}{\partial X}(0,\tau)=-L\frac{\alpha F}{\sigma_{eff}RT}I(p\tau),\quad \frac{\partial u}{\partial X}(1,\tau)=L\frac{\alpha F}{\kappa_{eff}RT}I(p\tau). \qquad (23)$$

E. *Numerical problem*

The system (19), (23) cannot be solved analytically, so a numerical approach is needed. The finite difference method was used by approximating both in space and time, on a generated mesh (Fig. 2) to solve the original problem.

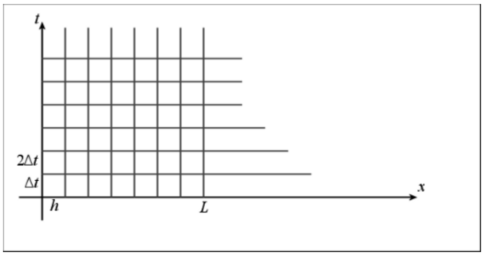

Fig. 2. Approximation in our numerical model

F. *The scaling method*

The simulation of the system (19) (22) (23) takes, very long running time, which does not allow the usage of it in real time. Our purpose is to find a method which transforms the original problem to a more stable parameter space. Two different transformations exist: transform the model parameters (the conductivities, the dual layer capacitance and the exchange current density) or transform the applied perturbation (total current density, *I*).

According to our model, the solution depends on *p,* the dimensionless time parameter (21) and the dimensionless exchange current density ν^2 (20). The simplest scaling method is based on multiplying the model parameters by the same number. This transformation leaves p and ν^2 unchanged, so as the main characteristic properties of the solution. This ttransformation (multiplication in (16) and (20)) results in the decrease of right-hand side in the boundary condition (23), which may stabilize the whole process. However, the solution of the problem (19), (22) and (23) will decrease, due to the change of the boundary conditions. It is the very reason why the obtained solution is multiplied in the rescaling process.

Due to this stabilization, larger time steps can be used in the model. This means that the problem can be solved within much shorter time than required in calculation of the precise solution.

There is another way to scale the parameters, namely decreasing the total current density (*I*). The problem with this method is that, in the computation of the energy consumed, it caused double scaling. This means if we scaled *I* with a specified number *S*, the energy must be rescaled with $\sim S^2$. Because of the double scaling, the model will compute with very small numbers, and this can be a source of data loss.

In Fig. 3, the process of the scaling method is showed. There are two steps which are not included in the normal solution of a numerical problem. The scaling process is (1→2), and the rescaling process is (4→5)

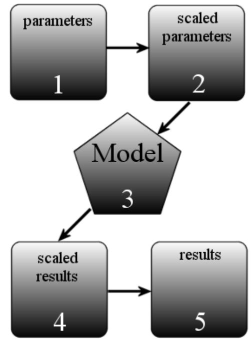

Fig. 3. The scaling method

G. *Characterization*

This system is usually solved by applying constant currents. These methods are suitable to characterize the steady state behavior of the fuel cell, while in practical situations fuel cells serve power according to the fluctuating power demand. For the overall characterization of these transient states, the system was solved using a high amplitude sine perturbation and the generated average energy was calculated and plotted in the frequency domain.

These energy spectra were characterized by the following method:

1. Calculation of the precise solution by running the model with the Crank-Nicolson Scheme (CNS) with a short enough time step and space divison.
2. Comparison of the errors of the different numerical models. This means that if a more stable method is needed, its accuracy must be analyzed.
3. Comparison of the errors of the scaling method introduced by applying the same time parameter p and the same dimensionless exchange current density ν^2.
4. Comparison of the running times of the scaling method. A much shorter period expected than in the precise solution.

III. Results

In this section the Implicit Euler Method (IEM) is compared with the precise solution first, then the model results with different methods are formulated.

A. *The Implicit Euler Method*

The CNS is not stable enough for longer time steps, therefore another numerical method should be applied. The IEM was used because of its better stability property. [5].

First of all, as it was mentioned before, the accuracy of the IEM was analyzed. In Fig. 4, the precise solution (y_{pr}) and the IEM were compared by calculating the relative error as

$$Err = \left| \frac{y_{pr} - IEM}{y_{pr}} \right|. \tag{24}$$

The difference is always less than 0.15%. The CNS, as it is the well-known, gives more a precise solution than the IEM, while IEM is more stable. Henceforth we use the IEM because of its stability, which is more substantial for our purposes.

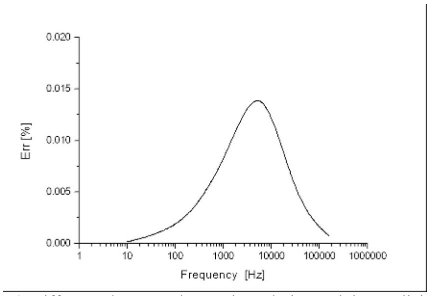

Fig. 4. Difference between the precise solution and the Implicit Euler method in consumed energy

B. Speed-up and stabilization

In the followings the stabilization of the model is formulated.

In Fig. 5, the results with two different time steps (1 µs and 10 µs) were compared using the IEM. The difference is negligible everywhere; however at longer time steps the model became unstable, hence the time required for computation cannot be decreased without loss of stability. The method was not appropriate at longer time steps because it became unstable.

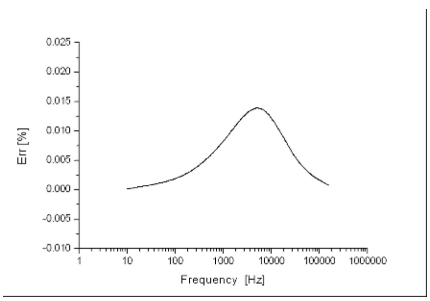

Fig. 5. Difference between the results with two different time steps (1µs and 10µs) in consumed energy

The instability can be conditional. In Fig. 6, a conditionally instable state of the model is shown. The small ticks on the curve are caused by the noise of the numerical method. On the increasing part of the curve the noise appears at around 0.5425 second, however, when the derivates become negative the noise starts to decreasing. The noise can be diminished by decreasing the length of the time step.

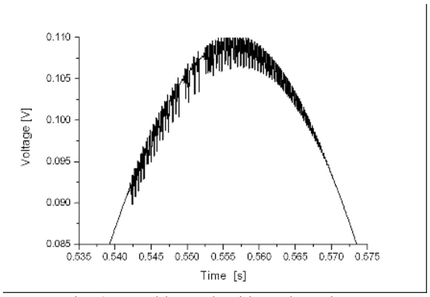

Fig. 6. Unstable result with too long time step

Hereinafter the results of the method presented in the section II./*F*. are introduced to fix the stability problem. The scaling factor has been introduced to elucidate the effect of the scaling method for the numerical solution. This factor is the multiplier of the parameters: the conductivities, the double layer capacitance and the exchange current density. For example if the scaling factor is ten, these parameters will be multiplied by ten before the computation.

In Fig. 7, the precise solution and the 10000 times scaled result were compared. The vertical axis is the consumed energy. For lower frequencies the difference is less than for higher frequencies.

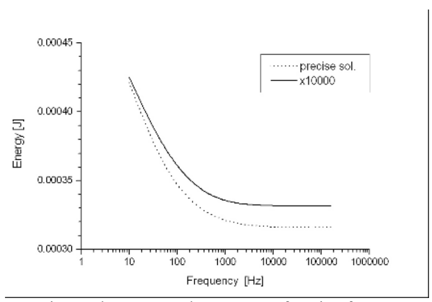

Fig. 7. The consumed energy as a function frequency

In Fig. 8, we compared the precise solution with two scaled solutions (100 times and 10000 times, still without using a larger time step) regarding the consumed energy. The difference is less than 5%, however, 10000 times scaling gave the remarkable opportunity of using a longer time step, which is the definite advantage of the scaling method.

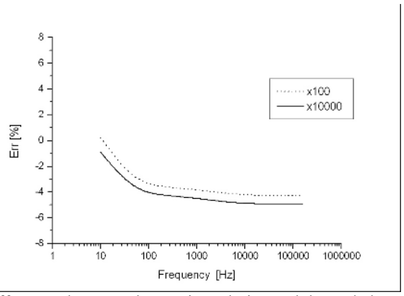

Fig. 8. Differences between the precise solution and the scaled results in the energy

In Fig. 9, three scaled results with three different time steps were compared by using the scaling factor equal to 10000. The figure shows that the scaling enables the usage of a much higher time step. The original computation became instable at 100 μs.

As it can be easily seen in the figure, the difference between the precise solution and these results is less than 5%.

Our next task is to analyze the necessary time for these results. The simulation was started with the longest stable time step and it was reduced when the reciprocal frequency reached the value of the time step (when 0.1/frequency<Δt).

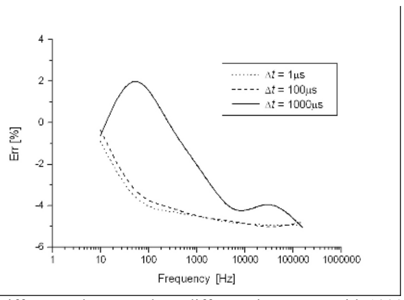

Fig. 9. Differences between three different time steps, with 10000 times scaling

In Fig. 10, the running times of the results presented above were compared. The scaled solutions can be ~10000 times faster than the precise one is, while the difference between the precise solution and the scaled solution was small and finally we are closer to the real time computation.

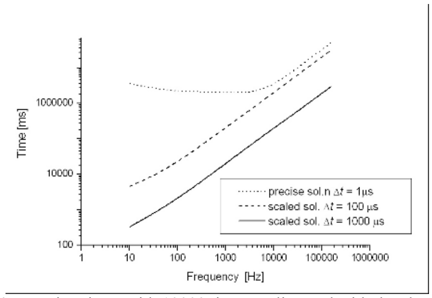

Fig. 10. Running times with 10000 times scaling and with the close solution with different time steps

IV. Conclusion

The fuel cells can be one of the future's main energy conversion devices. They do not emit pollutants or greenhouse gases and they are much more efficient than internal combustion engines. In order to achieve a more effective and energy-saving controlling and driving system for fuel cells, the simulations of the transient states are required. The simulation in this region takes, however very long running time, which does not allow the usage it in real time. Our purpose was to find such a solution which transforms the original problem to a more stable parameter space, which enables the usage of longer time steps, hence much shorter running time. Two different transformations were examined: the scaling of the governing material parameters and the scaling of the applied perturbation. Our results showed that the scaling and rescaling of the material parameters yielded closer solution to the original problem, than the scaling of the perturbation.

With the introduced scaling method we are closer to the real-time computation of a fuel cell system. With this method the calculation is not just fast, but also its accuracy is also acceptable for the practical usage. Our model may be used for real-time parameter estimation or for more effective and energy-saving drive system of a fuel cell.

Acknowledgment

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Model Driven Development of AJAX-Based User Interfaces

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***Abstract*—This paper presents an approach to develop user interfaces using MDSD (model driven software development) and UML. An UML profile is introduced to model user interfaces and to develop a sample AJAX (Asynchronous JavaScript And XML)-based web application. The paper covers common aspects and structures of user interfaces, introduces the main benefits of AJAX and outlines the architecture used to build a web application.**

***Index Terms*—MDSD, UML, AJAX, Widget, User Interface, Web Application.**

I. Introduction

User interfaces are the link between user and software and therefore a main factor for the acceptance of a software product. Hence, user interfaces should be intuitive to use and the experience and suggestions from the users should play a decisive role in development.

Graphical user interfaces consist of standardized components. They are mostly platform independent and can be described in a general manner. One arising problem is that standard components come with their own data model which you must use. Thus, you have to map your application model to the data model from the components you want to use.

Especially in the Web Technology World, the concept of w*idgets* becomes popular with tools like Konfabulator (also known as Yahoo Widget Engine, a JavaScript Runtime Environment for Windows and Mac OS X). Widgets are controls in a graphical or textual user interface. Style and layout of widgets are very impotant in graphical user interfaces. Examples for simple widgets are buttons and lists. A widget shows a specific set of data and handles user events. Therefore, in more general terms a widget displays some application data and provides actions for them.

One problem with using widgets in web applications in the past was the interaction mechanism. A traditional web application handles user actions by sending a request to the web server, which then evaluates the request and sends the answer in a response back to the web browser. So, the user workflow gets interrupted by a page reload. This problem is now solved with AJAX technology. AJAX-based web applications use JavaScript to enhance the interaction between user and web application. AJAX offers the possibility to obtain data from the web server in the background via a so called *XmlHttpRequest*-object. With DHTML (Dynamic HTML: HTML, CSS, JavaScript and Document Object Model DOM), the contents of a web page can be changed in the web browser without any page reload. Thus AJAX as a technology is well suited to enhance the user experience on web applications. AJAX offers the needed functionality to create interactive web pages using widgets - sometimes referred to as *rich clients*.

For building software in an efficient way, the concept of *model driven software development* (MDSD) [1] seams to be very interesting. MDSD is currently receiving a lot of attention. Using domain-specific languages (UML)-models are created that express the application structure or the behavior. The model is subsequently transformed into executable code by tools for automatic code generation.

In this paper a lightweight approach to model user interfaces is presented. It discusses the main aspects and the structure of user interfaces, provides an UML profile to model widgets and takes a closer look at some realization details. The paper ends with a discussion of related work, a summary and future prospects of the presented approach.

II. Architecture

The aspects covered in this paper are applied to a sample web application. The sample is based on the Java platform and uses the following, well known frameworks:

- hibernate framework [2] for the persistence layer,
- spring framework [3] for the application layer,
- DWR (Direct Web Remoting) [4] for the controller layer,
- and the dojo toolkit [5] for the client side presentation layer.

Nowadays, AJAX is popular and is used by many web applications, especially community pages. But one of the most important components of AJAX, namely the *XmlHttpRequest*-object [6], is not standardized yet by the W3C. Furthermore usage of *raw* AJAX is discouraged because of many cross browser inconsistencies. But there are many frameworks and libraries concealing browser inconsistencies and providing wrappers.

Using AJAX increases the logic on the client side. Depending on the amount of client side logic, AJAX-based web applications could be classified as follows:

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- **Server centric:** The logic on the client side routes user events to the server logic, which handles them and generates answers, which can easily be included in the actual web page.
- **Client centric:** The logic on client side handles the user events itself. The server is mainly used as application or data provider. Depending on the user event, the logic obtains data from the web server and changes the web page.

All traditional web applications are server centric. Creation of server centric web applications is usually done by using a web template language, e.g. JSP (Java Server Pages) or JSF (Java Server Faces). AJAX-based web applications can easily be built using AJAX-enhancements in the template language. The needed JavaScript code is then usually generated by the web framework.

The creation of client centric web applications has more influence on the architecture and therefore the client centric approach was used to create a sample web application. The presentation layer providing the guidance of users moves to the client side. An additional controller is needed on the client side for handling events and obtaining data from the web server. Because the fact, the sample web application built on top of the spring framework, the spring architecture model for web applications must be adopted to fit the needs of client centric web applications. This is shown in Figure 1. The presentation layer moves to the client side and an additional controller layer on client side as well as a communication layer connecting server and client side is needed.

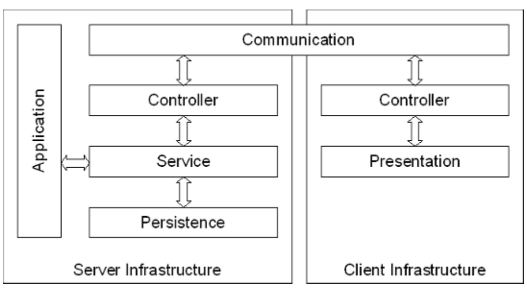

Fig. 1. Adopted Spring Web Application Architecture

The framework *DWR* (Direct Web Remoting) [4] seams to be appropriate for the server side controller and the communication layer. It maps Java objects to JavaScript objects and takes care of marshalling and unmarshalling. The key benefit is to automatically provide the functionality of the service methods on the client side without the need to implement a server side controller. These service methods are modeled according to the *Service Facade Pattern* [8].

On the client side the *dojo toolkit* [5] was used. It simplifies the access to DOM and CSS and provides a widget concept, which was used to realize the sample web application.

III. User Interface Aspects

Graphical user interfaces consist of controls named widgets. In web applications, widgets are responsible for the layout, style and actions and show content at a given area of a web page. Actions could be provided using buttons or links. Because a button is also a widget, widgets must have the possibility to nest them. Regarding the management of child widgets, two kinds of widgets can be determined:

- **Container Widgets** have child widgets, but do not manage them. They only provide an area for their content. An example is a layout widget. The management of the child widgets is done by a controller.
- **Composite Widgets** have child widgets and manage them themselves. E.g. the visibility of child widgets is controlled by the parent widget.

The life-cycle management of widgets includes well known tasks from programming: creation, initialization and destruction. Further tasks can be to obtain data and bind them to the widget and the change of the visibility of the widget.

The content, which is shown by a widget, can be classified into two kinds:

- **Static Content** does not change in the whole lifetime of a widget, e.g. a label shows static content.
- **Dynamic Content** is usually the main content a widget presents, in general application data from a database.

You usually design forms to enter or change application data and reports to present it.

The next aspect of user interfaces addresses the provided actions. Widgets offer different ways to initiate an action:

- Push a button
- Click on a link
- Mouse gestures
- Mouse and keyboard events

The first two possibilities are well known from traditional web applications. The other two are possible with JavaScript and DOM.

IV. Modeling Widgets

For the user interface aspects mentioned in section III, an UML profile was created to model widgets. The developed UML profile is shown in Figure 2. Note that the tagged values named *dojoAttachPoint* are platform specific aspects by now and should be removed or replaced in a general concept.

The figure illustrates that a widget consists of the aspects mentioned in Section III. The consist of semantic is modeled with associations between the stereotypes. The figure further shows that a widget inherits the UML meta-class *class* and the aspects a widget consists of inherit the UML meta-class *property*. How to model widgets using this profile is outlined in the following explanation.

The static content of widgets can be described directly within the UML model using attributes with the stereotype *description*. The property *init value* for attributes is used to hold the value. To present dynamic content, the widget uses a so called *context* to access the content.

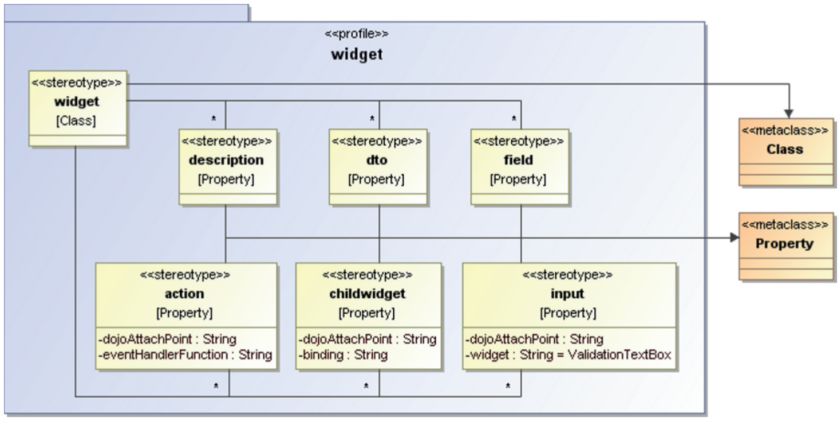

Fig. 2. UML Profile to model Widgets

The context is provided by data transfer objects. Data transfer objects are objects build according to the DTO pattern [9]. Each attribute with stereotype *dto* is part of this context. To represent data, which should not be modifiable by the user, the stereotype *field* is provided. The stereotype *input* is used for editable content. Both stereotypes are applied to attributes and the *init value* is used to specify the data according the following pattern: `<dto>.<property>`. The stereotype *input* provides the possibility to specify a special widget by name, which is used to change the content.

Actions are provided with the stereotype *action*. The stereotype *childwidget* is used to build composite widgets. The tagged value *binding* specifies the data, which is bound to the child widget.

V. Realization

As shown in Figure 1, a controller is needed on client side to handle user events, obtain data from the web server, bind data to the widgets, change the visibility of widgets according to the state of the web application and to map the state of the web application to an URL. The URL is needed to provide history and bookmark support for the web application. Using traditional web applications the web browser does it automatically, but it is not possible with AJAX-based web applications because of the lack of the page reload. The dojo toolkit provides a library for implementing history and bookmark support.

For the presentation layer the dojo widget concept is used. It describes a widget with three aspects:

- A **JavaScript object** that contain functions to create, initialize and destroy the widget and to change the content the widget presents. Event handler functions are also contained.
- A **HTML Template** that contains an HTML structure with placeholders for the content, which represents the widget.
- A **CSS Template** to describe the appearance of the widget.

The generation of the JavaScript objects is the main focus. The JavaScript objects contain the most content, compared to the other templates, based on a comparison using the widgets in the sample web application. It consists mainly of infrastructure code. Figure 3 shows three widgets including a composite widget.

The dojo widget concept consists of a couple of aspects. For example, it provides a way to merge content and the widget structure during widget creation. This is a good point to fill the widget with static content. For dynamic content DOM manipulation is used. The generated widget objects provide methods to keep the bound data and the view synchronized. If the widget is a composite widget, the child widgets are also updated to present the changed data. For actions, the corresponding event handler function is generated as a stub.

VI. Benefits

Our approach aims to describe user interfaces in a general manner. One advantage in the realized sample web application is to hide dojo specific aspects from the developer. The widget is modeled using common user interface aspects. The mapping to the dojo specific realization is done in transformations. Doing so, a lot of infrastructure code was generated.

Another aspect is the clear separation between programming and web design because of the usage of the dojo widget concept. It separates the design (HTML structure and CSS styles) and the programmatic development (JavaScript object). The model driven software development (MDSD) approach supports this concept even further by provideing a formal presentation model as a specification for the web designer.

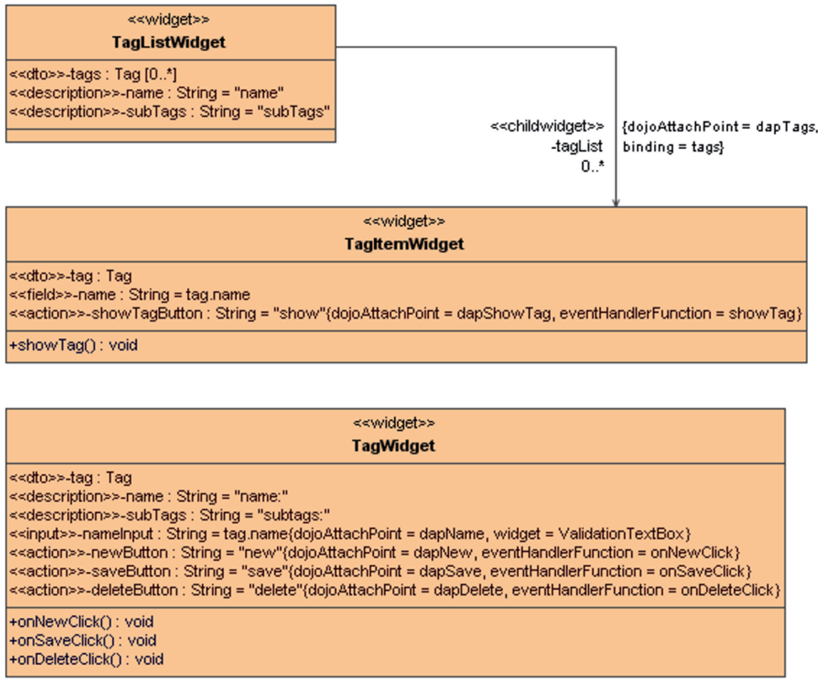

Fig. 3. Example of Modeled Widgets

The web designer can freely choose the layout and styles of the widget, which supports flexible widgets.

VII. Related Work

Approaches for model driven software development of web applications are very widespread. In [1] simple class to HTML mappings are shown. On the other end of the spectrum exist heavyweight and well integrated approaches like the UWE (UML based Web Engineering) project [10] and the CATWALK framework [11], which are both based on the web development framework Apache Cocoon [12].

Related work which is comparable, especially with the combination of the topics AJAX and MDSD, are not known to the authors.

VIII. Conclusion

The outlined approach still covers only the main aspects of user interfaces. Further work should be investigated for more completeness. Two further aspects are accessibility and internationalization. The dojo widget concept provides support for both accessibility and internationalization. E.g. for internationalization a tagged value could be used to indicate whether the content should be translated or not.

Another aspect of more completeness is to automate the controller, which is manually implemented for the most part in the current solution. For this purpose an activity or state chart seems to be appropriate. The URL describes the state of the web application. It contains data to retrieve the content for the widgets from the web server and information about visible and invisible widgets. With state machines another problem occurs. Every bookmarkable state needs a transition from the entry point. Because of the fact that transitions are always directed, every history-able state needs an additional transition in the other direction.

The last enhancement for more completeness to mention could be to determine a common set of events and event handlers. The so-called *hover*-events are common and often used. They always work in the same way but with different parameters. So they are a candidate for code generation.

Another target for further work is generality. The platform specific aspect *dojoAttachPoint*, which currently exists in the UML profile, should be substituted by a more common aspect. It is currently used to set an HTML element to which other HTML elements (e.g. child widgets) are added. It seams that such a concept is also quite common using web template languages like JSP or JSF, to split the templates according to their responsibility.

The main work for generality should be investigated to create transformations for another web template languages, e.g. JSF.

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Exploration and Management of Web Based Multimedia Information Resources

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***Abstract*- WWW is a huge multimedia information resource. It is composed of diverse heterogeneous unorganized information resources. Information resources include multimedia data from different diversities. Information exploration services are required for searching and browsing of multimedia information resources. Browsing mechanisms are satisfactory when information is organized in proper hierarchies. Search mechanisms are independent of organizations with in information resources. They just interact with object indexes for searching and only keep track of information objects with in information resources. Organization of only document indexes is required. Search mechanisms are ideal for the exploration web information resources. Mostly existing search mechanisms available for exploration of web resources are mono-modal. They perform search with in single modality of information. Combined search on more than one multimedia object types is at early stages of development. We explore with the help of multimedia object analysis that search with in all the subsets of multimedia object types is possible by using only four index types, two models of interactions and fifteen possible modes at most.**

Keywords: information resources, World Wide Web multimedia, organization, management, browse, multimodal search.

I. INTRODUCTION

Every day we interact with multimedia objects and search, browse, visualize, manage and store them. Our information needs are satisfied by the exploration of multimedia objects. Multimedia information retrieval system includes exploration and information management services [1]. Multimedia objects are managed with in multimedia digital resources [2]. Browsing of multimedia objects is strongly dependent on organization of information with in multimedia digital resources. Multimedia objects are searched by using their multimodal representations or indexes [3]. Multimedia information needs are well defined and to satisfy these needs, the diverse multimedia information recourses are available. There are challenges in existing multimedia information management; browsing and multimodal search mechanisms. These challenges motivate researchers to investigate new search, browse and information management techniques.

More than one modalities of information are associated with multimedia objects [3]. Multidimensional nature of multimedia object makes research in multimedia information retrieval systems more attractive and challenging. The field is immature and there are certain unresolved issues to be investigated in existing multimodal search, browse and information management techniques. Browsing of multimedia objects is strongly influenced by information organization. Searching of multimedia objects requires management of underlying multimodal representation of multimedia object types. Existing available multimodal search mechanisms have limitations. They cannot address appropriate multimodal representations of multimedia object types. Searching with in multiple media types using their multimodal representations is mostly not supported by existing search mechanisms. Limitations in existing multimodal search mechanisms require further investigation so they better fulfill multimedia information needs. In this article we discuss information exploration services, their association with information organization and propose a multimodal search framework for the exploration web based multimedia information resources.

II. INFORMATION EXPLORATION

Information exploration is a fundamental aspect of almost all information retrieval systems. Information exploration generally consists of three types of services; searching, browsing and information visualization [4]. Searching, browsing and visualization are distinct concepts but collectively fulfill user information needs in almost all types of information retrieval systems.

Search is an approach in which required information is quickly identified using keywords [4]. Generically information search can also be defined as a task in which user specifies information contents and system searches against them with in its information object representations. Due to unavailability of keywords user better navigates information using available contextual information about information organization [4].

Browsing as defined an "an approach to information seeking that is informal and opportunistic, and depends heavily on the information environment" [5]. More broadly we analyze that

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 500–506.

browsing is an activity in which user narrow down object spaces or domains[1] to fulfill his information needs.

Information visualization services enhance information exploration. They have beneficial effects on searching and browsing tasks [4]. Information visualization is defined as "a process, it transforms data, information, and knowledge into a visual form exploiting people's natural strengths in rapid visual pattern recognition" [6]. We define information visualization as a technique to decrease gap between human perception and information presented to the user.

Modern information retrieval systems mostly provide integrated searching, browsing and visualization services [4, 7]. ACM digital library for text retrieval and open video digital library for video retrieval [8] are good examples of integrated searching and browsing services. By using these information retrieval systems users browse collections, narrow down object domains and finally perform search with in selected object domains. Browsing techniques are mostly used to narrow down document domains that assist in more satisfactory, accurate and efficient retrievals.

III. BROWSING ,SEARCHING ASSOCIATION WITH INFORMATION MANAGEMENT TECHNIQUES

World Wide Web is a huge information resource. It consists of heterogeneous and distributed information resources. Diverse information resources collectively make WWW an information giant. Due to diversities in information resources available on WWW, information retrieval systems also show variance. Main aim of information retrieval systems is to retrieve useful information. Each information retrieval system whether it is general purpose, media specific or domain specific search engine, digital library or digital museum must be constituted by following three hierarchical layers.

- *Interaction Layer:* Provides Searching, browsing and information presentation interfaces.
- *Indexing Layer:* Provides Object modeling or indexing mechanisms
- *Organization Layer:* Provide mechanism for Objects organization with in information resources.

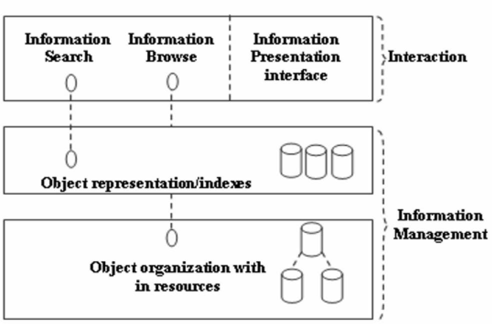

Fig 1: Browsing, Searching Mechanisms and their Association with Information Management Techniques.

First layer is for user interactions and last two layers cover information management mechanisms. Information search interface interacts with object representation and constitutes information search mechanism. Information browsing interface interacts with object organization with in repositories and constitutes information browsing mechanism. Importance of each layer depends upon type of information retrieval system developed, its operational domains and user behaviors. E.g. organization layer is absent in general purpose search engines.

Search engines using searching and browsing mechanism retrieve useful information. Search engines, if follow search approach then interacts with object models, and if follow a browse approach then interacts with object organizations with in repositories. Byron Marshall categorizes search engines as directories and indexes [1].

In directory based search engines information is categorized in predefined categories or cluster of objects. User browses these predefined clusters to fulfill his information needs. This directory based approach is also called top down approach because user starts interaction from highest level of organization [1]. Directory based approach is basically browsing and it is strongly influenced by the organization of objects with in data repositories. Browsing strategies consider object spaces organized as predefined structures or hierarchies. Information retrieval systems that incorporate browsing mechanisms must be aware of object organizations with in object spaces. In yahoo[2], IEEE[3], ACM[4], OVDL[5], WOMDA[6], Digital south Asia Library[7] and Hermitage Museum[8] documents are organized in the form of clusters or categories. User is able to browse with in these categories and some times browsing accomplish information needs with out any search task. DSpace is an open source system. DSpace provides basic features required for the construction of digital libraries [9]. Data model of DSpace addresses object organization with in digital repositories. Data model is constituted by different object types. DSpace provides searching and browsing services because data is organized in proper hierarchies.

Indexed based search engines are not concerned with object organizations with in repositories. They interact with document representations or models. We explore that for index based search engines underlying object spaces are considered flat. In flat object spaces organization of objects with in repositories is not important. Information management techniques for indexed based search engines keep object indexes along with their source links or URLs. Indexed based approaches are also called bottom up approaches [1]. We can also categorize them as information searching mechanisms.

WWW is a massive information resource. It is aggregation of different information resources. Information organization

[1] Object spaces or domains are actually underneath document organization with in collections

[2] http://www.yahoo.com
[3] http://www.ieeeexplorer.org a
[4] http://www.portal.acm.org
[5] http://www.open-video.org
[6] http://www.hcu.ox.ac.uk/jtap/
[7] http://dsal.uchicago.edu
[8] http://www.hermitagemuseum.org

with in its each constituent is mostly heterogeneous. Due to this heterogeneous organization of web resources they are mostly accessible via search mechanisms provided by general purpose search engines like Google[9], altheweb[10], and AltaVista[11] not by browse mechanisms. Mostly users are able to browse searched results. Top down browsing with in web resources is mostly not possible. However researchers try to provide exploration of web resources using browsing techniques. Relational browser categorizes web sites in to topics and provides their topic wise browsing [10]. Open Archive Initiative Protocol for Metadata Harvesting (OIA-PMH) harvest metadata of distributed repositories in a centralize location. So this centralized location provides searching and browsing mechanism of remote repositories metadata from a centralize location. University of Illinois Library at Urbana-Champaign (UIUC) is a Digital Gateway to cultural heritage materials. This project is based on OAI-PMH and harvest metadata from thirty nine metadata providers which includes museums, archives, academic and public libraries, historical societies, consortiums, and digital libraries resources. UIUC provides searching and discovery services [11].

Search, browse and information management mechanisms are interrelated. Clear distinction of search, browse mechanism and their association with information management techniques is necessary. During our investigation the following observations are made:

1. Browsing mechanisms are strictly dependent on information organization in data repositories. Browse mechanisms are based on the philosophy of top down or directory based approaches.

2. Searching mechanisms are independent of information organization in data repositories. Search mechanism interacts with object models or indexes and just keeps track of information object source URLs. They are based on the philosophy of bottom up approaches.

3. Information management techniques for search mechanisms manage indexes and only keep track of information objects by considering underlying information organization as flat.

4. Knowledge management techniques for browsing mechanisms must organize information objects in highly structured hierarchies. Information organization is mostly not flat.

5. Search mechanisms are satisfactory for the exploration of web resources; however browse mechanisms have application in the investigation of web resources. Searched results can be organized in browse-able clusters that ease their explorations.

IV. EXISTING INFORMATION EXPLORATION SERVICES

Multimedia information retrieval system research shows great advancements in recent years [12, 13, 14]. Researchers try to investigate new mechanisms for the exploration of multimedia digital information resources. Researchers deviate from mono-modal search mechanisms to multimodal search mechanisms [3, 15]. Interaction and modeling of a single modality for exploration of web resources is not sufficient to fulfill multimedia information needs. Recent research in multimedia information retrieval systems introduces new services for the exploration of web based multimedia information resources. Recent exploration services broadly classified into web based and non web based services. We discuss only existing web based exploration services.

Web based information exploration services are accessible via web. They can be broadly classified into research in general purpose, media specific and domain specific search mechanisms for digital information resources available on WWW. Information seekers are aware of this research because they mostly interact with information retrieval systems accessible via WWW.

A. General Purpose Search Engines and Information Exploration Services

Google, altheweb, AltaVista and lycose[12] are examples of general purpose information retrieval systems. All these information retrieval systems operate on web information resources. They are not constructed for a particular domain and provide search mechanism for more than one multimedia object types.

Evaluation: Existing search mechanisms available on WWW have limitations. By using general purpose search mechanisms like Google, altheweb, AltaVista and lycose user is able to formulate query and visualize results for one media type at a time. Our investigation reveals that indicated general purpose search mechanisms adopt mono-modal search mechanisms. They can only perform search with in text modality. Information retrieval functionality is mostly provided by using text present on page having multimedia object and file attributes associated with multimedia objects like file type, size, format and color. Existing general purpose search engines give illusion that they can perform search with in modalities of multimedia objects but their working is totally dependent on text retrieval techniques and pre-recorded attributes of multimedia objects. Due to this reason information needs are partially satisfied by using these search mechanisms.

B. Media and Domain Specific Search Engines and Information Exploration Services

Media specific and domain specific search mechanisms are also available on the WWW. They mostly provide mono-modal search mechanisms for specific domains [3]. ACM and IEEE are educational digital libraries. They provide specialized search mechanism for text retrieval with in particular educational domains. Terrasgalleria[13] provides specialized search mechanism for images. Digital South Asia Library provides retrieval facility for cultural heritage data of South Asia. Data is mostly stored in the form of images and

[9] http://www.google.com
[10] http://www.altheweb.com
[11] http://www.AltaVista.com
[12] http://www.lycose.com
[13] http:// Terrasgalleria.com

search facility is provided by annotated text modality. Open Video Digital Library (OVDL) provides search and browse mechanisms for video documents for pre-stored specified video repository. Video documents are organized in the form of cluster of genres like documentaries, educational, lectures, ephemerals, and historical. Each genre is a browse-able collection. Browsing mechanisms are provided by using video document genre, time, color (black or white), and sound (sound or silent). Search mechanisms are provided by using text modality of speech transcripts and bibliographic records. Hermitage Museum is accessible via web and provides searching and browsing of image based museum objects. User is able to browse collections and search using texture and color of image modality. WOMDA operates on multimedia data like manuscripts of poems, diaries, letters and information available in the form of audio, video and images revolves in the age of First World War poet Wilfred Own. WOMDA holds its own archives of data and also encourages public to upload related information [16]. Data is managed in the form of browse-able collections. WOMDA provides search facilities of audio, video, images and manuscripts using annotated text modality.

Evaluation: Domain specific and media specific search mechanisms provide advance search mechanisms. They provide search with in different multimedia modalities and information domains. Media specific search mechanism provides search facility with in a particular type of multimedia. They perform specialized search with in one multimedia type. User is mostly able to formulate query and visualize results for a specific multimedia type. These search mechanisms cannot discuss unification of search approaches for varying multimedia types. They are usually not expandable for multiple multimedia object type's retrieval. Their integration in general purpose search mechanisms that facilitate unified retrieval of all multimedia object types is not approachable. Domain specific search mechanisms rarely provide search with in multiple multimedia object types. Mostly they perform search with in specified domains and particular media types. However some search mechanism provides search with in domain specific multiple multimedia object types [16]. It is explored that their retrieval model is totally based on accompanying text modality of multimedia object types. Due to their limitation they operate for a specific type of media or with in specified domains and user information needs are partially satisfied.

V. USABILITY ISSUES OF EXISTING MULTIMEDIA SEARCH

We have a scenario of multimedia information need that elaborate usability problems in existing search mechanisms. Suppose Mr. X wants information about FIFA World Cup 2006 irrespective of which media type information belongs; this information exists in the form of text documents, video clips, audio files and posters. Existing search mechanisms cannot support retrieval of all these object types in one complete search. Existing search mechanisms either general purpose or media specific cannot support retrieval of subsets of more than one object types with in one complete search. By using existing search mechanisms user is enforced to perform separate searches for all types of objects so information needs are always partially satisfied.

Advance multimedia information retrieval techniques [17, 18] have been investigated in recent years. It is required that researchers should enhance these techniques by investigating and resolving the limitations. These advance retrieval techniques discuss basic indexing mechanism for multimedia data types. Their unification with in information retrieval frameworks is mostly not addressed, so satisfactory multimedia retrieval is not possible. Usable unification of these techniques in proper information retrieval framework should provide foundations for enhancement in general purpose web based search mechanisms to satisfy multimedia information needs.

VI. USABLE UNIFIED MULTIMODAL SEARCH FRAMEWORK FOR MULTIMEDIA INFORMATION EXPLORATION

To overcome usability problems in existing search mechanisms we investigate and propose an ontological based usable unified multimodal framework. The unified multimodal search framework will help users to retrieve multimedia information in a usable way. The framework is explained with the help of following multimedia object analysis.

A. Multimedia Object Analysis

Multimedia object or document in context of multimodal search mechanism is an interactive artifact that can be modeled or indexed, searched, browsed, visualized and retrieved. Multimedia objects can be classified into text, audio, video, images and graphics [19]. We investigate that multimedia objects have close interrelations. Audio, video, image, graphics and text are five basic multimedia object types. Image and graphics are interchangeable. We place graphics in broad category of image types. In this article hereafter we refer multimedia object types as image, audio, video and text. They can be represented interchangeably. Multimedia objects can be expressed as supersets and subsets of each other. Video can be decomposed into audio objects and visual streams. Audio object is composition of speech and sound. Speech can be represented as text objects and their context. Text consists of keywords and their context. Visual stream of video can be decomposed into scenes and their context. Scene is a video clip of continuous images having some specific context. Image objects can be decomposed into objects and their context. Fig. 1 depicts that video artifact is composition of text; image and audio object types. Video object owns all features of other multimedia object types.

Top down analysis of above hierarchy shows that video object type is decomposed into audio, image and text multimedia object types at intermediate levels and finally represented as

1. Text keywords and their context
2. Image objects and their context

Bottom up analysis of Fig. 1 shows initially text keywords, their context and image objects, their context composes image/graphics, audio and text object types and finally interpreted as video object type.

B. Multimodal Search Mechanism: Usable Unified Approach

We investigate from multimedia object analysis that video is a superset of all other multimedia objects. Video can be represented as audio and images. Textual representation of audio speech is possible [17]. Image and graphic objects can be represented by using content based image and textual annotation techniques [20, 21].

Video object is a multidimensional object. Its each dimension represents some specific searchable modality. Its dimensions include image modality, accompanying text modality and speech transcript modality. Image based modality is searchable via content based image indexing techniques [17].

Thijs Westerveld investigates that ordinary keyword based search by stemming and removing stop words from textual representation of speech transcripts along with combination of content based image search gives satisfactory video retrievals [17]. Advance text based indexing techniques have been investigated in recent years [22, 23].

Solution that provides search mechanism for video objects by using almost all of its modalities is a super solution for multimedia object types. Solution that provides search mechanism for text, image and audio objects are subsets of video search mechanism. Video search mechanism is a comprehensive method that incorporates image and text search mechanisms. Constitutes of video search mechanisms can be used for searching all multimedia object types.

1. Text objects can be searched by video search mechanism used for searching accompanying text and speech transcript modalities.

2. Video search mechanism used for searching image modalities can be used for searching image objects.

3. Audio speech transcripts and video speech transcripts are equivalent. Video search mechanism for searching video speech transcripts can be used for the search of audio objects.

Indexing text, accompanying text, speech transcript and image modalities are sufficient for searching with in all subsets of multimedia object types. By using image and text modalities multimedia information user formulates queries for all multimedia object types which enable the user to perform search with in all modalities of information.

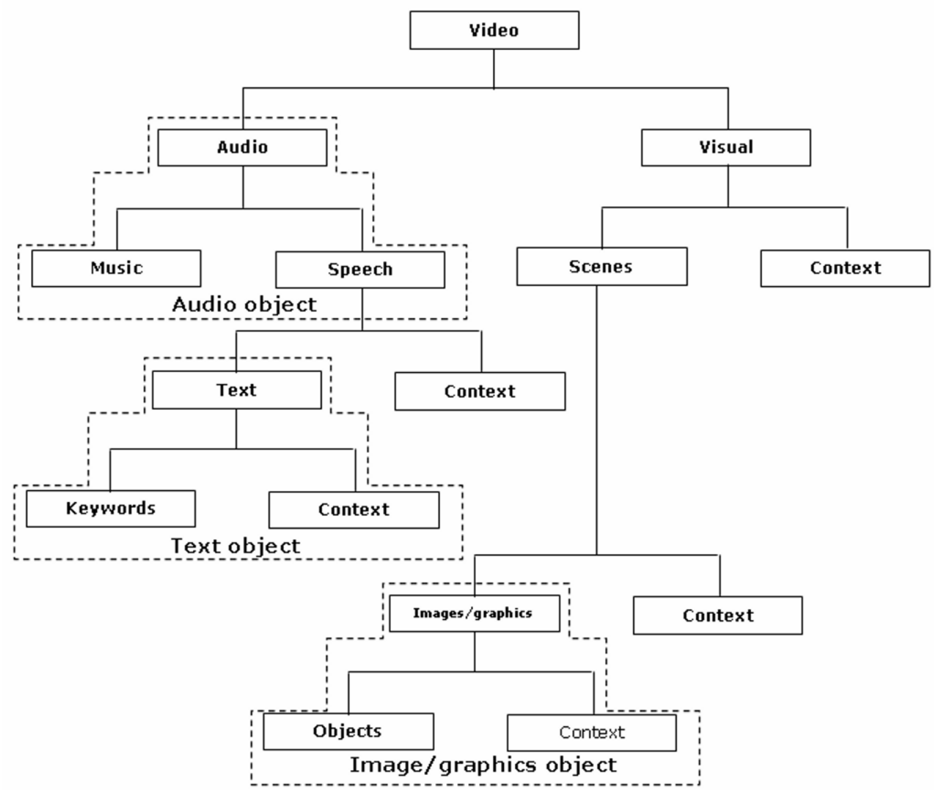

Fig2: Multimedia Object Decomposition Hierarchy, dotted region represents occurrence of text, audio, image/graphics object types with in video object type.

Modes	Multimedia object types	Indexes Involved in search	Modals of Interaction
1	Text	o Text Based Index	o Keywords
2	Image	o Accompanying Text Index o Image Feature based Index	o Keywords o Visual Image
3	Audio	o Accompanying Text Index o Vocal/Speech based index	o Keywords
4	Video	o Accompanying Text Index o Image Feature based Index o Vocal/Speech based index	o Keywords o Visual Image
5	Text, Image	o Text Based Index o Accompanying Text Index o Image Feature based Index	o Keywords o Visual Image
6	Text, audio	o Text Based Index o Accompanying Text Index o Vocal/Speech based index	o Keywords
7	Text, video	o Text Based Index o Accompanying Text Index o Vocal/Speech based index o Image Feature based Index	o Keywords o Visual Image
8	Image, Audio	o Accompanying Text Index o Vocal/Speech based index o Image Feature based Indexes	o Keywords o Visual Image
9	Image, Video	o Accompanying Text Index o Vocal/Speech based Index o Image Feature based Index	o Keywords o Visual Image
10	Audio, video	o Accompanying Text Index o Vocal/Speech based Index o Image Feature based Index	o Keywords o Visual Image
11	Text, Image, Audio	o Text Based Index o Accompanying Text Index o Vocal/Speech based Index o Image Feature based Index	o Keywords o Visual Image
12	Text, Image, video	o Text Based Index o Accompanying Text Index o Vocal/Speech based Index o Image Feature based Index	o Keywords o Visual Image
13	Image, Audio, video	o Accompanying Text Index o Vocal/Speech based Index o Image Feature based Index	o Keywords o Visual Image
14	Audio, video, Text	o Text Based Index o Accompanying Text Index o Vocal/Speech based Index o Image Feature based Index	o Keywords o Visual Image
15	Text, Audio, video, Image	o Text Based Index o Accompanying Text Index o Vocal/Speech based Index o Image Feature based Index	o Keywords o Visual Image

Table 1: Possible modes, indexes and models of interaction used in multimodal search framework

Searching with in subsets of multimedia object types is possible. Four multimedia object types can be represented in 2^4=16 combinations or subsets; one subset is empty so there are total fifteen combinations.

Searching with in subsets is possible by four index types, two models of interaction and fifteen modes at most. Four possible index types are:

1. Text based index for text artifacts.
2. Speech transcript text index for audio and video object types.
3. Image feature based index for image and video object types.
4. Accompanying text based index for audio, video and image objects having accompanying textual information.

We explain fifteen combinations in Table 1. User is able to formulate query for any subset of multimedia object types by using only text based and image based interactions or query formulation mechanisms. Our proposed multimodal search mechanism provides search mechanism with in all the subsets of multimedia objects types by using fifteen possible modes or layers. Table 1 demonstrates modes, possible index types and interactions against each mode of multimodal search mechanism.

A search mechanism that incorporates specified four possible index types and two models of interaction provides search facility with in all possible subsets of multimedia object types. Fig. 2 demonstrates a multimodal search mechanism that has the capability of searching with in any subset of multimedia objects types. One combination of multimedia object types, mode or a subset is activated at a time using four possible index types and two models of interaction at most. Fig. 2 demonstrates this multimodal search framework.

First dotted rectangle represents two interaction mechanisms via keywords and images; second dashed rectangle represents possible modes or subsets. One mode is activated at a time. User is able to perform search with in any subset of multimedia object types by using these fifteen modes. Third rectangle represents possible four index types. Search is always performed with in theses four index types at most.

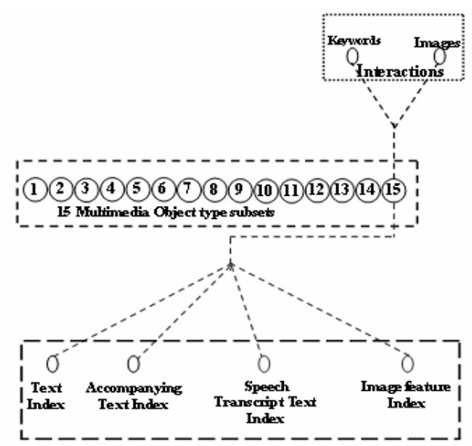

Fig 3: Framework for Multimodal Search Mechanism.

VII. CONCLUSION

Search and browse mechanisms are required for the exploration of multimedia information resources. Organization of web based information resources is unknown so they are mostly not browse-able. Search mechanisms are frequently used for the exploration of web based multimedia information resources. Existing search mechanisms whether they are general purpose, media specific or domain specific partially fulfill user information needs. User is able to perform search with in one media type at a time and they are mostly mono-modal. We investigate from multimedia object analysis that searching with in multiple multimedia types is possible. We have proposed a usable unified multimodal search framework by using two modalities of interaction, fifteen modes and four possible index types at most. By integrating our proposed search framework in web based search engines multimedia information users are able to perform multimodal search in all possible subsets of multimedia object types. Over proposed search framework provides web based multimedia information exploration service.

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GAMMAMAPPER - a Prototype Gamma Spectrometer-Data Logger

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Keywords: *ionizing radiation, spectrometer, data logger, encryption, , Vernam cipher, GPS, geographic coordinates, FAT.*

Abstract **In present study a portable autonomous gamma spectrometer-data logger is described. Device records all measured gamma events separately and binds data to geographic coordinates using a GPS engine. Data is stored on standard memory card with FAT file system. Device offers data protection by encrypting files. A standard I-button works as a key.**

I. INTRODUCTION

Ionizing radiation sources can be found in a wide range of occupational settings, including health care facilities, research institutions, nuclear reactors and their support facilities, nuclear weapon production facilities and other various manufacturing settings. These radiation sources can pose a considerable health risk to affected workers if not properly controlled. From the point of view of the occupational exposure, the radiation dose is a very important measure. But limiting only to measuring dose does not necessarily give enough information about possible hazards. Measurement of the energetic spectra of the radiation source allows distinguishing which isotopes cause the radiation, in what state is the source, whether the contamination of water or tissues occur, etc.

Recorded spectra are sometimes filtered, and the averaging nature of this process may mask short rises of radiation level (e.g. passage through a strong source).Therefore it is desirable to record separately every pulse output by the radiation detector and analyze data later. This method is not very suitable for the investigation of highly active radiation source since it produces too much data. In environmental research possible radiation sources are different and compatible to worked out method.

When studying environmental radioactive pollution, it is essential to have the measurement data binded to geographic coordinates. Nowadays a GPS-receiver is a handy tool, especially if the data should be collected from a moving vehicle.

Present study describes a prototype gamma spectrometer-data logger – “Gammamapper” – designed and built in Department of Physics of Tallinn University of Technology. The device is equipped with a CsI(Ti) scintillation probe and spectroscopic amplifier from Scionix. Every pulse output by the amplifier is captured by a peak detector, digitalized and stored on a memory card using standard FAT file system. Instrument’s main processor tracks geographic coordinates received by GPS engine. A separate low-cost micro controller prefilters GPS’s output stream looking for certain messages only. This “filter” also keeps track on the validity of geographic data and double-buffers it [1].

The main processor of the spectrometer protects the measurement results by encrypting of data.

II. DATA PROTECTION

Sometimes the results of experiments should be kept protected. Protection may include disabling accidental erasure of files and, even restricting access to files.

Traditionally protection is carried out in computers, where experiment data is stored. Encrypting algorithms are complicated but fast and very reliable.

Modern digital instruments offer many side-functions: they perform preliminary analysis of data, log measurements etc. Nevertheless these devices do not provide proper data protection. This is very important when sensitive results are treated (e.g. medical instruments or investigation of a crime scene).

Encryption is one solution to this kind of problem. The best solution is when the measuring device encrypts all data. In this case it is possible to protect encrypted files from accidental erasure also.

III. BLOCK DIAGRAM OF “GAMMAMAPPER”

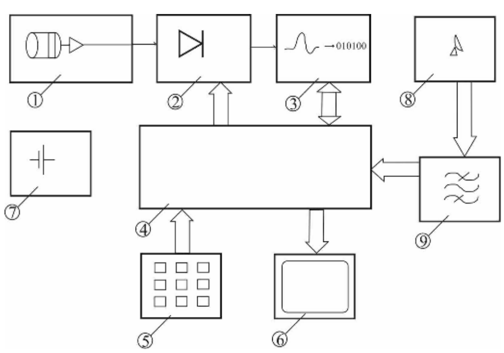

Figure. 1. “Gammamapper’s” block-diagram. 1 – gamma sensor, 2 – peak detector, 3- analog to digital converter, 4 – micro controller, 5 – keyboard, 6 – LCD display, 7 – power unit, 8 – GPS engine, 9 – GPS data filter.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 507–510.

Spectrometer uses Scionix's off-shelf gamma sensor (1) with built-in spectroscopic amplifier (see Figure.1.). This CsI(Ti) probe has compact size, improved ruggedness and significantly lower power consumption compared to standard designs implementing NaI(Ti) scintillation detectors and PMT tubes. Scionix's state-of-art sensor provides a bipolar semi-gaussian output signal with total pulse duration of 15 µs. Energy resolution measured at 662 keV is 8,7%.

Sensor's output pulses are fed to a peak-detector (2) built around fast operational amplifiers by Linear Technology. Detector is reset after every successive digitalization of input pulse.

Successive approximation type analog to digital converter (3) converts captured pulse's amplitude to digital form. This signal is handled by unit's controller.

Measuring process is controlled and all necessary conversions are carried out by firmware running on a standard off-shelf micro controller (4), manufactured by Microchip. This microprocessor has low power consumption and rich set of on-chip peripherals.

Control code also handles storing measurement data to compact flash or multimedia card. Standard file operations like deleting, viewing and moving are supported.

In order to reduce noise, controller is put asleep for most time of acquiring spectra. Every captured and digitalized pulse wakes it up and calls a short routine storing the event. If a preset count of pulses is reached, instruments encrypts data block (if encryption is enabled). After that it is saved on storage media. At the same time keyboard (5) is scanned and information display (6) showing progress of the experiment is refreshed.

Since spectrometers front end is a classical analog circuit, specific methods have to be employed to improve analog to digital converters differential and integral linearity specs. In present device a sliding scale linearization method suggested in [5] has been implemented.

Spectrometer uses a temperature sensor from Dallas to compensate drift of readings due to ambient temperature changes. Sensors readings are stored with spectral data to the same file. This approach should enable recalculating and correcting once measured spectral data in case a better correction algorithm is found.

When charge level is low Gammamapper's power unit (7) automatically terminates measurements. In that case to prevent loss of data and corruption of storage media's file system, all open files are closed. A special flag is set in files supplementary info area to notify premature termination of measurements.

After every preset amount of handled gamma events the spectrometer reads and stores geographic coordinates from a GPS engine (8). The engine (manufactured by Falcom) has been built on a new generation SirfStar III chipset with low power consumption and excellent sensitivity.

A separate low-power micro controller (9) filters GPS's data stream separating useful and valid information.

Measurement data files are transferred to a host computer for further analysis via micro controllers' built-in USB channel. The device has a special working mode where it outputs a block of data right after it has been collected. This mode in conjunction with special program is already used for teaching purposes.

Gammamapper's firmware also permits encrypting every single amplitude of incoming pulse "on the fly". High-speed operation and compact code of the device is achieved by using assembly programming language.

IV. Crypting Data

Scientific instruments can produce results at high speed. Therefore an encrypting algorithm should be compact and fast. One possible solution is to apply a Vernam cipher [2]. According to some authors [3, 6, 7] it is the only currently known unconditionally secure cipher that provides truly random key

There are many ways to generate crypto key. It is possible to use keypad to input the key, or read it from some kind of external memory etc. In this work a standard and widely spread memory key "I-button" producing 64-bit unique number was used [4].

As a result the input crypto key obtained following qualities:

- convenient – user does not have to remember or write down any numbers;
- fast – I-button can communicate with speed up to 16 kBits/s;
- cost-effective – I-button is widely used in various electronic phonolocks and easily available;
- power consumption - is reduced to minimum; the button derives power from data-line and only during reading. It is very economic in a battery-powered instrument.

In this prototype instrument data is saved on a CF- or MMC disk with sector length of 512 bytes. The process of ciphering data in "Gammamapper" is presented in Figure.2. While the processor of a spectrometer has collected a certain amount of results, a Vernam-cipher subroutine is called and the result is written to disk. This approach permits to carry out measurements at full speed. Only an additional small delay due to ciphering during disk write operations occurs.

File system of the spectrometer demands inserting a key (I-button) every time a new measurement is started (key is stored into RAM-memory) and also in case if an attempt is made to delete encrypted file. This should minimize the risk of accidental erasing valuable data files.

V. PC PROGRAM

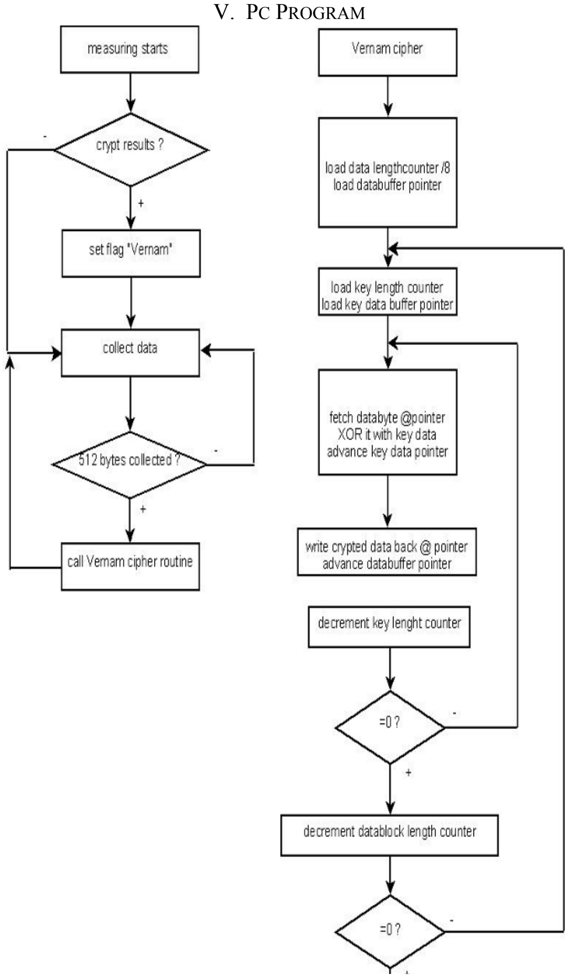

Figure. 2. The Vernam cipher flowchart

To support this system a small utility for visualization and analysis has been written. Figure.3. shows utility's working screens. This program uses Windows platform and allows plotting measured data onto geographical maps. To save computer memory only needed map quadrants are loaded. As the heart of utility's map engine a free GIS environment – Map Window GIS is applied..

On map user can select regions for which gammaspetra will be plotted. For better visualization on map waypoints can be colored according to energy of gamma particles what are detected in these points.

The spectra viewing tool of the program allows to add or subtract spectra (i.e. subtracting background) and format conversions for exporting data. It is helpful if more powerful analysis programs are used.

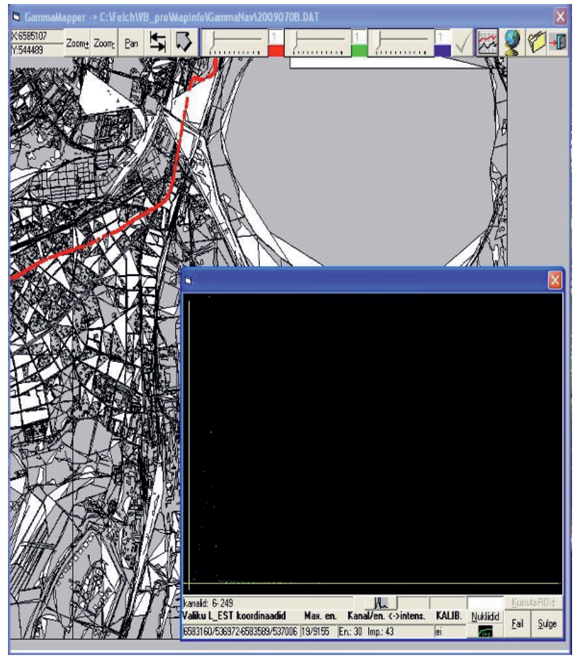

Figure. 3. The screen of Data analysis tool.

VI. CONCLUSION

Described in the present study a Gamma spectrometer-data logger has been tested in operation for about a year. More than 5000 hours of data has been collected. Instrument showed good stability of parameters during test period. Figure.4. reveals the inside of authors prototype.

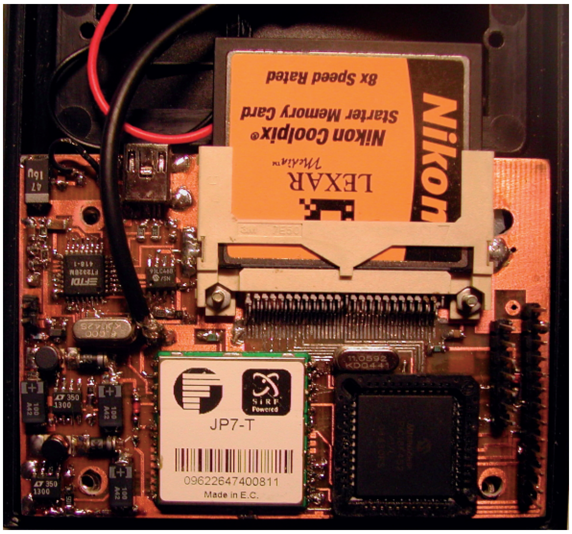

Figure.4. A glimpse to the " inside of " Gammamapper.

Testing showed that possible future versions of the device should include:

– bigger LCD screen and corresponding software for plotting spectra on it;

– possibility to display waypoints on instruments screen since the device has a built-in GPS receiver;

– touch screen instead of keypad to allow more convenient entering of comments to measurements;

– digital signal processor could replace traditional analog front-end circuitry.

A new version of spectrometer with indicated improvements is presently being designed.

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Extending the Consensus Problem

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Abstract – **The aim of this paper is to present a new definition of consensus, one which includes the classical problem of Byzantine Generals but extends to other problems as well. This new definition and its mathematical implications allow the future of other methods in solving this class of problems.**

I. THE BYZANTINE GENERALS PROBLEM

The Byzantine Generals Problem (BGP) is considered to be the problem of attaining consensus in the presence of uncertainties. As the original authors describe[1] this paradigm started with the need for a reliable computer system to operate even when one or more of its components crashed. A failed component may sometimes display an erratic behavior like sending differing information to distinct parts of the system.

The visual representation used to describe BGP is that of an army surrounding a city. The army is composed from many divisions, each division commanded by its own general. The generals can communicate only by messenger. Victory lies in the ability of the commanding generals to decide upon a common plan of action. Some of the generals are traitors and they try to prevent the attainment of an agreement. The algorithm used by the generals must guarantee that:

A All loyal generals decide upon the same plan of action.
B A small number of traitors cannot cause the loyal generals to adopt a bad plan.

All this requirements are equivalent [2] with the following:

Agreement	All nonfaulty generals decide on the same value.
Validity	If all generals start with the same initial value, then this is the decision value.
Termination	All nonfaulty generals eventually decide.

The real meaning of BGP [3] is not finding the traitors, but attaining the same final value among all the loyal generals.

Real fault tolerant systems use majority voting (Byzantine Generals) combined with error correction codes and redundant routing.

II. THE CONSENSUS

Defining consensus

In Computer Science, the consensus is presented [4] as a symmetric problem of reaching an agreement.

The processes involved in the consensus procedure must have enough initial states to allow them any possible outcome.

The most common way to describe consensus is through the correctness specifications [5], which are:

Agreement	All processes decide on the same value.
Validity	The decision value was the initial value of at least one process.
Termination	All nonfaulty process eventually decide, and that decision is final.

The final condition can be modified [6] in order to make consensus possible in more difficult situations.

Weak termination	Some of the nonfaulty process eventually decide, and that decision is final.

It is obvious that this specifications are equivalent with the ones made for BGP.

In real life though, consensus can have a slightly different meaning. Looking up the word in various dictionaries, the definition differs:

> In Webster the definition of consensus is depicted on multiple plans starting with a general agreement, or resolution accepted by all or at least the majority of individuals implicated; and ending with the cohesion of a group regarding sentiments/ believes.
>
> In Larousse the same word is defined as an accord of one or more individuals over one or more texts.
>
> In the Romanian Language Dictionary the term is described as an agreement or accord of people's wills.

Due to the general equivalence between the term accord and consensus, studying the definition of the word accord an interesting description appears:

> A musical accord is a combination of two, three or more sounds of different pitches, that are issued either simultaneously or succeeding, which form together an individual assembly, different from other accidental assemblies.

The idea of a combination, together with the temporal aspect led to the following proposal for a definition:

> The consensus represents the final result of one or more procedures over a initial set of values, where the final result of this proceedings is recognized and assimilated by a sufficient part of the debate participants.

Notes:

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1. The initial values can be totally or partially different, issued either simultaneously or succeeding, but belonging to the same general category.
2. The final values are not bound to belong to the same category as the initial values, but neither are they prohibited to.
3. The composition of the final result from the initial values as a mix or as a grouping of the initial values is a definite possibility.
4. The chronological aspect is not expressed explicitly in this definition, but it is not excluded either.
5. The "sufficient part" varies with each particular case. Nonetheless, the simple majority (50%+1) is usually that sufficient part of the participants.

The proposed mathematical expression of consensus

Let us define the function ψ which describes a process:

$$\psi : V_i^n \times R^+ \to V_f^n \tag{1}$$

$$\psi\left(\begin{bmatrix} x_0(0) \\ x_1(0) \\ \overline{x_{n-1}(0)} \\ t \end{bmatrix}\right) = \begin{bmatrix} y_0 \\ y_1 \\ \overline{y_{n-1}} \end{bmatrix} \tag{2}$$

where:

V_i – initial set of values

V_f – final set of values

n – total systems/processes participating at consensus

t - time

$x_i(0)$ – the initial value subjected to the debate by the system i, where $i \in \{0, 1, 2, \ldots, n-1\}$

Notes:

1. The initial values can be vectors themselves. For example, for a faulty process, the initial value transmitted to every other process can be different, and so in order to fully store the initial state of this process we describe it through a vector.
2. The exact functioning behind ψ does not concern us right now, the black box approach being the most general one. The function ψ varies with each particular case of consensus.

Compressing (2) becomes:

$$X^* = \begin{bmatrix} x_0(0) \\ x_1(0) \\ \overline{x_{n-1}(0)} \\ t \end{bmatrix} \quad Y = \begin{bmatrix} y_0 \\ y_1 \\ \overline{y_{n-1}} \end{bmatrix} \tag{3}$$

$$\Rightarrow$$

$$\psi\left(X^*\right) = Y \tag{4}$$

Let ν be the appearance frequency of an ordinary value k from the vector K who's values belong to the final values set.

$$\begin{aligned} &\nu : V_f \times V_f^n \to N \\ &\nu(k, K) = n_k \end{aligned} \tag{5}$$

where:

N – natural numbers set N={0, 1, 2, 3,…}

n_k – how many times the value k appears in K

We apply and maximize this function to the output vector, Y, of the ψ function defined in (1),(2):

$$\max(\nu(y_i, Y), i = \overline{0, n-1}) = \Theta \tag{6}$$

where:

Θ – is the maximum number of appearances made by any value from the output vector, Y, from the function ψ.

Finally we define a majority function – *maj* :

$$maj : V_f^n \to V_f \tag{7}$$

$$maj = k_j \tag{8}$$

where

$$\nu(k_j, K) \geq M,\ j \in \{0, 1, \ldots, n-1\} \tag{9}$$

where:

M – majority threshold.

Notes:

1. For a 50%+1 majority $M = \lceil n/2+1 \rceil$. Besides, if $M < \lceil n/2+1 \rceil$ there can be two ore more distinct values for which the majority is reached.
2. However, there are cases in which M can be as small as 2 [1] no matter how large is n.
3. Majority sets [3] can lead to more than one values for the *maj* function also.
4. These particulars hypothesis still do not affect the general idea behind this function: find out which value (values) has the most occurrences and that is the final value (values).

We assert that ψ is a consensus function if:

$$\Theta \geq M \tag{10}$$

In other words, the consensus function has as a part of its final result a number of identical values, number which is larger then a certain threshold.

Resuming all of the above , the consensus function can be defined by:

$$\begin{aligned} &\psi : V_i^n \times R^+ \to V_f^n \\ &\psi\left(\begin{bmatrix} x_0(0) \\ x_1(0) \\ \overline{x_{n-1}(0)} \\ t \end{bmatrix}\right) = \begin{bmatrix} y_0 \\ y_1 \\ \overline{y_{n-1}} \end{bmatrix} \\ &\max(\nu(y_i, Y), i = \overline{0, n-1}) = \Theta \\ &\Theta \geq M \end{aligned} \tag{11}$$

The proposed definition is fully described in (11). The initial set of values (V_i), the temporal aspect (t), the final values (V_f), the recognition and assimilation of the final result (Θ) by a sufficient part (M) of the participants are all present. Therefore we concise that (11) mathematically describes the consensus as expressed in the proposed definition.

The proposed mathematical expression of BGP

Based on the functions previously defined, the BGP can be expressed by simply restricting some of the terms, in other words by restricting the generality of (11).

The agreement proposition is achieved if:

$$y_0 = y_1 = \ldots = y_{n-1} = y \tag{12}$$

In other words the decided values must be the same for all of the processes.

Note: this means that a majority function is no longer necessary if there are no errors:

$$\max(\nu(y_i, Y), i = \overline{0, n-1}) = \Theta = n; \quad n \geq M \tag{13}$$

In the presence of errors a majority function is still required:

$$\max(\nu(y_i, Y), i = \overline{0, n-1}) = \Theta \tag{6}$$

$$\Theta \geq M \tag{10}$$

The validity specification is met if:

$$V_f = V_i = V \tag{14}$$

$$\exists x_i(0) = y; \; i = \overline{0, n-1} \tag{15}$$

The last condition, the termination, can be expressed:

$$\psi : V^n \times [0, t_{final}] \to V^n \tag{16}$$

$$t_{final} \neq \infty$$

The mathematical description of BGP is then:

$$\psi : V^n \times [0, t_{final}] \to V^n$$

$$t_{final} \neq \infty$$

$$\psi\left(\begin{bmatrix} x_0(0) \\ x_1(0) \\ x_{n-1}(0) \\ t \end{bmatrix}\right) = \begin{bmatrix} y_0 \\ y_1 \\ y_{n-1} \end{bmatrix} \tag{17}$$

$$\exists x_i(0) = y; \; i = \overline{0, n-1}$$

$$\max(\nu(y, Y)) = \Theta$$

$$\Theta \geq M$$

We see how easily this describes the problem. The different variations of BGP [1] due to different kinds of messages (oral or written) modify only M and the way the voting is made.

The mathematical expression of auctions

Auctions can be of many types [7] as shown in Table 1.

According to the classical definition of consensus, only the first price (closed) and Vickrey auctions can be considered to belong to the consensus class of problems.

For the remaining auctions we should remark that the initial values can be totally or partially different, issued either simultaneously or succeeding. The final result is recognized and assimilated by all the participants.

TABLE 1.
TYPES OF AUCTIONS

English	Ascending, buyers see all the auctioned amounts, buyers can enhance the bet
First price (closed)	Buyers send in an envelope the value they want to bet. Buyers can not enhance their bets Wins the buyer with the best price
Dutch	Descending, the buyers see the current price, they have to decide if they buy now or wait till the price declines more.
Double continuous	Like NASDAQ Buyers and sellers continuously see the market and prices, real time offers
Double closed (hidden)	Buyers and sellers submit simultaneously their offers in an envelope. The organizers open the offers and clarify the results.
Vickrey	Like the closed auction but the person who wins pays only the second price.

The fact that only the participant with the winning offer takes the object home is of no importance here, because all the other participants recognize the result. They all attain that final value, the fact that value does not match their internal maximum value is of no importance here.

The restrictions to (11) made in order to describe auctions refer to V_f, more precisely at what V_f should contain and to V_i, and at the fact that there should be a neutral value in it that means "no bet"

$$V_f = V_i \times V_i$$

$$y_i = (x_{win}, x_{i\max}) \tag{18}$$

$$\exists\, NB \in V_i, \forall v \in V_i$$

$$NB \xleftarrow{\text{overpowered by}} v \tag{19}$$

The majority function is no longer necessary because all the final values contain the win value.

Another application of consensus

An interesting approach can be depicted by different sub-systems that each record a different variable. However every sub-system must know, in the end, the general status of the assembly.

This, according to the classical definition of consensus in computer science does not belong to the consensus class of problems, because the decision value was not the initial value of at least one processes. The final result is a conclusion based on all the processes initial values and so belongs to a different category than the initial values.

According to the proposed definition, though, this problem can be considered a consensus problem.

The time can be restricted like in BGP, but according to [8] it should be left unbound. The restrictions that apply to (11) in order to define this class of problems are minimal, they refer only at V_f.

$$V_f = V_i^m \times W \tag{20}$$

where

m – the number of initial values taken into consideration

W – another kind of values

Note: m may or may not be equal to n. In some cases m can even be 0, the general status of the overall system being depicted only by W.

A visual example of this problem would be two persons trying to decide in which color should they paint a room. One persons wants yellow, the other one wants red. Let's assume that they settle to paint the room orange. Orange is not one of the first choices but it contains equally red and yellow.

Another more interesting example of how this problem can be applied is based on a classical tale: five blind people meet with an elephant for the first time. Each of them touch a different part of it and so, they all think differently about it. However, if they would listen one another they could build up a better representation of the elephant. In this case the input values are something like "tail=rope", "leg=tree", "body=mountain", "ear=leaf", "trunk=snake" and the output value should contain all of this and something more: the resolution that all this forms an elephant.

Some other possibilities

Let us consider the following hypothetical situation: the initial set of values is separated into two sub-sets: an active set (from which the initial values of the generals are selected) and a passive set (one that contains same kind of values as the active ones, of course with different meanings). E.g.:

V_i *active={'attack', 'retreat'}*

V_i *passive={'left', 'right', 'then'}*

Then the combinations can be infinite, e.g.:

('attack', 'left')

('attack', 'left', 'then', 'retreat')

('attack', 'left', 'then', 'retreat', 'then', 'attack')

This means that we consider the final set of values a σ-field over a composition of V_i . This would allow the use of the ergodic theory in solving this problems.

III. Conclusion

The proposed definition of consensus includes without limiting it the BGP, therefore extending the classical Computer Science definition.

The only difference between the problems included in the consensus class are the restriction of the general definition (11). This implies that any result that applies to (11) can also apply to any other problem.

The idea that the final value set is a σ-field over a composition of the initial values set creates definite possibilities in solving the problem with ergodic theory.

Acknowledgement

To my father, for everything he taught me.

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A Probabilistic Logics for RS Latch in Nanoscale

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***Abstract* - The device failure must be taken into account in the nano-scale design. This paper presents the probabilistic logic model to compute the probability distribution of the nano gate states. The characterization is based on the markov random field and statistic physics. The basic logic gates are probabilistically characterized. The effectiveness of the method is demonstrated by an 8-bit full adder. The analysis shows that the device probability distribution highly depends on the system structures and other performance parameters.**

***Index Terms* - Digital circuit, probability, probability logic, nanoelectronics.**

I. INTRODUCTION

The acceleration development of semiconductor industry provides the boom of information technology in the past decades, which gives the revolution change for the human life. However, the further development soon reaches its practice limitation. The traditional silicon-based systems will have feature sizes below thirty nanometers [1]. Nanoscale science and technologies will play a key role for future computing systems. On the other hand, nanotechnology, such as molecular electronics [2], has been demonstrating the potential for unprecedented levels of device density, low-power computing, and higher operating speed. It has brought us the promise of building multi-billion molecular-scale device systems. Although each nanoscale technology has its own characterization, the uncertainty of unreliable devices is common to all nanoelectronics technologies. The classic Boolean logic has exposed its defect in this area. A probabilistic modeling and design methodology for nanoelectronics has to be investigated to handle such a crucial challenge.

In the nano size device, we cannot simply assign the states as 0 or 1, but the probability of being 0 and 1. The new models should meet at least two basic aspects: First, it must describe the probability distributions of the device states effectively; second, it must be compatible to the Boolean logic for the large size chips. The evolution is analogous to the development of the quantum mechanism, the particles position and momentum can be exactly determined in the classic physics, but the concepts are replaced by the wavefunctions in the quantum physics, what we can do is to give the probability of the particles.

There are several models to compute the states probability distribution of the device [3, 4, 5]. One important method is to design the redundant logic in the form of error correction proposed by Von Neumann [6]. He theoretically proved that with an extreme high degree of redundancy, the unreliable logic units could be made reliable. Folling [7] proposed an architectural based on neural network which provides continuous adaptation to errors. The multiple connection paths are used as the synaptic weights of the neural network and a series of required input-output pairs are used for training the network. Bahar [8] suggested a new architecture based on the Markov Random Field (MRF). The MRF is a powerful approach for probability system analysis. It can be used in a network with immunity to architecture failures [9].

In this paper, we present a model of probability logic, which can be used in sequential digital circuit, the theory is based on statistic physics and Markov Random Field. We demonstrate the effectiveness of the method by a *RS* latch and *D*-type latch. The results show new features in nano size, which is the first report of the study on nanoscale sequence circuit with probability characterized.

The paper is organized as follows. In Section 2, we introduce the model of probability logic. Typical logic circuits are characterized and calculated in Section 3.

II. CONTINUOUS MODELING

In the Boolean logic, the symbol X_i can only take two values: 0 and 1. We define the state variable x_i for X_i, which can take any value in the interval [0, 1]. The state variable corresponds to the level of the node. If the level of a node is V, the state variable is $x = V/V_{DD}$, where V_{DD} is the source level. In a CMOS circuit, gates have the ability to tolerant the voltage fluctuation in a certain noise margin. For example, if $x \in [0,0.3]$, it is treated as logical 0; while if $x \in [0.7,1]$, it is treated as logical 1. The continuous variables can be used to model failure probability.

For the symbol X_i and X_j, the resolution of mapping to state variable x_i and x_j is given by [8]:

$$X_i \to x_i,\ X_i' \to 1 - x_i, \tag{1a}$$

$$X_i \wedge X_j \to x_i x_j, \tag{1b}$$

$$X_i \vee X_j \to x_i + x_j + x_i x_j. \tag{1c}$$

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If the probability distribution function (PDF) of the node X is known as $f(x)$, the probabilities of logical 0 and 1 and the corresponding failure probabilities are defined as:

$$P(X=0)=\int_0^{0.3} f(x)dx\,,\quad Err(x)=\int_{0.3}^{1} f(x)dx\,, \tag{2a}$$

$$P(X=1)=\int_{0.7}^{1} f(x)dx\,,\quad Err(x)=\int_0^{0.7} f(x)dx\,. \tag{2b}$$

In the Markov Random Field [10,11], the node's probability is only related with the nodes connected directly. Supposing in an M nodes graph, for node i, the N ($N \le M$) nodes set B_i is called *neighborhood of the node i*, if the nodes in B_i connect node i directly. Suppose each X_i takes the value from a range set L , then P is said to be *Markov random field on L*, if for every X_i,

$$P(X_i \mid \{\mathbf{X}-X_i\}) = P(X_i \mid B_i)\,, \tag{3}$$

Formula (3) means that the conditional probability of a node only depends on its neighborhood. As the state variables take value from interval [0,1], the range set L can be determined as $L=\{x | x\in[0,1]\}$. We define a clique as a set of nodes where each node in the set connects to other nodes in a graph. The advantage of MRF is that the probability of a node can be obtained by the clique formed by the node and its neighborhood. Fig. 1 shows the clique that the node Y is decided by the nodes X_1, X_2, …X_n.

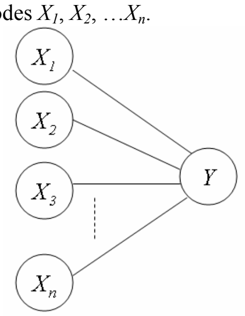

Fig. 1. The clique consists of Y and the n-node inputs.

Suppose that the probability density functions for these nodes are $f(x_1), f(x_2)\ldots f(x_n)$ and $f(y)$, respectively (Note that we assume the probability density functions with different variables can be different functions), where $x_1, x_2 \ldots, x_n, y$ are the state variables for the nodes X_1, X_2, …X_n and Y, $f(y)$ can be obtained from the proposed model.

Some concepts are borrowed from statistic physics [12] in the model. We construct an n-dimensional interactive space Ω which composed by the input nodes X_1, X_2, …X_n, all of them take value for L. Suppose these nodes are independent, the density function for the space can be expressed as:

$$\rho(\Omega)=f(x_1)f(x_2)\ldots f(x_n)\,, \tag{4}$$

which satisfies the relationship:

$$\int_\Omega \rho(\Omega)d\Omega = 1\,, \tag{5}$$

where $d\Omega = dx_1 dx_2 \ldots dx_n$.

We model the interaction of node Y and the input nodes as physics phenomenon. Suppose U_c are the clique potential, which scale the intense of the interaction, according to the statistic physics [12], the clique probability distribution can be obtained by Boltzmann formula:

$$f(x_1,x_2,\ldots x_n,y)= \frac{1}{Z}\exp\left(-U_c(x_1,x_2,\ldots x_n,y)/(k_cT)\right), \tag{6}$$

where kT is the temperature factor in physics view, but here it is just treated as a factor to control the sharpness of the curves. The term Z is called the partition function which is used to normalize the total probability to 1.

From physics point of view, $f(x_1,x_2,\ldots x_n,y)$ can be treated as a micro physical term, and f(y) is the output of the interaction, so it can be treated as the average of micro terms. The PDF of the output node Y, $f(y)$, can be obtained by average the clique probability distribution on the Ω space:

$$f(y)=\frac{1}{Z}\int_\Omega \exp\left(-U_c(x_1,x_2,\ldots x_n)/(k_cT)\right)\rho(\Omega)d\Omega\,, \tag{7}$$

The clique potential U_c plays important role in the calculation. Although it is not the real potential, it can be used to compare the difference of the relative potential levels. In the section III, it will be shown how to use the mapping resolution to translate the logic symbolic to algebraic equations, and how to obtain the clique potentials for basic gates.

III. Probabilistic Characterization of Gates

From the above discussion, we can see the key of the PDF computing is to search a suitable clique potential form, which is decided by the type of the gate. In this section, some basic gates are processed to show how to map the logic symbolic to algebra variables, and then the clique potential computing methods are discussed. From physics point of view, the valid input/output states of the combination circuit should have the low energy while the invalid states correspond to the high energy. The clique potential given in [8] is the negative sum over all the valid states. Here we adopt an equivalent expression which is the positive sum over all the invalid states, it also has the property that the potential of the valid states is lower than the invalid states. The only difference is the constant offset between each other, but it will be eliminated during normalization. The probabilistic characters for every kind of basic gates are also discussed in this section. Finally, a full adder and an 8-bit adder are discussed to show the effectiveness of the model.

A. The NAND gate

Theorem 1: For a NAND gate with input X_0, X_1, and output Y, if the PDF for the input are known as $f(x_0)$, $f(x_1)$, the PDF of Y is:

$$f(y)=\frac{1}{Z}\iint_{L^2}\exp\left(-\frac{U_{NAND}(x_0,x_1,y)}{k_cT}\right)f(x_0)f(x_1)dx_0dx_1\,, \tag{8}$$

where $U_{NAND} = 1 - y - x_0 x_1 + 2x_0 x_1 y$, is the NAND gate's clique potential.

Proof:

Fig. 2. shows a NAND gate with the logic express: $Y = \neg(X_0 \wedge X_1)$.

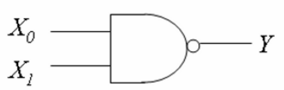

Fig.2. The structure of a NAND gate

TABLE I: THE TRUTH TABLE FOR NAND

X_0	X_1	Y	flag
0	0	0	0
0	0	1	1
0	1	0	0
0	1	1	1
1	0	0	0
1	0	1	1
1	1	1	0
1	1	0	1

In Table I, the states for (X_0, X_1, Y) with flag 1 are the valid states, while states with flag 0 mean the invalid states. According to the mapping resolution, the invalid states are mapped to: $000 \to (1-x_0)(1-x_1)(1-y)$, $010 \to (1-x_0)x_1(1-y)$, $100 \to x_0(1-x_1)(1-y)$ and $111 \to x_0 x_1 y$.

In our model, the clique potential is given by positive sum over all the invalid states:

$$U_{NAND} = 1 - y - x_0 x_1 + 2x_0 x_1 y. \tag{9}$$

Substitute the valid states to (9), $U_{NAND} = 0$, while $U_{NAND} = 1$ for the invalid states. The form of clique potential is satisfied with the requirement discussed above.

Substitute (9) to (7), then we obtain the (8).

Q.E.D.

B. The NOR gate

Theorem 1: For a NOR gate with input X_0, X_1, and output Y, if the PDF for the input are known as $f(x_0)$, $f(x_1)$, the PDF of Y is:

$$f(y) = \frac{1}{Z} \iint_{L^2} \exp\left(-\frac{U_{NOR}(x_0, x_1, y)}{k_c T} \right) f(x_0) f(x_1) dx_0 dx_1, \tag{10}$$

where $U_{Nor} = 1 - y + (x_0 + x_1 - x_0 x_1)(1 - 2y)$, is the NAND gate's clique potential.

Proof: Fig. 3. shows the structure of a NOR gate. The logic can be expressed as: $Y = X_0 \vee X_1$

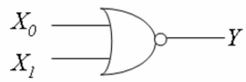

Fig.3. The structure of a NOR gate.

Similar with the discussion of the NAND gate, the valid states for (X_0, X_1, Y) are (001), (010), (100) and (110), and the invalid states are: (000), (011), (101) and (111), according to the mapping resolution, the clique potential can be obtained by summing over all invalid states:

$$\begin{aligned} U_{NOR} &= (1-x_0)(1-x_1)(1-y) + (1-x_0)x_1 y + x_0(1-x_1)y + x_0 x_1 y \\ &= 1 - y + (x_0 + x_1 - x_0 x_1)(2y - 1). \end{aligned} \tag{11}$$

Substitute the valid state to (11), $U_{NOR} = 0$, while $U_{NOR} = 1$ for the invalid states.

Substitute (11) to (7), then we obtain the (10).

Q.E.D.

Suppose both of the inputs are set to zero, if there is no fluctuation, the PDFs for X_0 and X_1 are:

$$\begin{aligned} f(x_0) &= \delta(x_0), \\ f(x_1) &= \delta(x_1). \end{aligned} \tag{12}$$

where $\delta(x)$ is called as Direc function. The properties of Direc function can be found in textbooks of digital signal processing [13]. Substitute (12) to (10), we can obtain the PDF of the output Y. If $k_c T$ is small, Y's PDF can be simply expressed as: $f(y) = \frac{1}{k_c T} \exp\left(-\frac{1-y}{k_c T} \right)$. We select the $k_c T$ as 0.05, 0.1, 0.2 respectively, the PDFs are plotted in Fig. 4.

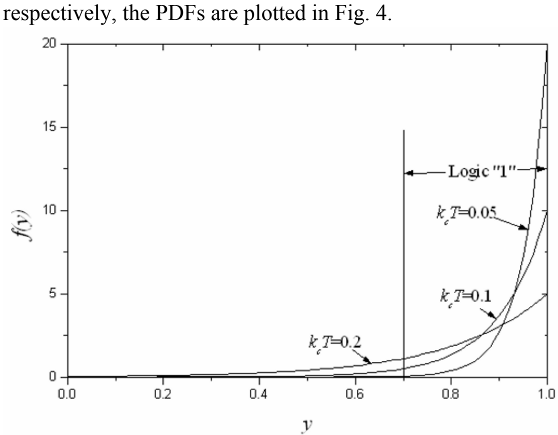

Fig. 4. The PDF of the output Y with different $k_c T$.

Fig. 4 shows the curves of the PDF with different $k_c T$. The low $k_c T$ corresponds to sharp curves while the curves become flat when $k_c T$ increases. According to the above discussion, the output Y can be treated as logic "1" when $y \geq 0.7$, so the probability of Y being "1" can be calculated by (2), which is 99.7% when $k_c T$ =0.05, 95% when $k_c T$ =0.1 and

77.7% when k_cT =0.2. The results show that failure probability becomes unwanted large with increasing k_cT .

C. The NOR type RS latch

Based on the above discussion, we can investigate the behavior of nanoscale sequential circuit. In this part, a NOR type RS latch will be studied.

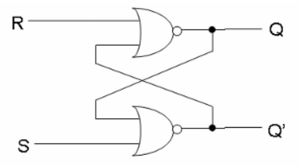

Fig.5. The NOR type RS latch.

Fig. 5 shows a NOR type RS latch, which composed by two NOR gates. R and S are the inputs and the Q and its complement Q' are the outputs.

If we set the S=0 and R=1 at beginning, Q will be set to zero and the Q' will be one, and S=0, R=0 in the following periods, the Q[*n*] will keep being 0 while Q'[*n*] keep being 1.

Suppose that x_S, x_R are the state variables for the S, R; $x_{Q[n]}, x_{Q'[n]}$ are the state variables for the Q[*n*] and Q'[*n*], the PDFs can be computed by the coupled equations:

$$f\left(x_{Q[n]}\right)=\frac{1}{Z}\iint \exp\left(-\frac{U_{NOR}\left(x_R, x_{Q'[n-1]}, x_{Q[n]}\right)}{k_cT}\right) f(x_R)f(x_{Q'[n-1]})dx_R dx_{Q'[n-1]} \ , \tag{13a}$$

$$f\left(x_{Q'[n]}\right)=\frac{1}{Z}\iint \exp\left(-\frac{U_{NOR}\left(x_s, x_{Q[n-1]}, x_{Q'[n]}\right)}{k_cT}\right) f(x_S)f(x_{Q[n-1]})dx_R dx_{Q[n-1]} \ . \tag{13b}$$

In the first period, we set the S=0 and R=1, the PDFs for them are:

$$f(x_S)=\delta(x_S),\ f(x_R)=\delta(x_R-1), \tag{14}$$

The *Q* will be zero and *Q'* be one in this period. In the following periods, we keep *S*=0 and *R*=0, the PDF for S is the same as (14), but the PDF for R should be changed to $f(x_R)=\delta(x_R)$. Using (13), the PDFs for *Q* and *Q'* in the next periods can be calculated step by step. We select the maximum periods number *n*=4, and k_cT are selected as 0.05, 0.01 and 0.02 respectively. The results are listed in Fig. 6-8.

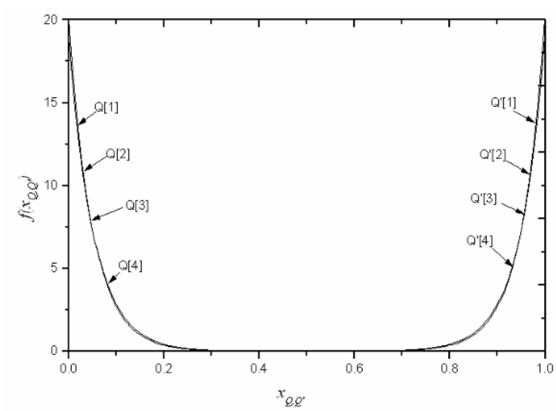

Fig. 6. The PDFs for *Q* and *Q'* in every periods, k_cT =0.05.

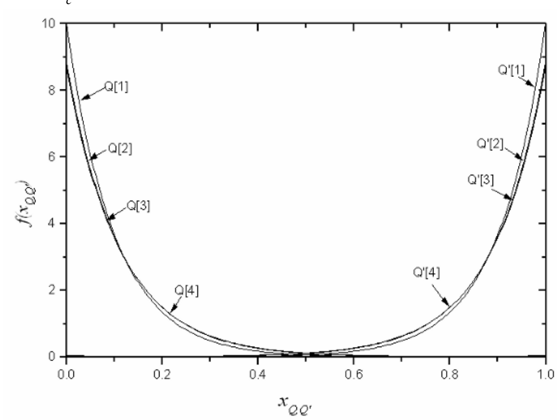

Fig. 7. The PDFs for *Q* and *Q'* in every periods, k_cT =0.1.

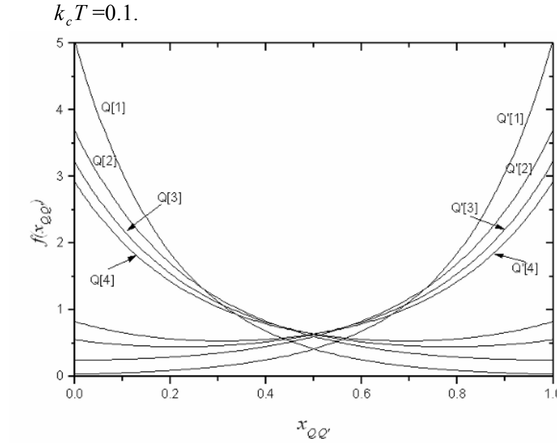

Fig. 8. The PDFs for *Q* and *Q'* in every periods, k_cT =0.2.

Fig. 6-8 list the PDFs for *Q* and *Q'* in every periods with different k_cT . In the case of k_cT =0.05, the curves have cliff shape, and the curves for *Q* and *Q*' in different stage are almost overlapped respectively. With the increasing of $k_c T$, the curves are becoming flat, especially for the PDFs of later stage. We can use the formula (2) to calculation the failure probabilities for every cases. The results are listed in table II.

TABLE II: THE FAILURE PROBABILITY FOR NODE Q IN DIFFERENT STAGE

	$k_c T$ =0.05	$k_c T$ =0.1	$k_c T$ =0.2
Q[1]	0.0024	0.05	0.218
Q[2]	0.0036	0.07	0.347
Q[3]	0.0037	0.08	0.415
Q[4]	0.0037	0.09	0.464

Because of the symmetry of Q and Q', Table II only lists the failure probabilities of Q. One can see the failure probabilities increase with both $k_c T$ and the stages, this is because the feedback of the outputs to the inputs, the failure probabilities are enlarged step by step. In the case of $k_c T$ =0.05, the failure probabilities for all the stages are negligible small, which means they are highly consistent with the Boolean logic. The failure probabilities for the case of $k_c T$ =0.1 are great than that of $k_c T$ =0.05, but they are also acceptable small. For the case of $k_c T$ =0.2, most of the failure probabilities are larger than 20%, and the highest failure probability even reaches 46.4%, which means the system are unreliable.

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An Intelligent Based Model for Urban Demand-Responsive Passenger Transportation

Xu JIN, Mhamed ITMI, Habib ABDULRAB

Abstract— **Agent-based urban traffic service has been looked as an efficient tool for traffic services. However, the main problem is how to build an agent-based model for it. This research presents a multi-agent based demand responsive passenger transport services model, which adopts a multi-agents coordination approach for urban traffic services control. In this paper, we propose a new demand responsive transportation services information intelligent control hybrid model based on multi-agent system that performs the basic interface, planning and supports services for managing different types of demand responsive transportation. In this research, we expose the main features and the behaviors exhibited of the multi-agent system. Based on this model, a simplified multi-agent based demand responsive transportation services system can be developed that is effective for reducing traffic congestion and air pollution.**

I. INTRODUCTION

The current transportation situation, main objectives of demand responsive transport services and advantages of the multi-agent concept are described in this section:

A. Actual Situation

In recent years, urban traffic congestion and air pollution have become huge problems in many cities across many countries. In order to reduce congestion and air pollution, many governments have invested in improving their city infrastructures. However, infrastructure improvements are very costly to undertake. Hence, existing infrastructure and vehicles have to be used more efficiently. Therefore, research on new traffic information control and traffic guidance strategies are particularly necessary and important. The application of new information technologies such as multi-agent technologies to urban traffic information control has made it possible to create and deploy more intelligent traffic management.

B. Objectives

To reduce traffic congestion, air pollution, accidents, financial costs, and other environmental damages. It is necessary to conduct further research on the various characteristics of traffic flow patterns. In general, road traffic system consists of many autonomous, such as vehicle users, public transportation systems, traffic lights and traffic management centre, which distribute over a large area and interact with one another to achieve an individual goal. Our objective is to increase the efficient passages of every vehicle, while at the same time reduce the number of vehicles on the street. This could result in reduction in air pollution caused by the vehicle, traffic congestion and financial cost.

C. Demand Responsive Transport (DRT) Services

Demand responsive transport services are planning computer systems in charge of the assignment and scheduling of client's traffic requests and using different vehicles available for these purposes. DRT services can provide rapid response transport services 'on demand' from the passengers, and offer greater flexibility in time and location for their clients. Moreover, it could also increase the number of passengers in every vehicle, thereby helping reduce environmental pollution and traffic congestion and financial cost.

D. Advantages of the Multi-agents Concept

A multi-agent system is an autonomous intelligent computer system, in which every agent always has a certain level of intelligence. The level of an agent's intelligence could vary from having pre-determined roles and responsibilities to a learning entity. A multi-agent system is an aggregate of agents, with the objective of decomposing a larger system into several smaller agent systems in which they communicate and cooperate with one other. So agent-based model can produce high-quality simulations for complex and large-scale system behaviors, such as the urban traffic system, a large-scale complex system with multiple entities and complicated relationships among them. Hence, the application of multi-agent system to model and study urban traffic information systems is highly suitable and can be very efficient.

In this paper, we propose a set of new demand responsive transportation information intelligent control hybrid model based on multi-agent systems that is able to automatically carry out traffic information control for DRT services. This paper is organized as follows: The next section describes related work on urban traffic simulation. Section 3 describes the problem to solve in greater detail. Section 4 describes the framework we have designed for the traffic information control based on MAS. In section 5 we define the agents for our problem domain. Section 6 introduces the agent planning sequence model. Section 7 defines the planning algorithm. Finally, Section 8 concludes the paper.

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II. SOME RELATED WORK

The application of new multi-agent technologies to urban traffic information systems has made it possible to create intelligent systems for traffic control and management: the so-called Intelligent Traffic Systems (ITS) [1] or Advanced Traffic Management Systems (ATMS). The basic task of ITS is to support road managers in traffic management tasks [2].

Because urban traffic networks have interrupted traffic flow, they have to effectively manage a high quantity of vehicles in many small road sections. On the other side, they also have to deal with a non-interrupted flow and use traffic sensors for traffic data information integration. These features make it difficult for real time traffic management. The urban traffic simulators can be classified into two main kinds: macroscopic simulator and microscopic simulators. Macroscopic simulators use mathematical models that describe the flows of all vehicles. In microscopic simulator, each element is modeled separately, which allow it to interact with other elements.

Multi-agent systems are an efficient tool for the basis of urban traffic simulator. Many researchers have made studies on this subject. In [3], Patrick A. M. Ehlert and Leon J. M. Rothkrantz designed driving agents, which can exhibit human-like behaviors ranging from slow and careful to fast and aggressive driving behaviors. In [4], J. Miguel Leitao uses autonomous and controllable agent to model both the traffic environment and the controlled vehicles. And their scripting language is based in a well-known graphical language, Grafcet. In [5], Joacbim Wahle and Michael Schreckenberg present and review a framework for online simulations and predictions, which are based on the combination of real-world traffic data and a multi-agent traffic flow model.

III. PROBLEM TO SLOVE

A. Existing Solution for DRT

1) Telematics-based DRT

In order to alleviate the problems encountered in traditional transit service several flexible services were studied and offered. Telematics-based DRT systems based on traditional telecommunication technology has played a role in providing equitable transportation service to elderly and handicapped persons who have difficulty in accessing regular public transit systems. Telematics-based DRT systems are based upon organization via a Travel Dispatch Centre using booking and reservation systems which have the capability to dynamically assign passengers to vehicles and optimize the routes. A schematic representation of telematics-based DRT services is shown in Figure 1.

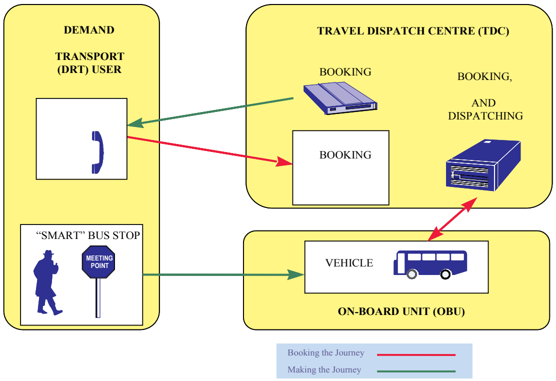

Fig. 1. Traditional Telematics Based DRT

Because it is based on traditional telecommunication technology, the telematics-based DRT services response to client is slow, and sometimes it is difficult to find the best solution for client, besides being unstable.

2) Intelligent Platform -based DRT

Intelligent Platform-based DRT systems are based upon an intelligent platform which has the capacity to dynamically assign passengers to vehicles and optimize the routes. These systems are used to provide real-time information on the status and location of the vehicle for the client. A schematic representation of Intelligent Platform-based DRT services is shown in Figure 2.

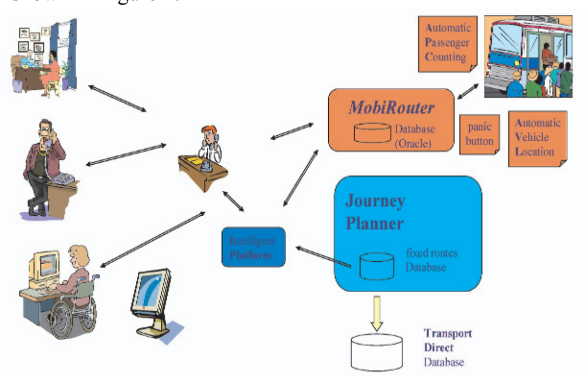

Fig. 2. Intelligent Platform Based DRT

The Intelligent Platform-based DRT services respond to clients in real time, enhances mobility and more reliable.

B. Approach for the Problem

We propose an approach based on multiple agents. Multi-agent systems can exhibit more complex behaviors such as: autonomy, learning, aggregation of agents: decompose the larger task to several small agents, and communicate and cooperate with each other.

Agent-based model has been used for simulation if complex large-scale system behaviors. So the application of multi-agent system to DRT system is suitable.

In short, the problem can be stated as follows: "How can we make agents to cooperate so as to give the client the best trip solution for his transportation request?" (See Fig.3)

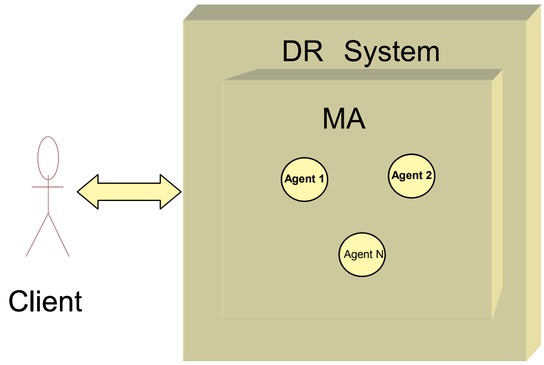

Fig.3. Agent Based DRT

In the following sections, we propose our agent-based hybrid model for demand responsive transportation information intelligent control simulation that perform the basic interface, planning and support services for managing different types of DRT services.

IV. FRAMEWORK OF TYSTEM

This section describes the agent framework used for urban demand responsive transportation information intelligent control. The system agent framework is composed of three layers. The first layer is an agent platform (Jade platform), on top of it is the multi-agent architecture and finally there is the urban demand responsive transportation information system. (See Fig. 4)

In the lowest layer, the Jade agent platform [6] provides a distributed environment organized in containers where agents can reside and communicate. The JADE platform provides an agent management system (AMS) in charge of agent identification and localization, a directory facilitator (DF) for identifying agents by their offered services and a message transport system devoted to support the communication between agents and containers.

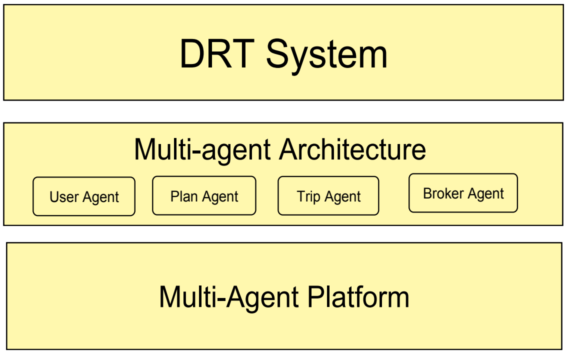

Fig. 4. System Agent Framework

On top of the JADE agent platform resides the multi-agent architecture [7], providing the base agents, structures and semantics for implementing urban demand responsive transportation information intelligent control system [8]. The agents are of four kinds as shown in Fig. 4: the user agent, which is in-charge of the communication with the different parties involved (vehicles, clients, other systems) by means of different devices [9]; the plan agents, which are in-charge of processing, assigning and scheduling the received trip requests encapsulate the assignment and scheduling of the transportation requests processed by the system; the trip agent, which is in-charge of modeling each real vehicle for the system [10]; and the broker agent, which is in-charge of the matching of transport requests with available vehicles [11].

Finally, over the first two layers an urban demand responsive transportation information system layer is implemented. By extending and implementing the base agents system provided by the architecture, a concrete trips planning and traffic information control system can be developed.

V. THE DEFINITION OF AGENTS

This section describes the main agents in the multi-agent architecture used for urban demand responsive transportation information intelligent control system.

A. *User Agent*

This agent represents a human user and their (allowed) interactions with system. It is responsible for capturing all the client's requirements. It provides communication and interoperability between the end user and the urban demand responsive transportation information intelligent transportation system.

B. *Plan Agent*

This agent is in-charge of processing, assigning and scheduling the received trip requests. It is responsible for coordination with the other agents and the administration of the transportation service. It helps in the collaboration among agents that provide support to related functions, such as the matching of request to vehicles, the geographical data access, the accountability of the transactions and the service payment among others.

C. *Trip Agent*

This agent is in-charge of modeling each real vehicle in the system. It manages the trip plan of the vehicle and provides interoperability between the vehicle and the DRT system. It makes proposals for the actual plan, processes change information by the planner agent, updates the plan and reschedules the remaining requests.

D. *Broker Agent*

This agent is in-charge of matching transport requests with available vehicles. It manages service descriptions coming from vehicles side and client's side. The broker agent only provides a list of possible vehicle candidates, but it does not perform the assignment of transport requests to vehicles as the plan agent.

VI. AGENT PLANNING SEQUENCE MODEL DESIGN

The agent planning sequence model design could be carried out by either a centralized model or a decentralized model. The differences among these two options are briefly described [12] and then we present our hybrid model for the solution.

The centralized model considers the optimization of the entire system as the most important thing. For this reason, it pursues the minimization of all disutility function that considers the agent operator (number of vehicles required, bus occupancy and slack times) [13] and the served users (effective waiting time, effective excess ride time).

The decentralized model is based on self-interested actors. This means that each agent seeks the maximization of its own utility. The hybrid model uses a third actor, the planner for applying filtering policies [14]. The agent behaves in a self-interested way as in the decentralized model approach [15]. The difference from the decentralized model is that the Planner filters the proposals by using different policies and hence is more stable.

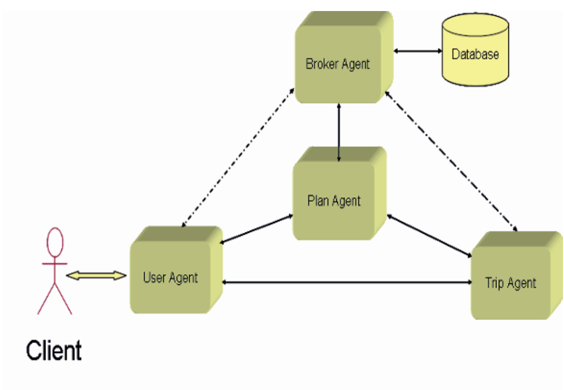

Fig. 5. Hybrid Model

In the hybrid model

1) A hybrid structure is used.

2) The plan agent applies filtering policies.

3) The Plan agent uses different policies to filter proposals received, whereby it can discard the worst proposal. So it can give the client the best solution. And the model has more stability. (See Fig. 5)

In this model the User agent's job is to represent the client and his decisions of the transportation request. So it also is responsible for the result. Since the real client do not directly communicate with the Plan agent, the User agent also constitutes a kind of client towards the transportation system, which is represented by the Plan agent. So when the clients change their requests in a dynamic scenario to deal with unexpected situations, the User agent also have responsibility for communicating the client about any subsequent changes to the original deal. These unexpected situations must also be communicated with the other agents. The User agent must communicate to the Plan agent about changes on the clients' desires (e.g. change the lieu, delays, and trip cancellations). The Plan agent will implement negotiation with the Trip agent and the Broker agent about the changed request. After filtering a list of proposals, the User agent gives the client (through his User agent) the most suitable trip solution offered by the Plan agent. At the same time, the User agent holds a request profile containing the client's preferences concerning the trip. It also holds some negotiation capabilities with the Plan agent.

The Plan agent processes all the client's requests coming through the User agent. It is the agent that in charge of executing a negotiation role in the layer. In addition, the Plan agent is in charge of implementing the assignment through filtering policies and the negotiation process. It holds a list containing the trip requests being processed and a list of filtering policies to apply to the trip solutions. Before the Trip agent gives the proposals to the Plan agent, it will communicate with the Broker agent that manages the trip plan of the vehicle and other traffic information. When the Plan agent receives the trip-proposals from the trip agent, it can implement different filtering policies that include: minimize the number of used vehicles, the waiting time, the travel time, and the total traveled distance. This point is very important for the client that the system just like a "Black Box" [16]. The clients propose the request to the User agent and get the best solution.

To sum up, a comparison of three models has shown that the Hybrid model is able to provide resolution for the gaps in performance between the centralized and decentralized models. The hybrid model gives better results for the clients, in terms of excess travel time when compared with the centralized model. On the other hand, when compared with decentralized model, it gives better results for waiting time. And the most important thing is the hybrid model is more stable than the centralized and decentralized models.

VII. PLANNING ALGORITHM

In the demand responsive transportation system, a general formulation of a client's demand function is:

$$Y = F(x_1 \bullet x_2 \ldots.. x_i) \quad (1)$$

Where Y is the dependent variable (level of demand) and x_i ($i = 1, \ldots, n$) are the explanatory variables. The demand function reflects the behavior of an individual client, whose preferences dictate the particular functional form of the relationship, or it may be formulated to explain the behavior of an aggregate group of individual client.

A particular specification of the demand function which is often used to analyze client's travel demand is expressed in terms of 'generalized costs' (time, money…). This concept is an attempt to summarize the 'cost' of a journey by adding together the various components of time or money spent. The operation system can be formulated as the following optimization problem:

$$\text{Minimize} \qquad AC_T = \beta\sum_{i=1}^{M} a_i \bullet q_i + m + c_0 \qquad (2)$$

Where:

AC_T is the all generalized cost of the journey at time T.

β is the value of coefficients.

M is a set of all passengers.

m is the monetary cost of the journey.

a_i is the value of time associated with time component i.

q_i is the time required to complete the journey divided into the various components i of traveling time: like a get-on delay time of passenger i, a get-off delay time of passenger i, the travel time.

c_o is the residual component of the 'cost' of making a journey which is not a function of the monetary cost or the time involved but is a 'cost' associated with user's special requirement component. This cost may represent many different aspects of travel, e.g. the effort involved in taking heavy or bulky baggage on a trip or that part of the discomfort or general inconvenience which is not a function of the time duration of the journey. In some models this component also represents the different propensities to travel of different segments of the population, grouped by income, sex, age etc and will take different average values in all these categories. The generalized cost function represents the traveler's willingness to pay to reduce the waiting, walking or trip time. The residual effort component is also measured in monetary units.

The User agent receives passenger's requests at time T and transfers them toward all others agents, and holds information of all passenger's requests. Plan agent exists for each vehicle and determines its route by cooperation with the trip agent and Broker agent. The route is determined as follows: 1) Finds some candidate routes by its local search, 2) decide each route by cooperation of agents, 3) do the above step until each route is decides, 4) when receive a new trip request T=T+1 do the above step. The flow of planning system is shown in Fig.6.

Plan agent operation algorithm

A. Terminology

P(t) : a set of passengers who are not assigned to any vehicles at time t. A request content of passenger $c \in Q$ is four components: (WT_{on}, WT_{off}, P_{on}, P_{off}), where P_{on} is get-on point, and P_{off} is get-off point. Q is the number of all passengers at time t.

$G_i(t)$: a set of passengers who are assigned to vehicle i at time t.

$G'_i(t)$: a set of passengers who have already got on bus i at time t.

F (t): a set of passengers who have already been transported at time t.

$STOP_i$: a sequence of points for the vehicle i which is determined by the below-mentioned routing algorithm.

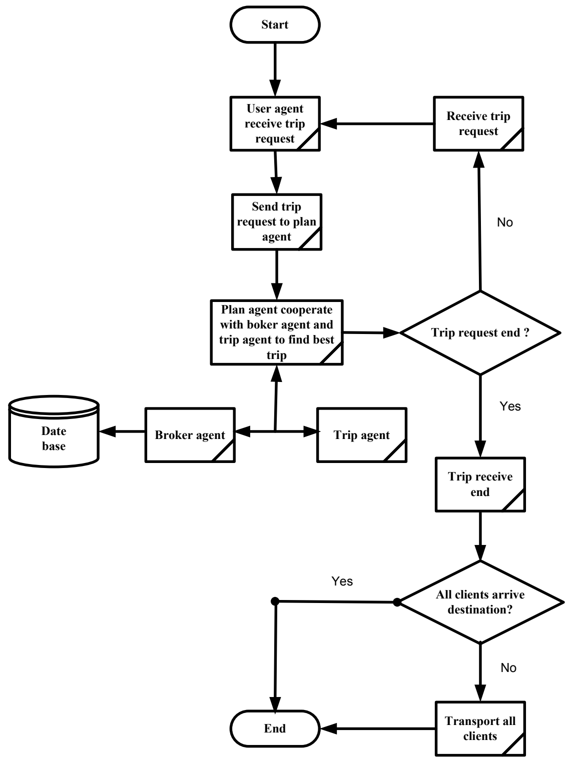

Fig. 6. Flow of the Planning System

$STOP_i = STOP_0, STOP_1, STOP_3, \ldots\ldots\ldots$

B. Agent cooperation plan algorithm

The following procedure explains how the routes of vehicles are determined by agents cooperated when a new request trip is received.

Step1: Broker agent set P (t), $G_i(t)$, $G'_i(t)$, F(t) for each vehicle i as follows:

$P(t) = Q - G_i(t) - F(t)$

$F(t) = F(t-1)$

$G_i(t) = G_i(t-1)$

$G'_i(t) = G'_i(t-1)$

At this step, broker agent in-charge of matching transport requests with available vehicles.

Step2: Trip agent finds the trip for passengers $G_i(t)$ that satisfies the following constraints by local search:

Minimize $\beta \sum_{i=1}^{M} a_i \bullet q_i + m + c_0$

Where

q_i is seted as the maximum waiting time(WT): a get-on delay time of passenger i, a get-off delay time of passenger i.

Select all routes which satisfy the constraints as candidate. At the first search, the variable c_0 is set to 100.

In the initial **as**signment process, Trip agent calculates the evaluation value of route in case the new passenger is inserted into the route according to a heuristic routing algorithm. The routing algorithm works as below: the Plan agent select the nearest stop point from the present location stops point but if the waiting time to get-on is longer than the maximum value of waiting time, the point is selected later. Each new passenger is assigned the vehicle which the waiting time is minimum.

Step3: In the cooperation process, Trip agent and Broker agent transfer some plans to Plan agent and each other to reduce the evaluation value. It is repeated until all passengers are transported at time t.

If there is no trip satisfied, change c_0 to satisfy the request of passengers and back to Step2.

VIII. Experiment

The test considered 200 users, 100 vehicles and 8stops. The demand scenarios were distributed uniformly in one-hour horizon. The results (see Table I) have shown that the approach gives better results for the clients, in terms of both, delay travel time and waiting time. On the other hand the hybrid model performs better from the vehicles' perspective.

TABLE I
Simulation Result

	result
Number of vehicles	99
Total distance traveled	2354.3 km
Number of Users	200
Completed number of trip request tasks	1703
Average waiting times	162.88 s
Average delays	13.62 s

IX. Conclution and Future Work

In this paper we have described a multi-agent architecture for the urban demand responsive transportation services information intelligent control system, and proposed a new hybrid model.

Because in the real system, the stability is the most important, and the multi-agent hybrid model is more efficient in the trip optimization and decision making process.

As future work is considered to continue optimizing the model with known benchmarks and to implement and evaluate a new version for demand responsive transportation system.

Acknowledgment

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Database Management System for Protein Structure Data

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Abstract-In this application system, we extended our domain specific Protein-QL and Protein-OODB into object-oriented database, relational database and XML database to manage protein structure data. We use EyeDB, Oracle and Oracle XML as the base databases to communicate with Protein-OODB. Our system uses RMI to return the results back to the clients so that transactions can be conveniently transformed between client and server. In this system, users can easily master our Protein-QL query language and get help from our online Help services. This system is implemented for protein domain, but it can be easily extended into other biological domains to build a Bio-OODBMS.

I. Introduction

In decade years, the protein data has been growing in an exponential rate since more and more advanced experimental techniques occur. So protein scientists have to solve the problem how to efficiently manage the huge amount of data of protein. Present databases can not meet this need since they neither match the structure of protein data nor have built-in data types and functions. In this application paper, we provide a system with built-in data types and built-in domain specific functional operators that based on Protein-OODB system to solve this problem. This new system has three components: a client API, a Middleware (including a RMI server, a protein domain specific language Protein-QL [10], Protein Algebra Architecture and a Protein-OODB), and multiple underlying databases. In order to use this system, users only need to master Protein-QL syntax which provides convenience for protein scientists to store, retrieve, and modify data, and defines some basic operations. Therefore, domain scientists can easily formulate complex requests for data without much learning. Overall, this application system is for a certain biological domain [5], but it can be easily extended into other biological domains to build a general Bio-OODBMS for biological applications.

The rest of the paper is organized as follows: Section II describes the architecture of database management system for protein structure data. Section III shows how to use this system. Related Work will be discussed in Section IV. Section V will show how to translate original query to specific query for each base database. Conclusion and future work are presented in Section VI.

II. The Architecture of Database Management System for Protein Structure Data

The architecture of database management system for protein structure data called Protein-DBMS is a three-layer architecture that consists of the following components: a client API, Middleware (including a RMI server, a query language for protein structures (Protein-QL), Protein Algebra Architecture and an object-oriented database for protein data (Protein-OODB)), and multiple underlying databases. Fig. 1 illustrates this architecture.

The clients can use this system to send domain specific requests and manage the protein data. The Client writes simple domain specific queries according to Protein-QL and sends them to the Server. The Server receives the queries and communicates with Protein-QL using RMI and checks the grammar of queries according to the syntax of Protein-QL. Then the interpreter between Protein-QL and Protein-OODB converts queries into specific database queries. Finally database sends the results back to the server. This system provides clients convenient access and is easily mastered. In addition, queries can be optimized by using Protein Algebra Architecture.

III. Protein-DBMS Applications

The detailed architecture of this application system is shown in Fig. 2.

- User API such as Visualization, Data Browser, Java Client, PQL Plus and PDB Expert provides very convenient service for different users.
- Clients can use our protein domain specific query language (Protein-QL) to request results from different databases that depend on the server provider.
- Handler is in charge of distributing queries to specific database server according to the client's request.
- Database ID is sent to Handler of Protein-OODB by Server Listener after client chooses database from which data results are generated. Then Handler decides to send queries to the right database according to Database ID.

The following will illustrate some examples to show how to use this system.

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 526–531.

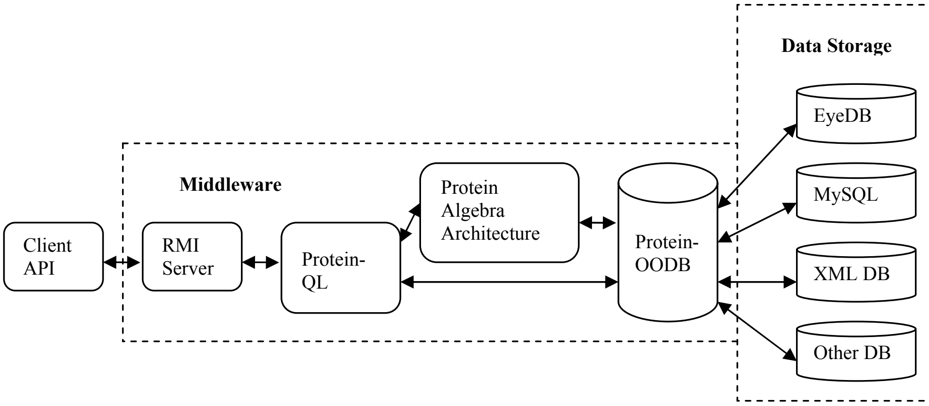

Fig. 1. The architecture of database management system for protein structure data

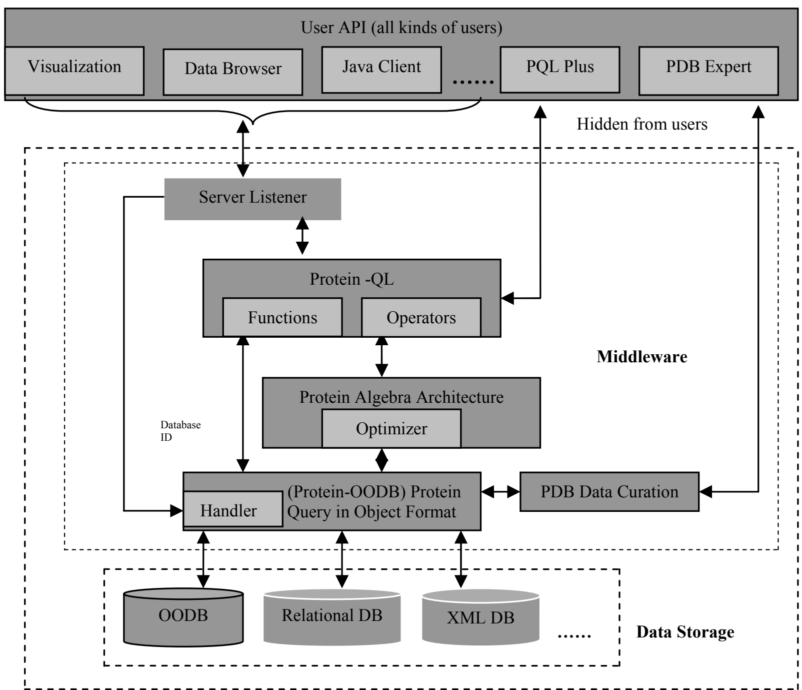

Fig. 2. The detailed architecture of database management system for protein structure data

A. Application by using Object-Oriented (EyeDB) as underlying database

Example 1: Gets the protein named "HIV-1" (Abbreviation "HIV-1 Protease").

```
(Protein)(Protein.proteinName = "HIV-1");
```

Then the system generates object query in EyeDB [1]:

```
select p from p in Protein where p.proteinName="HIV-1";
```

The result is shown in Fig. 3.

The PDB format result may be obtained using the service `PDBFormat(proteinName)`.

Example 2: Gets the secondary structure of protein named "HIV-1".

```
(Protein.secondary)(Protein.proteinName=
"HIV-1");
```

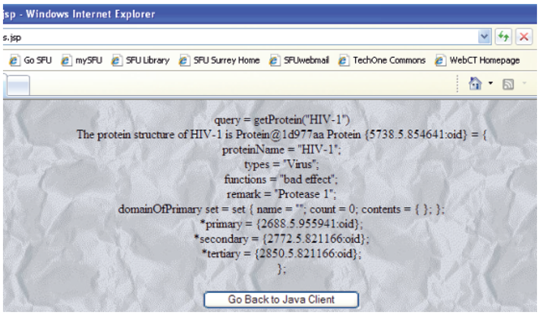

Fig. 3. The result for example 1

Our system will generate object query in EyeDB:

```
select p.secondary from p in Protein where
p.proteinName = "HIV-1";
```

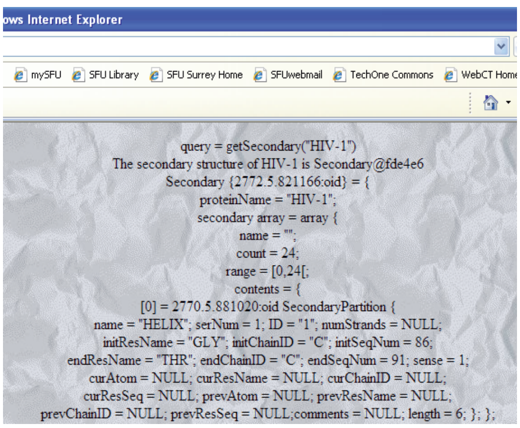

Fig. 4. The result for example 2

The result in PDB format using `PDBSecondary` service is shown in Fig. 5.

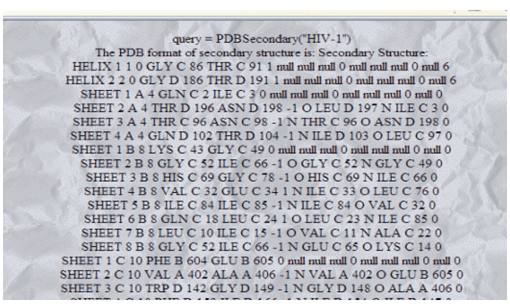
query = PDBSecondary("HIV-1")
The PDB format of secondary structure is: Secondary Structure:
HELIX 1 1 0 GLY C 86 THR C 91 1 null null null 0 null null null 0 null 6
HELIX 2 2 0 GLY D 186 THR D 191 1 null null null 0 null null null 0 null 6
SHEET 1 A 4 GLN C 2 ILE C 3 0 null null null 0 null null null 0 null 0
SHEET 2 A 4 THR D 196 ASN D 198 -1 O LEU D 197 N ILE C 3 0
SHEET 3 A 4 THR C 96 ASN C 98 -1 N THR C 96 O ASN D 198 0
SHEET 4 A 4 GLN D 102 THR D 104 -1 N ILE D 103 O LEU C 97 0
SHEET 1 B 8 LYS C 43 GLY C 49 0 null null null 0 null null null 0 null 0
SHEET 2 B 8 GLY C 52 ILE C 66 -1 O GLY C 52 N GLY C 49 0
SHEET 3 B 8 HIS C 69 GLY C 78 -1 O HIS C 69 N ILE C 66 0
SHEET 4 B 8 VAL C 32 GLU C 34 1 N ILE C 33 O LEU C 76 0
SHEET 5 B 8 ILE C 84 ILE C 85 -1 N ILE C 84 O VAL C 32 0
SHEET 6 B 8 GLN C 18 LEU C 24 1 O LEU C 23 N ILE C 85 0
SHEET 7 B 8 LEU C 10 ILE C 15 -1 O VAL C 11 N ALA C 22 0
SHEET 8 B 8 GLY C 52 ILE C 66 -1 N GLU C 65 O LYS C 14 0
SHEET 1 C 10 PHE B 604 GLU B 605 0 null null null 0 null null null 0 null 0
SHEET 2 C 10 VAL A 402 ALA A 406 -1 N VAL A 402 O GLU B 605 0
SHEET 3 C 10 TRP D 142 GLY D 149 -1 N GLY D 148 O ALA A 406 0

Fig. 5: The result for example 2 in PDB format

Example 3: delete a protein object whose name is "HIV-1"

```
delete (Protein)(Protein.proteinName =
"HIV-1");
```

Then the interpreter generates object query in EyeDB:

```
delete p from p in Protein where
p.proteinName="HIV-1";
```

The result is: `= 5738.5.854641:oid`.

B. Application by using Relational Database (Oracle) as underlying database

Example 4: Gets the protein named "HIV-1".

```
(Protein)(Protein.proteinName = "HIV-1");
```

The result is:

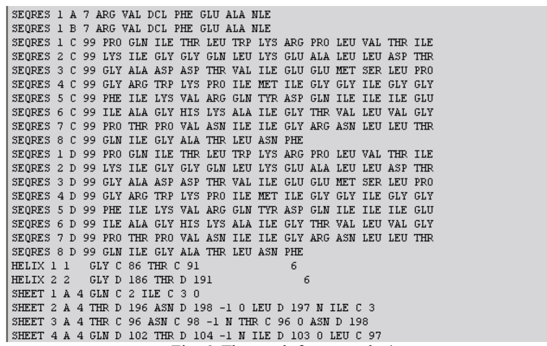
```
SEQRES 1 A 7 ARG VAL DCL PHE GLU ALA NLE
SEQRES 1 B 7 ARG VAL DCL PHE GLU ALA NLE
SEQRES 1 C 99 PRO GLN ILE THR LEU TRP LYS ARG PRO LEU VAL THR ILE
SEQRES 2 C 99 LYS ILE GLY GLY GLN LEU LYS GLU ALA LEU LEU ASP THR
SEQRES 3 C 99 GLY ALA ASP ASP THR VAL ILE GLU GLU MET SER LEU PRO
SEQRES 4 C 99 GLY ARG TRP LYS PRO ILE MET ILE GLY GLY ILE GLY GLY
SEQRES 5 C 99 PHE ILE LYS VAL ARG GLN TYR ASP GLN ILE ILE ILE GLU
SEQRES 6 C 99 ILE ALA GLY HIS LYS ALA ILE GLY THR VAL LEU VAL GLY
SEQRES 7 C 99 PRO THR PRO VAL ASN ILE ILE GLY ARG ASN LEU LEU THR
SEQRES 8 C 99 GLN ILE GLY ALA THR LEU ASN PHE
SEQRES 1 D 99 PRO GLN ILE THR LEU TRP LYS ARG PRO LEU VAL THR ILE
SEQRES 2 D 99 LYS ILE GLY GLY GLN LEU LYS GLU ALA LEU LEU ASP THR
SEQRES 3 D 99 GLY ALA ASP ASP THR VAL ILE GLU GLU MET SER LEU PRO
SEQRES 4 D 99 GLY ARG TRP LYS PRO ILE MET ILE GLY GLY ILE GLY GLY
SEQRES 5 D 99 PHE ILE LYS VAL ARG GLN TYR ASP GLN ILE ILE ILE GLU
SEQRES 6 D 99 ILE ALA GLY HIS LYS ALA ILE GLY THR VAL LEU VAL GLY
SEQRES 7 D 99 PRO THR PRO VAL ASN ILE ILE GLY ARG ASN LEU LEU THR
SEQRES 8 D 99 GLN ILE GLY ALA THR LEU ASN PHE
HELIX 1 1   GLY C 86 THR C 91               6
HELIX 2 2   GLY D 186 THR D 191               6
SHEET 1 A 4 GLN C 2 ILE C 3 0
SHEET 2 A 4 THR D 196 ASN D 198 -1 0 LEU D 197 N ILE C 3
SHEET 3 A 4 THR C 96 ASN C 98 -1 N THR C 96 0 ASN D 198
SHEET 4 A 4 GLN D 102 THR D 104 -1 N ILE D 103 0 LEU C 97
```

Fig. 6. The result for example 4

Example 5: Gets the secondary structure of protein named "HIV-1".

```
(Protein.secondary)(Protein.proteinName=
"HIV-1");
```

Our system will generate Protein-OODB query:

```
select p.secondary from p in Protein
where p.proteinName = "HIV-1";
```

It will generate query in MySQL:

```
select name, serNum,ID, numStrands,
initResName, initChainID, initSeqNum,
endResName, endChainID, endSeqNum, sense,
curAtom,curResName,curChainID, curResSeq,
prevAtom, prevResName, prevChainID,
prevResSeq, comments, length from
SecondaryPartition where
proteinName="HIV-1" order by sorder;
```

The result is shown in the following Fig. 7.

Example 6: delete a protein object whose name is "HIV-1"

```
delete (Protein)(Protein.proteinName =
"HIV-1");
```

Then the system generates object query in EyeDB:

```
delete p from p in Protein where
p.proteinName="HIV-1";
```

It will generate query in MySQL:

```
delete from Protein, PrimaryPartition,
SecondaryPartition, TertiaryPartition
where proteinName = "HIV-1";
```

Fig. 7. The result for example 5

Example 7: gets the sequence of protein named "HIV-1".

```
sequence("HIV-1");
```

The operators such as `sequence`, `noOfHelix`, `nearestNeighbor3D` are built in the query language so that the system will get the result directly from the database and does not need to translate them into specific queries.

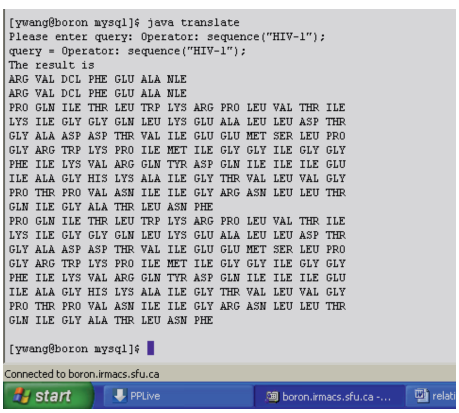

Fig. 8. The result for example 7

Example 8: gets the number of helix of protein named "HIV-1".

```
noOfHelix("HIV-1");
```

The Fig. 9 shows this result

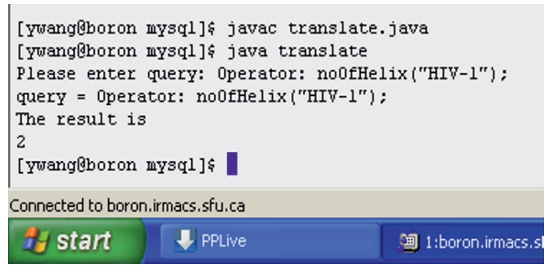

Fig. 9. The result for example 8

Example 9. gets the nearest neighbor of protein named "HIV-1".

```
nearestNeighbor3D("HIV-1");
```

The result is shown in the following snapshot:

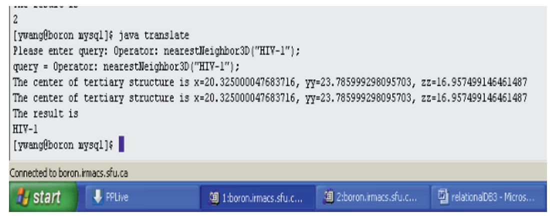

Fig. 10. The result for example 9

C. Application by using XML Database (Oracle) as underlying database

In order to make our work be applicable in other database systems, we only need to code one adapter if we want to connect our domain specific Protein-QL to other database. We can convert Protein-OODB queries to XML database such as Orcale 10g which can avoid translating Protein-QL to MySQL by making use of Protein-OODB. In this case, we need to convert Objects of EyeDB into String of object content in order to match data.

Example 10: Gets the protein named "HIV-1".

```
(Protein)(Protein.proteinName = "HIV-1");
```

Then the system generates Protein-OODB query:

```
select p from p in Protein where
p.proteinName="HIV-1";
```

Generated XPath by the interpreter is not shown here. The result is the same as Fig. 6.

Example 11: Gets the secondary structure of protein named "HIV-1".

```
(Protein.secondary)(Protein.proteinName=
"HIV-1");
```

Our system will generate Protein-OODB query:

```
select p.secondary from p in Protein
where p.proteinName = "HIV-1";
```

The result is the same as Fig. 7.

Example 12: delete a protein object whose name is “HIV-1”

```
delete (Protein)(Protein.proteinName =
"HIV-1");
```

Then the system generates object query in EyeDB:

```
delete p from p in Protein where
p.proteinName="HIV-1";
```

It will generate query in XPath:

```
delete from ProteinXML where
extractValue(OBJECT_VALUE,'/proteinName/t
ext()') like 'HIV-1';
delete from ProteinXML where
extractValue(OBJECT_VALUE,
'/PrimaryPartition/proteinName/text()')
like 'HIV-1';
delete from ProteinXML where
extractValue(OBJECT_VALUE,
'/SecondaryPartition/proteinName/text()')
like 'HIV-1';
delete from ProteinXML where
extractValue(OBJECT_VALUE,
'/TertiaryPartition/proteinName/text()')
like 'HIV-1';
```

Example 13: gets the sequence of protein named “HIV-1”.

```
sequence("HIV-1");
```

It will give the same result as Fig 8.

IV. Translation From Protein-QL To Other Query Languages

Fig. 2 shows the system how to recognize the destination database by using the Database ID sending from Server Listener. When the system gets Database ID, it will trigger the certain interpreter and go through certain database to get the result. For example, if users want to query data from the relational database, they only need to send Database ID--2 and Protein-QL query, and the system will carry result back to the users.

In order to add other databases by using different query language (QL) into this system, only certain interpreter according to QL is needed to add into our system that will make extension easy and save time.

Our Protein-QL can handle with all queries since users can access (or connect) all data to the queries from the Protein Extended ER (Fig. 11). We give an example here in Fig. 12 to get proteins that have the same subsequence `"ARG VAL DCL PHE"` as protein `"HIV-1"`. Here we can use function `location(proteinName, "ARG VAL DCL PHE")`. If the location returned is greater than zero, then this protein named “`proteinName`” has the same subsequence “`ARG VAL DCL PHE`” as protein “`HIV-1`”. The query of Protein-QL can be finished through the path of protein—primary-location.

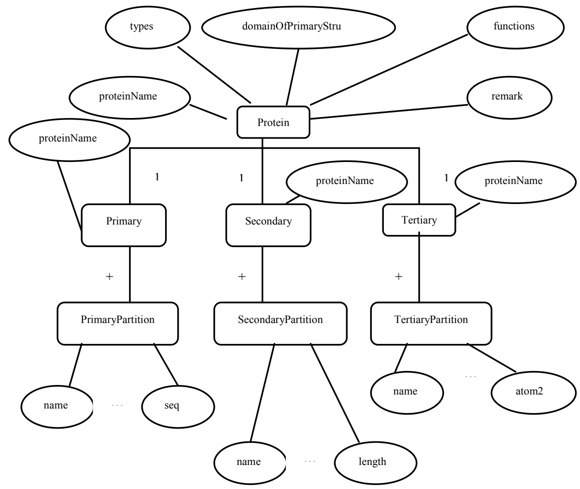

Fig. 11. Protein Extended ER

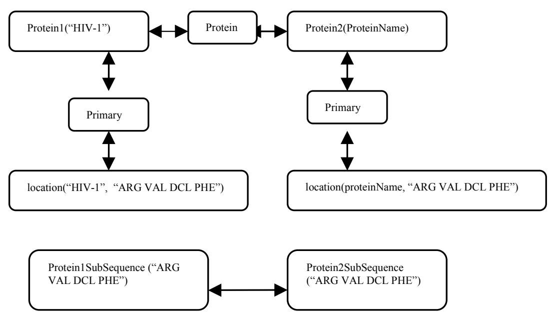

Fig. 12. The execution of Protein-QL

From above discussion, we know our Protein-QL is complete and can handle with all queries. And Protein-QL is translated into Protein-OODB exactly according to Protein-QL syntax, also our interpreters translate all Protein-OODB query formats into certain database query language, therefore our query language is sufficiently generic to interact with a wide variety of databases.

V. Related Work

A. BioVelo

As we know, users can use the BioVelo query language to write “*complex queries against Pathway/Genome Databases created using the Pathway Tools Software*” [11]. Users can extract lists of objects by writing queries in BioVelo according to specific conditions. This query language is based on a computer science concept called “*list comprehension*” [8] which needs bio-users to understand this computerized concept. Our query language is based on Objected-Oriented

DataBase with built-in operators and functions, users can easily understand and master it since those operators and functions are domain specific.

B. NeuroDM

In [12], authors proposed a novel neuron data model called NeuroDM with a neuron domain specific query language (NeuroQL). This NeuroDM defines a list of queries in neuroscience terms so that users can easily mater it. And NeuroQL consists of a list of queries that defined in neuroscience terms. NeuroQL enables neuroscientists to query the database using their own language without much syntactical restriction. We did similar work, but our application is for protein domain and we used OODB as our middleware part that reduced interpretation work and interpretation time among different language translation.

VI. CONCLUSION AND FUTURE WORK

Our application system provides wide services for different kinds of users and is independent of underlying databases, therefore users can request protein data in Object Oriented format, but data can be stored in multiple formats in OODB, relational DB, XML DB or other data storages. Presently, our system is implemented in protein domain, but it can be easily extended into other biological domains.

In the future, we plan to extend our system to accept XML input and output. In addition, we will make the system support a mapping that links terms used in the databases to their semantic meaning such that terms appearing in different databases, but representing the same data, can be treated as equals from the point of user API. The system uses ontology in protein domain knowledge to provide the necessary database semantic. The aim of the system is to create domain-specific database management that is user-oriented and independent of underlying databases and Protein-QL supports domain-specific objects, operators, functions, and ontology of the system is able to capture domain semantics to solve ambiguity and heterogeneity of data. We plan to make our new system be able to be easily applied to any other biological domains and users also can integrate this system to other domains shown in the following Fig. 13, we called that as Bio-DBMS. Presently, most protein data are stored in PDB format, so we plan to add PDB data curation system [7] into our current system so that the system can get much cleaner and more confident PDB data when clients use them.

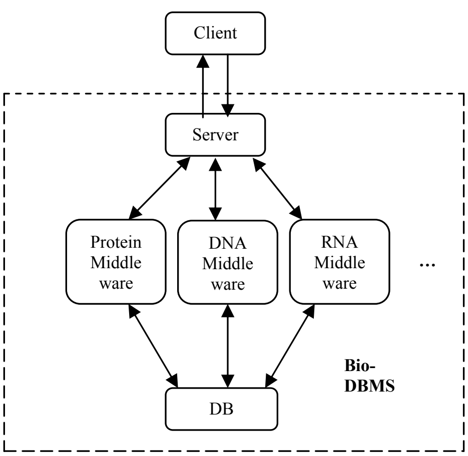

Fig. 13. The architecture of Bio-DBMS

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Ubiquitous-Healthcare based on Medical Device Communication Framework and PLC

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***Abstract*-A ubiquitous healthcare service based on a medical device communication framework ISO11073 and PLC (Power Line Communication) was studied, considering the ubiquitous computing and networking environment. We introduce the USN for e-healthcare service in smart environments. Beyond e-healthcare service, as an application for u(ubiquitous)-healthcare service with USN including PLC technology, we implemented the real-time health-monitoring service with a cost-effective Web server. We introduce the u-healthcare service based on the framework ISO11073/IEEE1073 (i.e. X73) with PLC technology for sink node or gateway without power limitation in USN environments.**

***Index Terms*—communication framework, ISO11073, PLC, QoS, u-healthcare, USN, X73**

I. INTRODUCTION

In the sensor network, especially in wireless sensor networks (WSN), the research mainly focused on routing, aggregation, and energy efficient data management algorithms inside one sensor network. The deployment, application development, and standardization aspects were usually not addressed [1].

We focused on the study of an application service, i.e. u-Healthcare service, among various services based on sensor networks composed of wired sensors or wireless sensors in USN. We defined the USN as a sensor network composed of ubiquitously prevalent wired-sensors as well as ubiquitously prevalent wireless-sensors, which were connected in the network.

Our main interest was to find a killer application based on PLC (Power Line Communication) for wired sink node or gateway, among various communication technologies related to sensor networks. Beyond e-healthcare service, the ubiquitous healthcare (u-healthcare) service will be more evolved in the emerging ubiquitous computing and networking society.

Let's discuss some researches related to smart environments and e-healthcare, because the wired-sensor networks are necessary in the smart environmets. 'Smart environments for all' was introduced in the special thematic session by Nussbaum [2], and he stressed that smart environments would have a lot of potential to increase the quality as well as the efficiency of healthcare. Development of smart home technologies dedicated to people with disabilities provided a challenge in determining accurate requirements and needs in dynamic situations; and Feki et al. [3] introduced the integration of context awareness and multimodal functionalities in the smart environments. Information such as the availability of resources, user profiles, location, input controls and services can be used to improve the interaction between users and their environments.

In the emerging USN, sensor devices could be severely miniaturized, harvest energy from the environment and communicate with other networks and devices integrated in our homes or our cities as smart environments. Recently, the new application of health monitoring and medical care based on USN is gaining popularity amongst researchers and gives good promises for future practical uses. We will introduce a framework of USN for u-healthcare service in smart environments.

The following sections are organized as follows. We will discuss the background of medical device communication framework and USN for ubiquitous healthcare service in the evolving computing and networking environment. We will present the X73 framework for healthcare service.

Beyond e-healthcare service, as a primitive application for ubiquitous healthcare service, we introduce our experience of the real-time health monitoring service using health-monitoring sensors and health information web server in the mobile Internet before the proliferation of ubiquitous computing and networking environment. For quality of service (QoS), we introduce an evaluation scheme for u-healthcare service in smart environments for point-of-care with PLC technology in sink node or gateway. Finally we conclude our study with consideration for further research.

II. BACKGROUND

The need for health monitoring in e-healthcare service is gradually increasing and its application is becoming feasible

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering,* 532–537.

with sensor network technology. By adopting tiny wireless sensor network devices in the ubiquitous networking environment with some specific health monitoring system, regular patients and elderly people can be observed at smart environments providing e-healthcare service. For this purpose, wearable vital sign sensors can be attached to people's body, allowing continuous communication transferring people's sensed physical status. Health monitoring with ubiquitous sensor networks (USN) in the ubiquitous networking environment has some outstanding features comparing with traditional medical healthcare systems.

Health monitoring based on USN in the ubiquitous networking environment provides a totally different healthcare system scenario. Wireless sensor nodes are initially small and generally use battery rather than power cables with power limitation problem. Sensor network's general features such as tiny sensor nodes, network construction and self-configuration make it possible to be used in medical care monitoring application, even with power limitation problem.

In the USN, we need to consider the advantage, the cost-effectiveness, and the security issues of the both (i.e. wired and wireless) sensor nodes/networks in the ubiquitous healthcare service, instead of insisting the use of wireless sensor/networks. We should consider the wired sensor/network based on PLC technologies, especially for the point-of-care medical service based on ISO11073/IEEE1073 in the USN environments.

The standard for point-of-care connectivity [4] establishes a set of specifications to allow seamless multi-vendor interoperability and communication between point-of-care devices, data concentrators, and clinical information systems. The document on point-of-care connectivity has been developed by the CLSI (Clinical and Laboratory Standards Institute) Subcommittee on point-of-care connectivity. The core of the standard is a group of three specifications developed by the Connectivity Industry Consortium (CIC). The specifications describe the attributes of an access point; the communication protocols between the device and the access point; and communications between a data manager and clinical information systems. The collaborative effort among providers and manufacturers has produced a set of specifications acceptable to both.

The most influential standards in the vendor community derive from bodies which have achieved an international influence and authority outside the formal International/European standards bodies, e.g. the USA based HL7 for messaging, DICOM for imaging, IEEE for medical device communications [5]. We introduce briefly as follows.

HL7, Health Level Seven by reference to the 7th layer of the OSI model, has been founded in 1987 by several vendors of software for the health care industry [6], and funded by American manufacturers of medical equipment and accredited by the American National Standards Institute (ANSI). It is a standard for the exchange of medical messages. Their goal was to develop messages consensual formats to facilitate a better interoperability of Hospital Information Systems (HIS). It develops its own syntax, in the seven levels of the protocol stack, for representing the information in a simple structure composed of segments and field labels (each one identified by its data type). Like DICOM, it exchanges the results of observations related to vital signs and biomedical signals, but it is not applicable to the interconnection of devices.

DICOM (Digital Imaging Communication) is a standards organization creating, and maintaining standards for communication of biomedical diagnostic and therapeutic information in disciplines using digital images and associated data. It was formed by the American College of Radiologists (ACR) and the National Electrical Manufacturers Association (NEMA). It occupies a privileged position in medical imaging since it is very widespread among the healthcare community and the manufacturers. It includes some directives for the exchange of biomedical signals, particularly ECG, but it is not applicable to the interconnection of monitoring devices.

ISO11073/IEEE1073 (known as X73), Standard for Medical Device Communications: a family of documents that defines the entire seven layer communications requirements for the 'Medical Information Bus' (MIB). This is a robust, reliable communication service designed for intensive care unit, operating room, and emergency room bedside devices. The X73 goal is to improve the interoperability and plug-and-play capacities of the different medical devices and medical information systems.

III. FRAMEWORK FOR U-HEALTHCARE

The following diagram [7] in Fig. 1 shows the upper layer communication stack in X73, or layered set of protocol and service components.

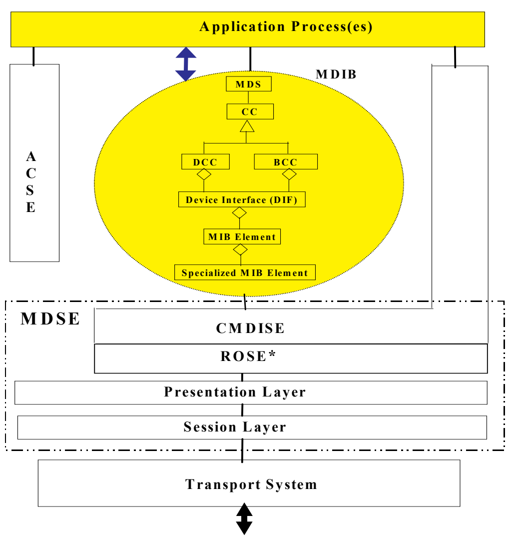

Fig. 1. Medical Device Communication Stack

ACSE (Association Control Service Element) is the

ISO/OSI standard for association control. CMDISE (Common Medical Device Information Service Element) is the object management service, in principle, a lightweight version of the ISO/OSI CMISE (Common Management Information Service Element).

The Remote Operation Service Element, ROSE, provides basic services used by the CMDISE (invoke an Operation, return the result of an operation, return an error, reject an operation). To comply with the definition of optimized encoding rules, a modified version of the ROSE is needed to work with the CMDISE.

In Fig.1, ACSE means Association Control Service Element, MDIB means Medical Data Information Base, and MDSE means Medical Device Service Element. Session Layer and Presentation Layer produce a minimized overhead only.

The ISO11073/IEEE1073 (known as X73) set of standards for Point of Care Medical Device Communication is the best-positioned international standard to provide interoperability with different sensors [6]. The X73 standard, currently in the development phase, is a single set of standards for complete connectivity among medical devices that contribute plug-and-play, transparency, and ease of use and configuration.

X73 extends to the seven levels of the Open System Interface (OSI) protocol stack and absorbs EN 13734 (VITAL) for the upper layers, EN 13735 (INTERMED) for the intermediate layers, and the older 1073 standards (1073.3 and 1073.4) for the lower layers. Thus, it gives a full solution from the cable itself and connector, to abstract representation of information and services.

X73 distinguishes four main standards groups: transport (e.g., wireless or cabled), services of general application (e.g., for events or polling), device data (e.g., object-oriented model and representation terminology), and network communication standards (e.g., gateway between the representation of data and IEEE1073, DICOM, or HL7 messages).

Toledo et al. [6] found a problem with the lack of X73 standardization of wireless connectivity, and they plan to fix it when required by using the lower layers of the wireless technology (Bluetooth, Zigbee, etc), then merging them as possible with the upper layers, which really follow the standard, waiting for the X73 to get updated to these technologies.

In the new and exciting field of Ubiquitous Sensor Networks (USN) we introduced, e-healthcare services for medical care applications appear as a promising research area with a wide range of possibilities. Beyond e-healthcare service, we introduce ubiquitous healthcare service. We introduce the health monitoring sensor network based on mobile Internet [9] as a primitive application for ubiquitous healthcare service.

The following use cases [10] reflect diversity, and there is throughout an assumption of use of existing or supplemented 11073 (X73) standard provisions.

UseCase#1: Mobile Wellness Monitoring of a Single Cardiac Patient;
UseCase#2: Wellness Weight Monitoring of a Single Patient Home;
UseCase#3: Chronic Respiratory Patient Management;
UseCase#4: Elderly patient follow-up;
UseCase#5: Cardiac Parameter Monitoring of a Single Patient Home;
UseCase#6: Secure Registration of Tele-monitoring Device;
UseCase#7: Body Area Network Monitoring;
UseCase#8: Critical cardiac alarms.

For elderly patient who lives alone or spends alone most of the day, the UseCase#4 was considered, and this case may be appropriate for the application of PLC based on wired sensor/network. Monitoring service controls his vital signs once a day and generates a summary of his/her level of activity. Vital signs are recorded using a pressure cuff (blood pressure and heart rate) and a digital thermometer. A wearable 3D accelerometer records patient movements during all the day and generates parameters that summarize patient activity (% time sitting, % time standing, % time laying, % time sleeping). All this information is sent to a gateway (embedded PC located at the home).

For u-healthcare with USN, we should consider both cases, which are the wired sensor/network and the wireless sensor/network. Considering the WSN, we should refer to the aforementioned UseCase#1, #3 and #7 based on X73.

We studied the real-time health-monitoring application, getting frequently personal information as well as writing analyzed information as a response to the request by clients or health-monitoring agents in the mobile Internet environment.

The service process of the u-healthcare system using portable device is as follows: Measurement of Biological Signal/ Analysis and Display of the Signal/ Data transfer and recording/ Diagnosis of transferred data. Looking at each phase of these four-step process, the measurement phase requires a composite function, network, non-expensive, light, unconfined and continuous measurement in order to acquire a great deal of various health indices.

The monitoring process requires module information from PDA (Portable Digital Assistant), laptop PC, cellular phone and other devices, and then the measured data shall be analyzed and transferred to applicable medical organization. When the data is transferred through a wireless network environment, the transferred data shall have interface with hospitals and/or other medical organization in order to prove the reliability of the analyzed data.

In order to minimize the size of the device design with low power, the analog circuitry must be reformatted as the digital circuitry as much as possible to be controlled by the program.

In general, the analog circuitry is used as signal amplifier and filters that eliminates 60Hz noise due to power line artifact and the ambient signal created by ambient light. Regarding to the elimination of ambient light, band rejection filter is normally used for ECG and de-multiplexing technique by means of multi-wavelength for SpO_2.

A USN platform for e-healthcare service, an evolvable network of tiny sensors, constitutes the particular USN design for smart environments of healthcare service to address the challenges imposed by a dynamic future based on personal mobility. Contrary to other architectures which rely on static behaviors, ANTS (**a**n evolvable **n**etwork of **t**iny **s**ensors developed by ICU University) [8] was built with the idea of providing adaptability for dynamically changing environment, particularly coupling with the needs of the inherently dynamic healthcare monitoring systems with advanced features in medical care environments.

Fig. 2. ANTS USN Platform in RFID/USN Exhibition

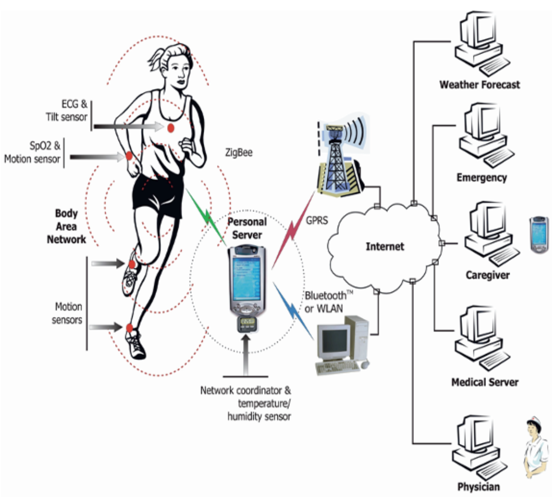

Fig. 3. Wireless Body Area Network for Patient Monitoring

ANTS system was designed to include wireless sensor networks at home environment to collect ECG data from the patient. In order to convey the data from wireless sensor networks at home to the Internet and to analyze and process the data, the project used wired and wireless networks including Internet. As depicted in Fig. 3, ANTS architecture builds all the necessary modules to provide a complete and effective system.

ANTS architecture includes, besides software considerations, its own hardware support based on four different hardware designs suitable for different sensor network requirements.

We need to consider the previous UseCase#7 based on X73 in the further research. For wireless communication to transmit personal health information, the design includes the 802.15.4 standard for radio communication, used in the Zigbee specification and offering promising reductions in power consumption. Sensors such as ECG, heartbeat, light, accelerometer and magnetometer are integrated into Sensor board. All these sensors can be operated individually or in collaboration with other sensors. If we want to use external health sensor or transmit collected health information to local computer system connected to Internet, we can use Interface board. It supports Serial/Parallel communication to communicate with external sensors or computers connected to Internet; we study the PLC adaptor for sensor node or gateway.

IV. EXPERIENCE FOR QoS

For the real-time health-monitoring network using mobile Internet in the USN environment, the dominating factor and the standard deviation of that random variable should be bounded within the deterministic response time. To be deterministic for real-time application, the estimation time should be bounded within deterministic time, the interchange of data between the sensor nodes in watch phone and the server should be automatic except the requested information by the user; therefore the Web server should be efficient and have high performance for the dedicated application if possible, and the exchanged data and analyzed information should be as simple as possible with a simplified and efficient format.

If possible, the bandwidth requirement for wireless or mobile Internet should be immune to network traffic conditions; also that will be ideal in case of the degradation caused by the other rich multimedia contents sharing the same network and server. The spent time in the Web server may be considered to be immune to the network and server condition. This system is based on wired or mobile Internet, and the monitored health data from sensors in a watch phone can be registered at any time using mobile Internet in the USN environment.

For the consistency of health-monitoring information and for a convenient user interface, we need a unified health-monitoring Web server for wired Internet and mobile Internet. We need to consider the health-information center accessibility to the doctor or nurse as well as to the disabled and elderly with the concept of X73 framework.

We used a single Web server as a health information Web server for cost-effectiveness and the simplicity of management. This method offers effectiveness and efficiency for the real-time health-monitoring network and utilization of resources, e.g. the bandwidth for communication and the size of disk storage of health information Web server for the patient. We also consider a reliability scheme.

We assume that the wrist phone with health-monitoring sensors wrote the health information regularly, which should be chosen carefully in further research with X73 framework, to the health information Web server through the mobile Internet, and processed the data and analyzes for the request of the results in real-time way. Depending upon the frequency of writing the health information, the workload of the health Web server may change, and the interval of regular writing may be considered as an arrival rate in the queuing performance analysis model for health-monitoring.

The transmission packet unit for billing by mobile communication service provider is about 0.5 Cents (U.S.); the packet size is 512 Bytes, which is the minimum packet size for billing in Korea. Therefore, if possible, the health-monitoring data for the wrist phone to the server through mobile Internet should be below 512 Bytes, and this size is also bounded much below 1.5Kbytes that is one emulated WML deck for performance evaluation of the health-monitoring Web server discussed later. As a reference, one SMS (Short Message Service) data, i.e. 80 Bytes, costs around 2.5 Cents (U.S.) in Korea. Therefore, we can see that one WML packet is much cheaper than on SMS message in terms of cost for wireless communication as a scheme in USN environment.

The health Web server should have the capability to show the appropriate health contents, i.e. the HTML contents for wired Internet as well as the mobile contents for many different kinds of mobile devices, e.g. WML, mHTML, etc. For the unified service, there are several constraints, compared to the contents for the wired Internet.

First of all, we should consider the various kinds of mobile devices as well as browsers for the mobile Internet, and each of those devices may have different capabilities in terms of images. We considered only text-based health-monitoring information from the wrist phone to the health-monitoring Web server and vice versa, to be immune to any type of Internet traffic load as well as to minimize the mobile communication cost for cost-effective health-monitoring services based on X73 framework with PLC technology.

Let's discuss more about u-Healthcare service considering important issues such as data quality of sensor nodes, real-time aggregation and estimation of random variables for comprehensive accessibility. We study to implement health-monitoring sensors in the wrist phone on the hand of the disabled and elderly people as a primitive example with single-hop sensor nodes; that single-hop sensor node even with PLC sink node/gateway is a starting point to get insights about important issues in terms of data analysis with random variables.

The wrist phone as a sensor node is a future product not implemented yet, thus we assume that the sensors for measuring the pulses, the strength of the pulses, and blood pressure, etc. would give the raw data about health of the disabled and elderly. Instead of the wrist phone, we used a commercial product which had the model name: WebDoc MX431 as shown in Fig. 4; and that model could be attached to the PC or PDA. We could measure the number of pulses per minute, the blood pressure, and the body fat.

We studied the real-time health-monitoring application, getting frequently personal information as well as writing analyzed information as a response to the request by clients or health-monitoring agents in the USN environment. We will discuss various aspects for QoS of a ubiquitous application service.

Fig. 4. Test Equipments (WebDoc MX431, PC, PDA, PLC Modems)

For the real-time health-monitoring network using mobile Internet in the USN environment, the dominating factor and the standard deviation of that random variable should be bounded within the deterministic response time. To be deterministic for real-time application, the estimation time should be bounded within deterministic time, the interchange of data between the watch phone and the server should be automatic except the requested information by the user.

Therefore the Web server should be efficient and have high performance for the dedicated application if possible, and the exchanged data and analyzed information should be as simple as possible with a simplified and efficient format. If possible, the bandwidth requirement for wireless or mobile Internet in the USN environment should be immune to network traffic conditions. That will also be ideal in terms of the degradation caused by the other rich multimedia contents sharing the same network and server.

This system is based on wired or mobile Internet, and the monitored health data from sensors in a watch phone can be registered at any time using mobile Internet with the domain name of test Web server for wired/mobile Internet in the USN environment. Actually this site has been used as a real-time information network server and we considered it as a health-monitoring server because it can be used as any Web server for testing.

We studied the important QoS performance metric, delay, from the user's perspectives for the disabled, considering the application of pervasive (or ubiquitous) computing technologies in healthcare. The preparation time for the disabled to get a service (e.g. medical device, mobile device, etc.) is *U* (ubiquitous environment time metric); the time spent by the disabled with medical device to do appropriate action for service is *D* (device time metric); the aggregate time to the medical healthcare server after the medical device for medical service is *S* (service time metric for communication and Web server); the time depending upon medical contents is *C* (contents time metric). If iteration of each variable is required, then subscript *i, j, k, l, m* can be used as U_i, D_j, S_k, C_l, P_m. Depending upon ubiquitous environment, device, service, and patient the iteration *i, j, k, l, m* may be different.

Among the above random variables, i.e. the performance metrics, (*U, D, S, C, P*) for the disabled or patient, the most dominating factor, i.e. the random variable, may be different depending upon u-healthcare environment, device, service, contents, and patient. We can represent the statistical values (i.e. mean, deviation, etc.) of random variables as performance metrics for QoS (quality of service) of real-time medical service, and the quality of u-healthcare service can be compared with the QoS performance metrics as comprehensive accessibility. Each performance metric can be used to enhance the QoS as follows. Ubiquitous environment time metric *U* will be shortened depending upon the proliferation of smart environments including PLC for u-healthcare. Device time metric *D,* i.e. handling speed of medical device, is one of important performance factors in any medical services for the disabled or patient. Service time metric *S* can be shortened according to the communication facility and Web server with DB in e-hospital. Contents time metric *C* can be shortened considering accessibility in contents design for u-healthcare service. Patient time metric *P* may be different from person to person depending upon the disability of the disabled.

We can order the dominating factors in the overall performance at the user's perspective. The user interface design for the disabled with a medical device in a ubiquitous healthcare environment with PLC technology (i.e. *U* becomes smaller as the ubiquity increases) is important to decrease the device time metric *D* that is heavily related to the UI (user interface) convenience of medical device for the disabled and handicapped users.

With more proliferation of ubiquitous network environment including PLC without power limitation, the time spent with ubiquitous device with sensor nodes will become more critical. We should consider this QoS concept in the implementation of X73 framework for u-healthcare service with USN including PLC technology.

V. CONCLUSIONS

A medical communication framework for ubiquitous healthcare service based on X73 with USN was introduced. We focused on u-Healthcare service based on wired sink nodes/gateways including PLC technology beyond WSN, and we showed comprehensive accessibility for QoS in the USN environments.

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A Performance Analysis of Integrated Social-Sensor Networking Service with USN

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***Abstract*-For performance analysis of integrated social-sensor networking service, a unified approach in a unified web service was studied. As a convenient and usable mobile web service for unified social-sensor networking service, the multi-lingual single-character domain names as indexing keys to metadata information in ubiquitous web service are convenient mobile interfaces when searching for social-sensor information and registering metadata information. We introduce the sketched design goals and experience of mobile interaction in Korea, Japan and China, with the implementation of integrated social-sensor networking service as an example of unified web service.**

***Index Terms*—performance metric, social network, sensor network, ubiquitous, web service**

I. INTRODUCTION

Mobile social networking services, based on mobile interaction, are emerging as a result of the evolution of mobile communication technology and multimedia mobile devices in the ubiquitous information society. Various sensor networks composed of nodes are being deployed, and we found an analogy between human networking and highly evolved sensor networking. We tried to find a unified approach for performance analysis and found an analogy between the social networking and sensor networking on the basis of web-based metadata directory service.

A social network is a social structure made of nodes which are generally individuals or organizations. The maximum size of social networks tends to be around 150 people and the average size around 124 (Hill and Dunbar, 2002) [1]. To better understand the concept of culture, and how it is related to human-computer interaction, Ford and Kotze [2] state that culture, being the patterns of thinking, feeling and acting, influences the way in which people communicate amongst themselves and with computers. We studied interaction with mobile devices as well as with sensors beyond 'with computers' because of the proliferation of mobile devices and applications with intelligence based on context-awareness. We focused on mobile social networking service as a specific example among various mobile applications to find a common solution.

We studied usable methods for accessing web service in a unified and ubiquitous way for various social communities. Researchers have begun to examine user's findings, and have thus been able to redefine the behavior as well as the limitations of existing search technologies [3]. Special mobile phones for mobile Internet interaction, i.e. Apple iPhone, Google phone by LG, or Yahoo phone by Samsung, are already available or will be available in the near future.

Web browsing on small-screen mobile devices will continue to be a major constraint for mobile users [4], and we need to consider new requirements for using mobile devices [5]. Therefore, we need to study usable mobile interaction schemes and unified web-based directory services. The scheme and service should be appropriate for mobile Internet and ubiquitous web sites, especially for the unified interaction service for user/manager in the social networking as well as in the sensor network management.

We show the useful results from the implementation of ubiquitous web service for usable mobile interaction as well as the possibility for the metadata USN directory service. To pervasively access a unified portal for information access to online social communities, the mobile user interaction, especially with mobile phones for mobile social networking service, should be convenient. We implemented the ubiquitous web portal accessible by mobile phone in Korea, Japan and China. The mobile interaction for real-time information access was successful on hi-speed trains such as the KTX and Sinkansen at over 300km/hour.

The mobile phone can be the sink node or the sensor node in a ubiquitous networking environment because the mobile service was available world-wide before the proliferation of sensor networks. The mobile device has several features that support the use in the middle of sensor networks [6]. Beside local connectivity with standalone sensor motes, connecting sensor networks to the Internet on a global scale creates endless opportunities for applications and services, as well as for new emerging models of operation. We studied the mobile interaction between a user and the unified IP-USN directory for highly evolved sensor web applications.

In the following sections, we will discuss the sketch of

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design goals for mobile interaction for a unified web service. Then, we will discuss the performance metrics for usability in accessing to a unified web site. Mobile (in Korea, Japan and China) interaction for a unified web service will be discussed on the basis of the implementation of a unified web site. Finally, we will conclude our study with a consideration of further research for a unified IP-USN directory service with metadata in the highly evolved sensor networking environment using COSMOS (Common System for Middleware on Sensor Network) [7] middleware platform.

II. SKETCH OF DESIGN GOALS

We need a sketching process to design experience or interaction as Buxton mentioned [8]. We sketched design goals for mobile interaction and application to a unified IP-USN metadata directory service to target usable and unified Web information service in ubiquitous computing and networking environments. Beyond social networking for humans, we are considering the highly evolved sensor networking for sensor nodes, because the concept of mobile interaction for social networking is similar to the highly evolved sensor networking of real-time information management for grouping of sensor nodes using web-based USN metadata directory service.

Most sensor network researchers would probably agree that we have placed too much attention on the networking of distributed sensing and too little on the tools necessary to manage, analyze, and understand the data [9]. However, with standards in place for networking and other infrastructure-related issues, or at least an emerging consensus as to how to solve these issues, the demands for a sensor web revolve around questions of information management. In terms of service discovery, service discovery protocols are designed to minimize administrative overhead and increase usability [10].

The sketched design goals for a unified web service must be considered on the basis of real service from a user's perspective. The primitive requirements for design goals are shown as follows. We need to consider usable mobile interaction for services in ubiquitous environments. We should consider a unified web-based directory service accessible with any device that provides unified information for universal services. For usability of real implementation, the targeted design goal is for the real-time service for information retrieval within a few seconds as well as for the real-time service of information registration within a few minutes. The functionality for simple text-to-speech in a few seconds with inexpensive service as well as with inexpensive investment is required for various services including Telematics. The continuous removal of information-garbage after refreshment keeping W3C guidelines for web-contents accessibility is also one of our design goals. In terms of resource utilization, it should consume less memory for storing information and communication bandwidth. It should be usable with any mobile phones, PC, IPTV, etc. and be easy to backup a web-based directory. For ubiquity, it should be usable on express train at speeds of over 300 km/hour (e.g. Sinkansen, KTX, etc). We need a web-based unified supra (i.e. super infrastructure) for various social networking of human beings as well as for sensor networking services for sensor nodes in the global perspective using IP-based Web. Polling schemes for information, instead of pushing and spam-style scheme, is required without addiction of service and should not be time-consuming for any user as a durable service. We don't need complex schemes for implementation. We need a service accessible to metadata information in convenient ways.

We tried to satisfy the aforementioned design concept for the implementation of a web-based directory portal for mobile interaction between human beings and service agents in a web-based directory server. We are trying to improve satisfaction for the design goals. Thus, we introduce the current status of our research to apply the insight of social networking to the advanced sensor networking for the u-City in the future, especially in the COSMOS project. The test-bed Web site is 'http://ktrip.net' has been implemented to satisfy the design goals. We studied the application to the unified IP-USN directory service based on metadata information management.

III. CONSIDERATION FOR WEB SERVICE

In previous work, the performance and UI (User Interface) issues in ubiquitous information networks were studied [11, 12]. This concept is applicable to the manager's mobile interaction for the metadata IP-USN directory service with metadata information management.

For information retrieval and access to online social networking services, the performance of ubiquitous web service with mobile Internet is important to provide QoS (Quality of Service) with cost-effective and inexpensive system solutions. We studied the requirement for performance of mobile interaction and its metrics in ubiquitous computing and networking environments. We introduce a social networking service, based on single-character multilingual domain names, and we show also the useful results of implementation of ubiquitous Web site for online social networking. We study the application of our concept to the unified metadata USN directory service in the COSMOS project [7] for Web-based sensor networking such as IP-USN.

For consistency of information when using online social networking services, in addition to usable user interfaces, we need a unified and ubiquitous web service for wired Internet and mobile Internet. Fast and convenient access to information as well as notification is required for online social community service including mobile social networking service. We need to write the information in the ubiquitous web site and make it accessible with single-character multilingual domain names instead of long URL strings for convenient web-based information management.

The complexity involved with providing consistent information access from portals or community sites has been increasing and the inconvenience of user interaction for information access has become challenging for even skilled users. We suggest a ubiquitous web service for usable mobile interaction, which is based on 'hand-board' with meaning of online whiteboard service using hand-sized information in PC or handheld phone, accessible with simple single-character multilingual domain names as a convenient mobile UI.

For ubiquitous information portal for social networking service, the unified information service is indispensable in online social networking. The ubiquitous Web server should have the capability of showing the unified contents, i.e. the HTML contents for wired Internet as well as the mobile contents for many different kinds of mobile Internet and various contents using different mobile markup languages, e.g. WML (Wireless Markup Language) and mHTML (mobile Hypertext Markup Language). Easy typing of the URL is also important especially for usable mobile Internet interaction. The single-character multilingual domain names, which could become the indexing key character for information retrieval from ubiquitous web server and for information registration into ubiquitous web-based directory, were considered as a convenient mobile user interface for online social networking services as well as in the metadata IP-USN directory services.

IV. A PERFORMANCE ANALYSIS

Depending upon the keypad stroke number, the level of convenience of mobile interaction could be estimated as a convenience metric as one of many usability metrics, which need further research for metadata directory service. Considering the convenience metric, i.e. defined as the keypad stroke number for a moment, we may understand the usability of the single character domain names instead of long URL-strings for access to unified and ubiquitous web service, especially in a mobile Internet environment. For a ubiquitous information network, even the input interaction of characters is important for retrieval interaction from a unified web-based directory. To access the unified web-based directory ubiquitously, the mobile interaction for information retrieval should be as convenient as possible for typing in the domain names or URLs for searching the right community web site and accessing to information in ubiquitous Web-based directory.

We can access dynamic information in a unified and ubiquitous web-based directory with multilingual single-character domain names as root nodes in information tree for usable mobile interaction in online social networking as well as in metadata IP-USN directory service for sensor networking. With multilingual single-character .net domains, we can access the required information, especially the unified and consistent information in a ubiquitous information network. If we looked at the mobile interaction for handheld phones for mobile information service, and even for the URL typing interaction for information access in the wired Internet. Multilingual single-character is very convenient and usable because it is like a root node in the tree of information-access hierarchy in web-based directory to generate any multilingual domain names or words for information access in China, Japan, Korea, and other countries using multi-lingual domain names.

Considering usable mobile interaction for mobile Internet for mobile social networking services, the development and application environment are very different from existing wired Internet environments, i.e. mainly based on MS Explorer as an Internet browser. Most web pages today are designed for desktop PCs, and viewing them on mobile web browsers is extremely difficult. Chen et al. [13] studied the adaptation of web pages for small-screen devices. It (e.g. mini-homepages, blogs, club, café, etc.) is very difficult to browse with mobile phones, thus we need to consider ubiquitous web service with usable mobile interaction and unified (for PC and mobile devices) online directories, i.e. so called 'hand-board' for text-based information adequate for fast TTS (text to speech) application.

We need to consider the accessibility to ubiquitous web-based directory sites as well as the performance of the mobile interaction for a unified web service. We used a single web server for metadata information access as a unified service for simplicity of information management and for cost-effectiveness in this study. We plan to implement the redundancy scheme to check the reliability of a unified USN directory service for the COSMOS project [7]. This method gives effectiveness and efficiency for the access of information and the utilization of resources, in terms of the bandwidth for communication and the size of disk storage to build ubiquitous web sites for mobile social networking service as well as for metadata IP-USN directory service.

We studied the important performance metric, e.g. delay, at the user's perspective. We studied the performance metric, e.g. delay, not only with the time in the network and server, but also with the spent time by user and the input interaction time with keypads for URLs or metadata information for notification/registration interaction with the web site for mobile social community services as well as for the metadata IP-USN directory service.

As *performance metrics* for usable mobile interaction, we assume that the random variables, the round-trip response time for a user's single interaction in a session, from user to the contents in DB or directory through wired/mobile Internet before next interaction with mobile phone is $\boldsymbol{R}$. That is composed of the preparation time for any user in online social networking to get mobile phones in the user/manager's hand for interaction is $\boldsymbol{U}$. The time spent by the user/manager with mobile phone to do appropriate interaction for service is $\boldsymbol{D}$. The aggregate interaction time to the Web-based directory server after the mobile device through wired/mobile Internet for mobile service is $\boldsymbol{S}$. The interaction time depending upon mobile contents in the metadata directory DB is $\boldsymbol{C}$.

For polling service, we may order the dominating factors in the overall performance of mobile interaction at the user's perspective as follows. In general, the relationship between delay metrics for mobile interaction with mobile phone could be $\boldsymbol{U}>\boldsymbol{D}>\boldsymbol{S}>\boldsymbol{C}$. Here, we need to decrease the major interaction times $\boldsymbol{U}$ and $\boldsymbol{D}$ for polling service, as well as the network and server interaction time $\boldsymbol{S}$ (or interaction time to contents or metadata in DB, $\boldsymbol{C}$). We need to decrease the times $\boldsymbol{U}$ and $\boldsymbol{D}$

for polling service in the mobile Internet for usable mobile interaction.

In Fig. 1, the overall time delay of several elements (i.e. user, device and Web-based directory server) are shown and discussed as follows: user's preparation time, U_1; time with user/manager's device, D_1 and D'_1; time in network and Web-based directory server, S_1 and S'_1; user's time for understanding and readiness, U_2; time with client's device, D_2 and D'_2; time in network and Web-based directory server, S_2 and S'_2; time for reading contents or metadata in Web-based metadata/information DB, $C_{1(Read)}$; time for writing contents or metadata in the web-based DB, $C_{1(Write)}$; user's preparation time for understanding and readiness, U_3; time for writing contents or metadata with user's device, D_3 and D'_3; time in network and web-based directory server, S_3 and S'_3; user's preparation time for understanding and readiness, U_4; time for finishing the session with user's device, D_4; time in network and web-based directory server, S_4. The random variables: *U, D, S, C* will be discussed in the following section based on experience in Korea, Japan and China.

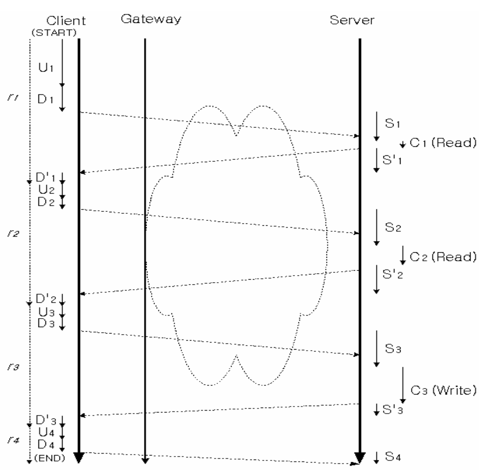

Fig. 1. Mobile Interaction Time for Sessions with a Mobile Device

To be time-deterministic for application with QoS (Quality of Service) in metadata IP-USN directory service, e.g. searching of required metadata in one minute, the mobile interaction with device should be usable, the web-based metadata directory server should be efficient and have high interaction performance for the dedicated directory service, and the metadata for interaction should be simplified in efficient format. The user's preparation time for metadata access, or ubiquity metric, ***U*** will be decreasing (i.e. being improved) depending upon the proliferation of ubiquitous computing and networking environments. The average keypad-press number for mobile phones is related to the interaction time related to the mobile UI agent.

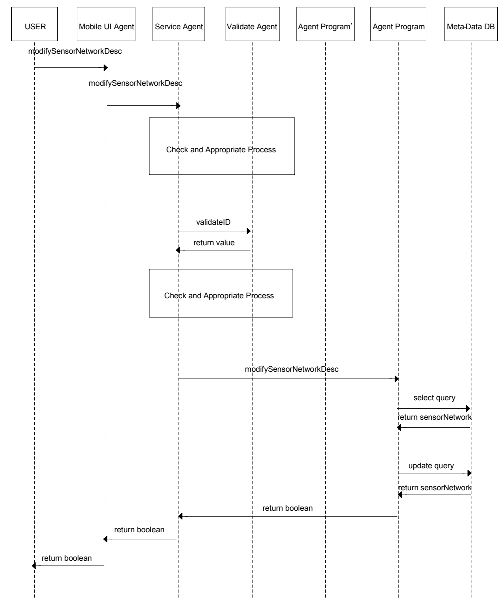

Fig. 2. Mobile Interaction Time for a Metadata Update Session in a Sensor Network

The aggregate interaction time is related to the convenience metric and *performance metric*, (as one of usability metrics) which can be improved with simple user interface and Web-based metadata USN directory for sensor Web, especially for usable mobile interaction in a unified web service. Fig.2 shows the detailed session for the metadata update for a sensor network, as one example among many of the different types of sessions such as registration, deletion, update and search in the USN Directory Service Component in the COSMOS middleware. To accomplish detailed interactions, we will estimate random variables including mobile interaction time between agent processes to further study performance metrics for QoS.

V. IMPLEMENTATION AND EXPERIENCE

For mobile interaction in a web service, the implemented system is based on wired or mobile Internet, many multilingual single-character domains for fast access to mobile social networking service. We completed the development of this application as a specific example among various applications including the unified IP-USN directory service in development with the analogy. The required information or advertisement can be registered anytime or anywhere using wired or mobile Internet with multilingual single-character domain names. We implemented a Web service for lecture group, i.e. composed of around 120 students, and alumni association, i.e. composed of around 60 members, among various social networking services. We focused on the social networking, then we applied sensor networking assuming the number of interacting nodes between 70 and 160, which is the number of sensor nodes in automobile.

We measured the time to type-in the 'ktrip.net' as a simple example to show the critical time in terms of performance. We measured also the aggregate time spent by web site and network, and we observed the time to read the right content after click the title in the displayed information list on the screen of mobile phone. We observed that the typing time for full domain names or long URL-string with mobile phone was serious in terms of interaction performance. Therefore, we need single-character domain names instead of long URL-string for mobile Internet UI with mobile phones as discussed. The typing time was faster than 3 seconds in most cases while multilingual single-character was completed in 1 second.

In terms of packet cost for interaction, one packet, i.e. 512 Bytes, costs 0.5 Cents (U.S.), the minimization of delivered packet number from the ubiquitous Web DB to the mobile phone is important for cost-effective interaction service with ubiquitous Web-based directory service. We considered both the cost-effective packet size and the number of packets for delivery in our implementation of the ubiquitous web portal for usable mobile interaction in mobile social networking service; and moreover the TTS (text to speech) functionality in interaction service was easy to be implemented because of the tiny size and text-based 'hand-board' contents for usable mobile interaction.

We did some experimental research with students as a mobile social network in a lecture group as follows. The number of students was around 800 over 7 semesters, i.e. around 114 students in each semester over 4 month period. This is similar to the number (124) of the averaged social network size [1]. They used a bulletin board or 'hand-board' using the site 'ktrip.net'. The cumulative number of clicks was around 47,000. This means that the average click number in one semester for one student was around 14 clicks per month. Therefore, each student clicked around 3~4 times a week. As an example of web services for online/offline social networking service, special application for information registration/retrieval from/into ubiquitous web-based metadata DB can be done with the notification bulletin board so-called 'hand-board' for alumni association in social networks; here the size of association members may be from tens of members to a couple of hundred members in online/offline social networking. In the deployment of real sensor networks, the appropriate group size of sensor nodes for each sensor network as well as the appropriate group size of sensor networks for the unified IP-USN directory service will be realized depending upon the application service such as u-City (ubiquitous city) scheduled for construction in the new Sejong City, Korea.

The polling of metadata in the directory can be considered similarly to the sampling theorem, the polling frequency $f_{poll} \geq 2 * f_{notify}$, where f_{poll} is the frequency of polling the ubiquitous record in the metadata DB table, i.e. a searched Web site for mobile social networking service, and f_{notify} is the frequency of notification in 'hand-board' by the association for mobile social networking service. In the IP-USN directory service, the polling rate by manager or components for the updated metadata information of a sensor network or a sensor node will be very similar to the above concept; and it will be helpful for the queuing analysis within the above arrival rate, i.e. as a practical polling rate, for the IP-USN directory service for further research.

The speed of on-line registration for advertisement/notification as well as the speed of access to special information with ubiquitous web DB is fast enough for real application. Moreover, the effectiveness of web DB usage for mobile social networking services can be anticipated if we consider the applications for various communities, mini-homepages, clubs, blogs, mobile blogs, UCC, and mobile UCC services based on rich media as far as the consumed disk storage and cost for operation and administration related to mobile social networking are concerned. We study the performance for real-time registration and search of sensor networks or sensor nodes, in the similar mobile Internet environment before proliferation of USN including IP-USN.

We implemented the Text to Speech (TTS) functionality for usable mobile interaction because the simple and text-based information used for 'hand-board' services was easy to implement TTS functionality based on speech synthesis. We will study the TTS application of the manager of metadata directory service for the highly evolved u-City. The conversion time of 1 Kbytes text-based information was around 1 second. In most cases, the information/metadata size for mobile social networking service and the record size in the metadata DB table for unified IP-USN directory service is enough within around 1 Kbytes on the basis of our experience and metadata design for a metadata IP-USN directory. This TTS functionality for interaction will be very helpful for the elderly. Also, the telematics service for auto drivers will be applicable with usable mobile interaction for listening to the contents in 'hand-board' with transformed audio contents.

According to the empirical results from the implementation of social networking services using wired Internet, the time *S* may be considered a rather short period (around 5~30 msec with Ping, which is related to the *S*. However, for a 5Kbytes web page, PC response time is around 2~3 seconds, which is related to the *S* and *C*, here *C* is much longer than *S*). Since mobile phones use mobile Internet (for short packets below 1.5Kbytes and even around 5Kbytes), the response time is around 12 seconds with little deviation through the WAP gateway. Thus, the time *S* is longer than *C*, where *S* includes the elapsed time at the gateway for the mobile Internet.

From the experiment of Web-based directory service based on the mobile Internet in Japan, with roaming service, we observed that the response time for a wired PC is rather fast and stable with little deviation as in Korea. The average response time for mobile phones for the first access to 'ktrip.net' was around 12[sec] with little deviation from what was observed in Korea. After the initial connection to 'ktrip.net', the reading time of registered information was around 2~3[sec]. The results in China were similar to the results in Japan and Korea. To conclude, the critical time the device time *D* with mobile phone in our experiment as found in Korea. The sum (*S*+*C*) was around 2~3[sec] and was not comparable to the time *D* that is at least over 30~60[sec] depending upon the amount of text-based information for writing with keypads during registration of information. The inconvenient interface for writing URLs or information with keypads caused a major bottleneck in degradation of overall performance for a unified web information service. This experience is applicable to the manager for real-time information management of metadata IP-USN directory service in the u-City.

We are also considering location-based service for mobile social networking services as well as sensor networking services for the u-City, using mobile phones. Then, the listed metadata information related to that specific location will decrease tremendously for usable mobile interaction in a unified Web service. The convenience and usefulness of multilingual single-character domain names will be remarkable for accessibility to the ubiquitous web sites for mobile social community service as well as for the metadata IP-USN directory service. With any mobile devices, the directory service for social interaction as well as for the sensor networking will be feasible in ubiquitous computing and networking environments for real-time metadata information management.

VI. CONCLUSIONS

A unified approach for performance analysis of integrated social-sensor networking service was studied. As a convenient and usable mobile HCI for mobile web service for online social-sensor networks, the multi-lingual single-character domain names as indexing keys to social-sensor information in ubiquitous web service are convenient mobile interfaces that allow one to search for social-sensor information and to register metadata information. We studied to apply to the metadata directory service because of the analogy. The convenience of multilingual single-character domain-names and the tiny 'hand-board' for usable mobile interaction was discussed with empirical results based on the implementation of a unified web service. We studied the design goals and experience in Korea, Japan and China for mobile interaction with the implementation of mobile social-sensor networking services, as an example of a unified web service. Unified web sites for the u-City will be studied with a unified USN directory service.

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Principles of Concurrent Processes Synchronization at Discrete Systems' Nonprocedural Simulative Models

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***Abstract*-The article is considering principles of non-procedural approach to developing the operational logic's control algorithms for simulators providing the possibility of building simulative models with complex functional mechanisms. These mechanisms based on transactions assembly and dividing means at the same time that from possibility of control signals generation by servicing facilities and requests' sources. The proposed technology application is illustrated with computing processes and schemes models' fragments.**

I. Introduction

While discrete systems and equipment architectures' simulation technology developing, the opus [1] suggested a non-procedural approach to the simulative models' creation and investigation. This approach implementation running, both an Automated Dialog Simulation System (ADSS) and its operational technology have been developed [2]. This project technology specific peculiarity consists in absence of need in programming at every stage of models' creation and investigation [3]. The researched object represents a set of discrete automates, simulating delays when input transactions processing. Such research object representation completely correlates with classical models of en-mass servicing, wide-scale implemented in discrete systems simulation [4]. Nevertheless, as to creation of complex adaptive operational algorithms' and non-conventional patterns of servicing processes' models on the basis of en-mass servicing theory apparatus it is essentially impossible. The suggested models represent just a coarse approximation of the real processes. These models have rather didactical and methodical value that practical one [4, 5]. To eliminate that deficiency within non-procedural approach frames, proposed is a mechanism of concurrent processes synchronization, providing the possibility of simulative models with complex functional algorithms creation. It is based onto using transactions assembly and dividing means as well as occurrence of servicing facilities' and request sources' control signals' generation [6].

II. Schemes of computational algorithms

The synchronization mechanisms' application can be sought as an example of multiphase multiple channels en-mass servicing system, implementing the concurrent computations algorithm. Schematically, that computational algorithm reveals as branched structure, which nods do allocate the general problem's partial tasks execution (Fig. 1). The links between top points do signify the initial conditions of computation process execution. The first phase includes *s* groups of concurrently executed processes P^1_{ij}.(i – set number; j – phase number). Generally, every process execution time is random value. Number of processes (n, m,…, k), concurrently executed at every set varies. At that, the q^{th} phase (q=2,…,r) computing algorithm actuation at hierarchical schemes is effected only upon all previous level's processes termination (Fig.1, a). Example given, the P^2_1 process initiates only when P^1_{11} … P^1_{1n} processes terminated.

When more complex synchronization schemes taking place, the process' initiation at q^{th} phase can depend upon external conditions, like, ex. g., process termination at structure's other branch. Let we consider the secondary link between P^1_{2m} and P^2_1 processes, shown at Fig.1,b. Such linking available creates a network synchronization scheme. In such a case when reference made to hierarchical system, the P^2_1, process initiation takes place after P^1_{2m} process termination. Such systems investigation specificity relates to dependency realizing the processes' asynchronousity influence onto system's reaction probability and time-dependent characteristics.

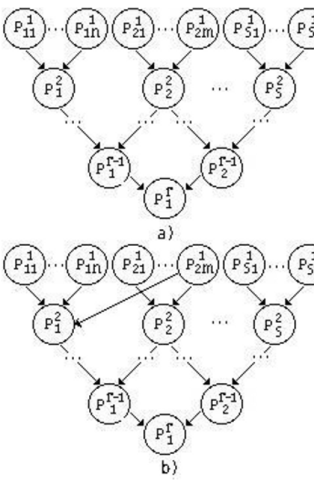

Fig.1. Schemes of computational algorithms' implementation: a) hierarchical; b) networking

K. Elleithy (ed.), *Innovations and Advanced Techniques in Systems, Computing Sciences and Software Engineering*, 544–549.

The suggested approach considers the following mechanisms of computing algorithms synchronization schemes (see above) modeling:

- controlled sources;
- transactions assembly;
- controlled sources joint using, transactions assembly and control signals;
- transactions assembly with consideration to several logical conditions.

III. SYNCHRONIZATION BY CONTROLLED SOURCES

Let we consider the first synchronization mechanism at the example of hierarchical multichannel multiphase network model, at which first phase the **s**-groups of concurrent processes are executed. At that such processes' number at the first and the last groups would be equal to n and k, correspondingly (Fig. 2).

While first phase, the concurrent processes simulation is executed by corresponding simulators' sets, identified with p_{ij} parameter (i – set number; j – channel number). Under **s**=2 condition, the second phase is represented with two identifiers, – p_1 and p_2, and the third set is characterized by the auxiliary facility p_3. Processes execution duration simulation P^I_{ij} occurs through transactions, arriving to input of corresponding simulators p_{ij}. Time intervals between incoming transactions follow the exponential dispersion principle. Input flows are simulated with the auxiliary of non-system requests' sources with corresponding numbers f_{ij}. Requests servicing follows the normal dispersion principle. At initial state every valve (simulators p_{ij}) is "open". Therefore the incoming requests are immediately transferred to servicing. Upon servicing, the transactions leave the system. At valves' output there are generated their locking signals (*"b"*). At the scheme these signals are shown with more fine crossed lines. Concurrently generated are the controlled source actuation signals $ctrl_i$, serving to generate the transaction purposed for simulating the second phase servicing process simulation.

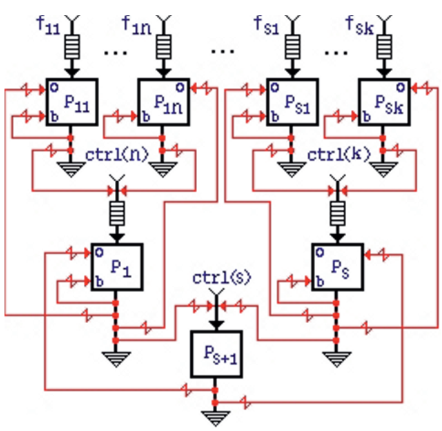

Fig. 2. Scheme of synchronization algorithm model based upon controlled sources

The request is generated only after receiving the "source actuation" signal from every valve's output at i^{th} set. This mechanism is given through setting the controlled source actuation threshold f_i. So, a transaction arrived to input of p_i (i=1, 2) facilities, simulates termination of all first phase's concurrent processes. At p_i facility's output there are generated the simulators' unlocking signals identified with p_{ij}. As a result, the p_{ij} facilities turn into the initial state. At the same time the source actuation signal $ctrl_3$ is generated thus initiating the third servicing phase process. Here the processing begins upon receiving «source actuation» signal from p_1 and p_2 facilities' outputs. In its turn, servicing at p_3 terminated, initiated is the facilities' unlocking at second phase. In such a way, the hierarchic systems concurrent processes' synchronization is simulated.

This synchronization method's benefit relates to simulative algorithm realization simplicity. But to obtain statistics on multilevel synchronization schemes' execution length, there exists a need for including supplementary facilities into the model [3].

An example of non-procedural model's initial text at n=3 and m=5 can be represented like:

```
  /* Group 1, phase 1, channel 1 */
f(p,11); if: (flow=11 exp(1.2)); qu: common;
   st: gauss(0.65,0.15);
   of: on*(((flow=1))) off;
f(p,12); if: (flow=12 exp(1.2));
   qu: pt - f(p,11);  st: pt - f(p,11);
   of: pt - f(p,11);
f(p,13); if: (flow=13 exp(1.2));
   qu: pt - f(p,11);  st: pt - f(p,11);
   of: pt - f(p,11);
      /* Group 2, phase 1, channel 1 */
f(p,21); if: (flow=21 exp(1.2)); qu: common;
   st: gauss(0.65,0.15);
   of: on*(((flow=2))) off;
f(p,22); if: (flow=22 exp(1.2));
   qu: pt - f(p,11);  st: pt - f(p,11);
   of: pt - f(p,11);
 …
f(p,25); if: (flow=25 exp(1.2));
   qu: pt - f(p,11);  st: pt - f(p,11);
   of: pt - f(p,11);
      /* Phase 2, channel 1 */
f(p,1); if: (flow=1 ctrl(3)); qu: common;
    st: exp(15); of: on*(((flow=3))) off;
      /* Phase 2, channel 2 */
f(p,2); if: (flow=1 ctrl(5)); qu: common;
    st: exp(15); of: on*(((flow=3))) off;
      /* Phase 3, channel 1 */
f(p,3); if: (flow=3 ctrl(2));  st: exp(15);
   of: on(f(p,1)2)off;
ts(10000);   /* Simulation time */
```

To reduce initial text volume, the model description uses the structural abbreviations specified in Table I.

TABLE I
THE LIST OF STRUCTURAL ABBREVIATIONS USED IN MODEL DESCRIPTION

Identifier	Full description
f	facility
if	input flow
exp	exponential
qu	queue
pt	pointer
st	simulation time
of	output flow
ctrl	control
wh	with histogram
whof	with histogram output flow
Const	constant

IV. SYNCHRONIZATION BY TRANSACTIONS ASSEMBLY

The second mechanism of non-procedural models concurrent processes' synchronization departs from using transactions assembly means; this approach would be sought at the example of computing operations organization scheme here above. The three-phases multichannel system simulative model's graphic representation refers to Fig. 3.

Differing from the previous model, here the transactions serviced at first phase do stay at the system, arriving for collection to the respective simulator p_i (i=1,…, **s**), representing the second phase. Here the initial valve is also locked. The second level first set of simulators servicing starts only if collection queue includes n transactions. Such mechanism is given by description of processes as transactions collection into one request ("*assembly=3*"). Therefore n transactions arrived to p_i input, the servicing process is initiated. At facilities' output the corresponding valves' p_{i1},…, p_{is} unlocking signals are generated. At that the initial valve p_i is locked. Restoring for next request servicing is possible only upon request processing at third phase termination. Upon collection, the issuing request from p_i output arrives to p_{s+1} facility input.

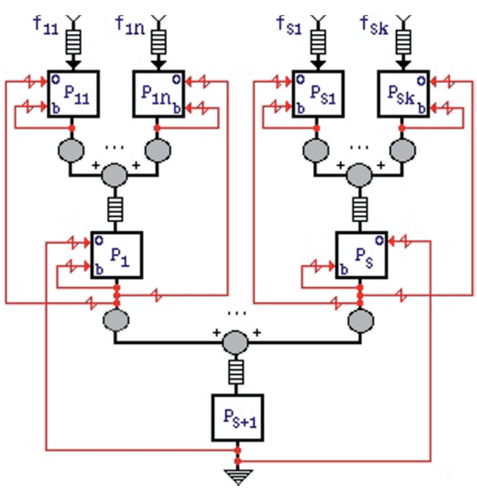

Fig. 3.Scheme of synchronization algorithm based onto transactions assembly principle

Correspondingly, this request processing initiation will involve after obtaining all p_i requests. The "**s**" requests number being arrived, the collection is effected. Incoming request immediately passes for servicing, upon which ending at p_{s+1} simulator's output the p_i facilities' unlocking signals are generated. As a result, provided is the possibility of second phase requests' processing activation. Here, the p_i simulators are initial state – restored.

This synchronization method used, the evaluated average time of requests' staying within system is calculated with standard statistical data collection means. For obtaining the probabilities-time computational cycle estimate duration when non-system sources description, a "*wh=12*" setting is given. At that such synchronization schemes delays calculation doesn't require no auxiliary facilities introduced into the model. An example of (n=3 and s=2) model's initial text can be written as:

```
 /* Group 1, phase 1, channel 1 */
f(p,11); if: (flow=1 exp(0.1))wh=12;
   qu: common;
   st: gauss(0.2,0.05); of: off;
f(p,12); if: (flow=2 exp(0.1))wh=12;
   qu: common; st:pt-f(p,1); of: off;
f(p,13); if: (flow=3 exp(0.1))wh=12;
   qu: common; st:pt-f(p,1); of: off;
      /* Group 2, phase 1, channel 1 */
f(p,21); if: (flow=4 exp(0.1))wh=12;
   qu: common; st:pt-f(p,1); of: off;
f(p,22); if: (flow=5 exp(0.1))wh=12;
   qu: common; st:pt-f(p,1); of: off;
f(p,23); if: (flow=6 exp(0.1))wh=12;
   qu: common; st:pt-f(p,1); of: off;
      /* Phase 2, channel 1 */
f(p,1); if: (flow=1-3 f(p,11)12-13)
        assembly=3;
   qu: common;
   st: exp(16); of:  on(f(p,11)12-13) off;
      /* Phase 2, channel 2 */
f(p,2); if: (flow=4-6 f(p,21)22-23)
         assembly=3;
    qu: common; st: pt-f(a,1);
    of:  on(f(p,21)22-23) off;
      /* Phase 3, channel 1 */
f(p,3)whof(10); if:(flow=1-6 f(p,1)2)
         assembly=2;
    st: exp(50); of: on(f(p,1)2);
ts(10000);
```

V. MIXED SYNCHRONIZATION

A model of more complex concurrent processes' synchronization variance is developed departing from mechanism of transactions assembly, the controlled sources' properties and external readiness signals (Fig. 4).

The two first synchronization ways relate to those previous models. The present algorithm scheme realizes both mechanisms at computational algorithm's first phase for the first and the last of simulators groups p_{ij}.

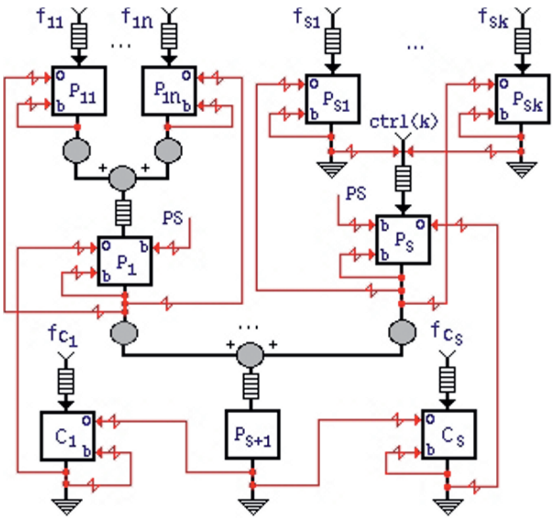

Fig. 4. Scheme of complex synchronization algorithm using external signals of readiness

Transactions servicing at the second phase initiates only upon receiving readiness signal from other simulators. These signals involve "actuated" state of p_i imitators. Readiness signals generation takes place at c_i simulators' output upon request processing termination. In such a way simulated is the dependency of initiation of the request processing with p_i ($i=1,...,s$) valves and concurrently operating c_i facilities. Here transactions arrive at p_1 valve's input upon transactions collection terminated. The collection is executed at n transactions arrived from the first group's simulators. Respectively, a request enters to p_s input after k signals of source actuation *ctrl(k)* generated and issuing from s^{th} group facilities' output.

Irrespectively to synchronization methods, at p_i output there are generated the corresponding simulators' group p_{ij} unlocking signals. Here the initial valve is locked and the request obtained from p_i output passes to p_{s+1} facility input. Accordingly, request' processing initiation takes place after requests' collection procedure from all p_i completing. Signals of a unblocking of facilities c_j (i=1,…,s) are generated after termination of processing. As a result, the c_j facilities restore the ability of readiness signal generation, allowing requests' processing at the second level.

Such synchronization mechanism used, the average time of requests' staying at system estimation is provided with standard statistical methods. Such model initial text example is following:

```
   /* Group 1, phase 1, channel 1 */
f(p,11); if: (flow=1 exp(0.1))wh=12;
      qu: common; st: gauss(0.2,0.05); of: off;
f(p,12); if: (flow=2 exp(0.1))wh=12;
      qu: common; st:pt-f(p,1); of: off;
f(p,13); if: (flow=3 exp(0.1))wh=12;
      qu: common; st:pt-f(p,1); of: off;
         /* Group 2, phase 1, channel 1 */
f(p,21); if: (flow=4 exp(0.1))wh=12;
      qu: common; st:pt-f(p,1);
      of:on*(((flow=7))) off;
f(p,22); if: (flow=5 exp(0.1))wh=12;
      qu: common; st:pt-f(p,1);
      of: pt-f(p,21);
f(p,23); if: (flow=6 exp(0.1))wh=12;
       qu: common; st:pt-f(p,1);
       of: pt-f(p,21);
         /* Phase 2, channel 1 */
f(p,1); if: (flow=1-3 f(p,11)12-13) assembly=3;
       qu: common; st: exp(16);
       of:  on(f(p,11)12-13) off;
         /* Phase 2, channel 2 */
f(p,2); if: (flow=7 ctrl(3)); qu: common;
       st: pt-f(a,1); of:  on(f(k,3)4) off;
         /* Phase 2, channel 1 */
f(p,3)whof(10); if:(flow=1-6 f(p,1)2)
              assembly=2;
       st: exp(50); of: on(f(c,1)2-4);
f(c,1); if: (flow=11 exp(0.1)); qu: common;
       st: exp(16); of:  on(f(p,1)) off;
f(c,2); if: (flow=12 exp(0.1)); qu: common;
       st: pt-f(c,1);   of:  on(f(p,2)) off;
ts(10000);
```

VI. Synchronization with Consideration of Logical Conditions

The fourth synchronization variance departs from transactions assembly mechanism and sequence of logical conditions satisfied (Fig. 5). Such condition can relate, e.g., to concurrently operated simulators' queues criteria like b_j (j=1,…, k). This synchronization algorithm can be sought as a model of "n" concurrently executed processes.

These processes' completion duration is modeled with p_i (i=1,...,n) simulators. The p_i facilities' inputs receive the Poissonian requests' flows with corresponding indexes f_1 - f_n. Requests' servicing is based onto normal distribution principle. Upon servicing completion the requests are passed to the p_{n+1} valve input, which valve serves for modeling the synchronization process with the use of requests' collection mechanism. The arriving to input requests are queued for assembly. Passing for servicing is effected only upon assembly procedure completion that is equal to "n" of concurrent processes termination. For including to collection by one separate request from every source, the p_i simulators and locked after servicing termination. When initial state, the p_{i+1} valve is locked with initial setting signal (*ps*). Therefore the request resulting from colleting procedure is queued. At the same time a signal of auxiliary condition checking is generated. Such condition consists in transactions presence at input storages to b_j (j=1,...,k) facilities.

This condition checking procedure simulation is fulfilled with $test_1$ simulator involving. It serves to organize a cyclic requesting of b_j facilities.

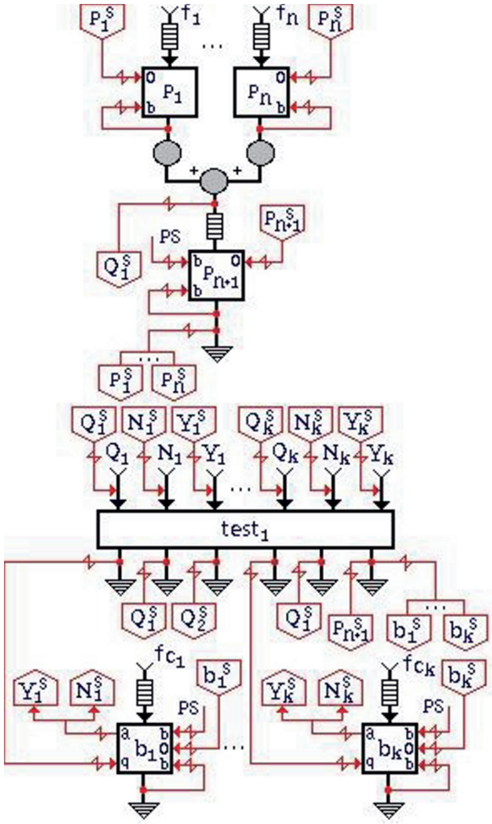

Fig. 5. Scheme of synchronization algorithm using requests collection mechanism and logical conditions satisfying.

When initial state, all b_j facilities are locked with "*ps*" signal. Requesting signals are generated by Q_j sources. At $test_1$ simulator output these sources generate requesting signals of "*q*" type, code 5 (determining the request presence at queue). A transaction being detected at the requested facility's input queue, then generated is a positive response, if contrary, the response will be negative. At first case the Y_j source is actuated. The generated request at $test_1$ simulator's output generates the requesting signal addressed to the next simulator, b_{j+1}. All responses received being positives at the $test_1$ simulator's output, the Y_k generates (P^s_{n+1}) signal of n+1th process initiation. Simultaneously, actuated are all b_j (j=1,...,k) simulators. The requests are evolved from queues and passed for servicing; processing completed, the b_j valves are locked. At that issue the auxiliary condition checking unit restores the initial value. A negative response obtained, generated is the signal of corresponding source N_j actuation. At $test_1$ simulator's output the obtained transaction generates the Q^s_1 signal of Q_1 source actuation. As a result the request cycle is repeated beginning with b_1 simulator; such process will be accomplished up to the instance of requests' setting into queues to every b_j facilities.

At checked condition fulfillment the p_{n+1} facility unlocking is effected, this signal being generated at the $test_1$ simulator's output through Y_k source. That signal arrival detected, the process of p_{n+1} process execution is initiated. The request, evolved from queue, passes for processing, upon which termination the facility p_{n+1} is unlocked. Concurrently the $p_1,...,p_n$ simulators unlocking signals are generated, after that the next cycle of "*n*" concurrent processes synchronization is started.

This model's initial text example can be represented as:

```
   /* Concurrent processes execution simulators */
f(p,1); if: (flow=1 equal(3,4))wh=12;
     qu: common;
     st: const(0.1); of: off;
f(p,2); if: (flow=2 equal(3,4))wh=12;
      qu: common;
     st:pt-f(p,1); of: off;
...
f(p,8); if: (flow=8 equal(3,4))wh=12;
     qu: common;
     st:pt-f(p,1); of: off;
/****** Collection simulator ******/
f(p,9)(1); if: (flow=1-8 f(p,1)2-8)
           on*(((flow=101))) assembly=8 ;
       qu: common; st: const(0.1);
       of: on(f(p,1)2-8) off;
/***** Storages simulators *************/
f(b,1)(1); if: (flow=51 gauss(3.5,0.8))wh=12; qu: common; st:
exp(1);   of: off;
f(b,2)(1); if: (flow=52 gauss(3.5,0.8))wh=12; qu: common; st:
pt-f(b,1); of: off;
f(b,3)(1); if: (flow=53 gauss(3.5,0.8))wh=12; qu: common; st:
pt-f(b,1); of: off;
f(b,4)(1); if: (flow=54 gauss(3.5,0.8))wh=12; qu: common; st:
pt-f(b,1); of: off;
/* Storages' cyclic requesting control facility*/
f(test,1); if:(flow=101-124 ctrl);
      st: (flow=101) const(0.1),
          (flow=102)const(10e-10);
      of: (flow=101) ask (5)
          ((f(b,1)) & (flow=102) (flow=103)),
         (flow=102) on*(((flow=104))),
         (flow=103) on*(((flow=101))),
         (flow=104) ask (5) ((f(b,2)) &
               (flow=105) (flow=106)),
         (flow=105) on*(((flow=107))),
         (flow=106) on*(((flow=101))),
         (flow=107) ask (5) ((f(b,3)) &
              (flow=108) (flow=109)),
         (flow=108) on*(((flow=110))),
         (flow=109) on*(((flow=101))),
         (flow=110) ask (5) ((f(b,4)) &
               (flow=111) (flow=112)),
         (flow=111) on(f(b,1)2-4,f(p,9)),
         (flow=112) on*(((flow=101)));
    ts(100000); /*  Simulation time  */
```

VII. Conclusion

The concurrent processes synchronization mechanisms described above do allow essentially fasten both creation and investigation of simulative models with complex algorithms of servicing facilities' functional logic control. The investigated objects can be represented with multiprocessor systems [8], multiple access protocols [6], communication protocols, based onto Frame relay protocol [9], ATM – commutators [7], local computing networks etc. For more information about simulation complex, see www.adss.onu.edu.ua.

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An Adaptive Train Traffic Controller

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***Abstract*- We present an implementation of a Train Traffic Controller Simulator, in which functional requirements can be adaptive, that is to say, modified, supplemented or even redefined through an approach based on finite state machines. Depending on its requirements, a Train Traffic Controller can assume simple or critical features in terms of safety or fault tolerance. The modeling based on an approach of state machines arranges the environment for adaptability to some functional requirements. This feature is better emphasized when we use automated tools of definition of state machine that permits an easy integration with the rest of the application.**

I. Introduction

A traditional area within the system application forms the Train Traffic Control of high and low density for critical mission. We have seen lots of Train Dispatchers Simulators or even traffic distribution. This work refers to a simulator of railway interlocking, that's to say, the part of the Train Traffic Control that is responsible for the train permissions under rules and restrictions well defined. A solution even if introductory about an adaptive tool related to some functional requirements of a rail interlocking, sets up an interesting contribution to this segment of the industry. The possibility for complementing of modifying functional rules into an environment of simulation makes possible the execution of tests and analyses which can cover the specificities of a specific railway interlocking.

Proprietary hardware and software forms real *Train Traffic Control Systems*. This way the functional requirements only can be evaluated through theoretical tests and other ones with its own specific hardware and software. Inside this approach there are some obvious cost restrictions to become the environment of wished simulation into a reality. There are some conceptual models of railway interlocking already defined in real application or academics, which can perfectly be useful as a base for the implementation of the simulator which is proposed in this paper. In [1] we can find a description more detailed about a model of Train Traffic Controller.

In this paper, it is presented with an introductory version of a *Train Traffic Control Simulator* based on the model described in [1]. As a matter of fact the simulator is implemented in Java language. The functional rules and criteria of completeness or requirements of general security are modeled from finite state machines (***FSM***). This approach allows uncoupling the code of the *controller module* in itself from the rest of the application. This way is possible to adapt a simulator to a new functional rule, which are essentially functions that implement the actions of the state machine or even its set of transition status. The use of a tool that helps to create *finite state machines*, such as (*SMC- State Machine Compiler*), has made easier to achieve the characteristic of adaptability. It is based on a language of scripts of easy manipulation and creates the code of state machine on the target programming language, (Java in this specific case).

In the following sections are described the fundamental aspects of a *Simulated Train Traffic Control Module* (interlocking) which logical model based on ***FSM***. We present an approach with a goal of achieving an adaptive simulator related to some functional requirements of a railway interlocking. To realize it, we use FSM and a specific tool (*SMC State Machine Compiler*) which are fundamental for the implementation. Additionally a model of simplified railway interlocking is described which fits as the base to the implementation of the simulator proposed. Some relevant aspects related to the continuing of this project are discussed and being tested some of those related to fault tolerance.

II. GENERAL CONSIDERATIONS ABOUT RAILWAY TRAFFIC CONTROL

As mentioned before, this paper is about a Train Traffic Simulator in the context of an interlocking. The Controller is responsible for the *movement authorities* of the trains. These authorizations mean the routes that allow the train circulation from one point to the other. The interlocking is also responsible for providing the *safe* detection of occupations on the part of the rail as well as providing the switch movement over the path of a route, maintaining the locking electromechanically. Figure 1 shows the typical context of a railway interlocking responsible for a crossing yard (*siding*).

Figure 2 shows a simplified schema of a typical control system and supervision. We want to emphasize a controller module, which is responsible for the implementation of control logic applied in case of railway interlocking. As shown in the figure 2 the controller module has a set of *Input* and *Output* variables as the following description:

- **Inputs**
  - $I_{controller}$ (Inputs from monitored devices in the field)
  - $CR_{controller}$ (Commands requests from dispatchers)
  - $Dist_{controller}$ (External disturbance which can to affect the controller's normal operation)

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- **Outputs**
  - $O_{controller}$ (Outputs which commands controlled devices in the field)
  - $MSG_{controller}$ (messages generated to the system's operators)

The simulator described of this paper, has its kernel representing the *controller module, see* figure 2. Essentially we aim to adaptability characteristic of the functional rules of an interlock. Depending on the operational characteristics of the railway, safety requirements, automation level and railway tracks topology (single or double lines), the interlock can assume several functional features.

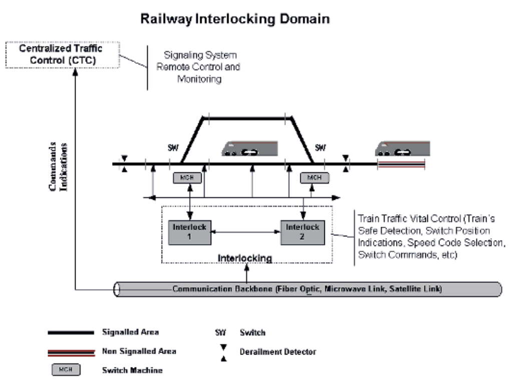

Fig. 1. Typical Single Line Railway Siding (Interlocking)

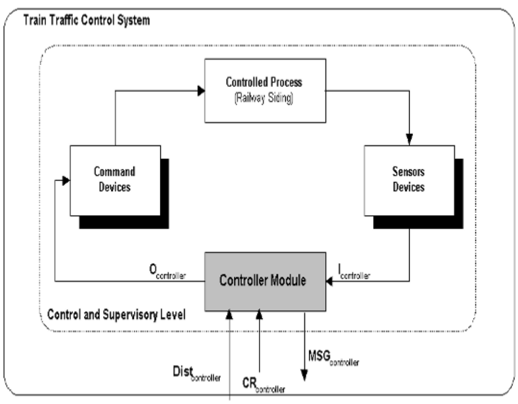

Fig. 2. System of Control and Supervision (Conceptual Model)

The main idea is creating a simulation environment of a controller module that might be reasonably adaptive to different characteristics of a specific application. As detailed in the section 3, the use of finite state machines, in the logical modeling of the simulator, permits this characteristic.

The following description gives details about the inputs and outputs of the controller module of the supervision and control model illustrated in figure 2. The rail elements related to all these inputs and outputs can be visualized in figure 1. It illustrates the railway business context in which the proposed simulator refers.

A. Inputs of the Controller Module ($I_{controller}$)

- Occupied and Unoccupied Track. Information that comes from the field to say whether the tracks are occupied by a train or not.
- Switch Machine Positions: (Normal Reverse and Locked). Information in switch positioning, indicating if the main element named Switch Point's in the reverse or normal position. The lock indication says if the switch is electrically locked which means that it would not be possible any other command on it.
- Blockage Indication (Open and Closed). Basically it indicates the blockage state for the route path to be followed. These routes are responsible for defining the authorized path for the train when it is getting in or getting out the siding. Open signaling means path authorization, while closed means blockage indicating that the path is forbidden.
- Speed Code Indication. Information that reflects the speed code selected for a certain stretch of the rail in agreement to a specific context.

B. Outputs of the Controller Module ($O_{controller}$)

- Switch Positioning Command. Command for choosing one of its positions, Normal or Reverse as required to the controller.
- Lockage Switch Machine Command. Command for putting the switch in state whose position can keep it locked for no possible further command.
- Speed Code Generation Command. Command that sends a track code to any stretch of the Rail for a specific speed, which must be detected by the train when it gets to this part of the rail.
- Command for Opening and Closing Blockage. Commands for putting an inputs and outputs blockage in a certain part of the railway, on an opening or closing state. This respectively can allow or forbid a route for any train.

C. *Commands Requests* ($CR_{controller}$)

- Controller Initialization. Request for activating the controller activation sequence
- Controller Termination. Request for activating the termination controller sequence.
- Route Request. Basically it means authorization for a train movement. A route establishes the path for a train, which can contain some elements previously defined, such as: stretch of via, switches and signalers. In this model an established route means a safe possible path for one train, because in the end of it means that it's is unoccupied as well as the via switch machine circuits over the route path. These switches are positioned conveniently and locked and all entrance lockage open and the specific speed code already established on the track. This last one will be detected through a subsystem installed in the locomotives.
- Request for Canceling Routes. These requests for canceling routes sent by the train traffic dispatcher, refers to routes previously established as shown in the item before.
- Request for Forbidden a Route. This request has as main goal, prevent routes to some parts of the Railway from being established, usually in unusual situation or maintenances.
- Requirement for Canceling the Prohibition to Align Routes. Train Traffic Dispatcher tries to cancel the forbidden condition to align routes described in the item before. In general as the removing of blockage and prohibitions has some regulations that involve some safety criteria, are recommended some levels of confirmations for these commands to be done.

III. LOGICAL ARCHITECTURE

It has been adopted a simple model of railway control system. Such a model comes from [1]. Essentially the controller module (see figure 2), can assume even nine states (as shown ahead) whose rules of transitions are dependents on variations of the inputs states, as detailed in the section before. Additionally to every state transition or event there might be or not an action in the foreseen outputs, also detailed in the section before.

The functional logic of the controller module, can be summarized as a set of rules of state transitions, as well as the actions resulting from transitions associated to an operational rules and safety requirements. The controller logical model can be easily transcript in finite state machines. To get it done, it is necessary to define formally the input and output predicates of the controller module as well as the state transition rules. Figure 3 shows the logical architecture of the simulator detailing the structure of the controller module in a state machine.

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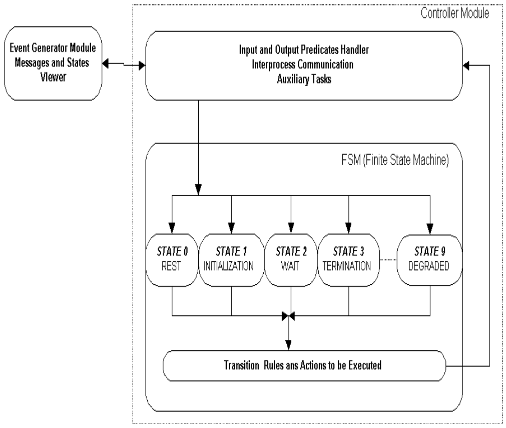

Fig. 3. Railway's Interlocking Simulator (Logical Model)

According to figure 3 the logical architecture of the rail traffic control simulator (interlocking) comes basically with the following structure.

- An Event Generator module is responsible to produce the simulator's inputs as well as the visualization of the output predicate's results. As described in section 2.
- The controller module itself is made up of a handler for the treatments of output and input predicates and also for the state machine of the simulator. The Handler basically executes a pre-process of the inputs variables in a way to send the transition events to the state machine. Additionally, some auxiliary tasks are done and mainly implement the actions designated to each state machine transition, in agreement to the rules of transitions and foreseen actions.
- The state machine essentially models the dynamic behavior of the simulator, putting it in a proper state in agreement to the rules of transitions established by the business model and rail interlocking safety requirements. These rules are usually expressed in *Boolean Expressions* that manipulate logically the controller's inputs and outputs in forms of predicates
- For the reasons of logical uncoupling and modular structuring, the set of actions foreseen in the rules of transitions of state machines can be implemented in a separate module. It aims to ease characteristic of adaptability of functional simulator requirements.

A brief description of each state predicted in the adopted rail control model. [1]

- Resting State. In this state the system, it is switched to off. There is no activation of the functions supervision and control.

- Initialization State. The track state variable and control are initialized as well as it is activated the functions of supervision.
- Waiting State. The system finds itself with all its functions expecting requirements from the operator, via entrance events or events generated by the controller module.
- Termination State. The state in which the system finds itself in process of termination or becoming off.
- State of Align Route. The state in which the system will process a requirements of routes.
- State of automatic route canceling. The system gets to this state when the train gets to the part of the rail where the route is involved
- State of Canceling by the Operator. In this state, the operator requires clearly to cancel a route previously aligned or being aligned.
- State of route alignment block. The system goes in thisstate when a block requirement is received from the operator
- Hazard State. State in which the system is not driven spontaneously. For the system to assume this state, it is necessary that some incoherent actions happen.
- State safe degraded. The system gets to this state when an event is detected taking the system to an unsafe condition and the treatment of the event can move thru the system.

IV. IMPLEMENTATION OF AN ADAPTIVE MODEL WITH FINITE STATE MACHINE.

The adaptive model, in which this article refers, means basically the possibility to modify itself without much effort or rules of the rail-interlocking simulator. See as an example the following rule in table 1.

TABLE I
INTERLOCKING BOOLEANS EXPRESSIONS

Current State	State of Align Route
Event	Route Granted
Output	Command to move switch machine and command to lock switch machine.
Following State	State to Align Route

It shows whether we want to change this rules in a way to include more actions in the output or even to associate this action to another or to go through another state. The state machine in the logical model system permits theoretically an easy adaptation even more if the appropriate tools are used to implement state machine integrated to the rest of the software.

Figure 4 shows the structure in levels of the simulator, emphasizing the levels in which some functional requirements can be adapted.

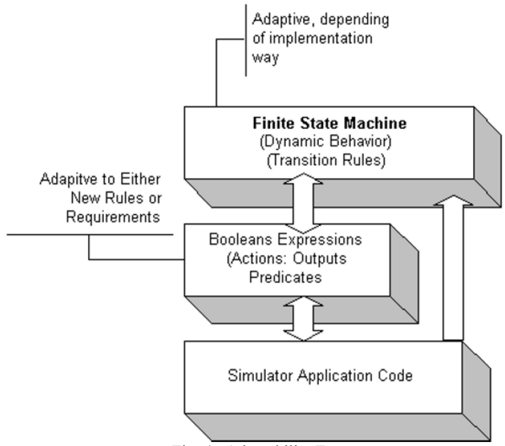

Fig. 4. Adaptability Features

A. Finite State Machine (Basic Definitions).

State machine consists in a powerful tool to the support of the dynamic modeling of general application in specific real time process control system. Essentially they model the dynamic behavior of process in which states well defined and its transitions of each other represents this behavior. From an initial state when variations occur on the inputs variable foreseen to the model, the state machine must process these variations taking the system to another state or remaining on the same. Normally are executed specific actions related to each transition predicted in the model of process. See below some basic definitions.

STATE is the condition that the system assumes in a certain moment.

EVENT is a relevant or notable occurrence happening from which transitions shall occur

TRANSITION is the relationship between two states indicating that an event has occurred. In the presence of a transition, the system will change from one to another. It is valid that a next state is the same one as the previous state in the event of a transition.

B. Implementation Aspects

This approach provides a simple and discernible way to write and implement logical control for many of complexes applications of the real world. [2]. The notation takes a state machines, and permits that complex logical controls represented by small diagrams can be however easy to be understand, verify, due to the simple rules that establish its fundamentals.

Notwithstanding it is necessary from a model to get to the real implementation of a state machine besides integrating it to the rest of the application. Implementation and definition tools of state machine are a lot useful in the development of application that makes the use of state machine. It is emphasized in this paper the compiler of state machine called ***SMC*** (***State Machine Compiler***). Information more detailed about this tool can be found on its guide on the website: ***http://www.oma.com.***

The compiler ***SMC*** truly developed by Robert C Martin and extended by Charles W. Rapp permits defining and implementing state machine in popular programming languages such as: *C++, Java and TCL.*

Basically the implementation process of state machines with the compiler is made though following steps.

Writing down the script ***SMC*** with the characteristic of the desirable state machine, it means that following the rules of compiler syntax SMC. To make all the transitions, states and requirements of actions predicted in the model of state machine clear. The SMC script file shall have the name like *< FSM_name>.sm*

To compile the script to a supported language (Java, C++, TCL), as a result, will have a code of a class of the target language.

To integrate the class the implements the state machine achieved from the script compilation. It's important to remember that all actions predicted should be implemented as a method of the application class, which can be called starting from the state machine.

Figure 5 below shows logically the previous flow, showing the SMC compiler role, within the process of definition and integration of a state machine in Java application..

As an initial result of adaptive rail interlocking simulator project, we present ahead a short sample of the implementation in its present level. Figure 6 shows the main GUI simulator equivalent to the Event Generator module and visualization already shown in the logical model of figure 3.

Basically we have three panels; the first forms the interactive part of the simulator. It is shown the layout of a rail-crossing yard as a reference to domain area of the interlock controller module being simulated.

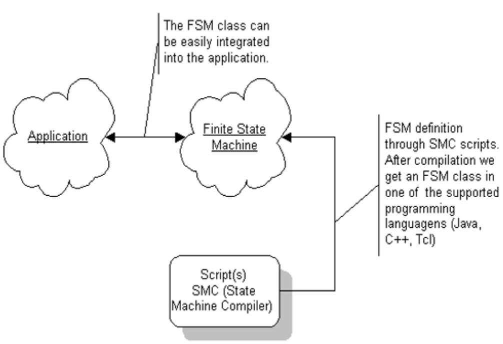

Fig. 5. Implementando uma Máquina de Estado com SMC

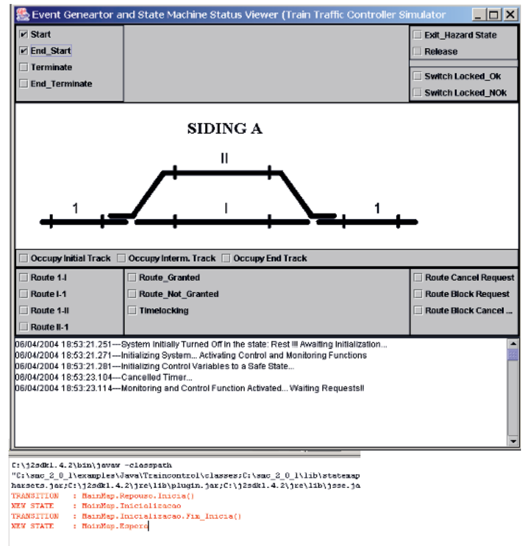

Fig. 6. Implementando Main Machine Interface (Event Generator Module)

The second one, which forms a massage area where massage allusive to actions related the exit of the simulator are visualized. The third Panel is a section where we can see the activity of the state machine of the controller module, that's to say, the event of received transitions and the respective transitions of state.

V. FUTURE WORK

The actual level of the described simulator in this paper represents an initial state that can evolve in several aspects. Two of them are particularly important to the effective consolidation of this simulator in the railway segment. Fault Tolerance support and improvements of adaptability characteristic are the goals to achieve in the future.

A. Fault Tolerance

The approach of state machines in fault tolerant system is usually based on its replication in distinct processors. We can use a configuration of triple modular redundancy (***TMR***) in which the state machine is replicated in three controller modules, see figure 7.

The principle in which the replication of the state machine as shown above, that's if these replicas executing in processors free of failure, starting from the same initial state and manipulating the same ordered group of input values, will produce equal outputs in each considerate replica.

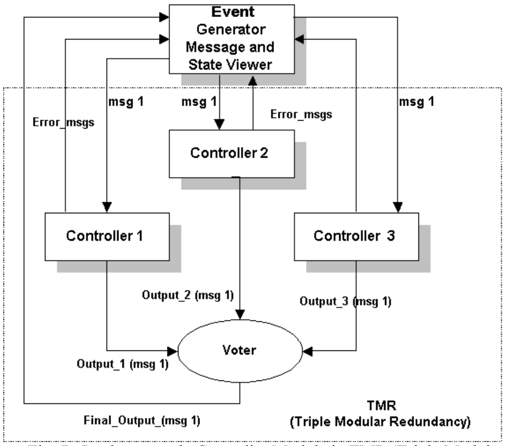

Fig. 7. Implementando Controller Module in TMR (Triple Modular Redundancy)

The usage of replication in fault tolerance requests that some conditions [3] are satisfied.

Entrance Consistency: All the replicas must receive the same inputs values.

Determinism of Replicas: From the same initial state the computation of the same ordered group of input data, must produces on all replicas in the absence of failures, the same ordered group of outputs values. The approach of state machines is the usual forms of guarantee this condition.

Figure 7 shows the logical architecture of the simulator in *Triple Modular Redundancy* (***TMR)***. In this case the Event Generator Module and State Viewer must send all information of control to the three controller modules as well as receive them. We have controller modules each one containing replicas of the same software version (state machine) that implements the control functions of signaling system interlocking simulator in itself.

Once generated an event (any one predicted), the same one has to be replicated to the three controller modules in asynchronous forms (because they are independent) should process the respective massages. The respective outputs also in an asynchronous form must be sent to the voter level, which will be done effectively a majority voting 2 in 3. It means that an output considered correct, will be generated if at least 2 of the controller outputs were identical. It must be guaranteed within appropriated techniques that the same information handling by voter is related to the associated inputs and outputs. Otherwise safety aspects would be compromised by absence of consistence. The process availability would also be compromised, because there would be frequent divergences in the voting process. Therefore the conditions presented before [1] need to be guaranteed.

B. *Extension Of Adaptability Characteristics*

The characteristics of rail traffic simulator adaptability presented in this paper, can be significantly improved if it is implemented an editor of Boolean Expressions, which represent the input and output predicates of the model. This module could also generate a *Java* code necessary for the implementation of the methods related to the actions of the outputs predicate invoked by the state machine. The state machine compiler (***SMC***), which has the source code free, would be a good reference for the implementation of this module. It presents analog characteristic, even for some other purposes.

VI. CONCLUSION

The conception of a *Rail Traffic Control Simulator* (***interlocking***) adaptive in terms of some functional requirements was possible with the use of state machines in its conceptual model. Suitable tools for the integrated implementation of state machine with application in popular programming languages, it's shown an interesting resource in the implementation even if preliminary of the prototype of this simulator.

The railway industry (signaling system segment) is an important area of critical mission application related to the control of process. This way the evolution of this implementation to support fault tolerance characteristics must be strongly considered. In general railway interlocking in real implementation is fault tolerant on hardware or software. There's a large amount of requirements related to the rail interlock, therefore an adaptive simulation tool in that sense will be relevant for that industry segment.

Improvements in the sense of turning this simulator more adaptive will be the objective of future researches.

ACKNOWLEDGMENT

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Resource Allocation and Cost Reduction by Means of Alternative Solutions

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Abstract - In the course of planning and implementation of projects it occurs frequently that after the preliminary calculation of the optimal resources allocation – in order to obtain minimal total cost – a project can not be realized at a price which is expected by the inviter of the tender.
In the following study we present a method by which, in case of a delimited maximal budget, we are able to determine the schedule, cost demands and resource requirements of a project to be implemented with the highest possible quality.

I. INTRODUCTION

In some projects the offered remuneration is lower than the estimated total costs. There are three possibilities to handle this problem. The first two is either to resign from the implementation, or to accept it knowing that we will loose money, but can regain it later in another project. In this latter case we have to determine the optimal resource allocation with minimal total cost. In the scheduling phase we can use some cost minimizing method. After that we can determine the lower and the upper bound of the start time of the activities. The next step is to determine a feasible solution and then the optimal resource allocation. The third possibility is to accomplish the project and find alternative implementation of activities that requires lower variable costs. During the search for alternative implementation of activities the most important aspect should be the quality, the decrease of costs is only the second one. If the total cost is lower than the offered remuneration we have to find the optimal resource allocation for the problem. Sometimes we cannot find alternative implementations, e.g. when a minimal quality level is given. In this case we can decide to either refuse or accept the implementation based on the extent of possible loss of money.

II. ALTERNATIVE SOLUTIONS

When in case of optimal resource allocation employed for minimal total cost during the preliminary calculation the expenses are higher than the budget earmarked for the implementation of the project then we have three options: we either give up the implementation of the project or we realize the project with losses or we replace some activities by new ones in order to reduce the costs. In the course of our analysis we will not consider the first option any longer since in this case we lose this business and there is no sense in making any further optimal resource planning. The second option is sometimes undertaken when they estimate that in spite of the initial losses the deficit will return during the implementation of the subsequent projects. In this case the allocation of the resources which is optimal for the given target function involving a minimal total cost shall be determined. (For the method refer to the study: Optimal resource allocation of active projects).

It is the third option we are going to study in this paper, i.e. we replace some activities to be realized with other activities in order to reduce the expenses.

In case of this third option, first of all, we have to make a list of alternative implementations of individual activities.

We should take into account three criteria when setting our objective (the order of the criteria also gives a priority during selection):

1. **first of all, we have to perform the given activities with the best possible quality**
2. **next the introduction of the alternative solution shall involve the greatest reduction costs**
3. **in the third place, the duration time of the activity and the demand of resources shall change at the slightest possible extent (they preferably get reduced).**

In addition to these criteria, for each activity a minimal quality requirement can be specified which in any case shall be met under the terms of the contract.

It is practical to arrange the alternative solutions belonging to a given activity in accordance with these criteria. From the list of the possible alternative activities to be implemented we should eliminate the ones which do not meet the quality requirements. During selection we reschedule the

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activities by using the alternative solution which is most suitable for the purpose and try to find the optimal solution again. The search of the alternative solutions is a selection problem determined for several target functions. We can find the optimal solution by the Branch & Bound method or by means of dynamical programming.

A The search of alternative solutions

The starting point of the method is the cost-optimal feasible allocation of resources or which is optimal for a given target function. We can find such a solution by determining a schedule with minimal total cost at the outset. If a non-critical feasible solution for this schedule is in existence, i.e. there is a feasible solution which does not amend the length of the critical path or does not overstep the limit of the resources then in accordance with the target function defined in Section II we should find alternative activities in a manner that during the employment of the alternative activities a non-critical solution must exist. If there is a non-critical solution then there is an optimal resources allocation as well and it can be found by a limited number of steps (see: Optimal resource allocation of active projects).

The method is illustrated on the following flowchart. In the first place we should make a timing to cope with the task. It requires the knowledge of the duration times and subsequent relations of the activities. We can carry out the scheduling by any method used in practice (CPM, MPM, Pert etc). Then we have to determine a solution with minimal total cost. This can be performed by any cost-optimising method used in practice (CPM/COST, MPM/COST, PERT/COST and so on). The utilization of the method requires the knowledge of cost demands, cost limits and the time/cost trade-off functions. We can estimate the time/cost trade-off functions or, in case of production of small series, we determine them by means of statistical methods with a given probability. (For the details of the determination refer to the study: Handling of uncertainty in the production management of small series). Then we should examine whether there is a non-critical feasible solution. If there is such a solution then, according to the previous paragraph, the resource allocation optimal for the given target function can be determined by a limited number of steps. If there is not such a solution then we have to find an alternative solution here as well, but the 2 and 3 priority points of the target functions shown in the previous chapter are to be switched.

The objective remains that we have to carry out the given activities with the highest possible quality, but in order to find a non-critical solution we have to realize an activity of that sort in an alternative manner, which has a fewer resource demands and/or shorter duration time.

If there is no other alternative solution then out of the solution list drawn up by the CPM/COST, MPM/COST and PERT/COST methods we have to pick out the solution which is next to the project plan with minimal total cost and to carry out the examination once again.

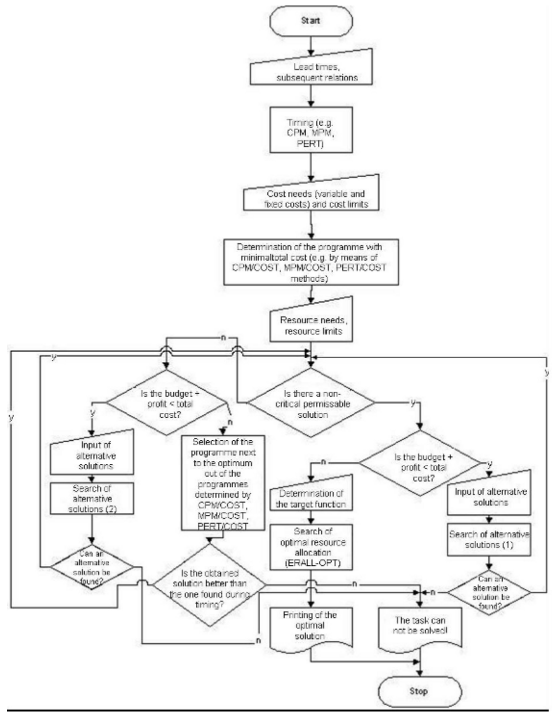

Fig. 1 Search of alternative solutions

Let us take a look at the following example:

Let us examine a project with the following activities which are shown in the table below.

Table 1

Activity	Normal time (t_{nij})	Rush time (t_{rij})	Normal cost (C_{nij}) (EUR)	Increase in costs/unit (ΔC_{ij}) (EUR)	Normal resources demand (R_{nij}) (person)	Increase in resource demand/unit (ΔR_{ij}) (person)
(1,2)	8	6	4500	100	3	0,5
(1,3)	6	5	12000	200	2	1
(1,4)	10	7	10500	150	2	1
(2,4)	6	4	7500	300	4	1
(2,5)	3	3	3500	-	2	-
(3,4)	7	6	1500	500	3	1
(4,5)	5	4	2500	650	1	1
Summary			42000	-	-	-

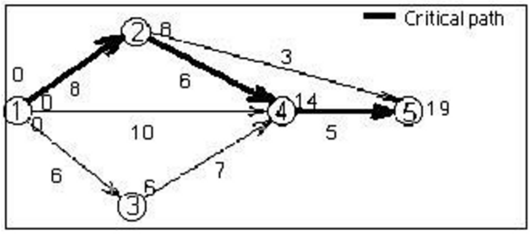

Fig. 2 CPM network diagram related to normal duration times

Let us suppose that the fix cost is 20000 EUR for 19 weeks. The reduction of the duration time results in a savings of 1000 EUR per week. In this case the project plan with minimal total cost is the project plan with minimal duration time, the total cost of the project plan is 43850 EUR (total variable costs) + 20000 EUR (total fixed costs) – 4 x 1000 EUR savings since we were able to complete the project earlier. Consequently, the total cost = 43850 + 20000 – 4000 = **59850 EUR**.

The following chart (table 2) demonstrates the possible solutions.

Table 2

Number of steps	Reduced activities	Reduction	Total variable costs (EUR)	Increase in costs/unit (EUR)	Total increase in variable costs	Total time of the project (week)
0	-	-	42000	-	-	19
1	(1,2)	1	42100	100	100	18
2	(1,2)+(1,3)	1	42400	100+200=300	300	17
3	(4,5)	1	43050	650	650	16
4	(2,4)+(3,4)	1	43850	500+300=800	800	15

Let us examine the case where the resources limit is 10 persons. Our available cost limit is 61000 EUR. In the second case the resources limit is 11 persons. Our available cost limit is 59000 EUR.

The following table 3 shows the activities which have alternative solutions.

Table 3

(1,2)			(3,4)			(4,5)		
q	vc/Δvc	r/Δr	Q	vc/Δvc	r/Δr	q	vc/Δvc	r/Δr
100	4500/100	3/0,5	100	1500/500	3/1	100	2500/650	1/1
90	4000/90	2/0,5	90	1000/90	3/1	96	2500/600	1/1,5

where ***q*** is the quality factor. When employing this method we do not have to use figures for the quality factors by all means. It is enough to arrange the alternative solutions in order of quality. We can, however, determine the target function easier when, according to some principle, we assign a quality factor to the alternative solutions.

If the quality of the implementation of some activities needs special attention then it is practical to designate our target function used for selection in a way that we weigh the quality factors of these important functions.

Let us examine the following case where the resource limit is 10 persons. Our available cost limit is 61000 EUR. The resource graph for 15 weeks is as follows:

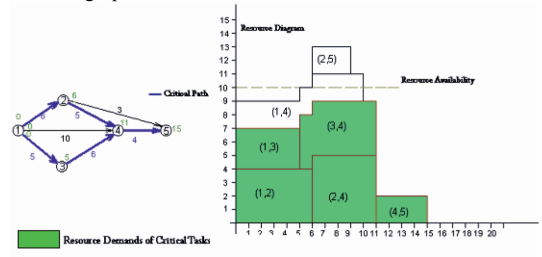

Fig. 3 Resource graph related to the shortest duration time of the project (there is not any non-critical solution here)

On the resource graph we can see that there is no non-critical solution. In order not to overstep the resource limit let us choose the 3rd production plan from the solutions obtained by CPM/COST.

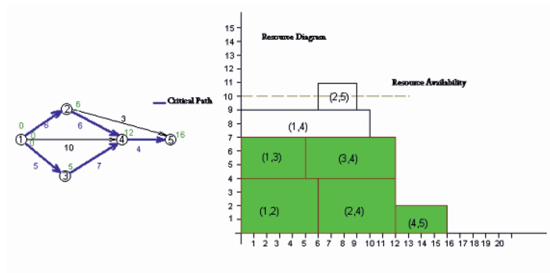

Fig. 4 Resource graph related to a longer duration time

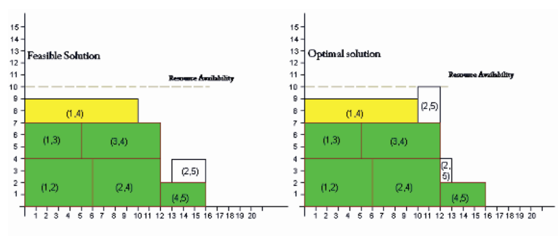

Fig. 5 Non-critical feasible and optimal resource allocation

During the determination of the optimal resource allocation the objective was to achieve a start as early as possible. At that time we did not have to reach a compromise concerning quality. The total cost: 43050 EUR (total variable costs) + 20000 EUR (total fixed costs) – 3000 EUR (savings in fixed costs due to the shorter duration time) = 60050 EUR < 61000 EUR (budget).

Unfortunately, in the second case we have to carry out the activity(ies) at a lower quality level in order not to overstep the limit of the planned costs.

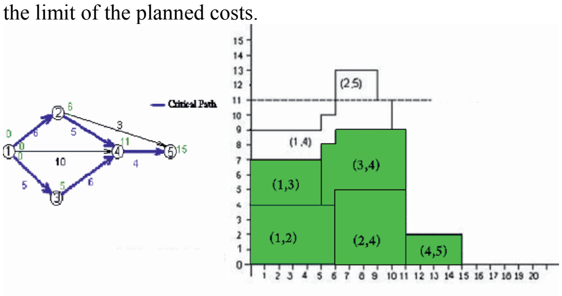

Fig. 6 Resource graph related to the shortest duration time of the project

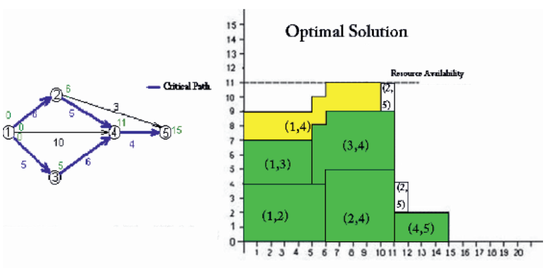

Fig. 7 Optimal resource allocation

The total cost is 59850 EUR as calculated earlier which exceeds the 59000 EUR budget. If the (3,4) activity is replaced by an alternative one then the resource demands and the duration time remain unchanged but the cost is reduced by 900 EUR. In this case the total cost will be 58850 EUR which is less than the available budget.

III. Summary

The tougher the competition for the implementation of a project, the greater the chance that we have to implement the project at a low price which does not cover our expenses. Companies are often not fully aware of the costs of the implementation of the given project and frequently undertake the realization of the project and find out only later that their expenses will be much higher than their preliminary estimation.

If the contractual incomes do not cover the expenses then we either give up this business or undertake it even if we know that we can realize the project with a loss or try to replace the activities by alternative solutions. In such a case we should take into account that the activities are to be implemented at the highest possible level.

IV. References

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