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Research and Development of Multi-Regional Monitoring Integration Technology Based on SIP Protocol

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Abstract: To solve the system overloading problem of sip protocol based on centralized monitoring and management system, we propose a new protocol system based on cross-regional stratification sip monitoring integrated management platform, it uses the automatic control technology, networking and communications technology, video / audio compression and transmission technology, sensors and integrated control technology and software engineering techniques. It is an integrated platform of biometrics, intelligent video analysis, anti-theft alarm techniques, 3G video management, valuables management, multifunction video and voice network, multimedia video services, and that it can be used with existing IP network, LAN, internet, 3G network, the telephone network. It is fully compatible and can be used with existing databases, monitoring systems, television networks for docking to achieve a variety of monitoring elements of centralized management, control and interaction. The innovative applications of CSS regional stratification mechanisms, ActiveX terminal buffering mechanism and VM resource allocation mechanism can provide users real-time information. And according to a variety of real-time information, it aggregated emergency solutions of the overall regional security or certain high-risk target key places of security.

Introduction

With video monitoring "Big Networking" trend started and the national standards formally implemented，it indicates that video surveillance "Big Networking" has become an inevitable trend. The technique is an integrated intelligent video monitoring management platform, it is based on a national cross-regional SIP standard protocol. It uses the automatic control technology, networking and communications technology, video / audio compression and transmission technology, sensors and integrated control technology and software engineering technology. It is a set of biometrics, intelligent video analysis, work order management, anti-theft alarm, 3G video management, valuables management, network video and voice-in-one multimedia video services .It is based on an integrated platform that can be used with existing IP network , LAN, internet, 3G network, the telephone network is fully compatible and can be used with existing databases, monitoring systems, television networks for docking, to achieve a variety of monitoring elements of centralized management, control and interaction, can provide users with real-time information, and based on a variety of real-time information aggregated emergency solutions for the overall regional security or certain high-risk target key areas of security.

Technical Backgrounds

1 SIP Services

SIP (Session Initiation Protocol) is known as Session Initiation Protocol, it is proposed by the IETF (Internet Engineering Task Force) organized in 1999 as an IP-based network, in particular it is a configuration such as the Internet network environment, real-time a signaling protocol network communication [1]. SIP protocol implementation is relatively simple, and it is the international field of multimedia session control application trends.

In the main communication of GA/T669.1-2008 "urban monitoring and alarming network system technology standards" for various interconnection agreements between the monitored area, SIP service can be two or more of the monitored area networking to meet the video and audio alarm information interoperability [2]. Therefore, this platform can provide external SIP service as a standard interface. While the intelligent scheduling system (GIS map bus) to provide interactive interface.

SIP service to control the transmission of information between systems in a monitor area. Networked systems based communication protocol to establish two transmission channels: Signaling / control channel and video and audio stream channel. Signaling and control channel is used to establish a session between the device and transmit control commands; the video and audio stream channel for the transmission of video and audio data, the compressed video and audio streams encoded using streaming media protocols RTP / RTCP transmission [3].

SIP protocol is used for central management server and streaming media servers, storage servers, and client session control protocol between its control principle, such as a simple real-time video transmission process [4] is shown in Figure 1.

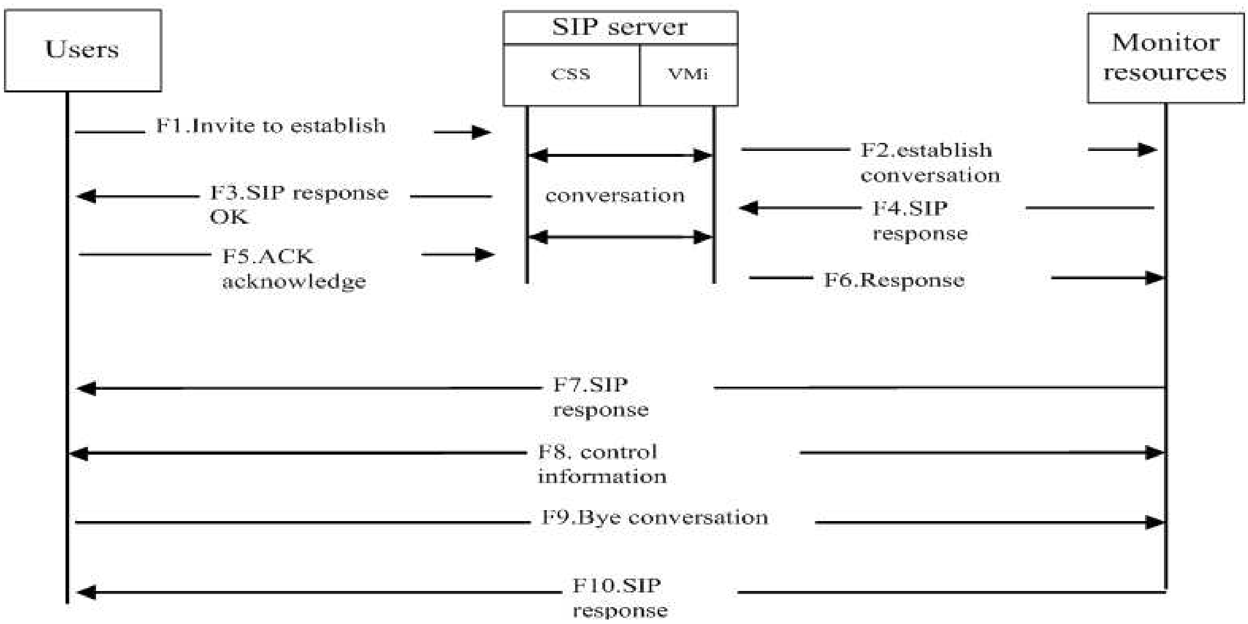

Fig.1.SIP Control Order Transmitted Flowchart

2 Major Problems to Architect the Platform

1) Core equipment overloading problem. It is not conducive to further expansion, and it can not meet the customers' business needs.

2) Network architecture complexity. All the video surveillance system is independent, decentralized, did not achieve interoperability, resource sharing.

3) Don't have a comprehensive management platform for integration. To achieve to strict management of image calling, equipment control, information management, for traffic control and other types of information, and so on.
4) System Compatibility is poor. It can not meet the interface unity, and it is not easy to be compatible with other systems.

Designs and Implementation

1 System Introduction

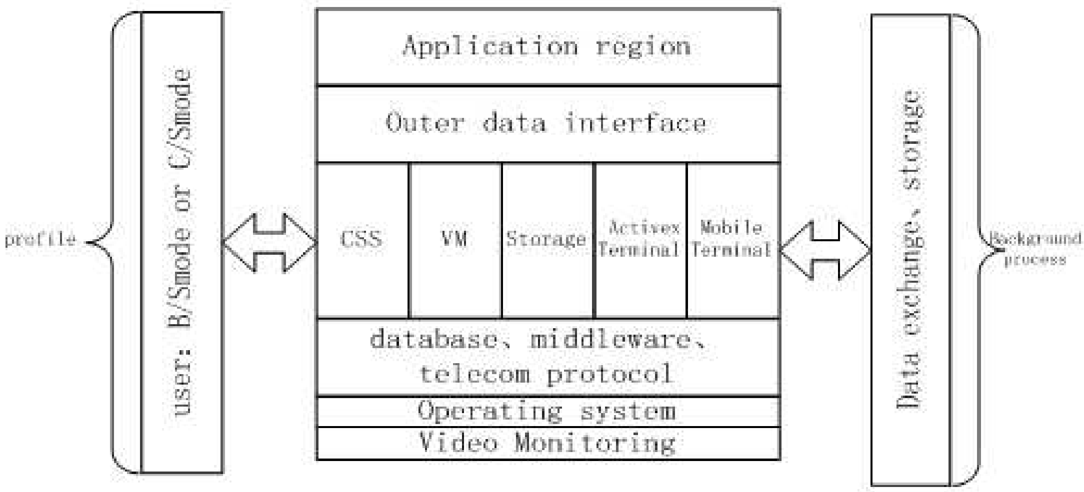

Fig.2 System Framework

The video surveillance system consists of front-end video monitoring, network video streaming servers, video monitoring clients and SIP servers, is show in Fig.2. Among them, the video monitoring end is composed of the video server and camera. It mainly completes audio video capture, compression, forward, and the transfer from collected data to the storage server, etc. CSS server system mainly comprises a SIP server system, the location server, the proxy server , registration server, control server and alarm servers. When the monitoring client send the camera monitoring request to find Internet video streaming server, and obtain the appropriate data stream, network video server via RTP protocol to establish a connection with video surveillance front, it is built in SIP-based signaling systems; it can also be seen as a UA client. When the video monitoring client and network video server is turned on SIP servers, it needs to be registered and regularly updated, so that the monitoring terminal can access to the location of the video server.

2 CCS Server

Central management server bears the manager's role in the whole system; its main function module has the following parts: SIP services, security authentication management, resource management, alarm management, logging and management, log management. SIP-based network monitoring in addition to the new CCS gateway needs to have the following sections:
1) Monitor resource layer: Contains all monitoring equipment and servers such as PTZ, matrix, DVR (Digital Video Recorder), DVS (digital video encoder) and so on.
2) SIP access layer: in the resource layer, based on a SIP-based networking protocol registration, authentication, on-demand, control, playback, forward, etc. functions.
3) Monitoring Management: Monitoring by the SIP network services, Web monitoring services, video signaling proxy forwarding services and other related services, to achieve front-end device

management, user management, video scheduling and other functions, it constitutes to the control system software management platform.

4) Monitoring application layer: The main face of the end users, administrators, and other with the system. Further comprising: user interfaces, the administrator interfaces, WEB application interfaces, Internet applications interfaces.

3 ActiveX Terminal Buffer Mechanisms

ActiveX controls the process of video, decoding, display, it receives the required dynamic link library and other documents into *. Cab file format, the *. Cab file format can store multiple compressed files for a Web page while loading. Generated ActiveX embedded in Web pages automatically, when you visit the client to download and install the client through the browser can display and operate. Video transmission receiver module is packaged in accordance with the RTP protocol in order to send the video to the client, the client receives the video according to the RTP protocol to unpack and play the video.

4 Video Mixer Video Streaming Server

SIP in CCS gateway is an application-layer control protocol, and Video Mixer network video streaming server via the RTP protocol achieved audio and video communication in three steps. At first, streaming media server for CCS server provides SIP negotiate session parameters, and then through the RTP / RTCP and RTP ActiveX client establishes a connection to transmit audio and video data packets, and finally discharged via SIP connections to end the session. RTP provides services mainly include transmission control, data series, time load identification and timestamp, the underlying data using the UDP protocol to transfer [5].

When creating an RTP connection, RTP and RTCP (Real-time Transport Control Protocol) will assign one port for each. The RTP session, each participant RTCP packets transmitted periodically. RTCP packets contains the number of packets sent, number of packets lost and other statistics, based on these information, the server can dynamically change the transmission rate and payload type. Two agreements with the use of feedback and make smaller overhead transmission efficiency optimization effectively.

Lower RTP protocol using UDP protocol, while UDP protocol is not guaranteed transmission quality, if a package is split fragmentation due to network congestion appeared lost, the receiver can not be easily re-UDP datagram, which causes the entire RTP packet is discarded. Therefore, the length of the RTP packet network MTU control within the allowable range. MTU is usually 1 500 byte, remove the RTP and UDP datagram packet headers, NALU best controlled at around 1 400 byte. Thus, when the need to send the NALU size exceeds 1 400 byte, the first frame image data which cut into several segments, and then follow the protocol RFC3984, each section will be packaged into an RTP data packet to send, an image was divided into multiple an RTP packet to be transmitted.

Cases and Analysis of Test Results

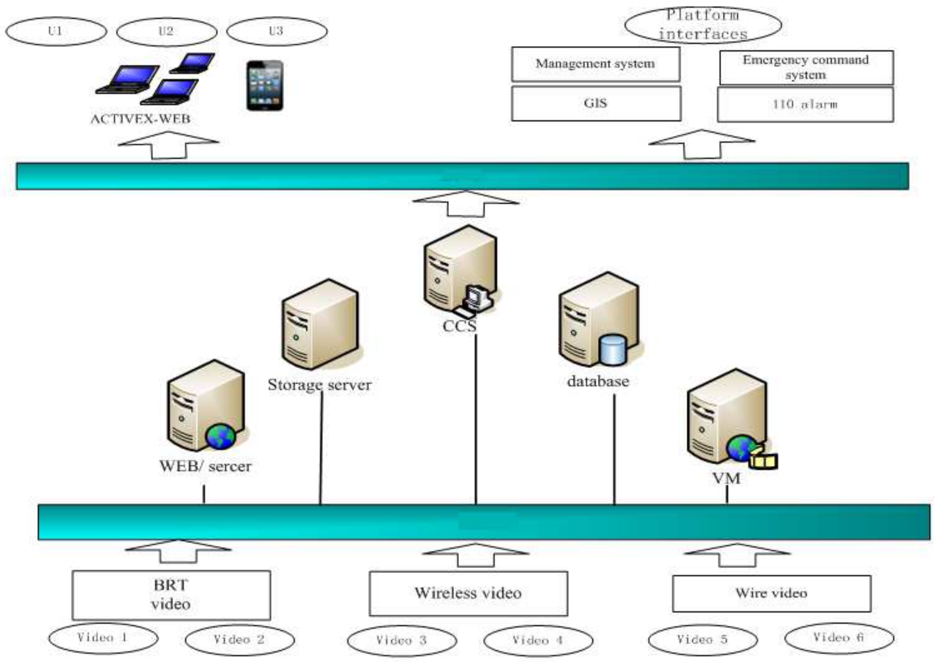

Fig.3 Video Platform

1 Video Stream Establishment

In cases we illustrate the video session establishment process, is show in Fig.3, assuming that the request for the video U1 promoters, requesting two and three-way video, video conversation detailed build process as follows, is show in Fig.1:

1) Start establishment of Client VM waits video requests, and F1 INVITE is sent to CCS requirements established by the U1 video connection requests, which explicitly given URI-list to request a two and three-way video connection.

2) When the CCS receives F1 message, according to the INVITE request URI-list and access to video according to VM resource allocation mechanism, according to equation (2) elect the lightest load, and satisfy the formula (1) of VM (i), finally the transmission F2-VM_NEW_SES-SION to the selected VM (i), the ready to initialize the video work is done.

3) Then, CCS returns F3 200 OK to the U1, U1 has successfully pleaded inform video, U1 before returning F5 ACK message acknowledgment. When U1 complete video receiver initialization, CCS re-transmit F6 VM_ENABLE_SOURCE to VM (i), and the U1 requests video turned to VM (i) deal. VM (i) return has been taken to transfer the video to U1.

4) When the video is established, the requesting VM start sending video images and voice streams, in order to simplify the complexity, Figure 1 represents the data flow direction of the arrow, and text What is included. VM (i) will send sound and video through the RTP and RTCP protocol to Audio Stream and Video Stream H264 encoding format sent to the U1 client; as for other clients, if CCS VM (i) maximum load of other people and then turn to load the video stream requests lowest VM (j) treatment, then, CCS will send the successful completion of the video message notification. Finally, all mixed VM (i) video streams all requests forwarded to the client, so that the requester can see the video image.

5) VM (i) in the transmission between the video when the U1 through Transmission Control Protocol to prevent network congestion, when the VM connected via CSS hears F9 to remove the session connection to U1 response message returned after removal F10 video session. If there are

users U2 and U3 request re-transmission of video and VM_ENABLE_SOURCE VM (j) of VM_ADD_SOURCE message to the CCS, and U2 and U3 to transfer images to a VM (j) handling, and VM (j) return has been taken to transfer video to U2 and U3.

2 VM Resource Allocation Calculations

Example shows how to allocate VM resources dynamically. Assuming a video initiator, inviting eight-channel video, the VM build number of a person's new video resources (n '= 8), the current state of the system shown in fig.3. The current system has two VM, numbered 1 and 2; 8 people 9 people and 10 people are requesting; 1000 video request numbers. The video data is composed of the equalization processing VM1 and VM2. First, determine the formula (1) is established, in this case, VM1 and VM2 satisfy the formula (1). Secondly, to meet the formula (1) under the condition of formula (2), select the minimum load to be responsible for the new VM1 requested video work forward.

$$n' \le c_i - \sum_{j \in R} x_{ij} n_j, i \in I \qquad (1) \qquad\qquad \min_{i \in VM} \frac{\sum_{j \in R} x_{ij} n_j}{c_i} \qquad (2)$$

3 Experiment Results and Analysis

SIP-based prototype system has been implemented to complete functional testing and acceptance, preliminary testing LAN average video session setup time is about 0.8 seconds, it is completely acceptable time limits. Experiments were random requests from eight, nine and 10-members with 1000 discovery channel video, the audio and streaming audio and video streaming is very smooth, VM1 load of 0.25, VM2 load of 0.5, consistent with their maximum capacity, VM dynamic allocation mechanism is more reasonable.

Conclusions

This paper presents layered monitoring and management platform system, Video Mixer modular as a CCS server manages multiple streaming media server and DVR resources. By monitoring system application implementation, verification system Video Mixer dynamic resource allocation functions. And the client based on ActiveX technology to achieve an audio and video surveillance performance. Using SIP for signaling control the streaming media technology and G.711, H.264 audio and video technology such as merging, making the system not only through the local area network but also operating within the WAN and has excellent codec efficiency and flexible network adaptability. After testing, the design of client-side performance is stable and reliable, the boot program starts normally, and the operating system is running stable. Finally, in order to achieve the CCS IXP465 embedded systems and part of VM, the limited hardware conditions, verify system with functionality and feasibility. SIP-based video surveillance client operation is simple, you can always access the video surveillance server, the video screen is clear and smooth, no connection limit, no streaking phenomenon.

As the system is based on a unified SIP protocol, simple structure, stable performance, low cost, etc., has the versatility, will have broad application prospects. For the future performance of the overall system will be more in-depth research and analysis, I believe it will help video surveillance fusion technology and industrial development.

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