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An evaluation of perceived urgency applied to amplitude modulated stimulus for military applications

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An evaluation of perceived urgency applied to amplitude modulated stimulus for military applications

by

Tyrone D. Moore

A thesis submitted to the graduate faculty

in partial fulfillment of the requirements for the degree of

MASTER OF SCIENCE

Major: Mechanical Engineering

Program of Study Committee:
Eliot Winer, Major Professor
Gloria Starns
Stephen Gilbert

Iowa State University
Ames, Iowa
2011

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ACKNOWLEDGEMENTS

I would like to take this opportunity to thank the following people for their contributions and support Dr. Eliot Winer, my major professor and chair of committee, for providing motivation and instruction, Dr. Gloria Starns, committee member, for invaluable support with university procedures, Dr. Stephen Gilbert, committee member, for his valuable perspectives and recommendations.

In addition, I would like to thank my parents, Mittie and Joesph, for their support, confidence, and motivation for me to pursue my passions. I would like to thank all my siblings for their continued support and motivation.

Finally, I would like to thank Battlespace and the U.S. Air Force and GMAP for funding my educational pursuits.

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CHAPTER 1. INTRODUCTION

The work presented in this thesis is intended to provide an overview of augmenting visual virtual reality environments with sound. Immersive virtual environments contain visual information spatially separated. This spatial separation decreases the ability to comprehensively asses the visual content in these environments. To aide in the creation of a cohesive image in virtual environments, augmentation is required. Augmentation is intended to add information or offload some of the visual information as aural information. Adding information may be done to enhance the realism of the virtual environment, or to provide the user with additional input, and offloading some of the visual information may be done to allow the user both cognitive access to all information being communicated in the virtual environment.

There has been significant work done on using aural cues in several environments, including virtual environments. In the field called sound quality there is a large body of work studying how sound is perceived in relation to the quality or annoyance of a device. This is contained in a more general knowledge framework which focuses on the response of humans to sound, a field called psychoacoustics. All of these are centered in a field involved with modeling and quantifying the physical parameters and behaviors of a sound within its environment, a field called acoustics.

The work presented in this thesis explores issues relating to the communication of information with sound. A review of pertinent literature and presentation of limited experimental work is intended to lay the ground work for future work.

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1.1. Human Senses as a Communication Tool of Information

Key elements of virtual reality consider all the senses available to communicate information, and the distribution of that information as efficiently and effectively as possible among those senses. A loss or limitation of information is caused by the over stimulation of one sense, leading to an inefficient translation of information. A review of the senses and the current instruments which access their systems provides a general background to the research described in this thesis.

From birth, humans experience phenomenon interpreted through five sensory modalities: touch, vision, audition, smell, and taste. The sensory modalities, which are yields of sensory organs, the physical devices such as limbs, eyes, ears, and the tongue, are receivers of information from the outside world. Our nervous system fused to sensory organs transforms stimulus into conceivable and interpretable sensations (Bernstein, 1893).

For example, the eye is solely an optical instrument, a recipient of light; the optic nerve is the conduit for transporting this light to the brain, and the brain is accountable for the transmission of light to an image. The sensory organs, in concert with the nervous system, provide meaning to our experiences and expedite the learning process. Berstein, (1893) states, an eye with an optic nerve destroyed, cannot present a picture, an ear with a severed auditory nerve, conducts no sound, and a leg with no nerve endings feels nothing. Essentially, the sensory organs are a device that through the automation of the nervous system assigns meaning to the data received by the device (Bernstein, 1893).

The hand is the most important organ to touch, as the eye to sight, ear to hearing, nose to smell, and the tongue to taste (Bernstein, 1893). Exploiting these organs induces a higher degree of arousal to the sensory organs; this creates a higher degree of stimulation to the

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nervous system. The more stimulation invoked by the nervous system corresponds to the unit of information being transmitted and received to the sensory organs. Nature naturally exploits these senses by appealing to the five sensory modalities. This work will focus on the audition modality to communicate data through sound stimuli.

If one considers a walk in the park, the sense of touch corresponds to the feeling of the ground below the feet, while the brain deciphers the surface type. The scenery appeals to the sense of sight, the air appeals to the sense of smell, and the ambient noise to sense of hearing. The summation of these senses contributes to our interpretation of the overall experience. Although only four of the five sensory modalities are being exploited one can assemble a comprehensive perception of the surroundings. Humans have used these fundamental observations in the construction of systems from rudimentary to complex. Blanchard, (2006) states a system is an entity that interacts with the outside world for a specific role or function. A system can range from consumer based product to a theoretical construct.

Products designed for everyday use have sought to exploit these senses, and has led to technological advances previously thought inconceivable. The transition from standard definition to high definition in television has brought more stimulation to the visual modality; The evolution from a stereo sound field to a Dolby™ digital surround sound field provides more stimulation to the audition modality. Independently, each of these technological advances provides more stimulation to its respective modality, but collectively they contribute to an overall enhanced experience. This experience is defined by the amount of content one receives to make a synthetic experience more realistic. The more realistic or engaged one feels is directly correlated to the degree of arousal or sensory stimulation.

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Consider driving a vehicle, where the modality of touch is exploited through our hands on the steering wheel, the accelerator and decelerator on the soles of our feet, and our limbs in the seat. This modality of touch is used to provide information regarding the status the vehicle and contributes to our sensed comfort, based the smoothness of the ride, the vibration in the steering wheel, and the response of the accelerator and decelerator. The modality of sound is provided by the ambient environment, which is composed of the engine noise, tires on the road, and other extraneous noises. The modality of vision is stimulated not only by the road, but also the interior of the vehicle and the control panel. The modality of smell is also incorporated into a vehicles operation; any smell that deviates from pleasant indicates a problem with the vehicles operation. Exploiting these senses provides the maximum stimulation to our nervous system by continuously streaming bits of information through our sensory organs. Based on this stream of information, we respond and make decisions regarding the driving experience.

Imagine dampening a modality from the driving experience, a reduction in the sense of touch can be achieved through utilizing cruise control. Cruise control is a rudimentary form of automation, which controls the vehicles velocity. In this scenario the sense of touch is removed from the soles of the feet, causing unanticipated changes in the accelerator and decelerator. The reduced sense of touch is mitigated by a heightened sense of the sight and sound. The sound of the engine and control panel now serve a more critical role as a continuous feedback device, to indicate changes in the vehicle's velocity. When designing a virtual environment a lesson from humans operating in the natural world is that multiple senses are used, and some would argue necessary to function effectively. Further, the goal of automation currently is to improve system performance, which consequently dampens the connection of a user to the system, an issue which will be discussed in Section 1.3.

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1.2. Virtual Environments and Automation

Virtual environments are computer generated synthetic environments being developed for applications such as training, engineering design visualization, and controlling Unmanned Aerial Vehicles (UAV's) in combat zones (Metzger, 2005). These environments contain complex sets of visual information, and currently rely primarily on the visual modality. This high concentration of visual content can overload the visual modality, while underutilizing all other modalities (Kaber, 2006). This requires introducing additional modalities such as sound (Kaber, 2006). The virtual environments typically contain a high degree of automation due to the complex systems being controlled. The automation is the sequence of events that transpires in the virtual environment (Lee et. el., 2004). The coupling of a virtual environment and automation presents the issue of sensory deprivation, which does not allow one to acquire the continuous streams of data in all modalities.

Automation is the technology that actively selects data, transforms information, makes decisions, or control processes (Lee et. el., 2004). Every facet of our life has some form of automation embedded into its architecture. A keyless entry remote used to unlock a vehicle requires an operator compression of an unlock button, this compression is transformed to a signal, the signal is sent to a receiver, which makes a decision to unlock the door, an auditory alert confirms the success of this process, and upon arrival at the vehicle the door is unlocked. Consider typing this sentence, each key depressed, involves a series of commands, where the data is transferred from the keyboard, thru a USB cable, to a motherboard, uploaded into software, and output on a screen. The complexity of a system is often ignored, since automation has become such common place, it is typically taken for granted, until a failure occurs.

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Automation allows humans to manage complex systems efficiently; however automation systems must be appropriately designed. Imperfect association between automation and people will yield catastrophic results (Lee et. el, 2004). For example, pilots trusting the ability of an autopilot, failed to intervene and take manual control even as the autopilot crashed an Airbus A320 (Sparaco, 1995). The link between trust and automation is determined by the reliability, predictability, and dependability of the system (Moray et. el, 1994). Reliability in an automation system refers to the consistency in which the system performs its intended function. Predictability is the accuracy and precision in which the function is performed. Dependability refers to how often the function is performed when prompted to do so.

Automation introduces a high degree of disassociation between the operator and the systems operation (Metzger et. el, 2005). This dissociation shifts the role of the operator from manual control, to a supervisory control. In this supervisory role, communicating any deviation from a systems ideal state relative to its actual state can provide operators with information to make critical decisions. Further, continuous feedback or trust that the automation system is functioning is important to the operator. Sound cues can be used to communicate and act as confirmation of the automation processes operation.

Every automation system cannot exploit all senses, and it is the systems maturity and complexity, which correlates to the modalities to be employed. Typically, through the evolution of a system different modalities are introduced. Consider the evolution from the transistor radio, black and white TV, standard definition TV, high definition TV, and now 3D TV, which has not only introduced different modalities and dimensions into the viewing experience, but enhanced the arousal of our visual and auditory senses. In a visually

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intensive task, such as flying an aircraft, additional modalities can be exploited to effectively communicate critical information to an operator.

In a typical aircraft, the cock pit contains an altimeter, attitude indicator, airspeed indicator, magnetic compass, heading indicator, turn indicator, vertical speed indicator, course deviation indicator, and radio magnetic indicator (Ahlstrom et. el, 2002). This instrumentation requires the operator's attention at any given time within a flight. Some of the instrumentation is embedded into a flights automation system, while others require the operator's manual focus. In addition to the actual flight operation of an aircraft, a pilot can become visually saturated with instrumentation, leading to a loss of critical visual information. To minimize this risk an additional modality, such as audition, can be introduced, to communicate some of the information currently contained as visual information.

Monitoring the flight instrumentation devices can be offloaded from the visual modality to the audition modality (Blattner, 1989). Sound can be embedded into a system's architecture to communicate deviations from an ideal operational state. Deviations from this state can be mapped to the dynamics of a system to provide a continuous feedback or a system status to an operator (Edworthy et. el, 1999). This alternate modality can reduce the risk of visual saturation. If the sound is appropriately mapped to the dynamic behavior of the system, then an operator in a supervisory role can be continually receiving data regarding variations in the systems behavior (Bliss et. el., 2000). This data can be used to make critical decisions and mitigate the risk of catastrophic failures.

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1.3. An Application: Autonomous Vehicle Control

The application for the work in this thesis is augmenting the virtual environment used to control autonomous vehicles. As the technology increases to display information, the capabilities of autonomous vehicle control increases. In addition to computer, visual, and tactical displays being continually advanced, sound is an additional communication medium that needs to be developed.

Historically, the operation of aerial vehicles required two operators: mission intelligence and a navigator (Parunak, 2003). Advances in control systems, now allow the operation of multiple aerial vehicles by a single operator (Parunak, 2003). It is projected by the year 2020, that remotely piloted vehicles will be operated fully autonomous (Parunak, 2003). Advances in automation will lead to a further disassociation when automation executes an unexpected action, or fails to execute an action (Donmez et. el, 2009). This scenario decreases operator confidence in the automation system (Lee et. el., 2004). To place the operator at ease, or in a state of awareness, an auditory monitoring system can be used to compensate for this dissociation. An operator's awareness of the situational status at any juncture in the automation sequence will provide a pilot with real time data (Moray et. el, 1999). The concept of introducing an additional modality to compensate for a loss of information can be extended into any automation system or system in general.

When controlling a UAV, multiple systems and information sources are monitored (Parunak et. el., 2003). For some of these systems and information sources it is important or comforting for a UAV operator(s) to know the system is operating as expected, or the information on which a decision was based has not changed (Stephen et. el., 2006). Likewise, the operator needs to learn of changes before they reach the alarm (emergency)

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stage (Bliss et. el., 2000). The alarm state indicates a critical operational state in which a failure is imminent, such as an attack on the UAV.

In a combat setting the status of threats and mission objectives are in constant flux. This dynamic behavior of the combat setting requires precise and timely information. UAV's serves as the first line of defense, gathering information for mission intelligence for the development combat strategies. These vehicles survey terrain, identifying any potential threats prior to troops being deployed in the area. If the UAV is lost or intelligence relayed through the UAV is inaccurate it will likely result in the loss of human life.

Since UAV's are remotely operated, the view is only through the lens of the camera attached to the vehicle, which creates a skewed perspective. A lens provides a small percentage of the information, compared to a pilot in the aircraft. This limits the sense of realism, which translates to a loss of intelligence that can be gathered. To emulate the environment surveyed by the UAV, immersive virtual environments can serve as a tool to reconstruct an environment from a pilot's perspective in an aircraft. Battle Space is a project aimed at constructing a virtual environment which allows the operation of UAVs embedded within this environment.

A virtual environment has the ability to reconstruct any terrain, travel through this terrain, and update information regarding this terrain. An advance in remotely piloted vehicles allows information to be sent to a virtual environment from multiple sources. Thus, a squadron of UAVs will soon have the ability to communicate with one another as would a squadron of human pilots. In addition to the spatial separation of threats, UAVs, as represented in the actual environment, will have the ability to be reproduced in the virtual environment. Battle Space is building the framework for this reality.

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Within the application of autonomously controlling vehicles, two issues were identified by researchers as potential starting points. One is the communication of the health of a system, and the other is to monitor a secondary task. System health includes: proximity to threats, fuel constraints, path taken by the UAV, and operation of the communication with the UAV. The secondary task includes for example, when pilots are controlling multiple UAVs. After one UAV is sent on a mission, the attention of the pilot is then set on launching a second UAV on a mission. While performing the task associated with second UAV, the pilot needs a mechanism to monitor the status of the first UAV. For example, that all is proceeding as planned or there has been status change in the first UAV. The monitoring of the first UAV is a secondary task, which must be done in such a manner as to not degrade the capability of the operator to launch the second UAV. Sound is being proposed to monitor system health and communicate the status of a secondary task.

1.4. Examples of Communicating with Sound

In the natural world we interpret sound by two quantities, (1) quality commonly referred to as its pleasantness, and (2) its location. Consider the example of the walk in the park each of the sound sources are at locations spatially separated from one another. A bird chirping in the tree stimulates our aural senses in two ways. First, the frequency and temporal content can help us identify the bird, or just feel relaxed on a spring day, but the sound also contains information that our aural system can use to identify its location. Immersive virtual environments are being constructed to mirror the natural world with a high degree of accuracy. The examples of using sound in an immersive environment can be

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separated into (1) spatial sound, commonly called 3D audio, and (2) the perceived quality of a sound.

3D sound can be employed to promote interaction in a 3D environment, 3D audio employs the knowledge of existing auditory display techniques within a spatial reference frame. Localization, manipulation assistance, movement assistance, spatial alerting, and spatial tracking are the five general groups in which 3D audio can be employed. Localization refers to the identifying the spatial location of a sound, such as identifying a bird in a tree or a person walking behind you. Manipulation assistance refers to using spatialized stimulus to assist in user related task. Movement assistance refers to using 3D audio to simulate performing well defined movements or spatial tracking. Spatial alerting refers to communicating a sense of urgency spatially. This will employ knowledge of existing sound quality principles in conjunction with the head related transfer function. Lastly, spatial tracking could be employed for object tracking beyond the visual field, essentially using sound as the only modality to communicate data (Gonzalez et. el, 2010).

3D audio allows the navigation of an environment through sound, just as in the natural world. This allows an additional degree of freedom in the construction of sounds, the degree of localization. 3D audio provides a spatial reference frame which allows the communication of sound source localization. 3D audio falls with the domain of "audio annotation" is an extension of Spatial Audio Display that was first developed in the late 1980's. Accurate 3D audio is difficult to accomplish in practice, it not only models the sound pressure generated at a person's ear, but also models the human response, and the sensation produced by each sound source (Brungart, et. el, 1998).

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In order to model how sound is heard in a spatial environment, modeling of the head, shoulder, outer ear, middle ear, and inner ear must be done. There are several methods that have been employed to accomplish such a feat, such as the head related transfer function method (Brungart et. el., 1998). The general methodology is to determine the spatial location of a sound; this is accomplished through creating a single sound stimulus, and applying filters to the stimulus to simulate different spatial locations. The filtering applied represents a spatial location for a side incidental wave, which is the method of delivery for head phones. There currently is no consistent standard on the filtering applied for spatial locations, only observations based upon experimental data.

Spatial audio and virtual environments perfectly align with one another in terms of application. Since virtual environments are 3D, localizing content in this environment represents the natural world. The degree of accuracy in which this environment can be reproduced, aligns with the degree of realism. Virtual environment have the ability to reproduce natural environments visually with a high degree of accuracy. 3D audio is suboptimal at best, the concept of 3D audio is its developmental stages. If one considers the desire of military forces to have autonomous UAV's, then the representation of a natural environment in a virtual environment will only facilitate this process.

To date there has been several documented successes of implementing 3D in a military application. A practical application of 3D audio was applied at Edwards Air Force Base, where under laboratory conditions pilots could spatially locate sounds, but in flight conditions they were able to identify only a small percentage of the same sounds, due to the high load (Ericson, 2005). 3D audio does have vast potential aircraft applications, but an

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observation such as this reveals the requirement to consider additional factors such as cognitive load.

Aural cues applied to military setting are not limited to the 3D, but can also be implemented through closed ear headphones. For example, spatial disorientation is the leading cause of accidents for pilots; this disorientation causes pilots to become confused about their physical orientation relative to earth (Brungart et. el., 2008). Brungart et. el., (2008) found through altering auditory stimulus, the attitude indicator could be offloaded from the visual modality to the audition modality. This provided a continuous stream of data which communicated to the pilot the pitch and roll of the aircraft.

The study included four objectives (1) Provide information about the pitch and roll of the aircraft, the sound characteristics were modified to communicate a pitch up and pitch down condition. (2) The indicator should have an intuitive "anchor point for a straight or level flight. The intuitive nature of this anchor point was required as to no induce an additional cognitive load in the interpretation of the anchor point. (3) The indicator should easily distinguish between pitch up and pitch down orientation, this is to address misinterpretation discussed in Chapter 2. (4) The indicator should be a sound source which can be tolerated for long durations; thus reducing the effects of fatigue and annoyance.

The findings of this study showed encouraging results, the procedure used in this study is employed to satisfy multiple criteria in the construction of auditory displays. This research is applied to a specific purpose, which provides a context for the auditory display system. The cognitive load is represented accurately as well as with the appropriate instrumentation. It is important to note that auditory displays are generally context dependent.

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1.5. Problem Statement and Thesis Outline

The goal of this research was to provide a basis for future development of a system to communicate to a pilot in a Battle Space application using aural cues. To both determine the means and type of information to communicate. To accomplish this goal the research focused on (1) Identifying methods within and outside of the virtual reality community to use aural information as a communication means, (2) Develop a framework of issues to be considered in developing and communicating capabilities, (3) Explore a portion of the experimental work that will be needed to implement an aural communication in a virtual environment.

This thesis is organized with Chapter 2 providing the literature review of all the work that was identified to influence the design of an aural communication and multi sense communication system. Chapter 3 presents issues of implementing a system, with a focus on typical equipment and acoustic environment. Chapter 4 will focus on the efforts to characterize, and thus calibrate a system. Chapter 5, 6, and 7 presents the design, data analysis, and results of one experiment to provide an initial assessment of communicating urgency as a representative aural communication system for battle space. Chapter 8 will summarize the research and propose future work to develop an aural communication system for autonomous vehicle control.

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CHAPTER 2. LITERATURE REVIEW

There is a significant body of literature that presents the primary elements and examples of aural communication systems. A literature review on the characteristics and design of aural displays is presented, followed by other work on creating, implementing, and studying aural communication and quantifying the reaction of these communications. The compilation of this literature is provided to expand the perspectives and research community to be included in the study of aural communication systems. Most notably, the work within the noise control community, referred to as sound quality is explored. Comparing these works provides not only a cross disciplinary view to motivate future efforts, but also highlight issues to be addressed in future work. The background works are organized as (1) auditory displays, (2) auditory alarm theory, (3) sonification, (4) sound quality, (5) audio context, and (6) cognitive load.

2.1. Auditory Displays General View

In offloading from the modality of vision to a modality such of sound there are several key factors that must be considered in order for this conversion be successful. (1) This conversion requires an efficiency of 100%, thus the sound stimulus must accurately communicate the data as if the information was presented visually. This requirement must be satisfied so a loss of information is not contributed to the modality used to articulate the information. (2) The cognitive processing required to interpret the auditory data must be less or equal to the cognitive processing of the visual data. If the cognitive load introduced is greater than the previous modality, than an inefficient processing of information will occur. (3) The quality of the sound stimulus employed to offload the data, must induce a sensation

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comparable to that of the visual data. This ensures the same response by the subjects regardless of the modality employed. (4) Lastly, the environment in which the auditory stimulus is broadcasted into must not influence the meaning of the data; a pure representation of the visual data must be delivered through sound. This ensures the acoustics of an environment does not negatively influence the perception of the sound stimuli. To satisfy these criteria a general understanding of the characteristics of an auditory display is used to create the framework for the construction of an appropriately designed auditory display.

Data historically communicated through a visual modality can be off loaded to the modality of sound (Elridge, 2006; Edworthy et. el., 1999). The visual modality is limited to visual data in the direct sight field and peripheral field (Zwicker, 2007). This limitation can be relaxed with sound due to its multi-directional nature (Hartmann, 1998). Thus the primary advantages of auditory displays versus visual displays are: auditory displays do not require a user in a specific spatial orientation, or attentional focus (Elridge, 2006).

An auditory display is an auditory representation of data (Elridge et. el., 2006), communicated through sound to exploit the human hearing system. Auditory displays can be hypothesized from numerous theoretical frameworks such as auditory alarm theory, sonification, and lastly through its sound quality. At a fundamental level each of these approaches requires a sound representing an event in the natural world, typically referred to as its referent (Edworthy, 1999). Each event in the natural world has a respective referent; this is referred to as a complex auditory display. In the literature to date there has been very little cross pollination amongst the theoretical frameworks to construct auditory displays.

The distinction of auditory data from other forms of data lies in the differences on how one hears and interprets the sound data. Sound data is dynamic in that there is not a

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standardized set of sounds each with a specific purpose, and so it is not possible to compose a set of sounds to communicate a universal message. For example, there is no one universal interpretation of a 1kHz pure tone signal at 70dB, or any pure tone at any sound level. It is currently not possible to develop a universal message with sound, but a methodology to construct an auditory display through embedding data into sound is possible. Sound quality, alarm design, and sonification are the three domains within the auditory display umbrella, to postulate a universal interpretation of auditory data.

Each domain embeds data into sound or quantifies human's interpretation of sound. Although each domain has its own nomenclature and role its function is synonymous, to exploit the human hearing system when communicating auditory data. Each component of auditory data serves a prescribed function; this function is dependent upon the realm of operation. However, regardless of the function, the auditory data must possess the following requirements.

The auditory data shall accurately reflect the phenomena it is attempting to portray. This portrayal can either be abstract or tangible in nature. If this condition is violated it will result in a misalignment between an event in the natural world and the auditory display. Imagine an alarm sound being used at a spa to induce a state of relaxation or tranquility, the sound and desired state conflict since an alarm induces a state of panic. Thus alignment is necessary to ensure the sound promotes the desired action and response.

The variables of the physical phenomena must be embedded into individual acoustical parameters of the auditory data. This requirement is interpreted as the physical variables of the natural realm must be fully quantified; this allows mapping the variables in the auditory realm, an elaborate discussion of this requirement is later in the chapter. Lastly,

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the auditory data must be governed and constructed in alignment with the existing finding in sound quality, alarm design, and sonification. The consequence of violating these criteria will be discussed later in this chapter.

Mapping the variables in the physical realm to the acoustical parameters in an auditory signal is a critical step in establishing an auditory data stream. The physical parameter amplitude is best designated for absolute quantities, such as magnitudes when a minimum and maximum threshold exists (Belz et. el., 1999). The physical parameter frequency is designated for quantities that are cyclical in nature for a range of variables which repeat over a temporal dimension (Belz et. el., 1999). The physical parameter phase is designated for quantities outside a desired operating range (Belz et. el., 1999). Appropriately building the data stream for their physical quantities will lead to a more intuitive interpretation of the auditory data (Elridge et. el., 2006), which are the building blocks for an auditory display. An in depth view of auditory alarm design, sound quality, and sonification will provide insight to their similarities.

The construction of an auditory display creates a monitoring system, which are the foundation of auditory displays. Walker and Kramer define an auditory monitoring system as any non-speech sound used to present information to a listener (Walker et. el, 2004). Auditory monitoring systems can be either discrete or continuous units of data communicated through sound (Yost et. el, 2008). A discrete auditory monitoring system is commonly referred to as an alarm, while a continuous system is referred to as a sonification (Spain et. el., 2008). The auditory alarm design community has developed robust guidelines for auditory alarm displays (Edworthy et. el., 1999). Presently, a robust set of guidelines do not exist to develop sonification displays (Spain et. el., 2008). Sonification can be thought of as

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an infinite alarm, it attempts to maximize the information communicated to the user by providing recurrent auditory feedback. One view is that constructing a series of discrete alarms creates a sonification.

Evidence has shown that continuous auditory alerts help identify changes from a normal operating condition, under a multiple task situation, more readily than a visual display alone (Metzger et. el., 2008). Sound has the advantage of being multi-directional, and can take several forms from simple, complex tones, auditory icons, and speech warning (Wogalter et. el., 2002). Wee, (2003) investigated the feasibility of replacing alarms with sonification displays. Although there were no performance differences between alarms and sonification, users rated the continuous auditory display easier to monitor than a visual display or alarm (Spain et. el., 2008; Wee, 2003). This stems from continuous auditory displays providing a stream of data, which gives the operator more information to make better decisions. Psychologically, it creates an atmosphere of assurance, which fosters trust in the sonification (Spain et. el., 2008).

This continuous stream of audio data, when designed appropriately (Edworthy 1999), will boost the overall system performance, operator confidence, and improve safety (Edworthy et. el 1991, Wolgalter et. el., 2002). While inappropriate design will cause problems (Haas, 1972), and cause degradation of the same parameters. This is due to a serious mismatch between the perceived acoustics (psychoacoustic) (Edworthy, 1999), and its intended situational acoustics (for example the urgency associated with the state or condition that the signal represents). The listener may not accurately perceive the acoustics for the situation it is trying to communicate (Edworthy et al., 1991). To mitigate this

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mismatch a user centered approach must be adapted for auditory displays (Stanton et. al., 1998).

The user centered approach to auditory displays and auditory monitoring are postulated in the theory of auditory affordances, based upon Gibson's (1977), theory of affordances. According to Gibson, we do not experience the world directly, but a mental representation of the world. This allows one to interpret the world through our own activities and experiences (Stanton et. al., 1999). This approach to understanding human perception can be extended to perception of sound events, for example auditory warnings (Stanton et. al., 1999). Edworthy, (1999) states this theory have four main propositions:

1. We are surrounded by sounds. They are a part of everyday life.
2. We are introduced to sounds. The function of sound is explained to us when people teach us how to use a sound.
3. We learn about sounds by seeing other people respond to them. We acquire knowledge about how to respond to sound events by watching others.
4. A sound has a definite function. Each sound has the potential to be ascribed to a function.

A well-constructed auditory monitoring system shall have a high degree of alignment with the incident in the natural world it is intending to communicate (Edworthy, 1999). If the communication of the auditory display conflicts with the event in the natural world it represents, the auditory display will excite confusion (Edworthy, 1999). This confusion will manifest itself in a lack of interpretation of the auditory display, rendering the

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communications intelligible. This intelligibility will communicate a message that deviates from its intended rationale, fostering a lack of trust in the auditory display (Edworthy, 1999). In this case the auditory display will either be ignored, or become a source of annoyance (Patterson, 1982; Stanton et. el 1999). At present the mapping of an auditory display to its rationale appears, is at best, suboptimal, and, at worst, random (Stanton et. el 1999).

The mapping of the auditory display to a system is difficult, since one cannot anticipate all the changes in a system over its life cycle (Elridge, 2006; Edworthy, 1999; Blanchard, 2006). The system complexity dictates the type of auditory display employed, and the evolution of a complex system over its life cycle requires modifications of the auditory display. In a manufacturing environment typically auditory displays and control systems embedded into a respective systems operation. The control systems role is to facilitate the operation of the systems intended function. Improvements in the control system can lead to an increase in the productivity and efficiency of the process. Typically, resources are dedicated to the monitoring of the control system at multiple levels within a manufacturing organization. This allows the constant refinement of the control system leading to a more effective overall system.

Auditory displays have historically remained static throughout a systems life cycle leading to less effective auditory displays. This results in the dynamic behavior of the system being described by a static auditory display, and a misalignment of the event being represented by the auditory display. An operator in a manufacturing environment may require auditory cues from the primary system function in the training phase of systems operations. Through experience non-auditory cues may become a more effective mechanism for a communicating the same systems dynamics. This requires modifying the existing

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auditory display from a primary system function to a secondary system function. The mapping of the auditory display to a tangible event is the most difficult task in the auditory display process (Edworthy, 1999). This concept, developed under the theory of affordances, based on the idea of a direct mapping between a sound and an event. Edworthy, (1999) adopted this concept and created a sound theory of affordances, from which a methodology was developed to create this mapping, which states:

> ***Establish the need for warnings:*** *This requires identifying the referent. The referents are the situations or events to be represented by the auditory display.*
>
> ***Determine if the event will be represented by abstract or natural sounds:*** *A natural sounds is sampled from the environment, once sampled, it can be modified appropriately. Abstract sounds are unnatural sounds generated through instrumentation.*
>
> ***Generate Trial Sounds:*** *Create sounds that accurately portray the situation or event. Recall the guidelines from the auditory data section.*
>
> ***Appropriateness Ranking Test:*** *Rank the appropriateness of each sound to the referent.*
>
> ***Design Trial Warning/Alarm Set:*** *Assign a sound to each referent; allocate one main sound and multiple reserve sounds.*

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Learning/Confusion Test: *This seeks to establish the learn ability of the sound for its respective referent.*

Urgency Mapping Test: *This is mapping of the situational urgency of each referent. This resolves discrepancies between the situational urgency of a referent and its perceived urgency.*

This section has outlined the general approach of different domains to construct an auditory display. Several fields have established a methodology to communicate data through sound, an in depth investigation of their methodology will be explored in the upcoming sections. The key observation is all the items presented in this section can be generalized across all other theoretical domains.

2.2. Auditory Alarm Theory

Alarm design theory has developed robust guidelines and nomenclature in their quantification of auditory displays. The building block of an auditory display in alarm design theory is a pulse. A pulse is a complex tone, with a prescribed onset and offset at the beginning and end of its amplitude envelope (Patterson, 1982). The duration of the pulse is on the order of milliseconds (Patterson, 1982). The pulse onset and offset are employed to prevent startling; the lack of an onset or offset is equivalent to a sudden sound being introduced to your environment at full volume (Patterson, 1982). The complex tone can

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contain a starting frequency, intermediate frequency and ending frequency (Patterson, 1982). The pulse can vary in its duration and amplitude.

A series of pulses in defined as a burst, a burst introduces the inter pulse interval, which is the time between each pulse. The inter pulse interval can be manipulated to modify the frequency of each pulse. In summary, alarm design theory has developed the concept of the pulse, embedded in the pulse are the: starting frequency, intermediate frequency, and ending frequency, onset, offset, and inter-pulse interval. This provides six variables to be mapped to parameters in the natural world. Auditory alarm theory was developed for events in which there is a high degree of urgency, such as an imminent failure or danger, typically used in the construction of alarms and sirens. In the construction of urgency there are several parameters identified by Patterson, (1982), to increase the perception of urgency, including the fundamental frequency, harmonic series, delayed harmonics, amplitude envelope, speed, number of repeating units, rhythm, pitch, pitch range, and musical structure.

Historically, these alarms were reported as too loud, too high pitched, irritating, inappropriate, or confusing (Stanton, 1999). To mitigate these issues employing guidelines provided by Patterson, (1982), and Edworthy, (1991), can serve as a basis for your auditory display construction. This study has verified the methodology outlined above, and its ability to produce robust results. The ECG monitor is only one example that exemplifies the procedure for mapping sound to an event, which will be discussed later. Since sound event mapping is situation specific, sonification displays are observed in many applications. In general, the situation should dictate the type of sonification used, and the theory of affordances provides the methodology to construct auditory displays for its respective referent.

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A limitations to this approach, is that it is only suitable for an abstract representation of an event. If we consider natural sounds, the variables are difficult to manipulate (Phil, 2005), applying an onset or offset to a natural sound can decrease its intelligibility due the sound synthesis required for the manipulation of the signal (Hellier et. al., 2002). This loss of intelligibility, results in a loss of the signals integrity, which violates the 100% efficiency principle introduced previously.

In attempting to reconstruct a pulse as described in the literature of Patterson's, (1982) article Guidelines for Auditory Warning on Civil Aircraft, there are generalities on the approach, but there no mathematical equations for the construction of a pulse. In articles referenced by Edworthy, 1997, On Using Psychophysics Techniques to Urgency Mapping in in Auditory Warning it employs this approach based on the guidelines outlined by Patterson. Edworthy's findings in this study were that increasing the inter-pulse interval and frequency increased the perception of urgency, these results were consistent with Patterson's observation. However, none of the literature that was used provided clear mathematical modeling of the pulse. This creates difficulty in duplicating or building upon the results of the existing literature, since the basis under which the pulse is being constructed is different. The limitations and lack of an appropriate model, has the potential to lead to a different approach of investigating auditory displays.

2.3. Sonification

The most common forms of sonification are: earcons, auditory icons, audification, and model based sonification, and parameter mapping (Yost et. el, 2008). The auditory stimulus in a sonification display can either be tangible or abstract in nature (Walker, 2004).

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If the representation is abstract then many of the finding from alarm design theory can be employed, such as the ECG machine. A practical application of successfully implementing a sonification in a multi-faceted environment is observed in an ECG monitor (Stanton 1999). The ECG is an electrocardiogram which measures and records heart activity. Heart beat is mapped inter-pulse interval and amplitude to faintness of the heartbeat. A nurse can simultaneously monitor the heart rate through the ECG, while performing tasks associated with caring for a patient. This is an abstract representation of a sonification display.

If the representation is abstract then the concept of an earcon must be introduced. An earcon, is the building block of a sonification display (Spain et. el, 2008), used to embed discrete units of information into an auditory signal. The organization of the blocks dictates the interpretation of the auditory data, and the units of information are the variables which exist in the natural world. The pulse terminology from alarm design theory, described in terms of sonification, is a one-element earcon (Belz et. al., 1999). The concept of an earcon and pulse are synonymous, their distinction lies only in the domain in which they were conceptualized. Earcons are abstract acoustic motifs used to communicate complex messages (Elridge, 2006).

Auditory icons are tangible acoustic motifs used metaphorically to articulate the real world phenomena, metaphorical representations of events in the real world. The most well-known is the “wastebasket" sound, which is a simple two-note "earcon" for indicating file deletion proposed by (Sumikawa et al., 1988). Not only does the wastebasket sound convey more information about the event and objects involved in it, but it does so in a more intuitive way. (Buxton et. el., 2010). The wealth in auditory icons is that it communicates information regarding the content or features of the natural world through sound.

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Audification is the translation of raw data into an audio stream (Elridge, 2006). This is process of turning an auditory signal into a sound. This is focused on the implementation, rather than the construction of the auditory data. Model based sonification is extracting information from an acoustical signal based on our interactions with the real world (Elridge, 2006). Lastly, parameter mapping is correlating data dimensions onto auditory display dimensions (Elridge, 2006). These ideas will be discussed later in Chapter 2, the focus of section 2.4, 2.5, and 2.6 are the types of auditory data. Each of these displays serves a distinct purpose and is application dependent. Sonification expands the field of auditory displays vastly with concepts such auditory icons. Encompassing many types of sounds, methods for their construction, provides the flexibility required to construct an auditory display. In sonification and alarm design theory, both seeks to exploit the sensations perceived by sound.

Existing sonification displays have been developed to accommodate theories in auditory perception and psychoacoustics (Spain 2008). This has consistently correlated a human perception to physical parameters of a sound (Edworthy 1991). Manipulating the physical parameters volume, pitch range, pulse range, harmonic series, fundamental frequency, and rhythm has been consistently mapped to the state of a system (Patterson, 1982; Edworthy, 1999; Wogalter, 2002; Suied, 2008; Spain, 2008). Sonification design supports pre-attentive awareness by taking advantage of the resolution in the human hearing system (Moore, 1981). Pre-attentive referencing refers to the ability to attend to information, without using limited focal resources (Jacobson, 2003). Under normal operating conditions, the continuous signal fades out of focal attention and can be monitored pre-attentively

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without disrupting attention (Jacobson, 2003; Spain, 2008). This allows an operator to focus on a primary task while simultaneously monitoring secondary and tertiary tasks.

Finally, sound event mapping is situation specific, thus the situation should dictate the type of sonification used, and the theory of affordances provides the methodology to construct auditory displays for its respective referent.

2.4. Sound Quality

Sound quality refers to a discipline within the engineering noise control community to develop means to relate measured sound to how people perceive sound. There is an extensive body of literature which has developed measures which could be used to develop sounds in auditory displays. The theoretical frameworks of alarm design theory or sonification can benefit from the observations found in sound quality. Quality in its natural sense provides a perspective the integrity of an entity. If an entity has a high degree of quality it is perceived as producing a more robust and desirable outcome from the perspective of the user. If one considers the quality associated with an entry level vehicle as compared to with a luxury vehicle, there are inherent differences in their construction. Consider closing the door of vehicle such as a BMW as compared to a Toyota. Each vehicle has an inherent quality that associated with its construction; each choice in its construction material contributes to its quality.

A consumer purchasing a luxury vehicle will not have the same expectations as a user purchasing an entry level vehicle. To provide a context into the concept of sound quality, if one considers consider the acoustical quality differences between a high end stereo system, and an entry level system. Such as the transition from mono to stereo, stereo to Dolby

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surround, Dolby surround to Dolby digital, and Dolby digital to 3D surround sound fields. Each evolution of the sound system strives to increase quality. Since man-kind has developed consciousness they have strived to develop mechanisms to improve the quality of experiences, thus improving the quality of life.

The advent of the air conditioner made it possible to work in severe heat. The evolution of the vehicle reduced the travel time from weeks to hours. This method of travel was surpassed by the aircraft where months were dissolved to hours. Modern history has embarked man on a journey that encompasses improvements as vast as the human consciousness. Sound is no exception, improving the quality of sound as perceived by humans improves the quality of the experience in which sounds are introduced. This experience can be described by the sound quality parameters. Sound has the ability to induce a sensation; these sensations contribute to our perception of an event.

Sensation is defined as a mental process (as seeing, hearing, or smelling) resulting from the immediate external stimulation of a sense organ often as distinguished from a conscious awareness of the sensory process (Merriam –Webster, 2010). Fundamentally, an auditory display seeks to exploit our senses to induce some sensation. Sound Quality has defined the robust parameters loudness, roughness, tonality, and fluctuation strength which are sensation quantities that based upon the human hearing system. This is different from the traditional method of auditory display construction which is based upon the physical attributes of the sound.

Fastl and Zwicker, (2007) states loudness belongs to the category of intensity sensations. An intensity sensation of a sound correlates to a signals amplitude and sound level. Loudness is a measure of how sounds are perceived at different frequencies relative to

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one another. Fastl and Zwicker states the sensation stimulus loudness can be measured by answering the question how much louder or softer a sound is a heard relative to a standard sound. The loudness level of a sound is the sound pressure level of a 1-kHz tone in a plane wave and frontal incident as the sound; its unit a phon (Fastl, 2007). Conceptually loudness is a simple concept to understand, but its implementation is unclear. A standard has been established for calculating loudness, ISO 532:1975, but its implementation raises several questions, loudness is described as

$$N = \int_0^{24Bark} N' dz, \qquad [2.1]$$

where N' is the specific loudness, $N' = .08\left(\frac{E_{TQ}}{E_o}\right)^{.23}\left[\left(.5 + .5\frac{E}{E_{TQ}}\right)^{.23} - 1\right]\frac{sone}{bark}$, where

E_{TQ} is the excitation at threshold in quiet, E_o is the excitation that corresponds to the reference intensity $I_o = 10^{-12}$ W/m^2 , lastly E is the excitation level at the respective critical band.

The model requires applying the effects of the human hearing systems which can be accomplished with a fixed filter that approximates the effects of the ear drum. After the effects of the ear drums has been compensated, a secondary fixed filter can be applied which compensates for the transfer function of middle ear, each of these quantities are well defined, and typically approximated as the by the third octave bands. After the effects for the human hearing system has been considered, one must determine the excitation pattern of the stimulus across the basilar membrane (Moore, 1989). The basilar membrane behaves as a

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Fourier analyzer breaking down the complex sound into its sinusoidal components (Moore, 1989. The basilar membrane decomposes frequencies in bands, generally modeled as the third octave band. The loudness scale is based on the critical band rate, the Bark

In each of the critical bands the excitation along the basilar membrane must be quantified. In the calculation of loudness, this is quantified by a quantity coined ΔL or E, which has conflicting definitions according to a review in the Fastl text. On page 85 of Psycho-Acoustics, this quantity is the defined as the masking depth of tone relative to another. Consider when walking on a busy street engaged in a conversation you must increase the sound level of your voice to overcome the background level of the traffic, the traffic is masking your voice. The degree to which you must increase your voice relative to the background level is the masking depth. The masking depth is presented as the first definition.

Further, imagine within this walk there are variations in traffic flow, which causes fluctuations in the background level changing the masking depth. Thus, within each critical band the masking depth is considered only at its maximum and minimum for a respective signal. This would be considered specific loudness. A later definition was the excitation level is simply the difference between the maximum and minimum excitation level at each respective critical band. Although, the concept is presented in detail the methodology for calculating this excitation level is not provided. In the construction of an auditory display the tools must respond to events in the natural world, this model is not sufficient. A real-time system must be able to dynamically update the excitation levels in the calculation of roughness.

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The next parameter is auditory roughness was introduced in the acoustics and psychoacoustics literature by Helmholtz, (1885) to describe the buzzing, harsh, raspy sound quality of narrow harmonic intervals, and dissonance (Vassilakis, 2005). Roughness is created when signals at different frequencies resulting in complex signal fluctuate between a maximum and minimum; fluctuations between 15-150Hz quantify roughness (Vassilakis, 2007). Roughness is defined in the units of asper, where one asper is defined as 1kHz tone, 100% modulated with an amplitude modulation of 70Hz, at 60dB. The roughness is defined as

$$R \sim \int_0^{24Bark} \Delta L_E(z)\, dz, \qquad [2.2]$$

where ΔL_E is the difference between the minimum and maximum excitation level for each critical band, and the roughness is evaluated over all the critical bands.

Vassilakis's, (2011) study investigated the existing models of roughness, relative to the observed roughness of sound stimuli. The findings of the study are the existing models lack the ability to quantify observed roughness, and the model proposed is the best model for quantifying roughness. The issue is the stimuli used is unknown, the roughness model could be a function of the stimuli employed. Implementing sound quality parameters provides a unique perspective to the construction of auditory displays but raises many questions. The final quantity is the fluctuation strength of the amplitude modulated broad –band noise. Fluctuation strength is dependent upon the modulation frequency, sound level, modulation depth, center frequency, and frequency deviation (Fastl, 2007). At modulations less than 20Hz, introduces the sensation of fluctuation strength (Fastl, 2007).

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An additional sound quality parameter is tonality, which is a count of the number of pure tones in a signal. The final parameter is sharpness, which is most related to the density of a sound (Fastl et. el., 2007). Sharpness is evaluated in the same manner as loudness; the most important variable influencing sharpness is the spectral envelope of the sound. Sharpness increases at levels from 30dB to 90dB by a factor of two, the dependence on level can be ignored as a first approximation, and there is a critical band dependence, which means as long as the bandwidth is smaller than the critical band, no difference in sharpness can be detected (Fastl et. el., 2007). The unit for sharpness is defined as an, acum, and is defined as a narrow-band noise one critical band wide at a centre frequency of 1kHz having a level of 60dB (Fastl et. el., 2007). Sharpness is defined as

$$S = .11\frac{\int_0^{24Bark} N' g(z) z\, dz}{\int_0^{24Bark} N' g(z) dz}, \qquad [2.3]$$

where g(z) is weighting factor that has a value of one up to 16 bark, and increases up to 4 at 24 bark.

Sound quality parameters such as loudness, roughness, and fluctuation strength can be combined to determine sensations such as pleasantness and annoyance. In a study conducted by Schutte (2009), he investigated 16 sounds for an aircraft takeoff, and 15 sounds for an aircraft landing. Each sounds was presented to a subject in 18 pairs, with a reference sound, and the response variable for each treatment was a free verbalization. The subjects provided semantic descriptors aimed at grouping the sounds into homogenous clusters (Schutte, 2009). The verbalizations focused on the characteristics of the sounds, where the

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nouns and adjectives were extracted to create a categorization, this procedure consisted of classifying words as clusters through a linguistic analysis.

In this particular study the results were inconclusive primarily due to the large number of adjectives and nouns from the group of subjects. However, if the results were consistent, in that a high degree of consistency amongst adjectives and nouns, these descriptions could be used to quantify the aircraft takeoff and landing sounds in terms of its respective adjective. If these adjectives were mapped to the sound quality parameters a mathematical relationship could be developed. For example, through the linguistic analysis if it were determined the sounds were most frequently verbalized as annoying. Then a degree of annoyance could be used to quantify the sounds, measurements of the sound quality parameters could be performed, and mapping the semantic description with its sound quality parameter could create the scale of annoyance in terms of its sound quality parameters.

2.5 Audio Content

Once the sound is mapped, the interpretation of the sound must be considered. Interpretation of sound stimulus is influenced by ones exposure to auditory signals, life experiences, and context in which they interpreted urgency. Humans have varying experiences which expose them to different types of auditory signals. Factors such as geographical location, social status, preference in music, and other extraneous factors all contribute to the processing of an auditory signal. The life experience of an individual is a key component in the processing of auditory signals. If an individual is frequently exposed to loud noise compared to an individual who has sensitive hearing to sound, each scenario will influence the perception of a sound event. In short, the experience that varies from one

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individual to the next will have an impact on how sound events are perceived. To capture these differences in a comprehensive manner a context and scenario must be established in the perception of urgency.

A scenario is defined as an imagined sequence of possible events, or an imagined set of circumstances. Context is defined as the circumstances or events that form the environment within which something exists or takes place. The context is the domain in which urgency will be evaluated, this will vary by application. The context will change for a military application, civilian application, and manufacturing application. The context is the overarching discipline in which the research question lies, when evaluating a sound event. Within a context a series of scenarios or events will be evaluated, each scenario must provide a different set of circumstances. Each circumstance must have the ability to embed a secondary and tertiary event within the primary circumstance. The ability to incorporate a hierarchically arrangement of a series circumstances must be incorporated in the primary event.

A component in research is an assurance that a subject understands the context of the research. A subject must possess a reference frame for an event, that reference frame will be dependent upon the subject's background and life experience. This background and life experience will be the factor that motivates the subject's decision making processes and actions in an event. The speed in which decisions are made and actions performed can influence the subject based on the familiarity with the reference frame. To handle this situation in research, a context must be developed that is applicable to the majority of the population of interest. The presentation of a context of this nature will yield each subjects interpreting an event from the same reference frame.

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In most research applications the context of the experiment is similar to the actual environment of interest. This observation is confirmed by the works of Marshall and Lee, (2007), which focused on auditory alerts in vehicle information systems. The researchers investigated alarms used to trigger actions. The first event was pressing the brakes in a vehicle; this action was triggered by an alarm to indicate an imminent collision. The second event was a vocal action; triggered by an alarm to indicate an email was received. This research was conducted in a driving simulator, which aligns the experimental and natural world. The results of this experiment can be directly applied to alarms in vehicle information systems.

Establishing a context requires a well-defined procedure. The observations in the study fundamentally led to the development to the following criteria for creating a context to construct an auditory display:

1. Provide a dynamic environment, so the subject is engaged in a task representative of the context operator. The requirement of this task is provide the same cognitive load as a dynamic control environment.
2. Provide representative instrumentation. The instrumentation will be a factor since it will provide the context for the actual environment.
3. Identify the appropriate population for the context.
4. Replace the interval in which there is no cognitive load with a task of the same load.

The relationship between the context and scenario should be a parent child relationship. The parent is always the context, and the child should be the primary scenario,

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while the grand child should correspond to the secondary scenario. Each successive scenario should follow the respective pattern. Within a context the number the children scenarios will increase with the complexity of the overall context. A base auditory signal must be established for each context. Within the children scenarios the acoustic parameters of the signal would need to be modified to communicate the status of the successive children scenarios. This should provide a comprehensive picture of the primary scenario. The primary scenario is of utmost importance for the overall perception of a sound event.

The domain in which the sound event concept is conceived will influence the perception of the sound event. This is based upon the events unique to a given context, for example applying the concepts of urgency from the theoretical domain to a contextual domain is the first step in creating a real world auditory display for perceived urgency. When applying the contextual factor for urgency there will be a variation in the scale of perceived urgency due to the inherent nature of the context. Auditory signals identified as very urgent in one context may only be slightly urgent in another context. This variation raises the question if there is an interaction between context and urgency. Once a context is established the cognitive effects of the context must be considered.

2.6. Cognitive Load

Sound has the advantage of being able to be interpreted at a pre-attentive level, which allows one to decrease the overall cognitive load of a subject engaged in a task. If we revisit the pilot example, the pilot can reduce the strain of constantly scanning multiple instruments in the visual modality, if some of the instrumentation is offloaded to a modality that facilitates the same information in a more effective manner. While this can be achieved

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by an auditory display, the total work load and task demand within an event will have an influence on the ability of a subject to interpret the sound stimulus.

Cognitive load is the load on the working memory during instruction. Cognitive load can be decomposed into three components: intrinsic, extraneous, and germane. An intrinsic load is defined as a load that has an inherent difficulty associated with it (Sweller, 2010). The extraneous load is the working memory load experienced by learners as they interact with instructional materials (Sweller, 2010). Germane load is the load on the working memory when receiving instructions (Sweller, 2010). The sum of each of these components defines the total cognitive load. In considering total cognitive load the factors: audio, visual, primary task, and error needs to be considered. The audio factor considers any auditory stimuli, or visual is any visual stimuli that contributes to the cognitive load. The primary task is any action that a user must primarily perform that contributes to the cognitive load. Lastly, the background task error is any task not associated with the primary task that contributes to the cognitive load. An identification of the existing load in the real world and experimental setting will need to be performed to ensure there is alignment between the two. In areas in which there is not an adequate background task, a supplemental task will need to be introduced to compensate for this lack of load. This will provide a representative cognitive load of the actual environment.

In considering the cognitive load for the primary task, an analysis must be performed to determine the requirements of a primary task and its relationship to the context. In order to perform this analysis existing literature can be analyzed. For example, in a study conducted by Sommerich, et al, (2004), examined the cognitive load due to the coordination of multiple and simultaneous task. The study examined three aspects of cognitive task

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including: skill-, rule-, and knowledge based tasks. Skill based task are characterized by the use of rote knowledge, typically involving performance controlled by behavior in the short term memory. Rule based tasks are characterized by the use of propositions in the long term memory, which is described as a process of mentally mapping environmental characteristics and task goals to actions (Sommerich, et al, 2004). Rule-based task occur in a familiar setting in which definite rules of performance are defined. These three types of behaviors should not be considered dependently but rather falling on a continuum, each type flowing into the next.

In order to assess the impact of varied cognitive loads in a dual task scenario three classes of workload techniques can be used. They are performance, subjective and physiological workload assessments. The speed and accuracy with which a task is performed can be used as the performance indicator (Sommerich, et al, 2004). The subjective measurement involves the operator giving his or her opinion. Physiological measures are provided by heart rate variability and electro-encephalogram signals.

In the research by Sommerich, et al (2004), the cognitive load and cervicobrachial muscle response were evaluated and related while research participants performed a typing task in conjunction with several cognitive tasks. Based on previous studies, if the primary task demands are significant in terms of required cognitive resources due to task complexity, efficiency and productivity of a secondary task are likely to decline. The typing task was considered a true secondary task, because participants were required cognitively to simultaneously process the skill-, rule-, and knowledge based task. Manipulating the cognitive load in a dual task scenario, also involving psychomotor task performance, may

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cause corresponding changes in the perceived stress and muscle tension, as well as primary and secondary task performance.

The skill based task involved participants reading, rehearsing, and recalling lists of two to seven familiar, but unrelated nouns. For the rule based task condition, participants were posed with a geography task that required the use of a definite decision rule. The knowledge based task condition required participants to learn a complex schedule of daily work activities and modify the schedule to include additional activities. The skill based task revealed a significant impact on cognitive load. Secondary task performance decreased 23% from the baseline condition when typing was performed relative to a skill based task. Lastly, the muscle activity varied according to task, the skill based found a 6% difference.

The above experiment structure can be applied to any context. In a representative context, of the primary task can be the typing task, in which the user is engaged mentally and physically. The typing simulates a user at a workstation, which represents a task in an actual event. The research also showed that for the majority of the experimental duration, the typing task which was the continuous task and the discrete tasks were the skill, rule, and knowledge based task. If a primary task does not induce the proper cognitive load, a secondary task may be considered.

The analysis of cognitive load must be performed independently in a conceptual manner then reintegrated in the context. In order to perform this analysis, the literature must be decomposed into elementary components. A mapping of these components to Battle Space will provide insight to achieving a cognitive load representative of the real environment. In a study by Lee et. el, (2004), (they examine the effects of cellular cell phone communication in driving performance. While driving each user was asked to

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perform two secondary tasks in two conditions: (1) without a task, (2) mathematical addition via cell phone both of which were carried out in rural and urban road.

If the workload is too low complacency will set in, if the demand exceeds the effective capacity the operator is more error prone. In a driving environment with vehicle information systems the threshold for this overload can be reached. An assessment of the driver workload while performing driver irrelevant task is of considerable interest.

Workload is defined as task demand, effort expanded, and the results of that effort. This workload can be examined by various stressors that influence the performance and responses of the human operator. Workload assessment techniques can be classified into four broad categories related to primary and secondary task, subjective rating, and physiological parameters.

The primary task performance is this case is the speed control of the driver. If the driver perceives a decrease in attentional capacity then he or she can reduce the vehicle speed as a compensatory behavior to facilitate relocation of mental resources. Imposing a secondary task measures the residual resources or capacity not utilized by the primary task which was mathematical addition. In addition, physiological data for subject were collected, as the heart rate was measured as an indicator of workload.

Prior to the start of the experiment the subject provided a baseline heart rate measurement after a five minute resting period. The change in heart rate was then considered the unit of measure for the physiological data. Indicators of driving performance, task performance, and physiological responses were measured. An analysis of driving speed in the two road situations were performed, and the heart rate for the different secondary task were assessed. A significance decrease, 5.8%, in driving speed was observed with the

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mathematical addition task via cellular phone. In addition the physiological cost was also increased with phone tasks from 90.8 to 87.7, heart beats per minute and the incremental heart rate while performing the secondary task was significantly higher. However for the different situations the correct answer rate of the math problems was not a significant difference.

The methodology employed by Sommerich, et al, (2004) was used to devise a set for a military application for a primary and secondary task. The primary task is the continuous monitoring of the health of the UAV, through and auditory display; a secondary task can be updating mission parameters needs to be identified. In general the task must comply with the following guidelines.

1. A balance between the operator workload must be established. The subject engaged in the secondary task must not become complacent or cognitively overloaded.
2. The secondary task should apply an additional load cognitively, but not consume the user attention relative to the primary task.
3. The secondary task must have a measurable performance parameter.
4. The workload of the secondary task must small compared to the primary task.

Based on the requirements written above, a secondary task needs to be allocated to meet these requirements. If an abstract representation of a task is need, a representative task such as a mathematical addition task may be suitable. This task has been confirmed to induce the requirements listed above. The only issue regarding the application of this task is

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the level of sophistication of the problems is unknown. To compensate for this issue the level of difficulty in the problems will need to vary, this can be accomplished by a carefully planned study. Each parameter of skill, knowledge, and rule based task can used.

2.7. Literature Summary

The literature review has been provided primarily for future work. The particular issue to be explained with the rest of the thesis are (1) Identifying a typical audio display system, and (2) Implementing an example experiment to evaluate calibration of an audio display system, and developing a correlation between sound parameters. (3) While auditory displays have been developed and tested the available documentation does not provide sufficient detail regarding the system design, calibration, and acoustic environment. Thus the work in the rest of this thesis is intended to provide documented system and analysis design details for future work.

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CHAPTER 3. AUDITORY DISPLAY SYSTEM MODEL FOR UAV

The literature review provides a basis to design a concept for auditory display systems (ADS) that will be applied to a virtual environment that is used to control autonomous vehicles. Based on the literature review, the auditory display must be developed for the application. Information and success in other applications should be considered, but to start, all relationships between the sounds and the desired information that is communicated should be reestablished. Further, with no auditory displays having been designed using sound quality parameters the applicability and models for their use must be established. This chapter provides a system model for such an ADS system.

Applying the lessons from literature on cognitive load will be critical for future developments of an auditory display. In the case of a UAV application the primary and secondary task can be considered. The methodology employed by Sommerich, et al, (2004) was used for a military application, which involved a primary and secondary task. The primary task is the updating of mission parameters, and the secondary task is the continuous monitoring, and the health of the UAV, through an auditory display. In general the task must comply with the following guidelines.

1. A balance between the operator workload must be established. The subject engaged in the secondary task must not become complacent or cognitively overloaded.
2. The secondary task should apply an additional load cognitively, but not consume the user attention relative to the primary task.
3. The secondary task must have a measurable performance parameter.

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4. The workload of the secondary task must small compared to the primary task.

Based on the requirements outlined, a secondary task needs to be allocated to meet these requirements. If an abstract representation of a task is needed. A task similar in nature to the one presented by (Sommerich, et al, 2004), a mathematical addition task, may be suitable. This task has been confirmed to induce the requirements listed above. The only issue regarding the application of this task is the level of sophistication of the problems is unknown. To compensate for this issue the level of difficulty in the problems will need to vary, this can be accomplished by a carefully planned study. Each parameter of skill, knowledge, and rule based task can tested, if desired by using math problems. The number of task introduced should align with the appropriate cognitive load for the event in the natural world. This methodology ensures the appropriate cognitive load is induced, this load will be representative of the load in the natural world. The cognitive load is only feasible if an auditory display system appropriately designed.

3.1. System Model of an Example Auditory Display

The auditory display system, to be designed is an assemblage or combination of elements or parts forming a complex or unitary whole (Blanchard et. el., 2006), a complex scheme of auditory signals comprised to communicate phenomena experienced in the natural world, with the goal of dictating the desired behavior of a subject through the manipulation of the auditory signal(s). A framework of a systems approach is proposed. There have been many factors considered in the construction of an auditory display: the auditory data, auditory monitoring, alarm design theory, sonification, sound quality parameters, context,

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and cognitive load. Thus far the resources or general approach to implement an auditory display has not been considered, a systems approach provides the tool to aide in its construction.

ADS can be classified as a system composed of components, attributes, and relationships (Blanchard et. el., 2006). Blanchard et. el defines the characteristics of a system in the following manner.

1. **Components** are the operating parts of a system consisting of input, process, and output. Each system component may assume a variety of values to describe a system state as set by some control action and one or more restrictions.
2. **Attributes** are the properties of discernible manifestations of the components of a system.
3. **Relationships** are the links between the components and attributes.

A decomposition of the systems orientated approach applied to ADS is presented in Figure 3.1. Hierarchically, the top level being the system level, ADS comprised of multiple referents with auditory signals that define the referent are at this level. The second level is the subsystem level, in which each individual referent is contained. Embedded within each referent are auditory signals. Embedded in the auditory signals are the variables that are mapped to the phenomena in the natural world. The second level primary objective is achieving the appropriate mapping between the referent and its respective state.

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The third level is the component level, this level is defined the three distinct components: signal processing, environmental acoustics, and the effects of the human hearing system. The signal processing component involves the transformation of the auditory signal at the sub-system level into a sound. An ADS is unique to most systems in that everything that remains outside the boundaries of the system is considered the environment. Typically the environment is not considered as being part of the system, but rather the space in which the system resides. In ADS the environment is considered as being part of the system, since it influences the acoustical properties of sound which propagates through the environment for the sound producing device and the person.

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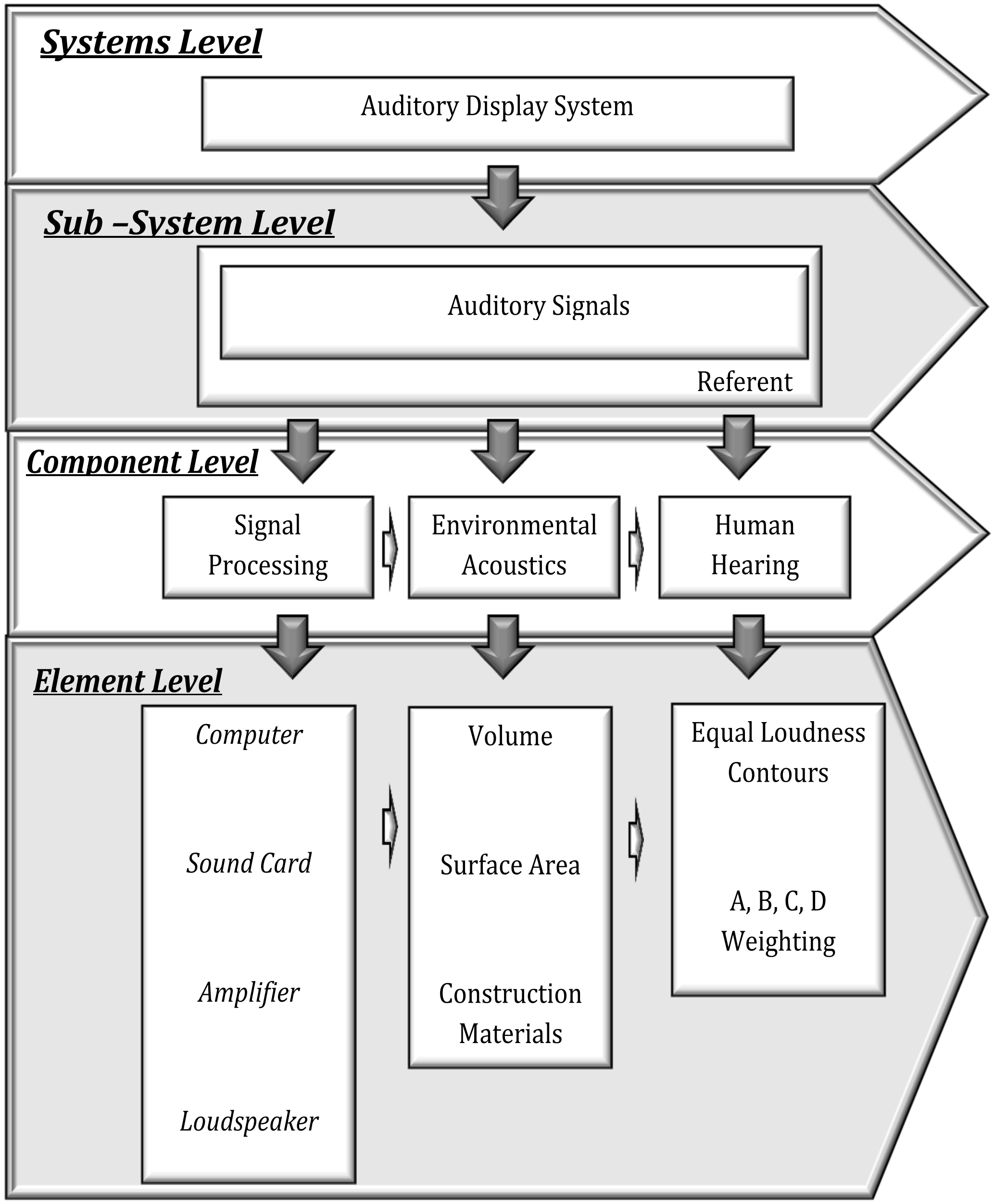

Figure 3.1. System approach of a top down view for auditory display system model.

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The effects of the human hearing system are the final component, as sound is transferred from the environment to the ear it undergoes another series of transformations. Within each component there are elements which provide the attributes for the components.

The fourth and final level contains the elements for each of the components. These elements serve as the elementary factors that comprise a component. The elements for the signal processing component include but are not limited to computers, amplifiers, media players, and loudspeaker, primarily any element that is involved in the transformation of the auditory signal into a broadcasted sound. The elements for the environmental acoustic component are the spatial area quantities and the construction materials at the surface boundaries. The elements for the human hearing system component include but are not limited to Equal Loudness Contours, and A-Weighting. The primary functions of these elements are to modify sounds to mimic the human hearing system.

Each level of the system is designed to serve a function for the overall operation of the ADS. The subsystem level serves as an input relationship to the component level indicated by the arrows, as the dynamics are changed in the natural world, this information must be transferred and transcend to the referent. The referent then serves as an input to the signal processing component to become an audible sound. The auditory signal of the referent flows through the signal processing elements until its output into the ambient environment. The loudspeaker or audio output device serves as an input into the environment. Once output the environmental acoustics component is activated, quantifying the environmental acoustics elements ensures the signal is heard as it is intended upon reaching the ear. Lastly, once the ear is reached the human hearing system component must be quantified through its elements.

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An ADS is an hierarchical arrangement of interrelated components working together to communicate the state of a system in the natural world through sound (Blanchard et. el., 2006). The subsystem must have the following characteristics:

1. The behavior of the each referent has an effect on the properties and behavior of the overall ADS
2. The properties and behavior of each referent depends on the properties and behavior of other referents
3. Each referent has the two properties listed above; single referents can only be used to describe one set of phenomena.

These characteristics allow the individual referent to be developed as a subsystem; the subsystems can then be integrated into the overall ADS. In construction of ADS a clear understanding of the function of the system must be established. ADS functions can either be a continuous monitoring system, a task triggering, or an alarm system. These functions can also be subsystems of an overall system that communicates the state of a system across the entire spectrum of urgency. Once the function is established a system can be classified to provide insight into their application sound *(Blanchard et. el., 2006).*

An ADS is a human-made systems which humans have intervened through components, attributes and relationships (Blanchard et. el., 2006). Humans establish the relationship between the acoustic parameters of the auditory signal and the event in the natural world. This mapping processing described in Chapter 2, defines the procedure for the construction of a referent. Human systems require engineering, modeling, and evolution to

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be efficient. Just as natural systems evolve and adapts to its environment, a human made system must be designed with these capabilities.

Blanchard sates an ADS is a physical systems those that manifest themselves in a physical form (Blanchard et. el., 2006). To construct a physical system, one must model the ADS with a conceptual system. The conceptual system is one in which symbols represent the attributes of the components (Blanchard et. el., 2006). Conceptually the ADS can be modeled with a block diagram, a full presentation of the conceptual model will be provided in Chapter 4.

Blanchard states an ADS system is dynamic systems which combine components with activities (Blanchard et. el., 2006). Dynamic systems are allowed to change as it interacts with its environment, as the referent changes, the signal changes, and the behavior of the user is modified due to changes in the auditory signal. Lastly, an ADS is an open system which interacts with its environment. Open systems allow the information to cross the boundaries of the system (Blanchard et. el., 2006).

The construct of ADS are complex and requires the knowledge from several distinct fields. The contribution from each field increases the likelihood of success for creating ADS. A life cycle approach must be considered and accommodations for the system to evolve must be incorporated into the original ADS. As the system in the natural world evolves the ADS should reflect this evolution. Lastly, in the construction of an auditory system a top down will ensure that lower level systems do not take precedence of the overall ADS functions. Optimizing each individual referent independently will lead to an overall sub optimized system.

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In a systems approach to the design of an ADS there are several requirements that must be fulfilled. These requirements ensure appropriately designed ADS which align with the dynamics of the natural world with its respective referent, the fundamental approach as developed by Blanchard, (2006).

1. *Identify the primary function of the ADS.* In devising the function for the ADS employ a top down approach, input from the process expert is critical is establishing the scenarios and events which take place in the natural world.
2. *Determine how the system will be engineered.* ADS are constructs are human made systems designed to explain phenomena in natural systems. The entire life cycle of the natural system must be considered in the engineering of the ADS so alignment exists between the two systems.
3. *Identify relationships at various levels of the system.* The relationship amongst and within the various levels of the system must be identified, documented, and quantified. Failure to identify key relationships will lead to inefficient ADS.
4. *Identify the interaction with the environment.* An auditory display system is an open system which interacts with its relationship. Interactions between the primary signal and reverberant field must be appropriately quantified. Interaction with the natural system and ADS must be aligned. Any interaction the ADS have with any internal or external factor must be addressed.
5. *Identify sources of error.* Sources of error lead to a loss of intelligibility and misinterpretation. These errors can minimize by modeling and calibrations.

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6. *Identify the appropriate resources.* ADS required knowledge from several distinct fields, each field must be appropriately represented in order to appropriately the ADS.

Consideration of each of the items outlined is critical for the successful implement of ADS. If one of the criteria outlined is not satisfied it will surface as a failure in the ADS. This failure can be minor or catastrophic nature, but can always be avoided with a systems approach.

The existing literature provides a detailed outline of the construction of an auditory display, but does not provide the same level of detail in regards to the acoustics of the environment in which the auditory display is constructed for. Since the environment effects the interpretation of any sound wave, an analysis of the impact of the environment with the auditory display merits investigation. There has been little evidence identified in the literature reviewed that the acoustic environment was first accurately characterized prior to the conducting the analysis for the auditory. The fact the acoustics and auditory display were treated independently can confound the results in the mapping process. This raises the question if the results are due to the acoustics of the signal or the acoustics of the environment.

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CHAPTER 4. AUDITORY DISPLAY CONSTRUCTION

The auditory display system developed for this thesis is one to communicate the urgency for a UAV operator to attend to a background task while performing a primary task. Before developing for a complete system, the ability to communicate urgency must be established. The work presented for the remainder of this thesis focuses on the factors which determine if and how urgency can be communicated to the UAV operator.

Urgency is being used to represent the status of the UAV, which oscillates between a normal operational state and an urgent state where an alarm or emergency action is needed. Since there is extensive research on alarm system design, the goal is to develop the physical system with the appropriate equipment, and provide initial data on the viability of communicating different levels of urgency, in particular below alarms levels.

The system being developed can be viewed in several ways. In addition to the perspective presented in Chapter 3, Figure 4.1 shows a system perspective commonly used to develop signal processing and control systems. This system model shows an input, "desired action", and at the far left is modified by elements in the system to produce an output, "action", at the far right. A critical aspect of the model in Figure 4.1 is that the human is contained in the model. This is done because there are existing models that relate the sound exposed to a person's ear, and the resulting perception of this sound. Figure 4.2 is a frequency domain transfer function block diagram of the signal in the ADS.

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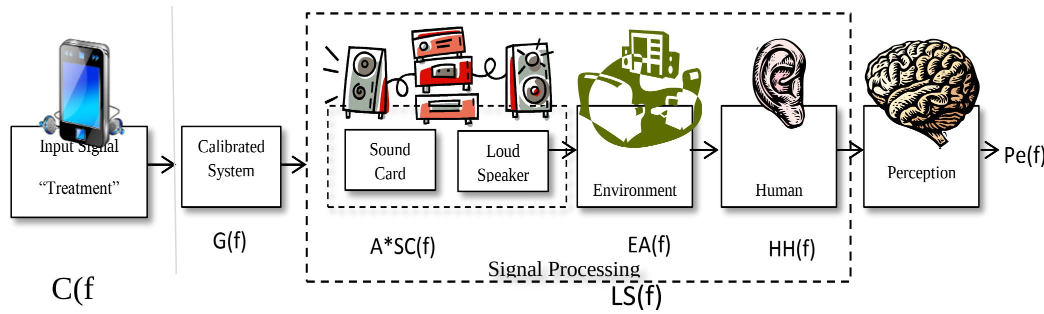

Figure 4.1 - Graphical symbolic representation of the auditory display, including sound producing equipment, environment, human hearing, and perception.

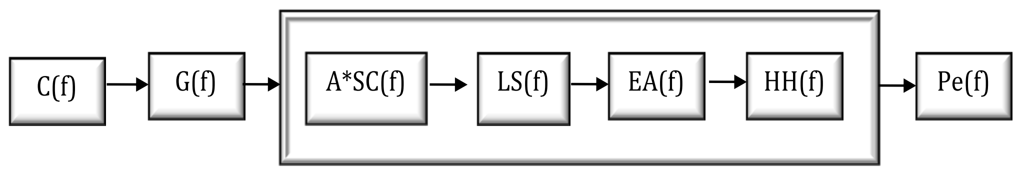

Figure 4.2 –Conventional block diagram representation of the auditory display, including sound producing equipment, environment, human hearing, and perception.

In terms of the signal processing elements, each block in Figure 4.1 is represented by a transfer function, where the transfer function, which relates the input and output of the that block to a known physical parameter, the quantities in Figure 4.2 are:

- *C(f)* - is the intended auditory signal,
- *Pe(f)* - is the sound as perceived by the subject,
- *A*SC(f)* - is the transfer function of the audio system electronics,
- *LS(f)* - is the transfer function of the loudspeaker,
- *RA(f)* - is the transfer function of the room,
- *HH(f)* - is the transfer function of the human hearing,

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- *G(f)* - is the calibration transfer function.

The reason for the signal processing systems approach is to develop a procedure to calibrate signals used in the auditory display. Existing literature mentions calibration being performed, but omits the methodology employed to produce repeatable experiments. This omission creates an uncertainty in the interpretation of the results and questions the integrity of the calibration process. Further, accurate characterization of each component in a system will ensure the signal processing, environmental, and human hearing effects are not a factor in the reproduction of a sound.

The block diagram in Figure 4.2 does appear to be unique to other systems documented in ADS related literature, in that the human and room responses are included. The human hearing and room responses are included here to ensure that changes perceived by the user are caused by the change in the input signal, "the referent", and not the manner in which a human perceive sound or how the room alters sound. As an example, several studies have shown that people rate a higher frequency alarm as indicating a higher level of urgency but also indicate that a louder alarm indicates a higher urgency (Patterson, 1982; Edworthy, 1991). However humans experience higher frequencies as louder. Therefore, what is not clear in these results is if the higher frequency alarm is perceived as being more urgent, because the human have a greater sensitivity to higher frequencies, and thus hear the higher frequencies as being louder or if a higher frequency by itself is perceived as more urgent. Thus, the human hearing response and changes in a referent must be aligned, and factors that contribute to this response must be identified so the sound that reaches the human ear is the sound as it was originally designed. Therefore, the goal of modeling the auditory display

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system with Figure 4.2 is to directly relate the referent to perception without equipment, room acoustics, or the physiology or psychology of the human hearing system adding uncertainty to the results. Figure 4.3 provides the system model presented in Chapter 3 with a common notation to the ADS system modeled with Figure 4.2.

This allows a relationship between a referent and perception to be implemented with different equipment, and in different room environments. Further, it can be implemented with a diverse range of subjects, and clustering subjects may be achievable based on the finding in the 3D audio applications, and the head related transfer function. The head related transfer function creates filters to spatial locations based on subjects being groups on similarities.

As previously mentioned, conceptual systems are represented as symbols, with its attributes components. A conceptual system is required to quantify the factors at the component and element levels of the system. These levels introduce error, negatively impacting the efficiency of the overall ADS. To date a standardized approach to modeling ADS components has not been established. To relate the following work to the system presented in Chapter 3, Figure 4.3 represents the components of the auditory display system used in the initial experimental work. The remainder of this chapter will provide the background for each element in the system.

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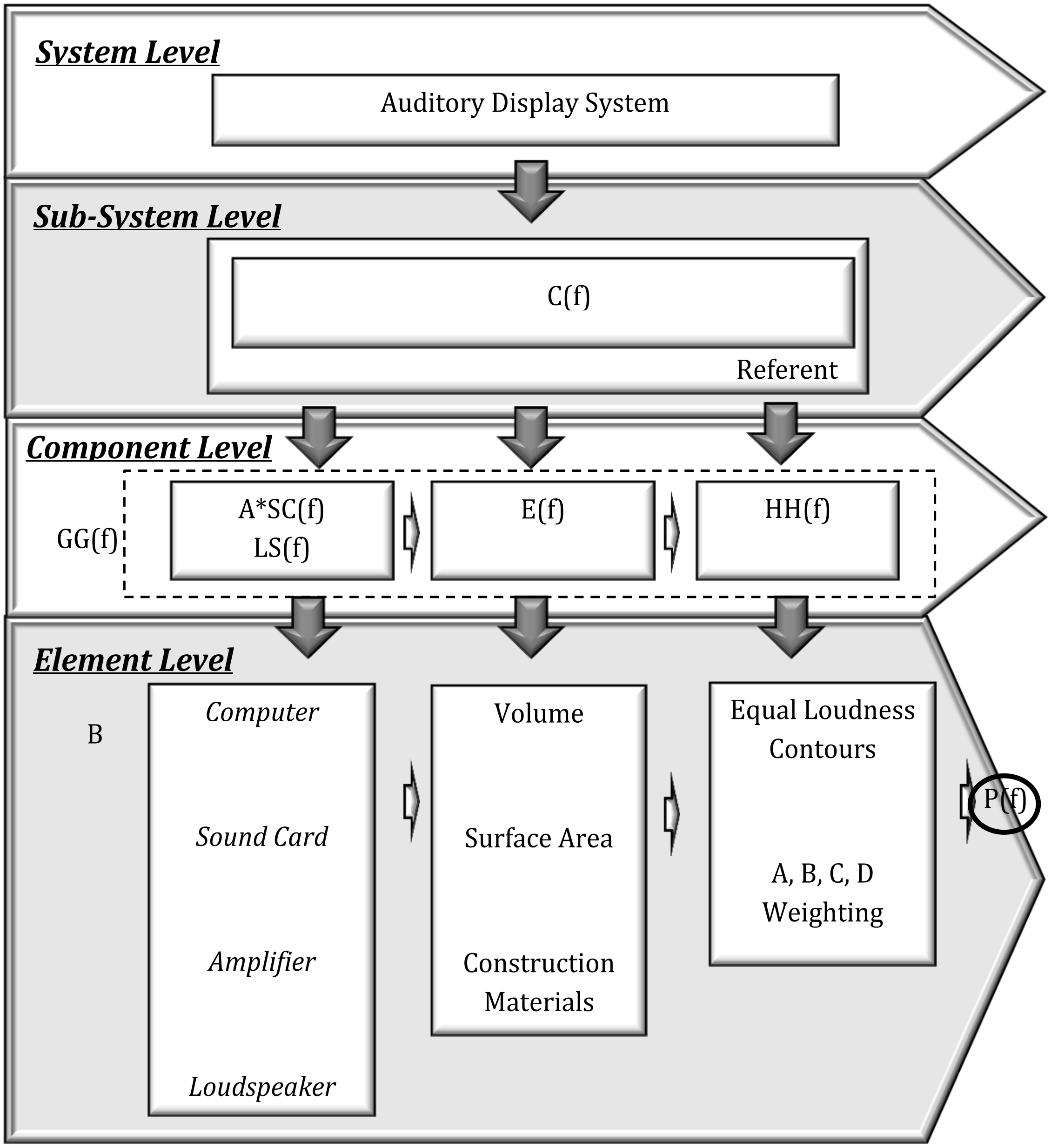

Figure 4.3.- Figure 3.1 filled in with elements from the Conventional block diagram representation of the auditory display Figure 4.2.

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Each of the transfer functions in Figure 4.2 is represented in the frequency domain, which is related to the time signals through the forward Fourier transform. Using, the intended auditory signal, C (t), as an example the Fourier transform is defined as,

$$C(f) = \int_{\infty}^{\infty} C(t) * e^{-j\omega t}\, dt \qquad [4.1]$$

where C(f), represents a signal in the frequency domain, and $\omega = 2\pi f$, where f is frequency and $j = \sqrt{-1}$ for equation 4.1 and 4.2. The frequency domain can be converted back to the time domain through an inverse Fourier transformed as described by,

$$C(t) = \frac{1}{2\pi} \int_{\infty}^{\infty} C(\omega) * e^{j\omega t}\, d\omega \qquad [4.2]$$

where t is the time. Equation 4.1 & 4.2 are required to convert between the frequency and time domains, and manipulated in both domains. For example, the calibration will be performed in the frequency domain, but the calibrated sound is broadcasted to a subject in the time domain. Further, measurements for calibration will be taken in time domain, but the calibrations corrections will be applied in the frequency domain.

4.1. Calibration

The combination of the calibration process, environmental acoustics, and human hearing components can be quantified by a single term, G(f). Using Figure 4.2, the relationship between the intended auditory signal, C(f), and the sound perceived by the subject, Pe(f), is

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$$Pe(f) = [G(f)][A*Sc(f)][LS(f)][RA(f)][HH(f)][C(f)] \ . \qquad [4.3]$$

In this model, the transfer functions or variables that can be controlled are (1) the intended signal, C(f), (2) the calibration transfer function, G(f), and (3) the audio system electronics amplification, A. Equation 4.3 is used to relate the calibrated transfer function to the system response. To begin, it is the intent of the auditory display to have the intended auditory signal, C(f), be equal to the sound perceived by the subject, Pe(f),

$$Pe(f) = [C(f)] \qquad [4.4]$$

This relationship ensures the designed signal is the same signal, which is perceived by the subject. Using Equation 4.3 and 4.4, solving for the calibration transfer function, G(f), yields

$$[G(f)] = \frac{1}{[A*Sc(f)][LS(f)][RA(f)][HH(f)]} . \qquad [4.5]$$

The calibration transfer function is then used by multiplying the Fourier transform of the auditory signal, S(f), by the calibration transfer function, G(f), and taking the inverse Fourier transform yields,

$$S_c(t) = \frac{1}{2\pi} \int_{\infty}^{\infty} G(f) * S(f) * e^{j\omega t} \, d\omega \qquad [4.6]$$

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where $S_c(t)$ represents the calibrated auditory signal. When this signal is input to the system electronics as the treatment, one expects the perceived signal to be aligned with the referent. In order to determine the calibrated transfer function G(f), each of the individual attributes of the transfer function must be accurately quantified.

4.2. Misinterpretation, Intelligibility, and Criteria

When designing an auditory display system and determining the appropriate representation for the transfer function in Equation 4.5, sources of error must be considered. The different errors can be classified by their origin: causing a user to misinterpret the intended communication or making unintelligible the characteristics of the sound being used to communicate information. In the case of misinterpretation, the user hears the characteristics of the sound which are designed to communicate information to the user, but the sound characteristics are not interpreted as intended. For example, if frequency is intended to communicate a variable but a human hears a higher frequency as being louder, based solely on the characteristics of the human hearing system, then the interpretation of the frequency change is a change in its sound level and not the frequency.

Intelligibility is different in that the sound is designed correctly, but something such as background noise, interferes with the sound so that the subject does not hear the sound as designed. An analogy is when talking to someone at a party with a high background level; it is difficult to understand what they are saying. The psychoacoustic community defines these phenomena as masking, this occurs when a primary signal is masked by secondary signal, and the primary signal is the signal of interest. Masking occurs in ADS due to the sound

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field. A brief description of the acoustic sound fields are presented later in this chapter, and a full description of masking and its impact will be provided in Chapter 5.

If one considers the conceptual system view outlined in Chapter 3, the errors can manifest themselves from sources at the subsystem and component level. The error at the component level is primarily due to the unsuitable mapping in the construction of the referent. The mapping procedure identified by Edworthy, outlined in Chapter 2 can be employed to minimize the error associated with mapping. Mapping is a subjective process so quantifying its error can be difficult. The sources of error at the component level are quantifiable through the implementation of mathematical models and well defined theory. This ability to model the component level will minimize error associated with its attributes.

In the calibration efforts, criteria must be applied, they are referred to as “just noticeable differences”. Thus, one cannot expect a systems calibration to accuracy greater than the just noticeable differences to be perceived by the subjects. Just noticeable differences quantify the minimum perceptible difference in amplitude, frequency, phase, and masking characteristics of an auditory signal. The human hearing system has the ability to resolve acoustic characteristic within a defined threshold, just noticeable differences quantifies this resolution.

As mentioned previously loudness is an intensity sensation, which corresponds to the amplitude variation of a signal (Fastl, 2007). The criteria for just noticeable differences in amplitude variation of a pure tone are 1%, for sound levels up to 100dB (Fastl, 2007). The practical implications of this observation is a sound which deviate in amplitude by more than 1%, and this deviation is not intentional then sounds may be evaluated strictly on this

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deviation. Amplitude variations that are not embedded in the ADS must be eliminated or a UAV pilot may begin to interpret sound sources based purely on error.

The just noticeable differences in amplitude variation of a pure tone are 1dB (Fastl, 2007). Thus the calibration should be performed to an accuracy of 1dB . In general the just noticeable difference for frequency content is 2Δf, where Δf is the frequency difference between the carrier frequency of the signal and modulation frequency. The just noticeable difference criteria will be used in the development of the transfer functions for the calibration described in equation 4.5 and 4.6.

4.3. Modeling Each Transfer Function

Each of the transfer function is Figure 4.2 will be modeled for the purposes of calibration and minimizing their impact on intelligibility. In the following section, each component will be discussed, presented with a theoretical model or the basis of the measurements needed to experimentally determine the model.

4.3.1. Computer, Amplifier, and Loudspeaker

The transfer function for the computer and amplifier, A*Sc(f), and the loudspeakers, LS(f), can be determined individually, however were combined as a single component,

$$[A*Sc(f)][LS(f)] = A*Sc*LS(f). \qquad [4.7]$$

The lack of a theoretical model did not allow the decomposition of the individual component, thus measurements were preferred to determine the combined transfer function. Measurements could be used to obtain data or expressions to measure $S_c(f)$ and LS(f)

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separately, but due to complexity were not performed. Modeling $S_c(f)$ and LS(f) separately would be useful if different components were used for each of the attributes of the transfer function.

The transfer function measurement requires that the input signal, In(f), and the output signal, Out(f), be measured or known simultaneously. The transfer function, A* $S_c(f)$ *LS(f) can be calculated

$$A*Sc*LS(f)=\frac{Out(f)}{In(f)}\,. \qquad [4.8]$$

In the case of this work a known input was used along with a measured output. One would expect higher accuracy with measuring both the input and output, and therefore should be considered in future work.

Critical issues to the measurements are the input signal and the measurement. The input needs to be a broad band signal, which excites the elements over a large range of frequencies. A continuous noise signal was chosen, because of the measurement equipment being used. Within the category of broad band signals, there are many signals to choose from, two being white noise, which has an equal amplitude at every frequency and pink noise, which has equal amplitude in each octave band, thus have more signal at low frequencies.

White noise was chosen because it could be easily created with a random white noise generator, creating a .wav file to be input to the computer's sound card. There were some measurements challenges caused by this signal choice, which will be discussed in the next

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chapter, therefore future work should examine different input signal options. There are multiple choices of the spatial location to measure the output signal. The computer and loudspeaker can be placed in an ideal environment such as an anechoic chamber, or an environment representative of the natural environment in which the auditory display will be implemented. Issues related to the frequency resolution of the output measurement and connection to the room acoustics are presented later in this chapter.

4.3.2. Environmental Room Acoustics

The acoustics of the room impacts both the calibration, misinterpretation and intelligibility of the audio display. In many testing environments and applications, headphones are used to avoid these issues. Many subjects prefer not to use headphones for a long duration for typical applications, and this deserves greater exploration. Further, headphones have a transfer function associated with them and are not a guaranteed solution to ignoring room effects. For this research, the impact of the room is studied to provide a model which identifies issues for future research.

Quantifying a room in terms of its reverberant time allows a comparison amongst different environment. Reverberation time is the time for sound to decay in a room, this is a common quantity used to describe a room, but is not sufficient for designing an ADS. Each room has a varying geometry, length characteristics, and construction materials which quantifies its reverberation time. The more absorbent the material at the boundaries, the lower the reverberation time for the environment. The less absorbent the material at the boundaries, the higher the reverberation time and when comparing two rooms, the volume and surface impacts the reverberation time. In a single room the volume and surface area are

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static; therefore the dynamic variable is the coefficient of absorption. Adding or removing dampening material such as furniture or people can change the absorption coefficient of a room increasing or decreasing the reverberation time. A mathematical description of the reverberation time is provided in Chapter 5.

As autonomously controlled vehicle control systems are embedded in virtual environments, sound output into these environments will be through loudspeakers. The loudspeaker will not only communicate the frequency content of the signal, but also the spatial location of a sound. Considering the environment as these communication systems are implemented will quantify all the factors associated with sound. The impact of the room is quantified in three ways:

1. The absorption and room dimensions which varies with frequency can change the sound amplitude and the frequency content in the environment, leading to a change in perception as described by the transfer function EA(f) in Figure 4.2.
2. The sound caused by reflections interfering (masking) the sound used to communicated.
3. Background noise unrelated to the audio display which masks the audio display sounds.

4.3.2.1. Room Calibration

The acoustics of a sound in a room begins when an auditory signal is radiated by a sound source. The room creates two distinct sound fields: the direct sound field and

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reverberant sound field. The direct sound field is the sound field that propagates directly to a person and has not undergone any reflections or absorption at the boundaries. In a simplified model of a loudspeaker the sound level of the direct field decays 6dB with doubling the distance from a source. The sound level decay is experienced until the reverberant sound field is reached. The reverberant field is the sound field which has undergone absorption and reflections at the boundaries. The sound level in the reverberant field is idealized as constant, however sound level typically fluctuations exist. Modeling the reverberant field with an energy balance where the input and output energy are equal, results in a differential equation for the root mean square pressure of the reverberant field in a room, P_{rms}^2, is

$$\frac{V}{\rho c^2}\frac{d(P_{RMS}^2)}{dt}+\frac{\alpha S}{4V\rho c}P_{RMS}^2=W_{in}(t)\,. \qquad [4.9]$$

where V is the room volume, ρ is the air density, c is the speed of sound in air, S is the surface area of the room, α is the room absorption coefficient, and $W_{in}(t)$ is the input sound, in our case from the loud speakers.

Solving the differential equation, Equation 4.9 for the steady state sound, $\frac{d(P_{RMS}^2)}{dt}=0$, and adding in the direct sound, the total sound pressure in the room yields,

$$P_{rms}^2=\frac{W_{in}}{2\pi\rho c}\left[\frac{1}{4\pi r^2}+\frac{4}{R}\right], \qquad [4.10]$$

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where r is the distance from the sound source, and $R = \frac{S\alpha}{1-\alpha}$ is commonly called the room constant. Equation 4.9 shows the division of the sound field into the direct field, the first term, which is dependent on the distance from the sound source, the reverberant sound field, which is independent of distance from the source.

Graphically the steady state sound in a room can be represented as shown in Figure 4.4, where the direct field is shown to decay with distance and the reverberant field is constant everywhere. When close to the loudspeaker, the sound is dominated by the direct field and when far from the source, the sound is dominated by the reverberant field.

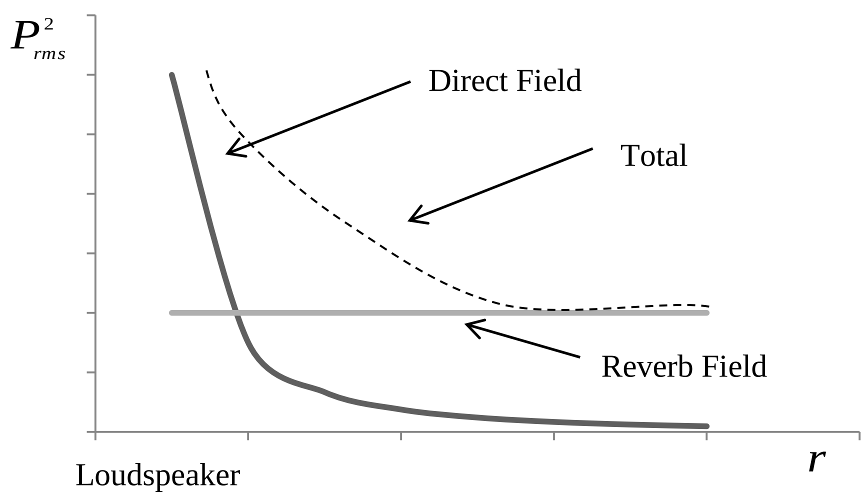

Figure 4.4 – Loudspeaker output to ambient environment, direct, reverberant, and total sound field as described by Equation 4.10.

The reverberant room model, Equation 4.10 and Figure 4.4 raises two issues related to calibration when designing an auditory display system.

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1. In the direct field, sound level changes with distance, thus a subject may hear a sound change which is only created by the subject moving their head.
2. The acoustic parameters of the room vary the amplitude of the sound.

First consider the spatial variation of sound in the direct field. In order to ensure that the sound level is not significantly altered by head position, the user can be placed in the reverberant field of the loudspeakers. Next consider the sound level variation caused by the room acoustics, e.g. sound absorption. This should also be measured using an input/output measurement technique. Chapter 5 will discuss how this attribute can be analyzed.

4.3.2.2. Room Acoustics and Intelligibility

Reverberation and background noise impacts intelligibility of the signals of an audio display by masking the signal's sound. Reverberation is a particular issue when time variations in a sound are used as communication variables. If the signal used for this research is created by amplitude modulation, the reverberant field may interfere with hearing the lower sound level segments of the signal caused by the modulation. A general model is developed for the specific signal and treatment used in this experimental work. The particular sound is an amplitude modulated sound which as a sound power which is

$$W_{in}(t) = W_{in}(1 \quad A_m) + A_m * \cos(\omega_m t) \qquad [4.11]$$

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where A_m is the amplitude of modulation, and ω_m is frequency of modulation. Equation [4.11] is substituted into the differential equation, equation 4.10 and solved for the root mean squared sound pressure, P_{rms}^2 of the reverberant sound field yielding,

$$P_{RMS}^2 = \frac{\left[\frac{\left[\frac{A_m \rho c^2}{V} * e^{\frac{\alpha Sct}{4V}}\left[\frac{4\rho c}{\alpha S} * e^{\frac{\alpha Sct}{4V}} - 1\right]\right]}{\left[(\omega_m)^2 + \left(\frac{\alpha Sct}{4V}\right)^2\right]}\right] * \left[\omega_m \sin(\omega_m t) + \frac{\alpha Sct}{4V}\cos(\omega_m t)\right]}{2 * e^{\frac{\alpha Sct}{4V}}} \qquad [4.12]$$

As an example, Figure 4.5 shows the predicted reverberant field from Equation 4.12, along with the direct field. (Details for the calculations are provided in Chapter 5). The red line in Figure 4.5 represents the reverberant field, determined by the reverberation model Equation 4.11, and the blue line represents the modulated signal in Equation 4.11. It is important to note in the Figure 4.5 and 4.6, only the envelope of the auditory signal is plotted. Figure 4.6 shows some data for the same signal but different modulation amplitude, A_m. Comparing the two figures, in the case of Figure 4.5, the reverberant field will not be masked by the direct sound. However, Figure 4.6, reveals the reverberant field will mask the direct sound at the low levels, making it so that the person will not hear the amplitude modulation as intended. To quantify the masking, a quantity ΔdB is defined as

$$\Delta dB = (Min(L_D) - Max(L_R) \qquad [4.13]$$

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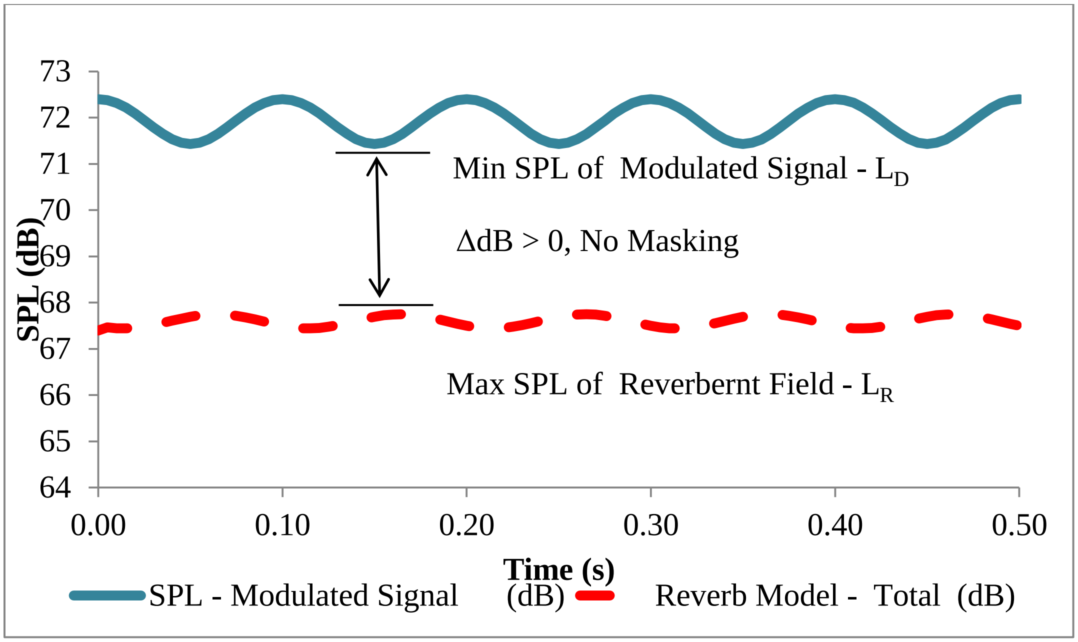

Figure 4.5 – Modulated signal envelope, indicated with blue line, and reverberant field envelope, indicated by red line, where no masking is present.

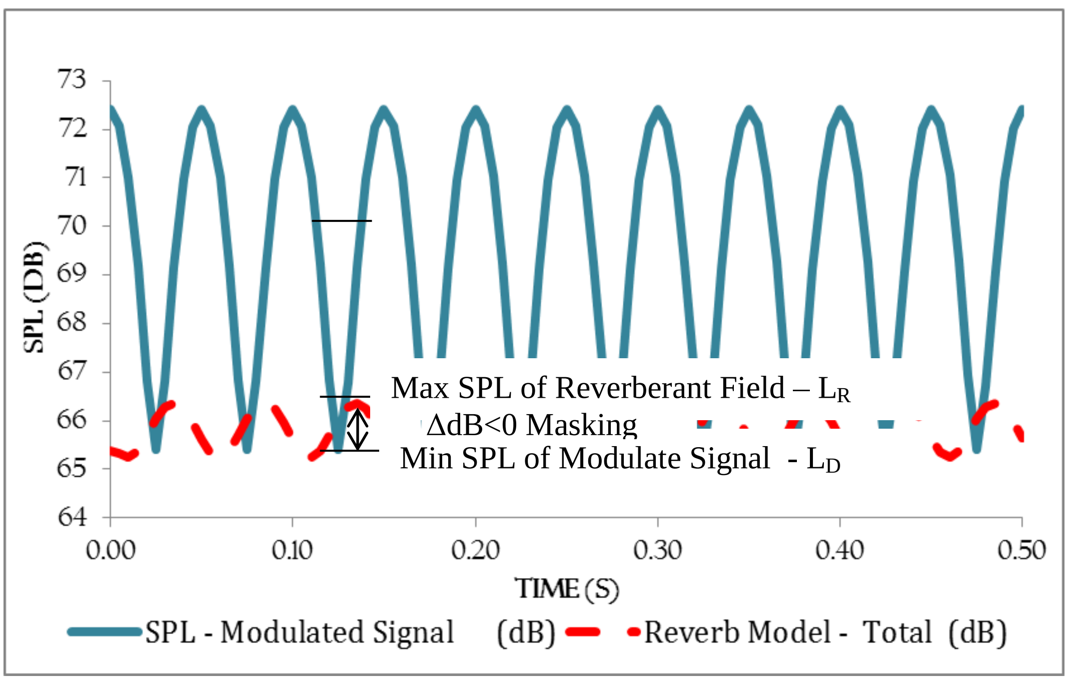

Figure 4.6 – Modulated signal envelope, indicated with blue line, and reverberant field envelope, indicated by red line, where masking is present.

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Where Min(L_D) is the minimum sound level of the direct sound field and Max(L_R) is the maximum of the reverberant field. When ΔdB is positive ($\Delta dB > 0$), as in Figure 4.5 , no masking is present and when ΔdB is negative ($\Delta dB < 0$), as in Figure 4.6, there is masking. Graphically, no masking is indicated by a lack of interaction between the two signals. Graphically, masking is indicated by an interaction between the reverberant field and the modulated signal. While not explored in this thesis, ΔdB depends not only on the signal modulation, A_m and ω_m, but also the room characteristics, α, S, V, and the distance, r, from the source to the user. Future work should examine these relationships. The final issue is background noise, which has the potential to mask the audio signal and can only be corrected by reducing the background noise or increasing the volume of the auditory display's signal.

4.3.3. Human Hearing

The final component in an auditory display system is compensating for the human hearing system. This component is difficult to quantify due to the difference amongst humans and their psychology in the interpretation of sounds. The human hearing system is complex in nature; to date there is little agreement if all humans interpret sounds in the same manner. Interpretation of sounds is confounded with: sensitivity to hearing, experience, and exposure. The experience encountered over a life time provides the framework for the interpretation of sound events from our environment. The greatest source of error will be the differences amongst humans contributed by their hearing system.

The effects of human hearing system can be modeled with well-established experimentally determined standards. The standards used in this experiment were the A-

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Weighting Factor and Equal Loudness Contours. Both are explored, since they are developed for different purposes; A-weighting was developed to model the susceptibility for hearing damage while Equal Loudness Levels were designed to predict sound level perception. Implementing either standard can be accomplished by either applying a curve fit to data points or through tabular data. Each standard provides the transfer function for the quantity, HH(f), in either a graphical, tabular, or equation form.

The International standard IEC 61672:2003, gives the A-Weighting Factor in equation form. It is critical to know that Equation 4.14 applies the A-Weight at each centre frequency of the third octave band. Thus if Equation 4.11 is to be used for the transfer function HH(f), some additional manipulations will be needed and will be described in Section 4.4.

$$R_A(f) = \frac{12200^2 * f^4}{(f^2 + 20.6^2)(f^2 + 737.9^2)(f^2 + 12200^2)(\sqrt{(f^2 + 107.7^2)}} \qquad [4.14]$$

Equal Loudness Contours are a curves corresponding to a subjective evaluation of loudness, average over a large population. This is a magnitude estimation parameter that determines the relationship between a physical sound level and judged volume (Moore, 1982). The Equal Loudness Contours curves used in this research comes from ISO 226:2003. The 40 phon contour was selected for this experiment. Figure 4.7 contains two curves considered the 40 phone curve, the blue curve is the original 40 phon curve, and the red curves are the corrected curves which began being used after 1996. Thus there is variability in curves or models, even from standards. Future work should address these

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issues. For the purposes of the research presented in this thesis, the data for the ISO Equal Loudness Contours is tabulated and used to develop an equation through a curve fitting process which will be presented in Chapter 5.

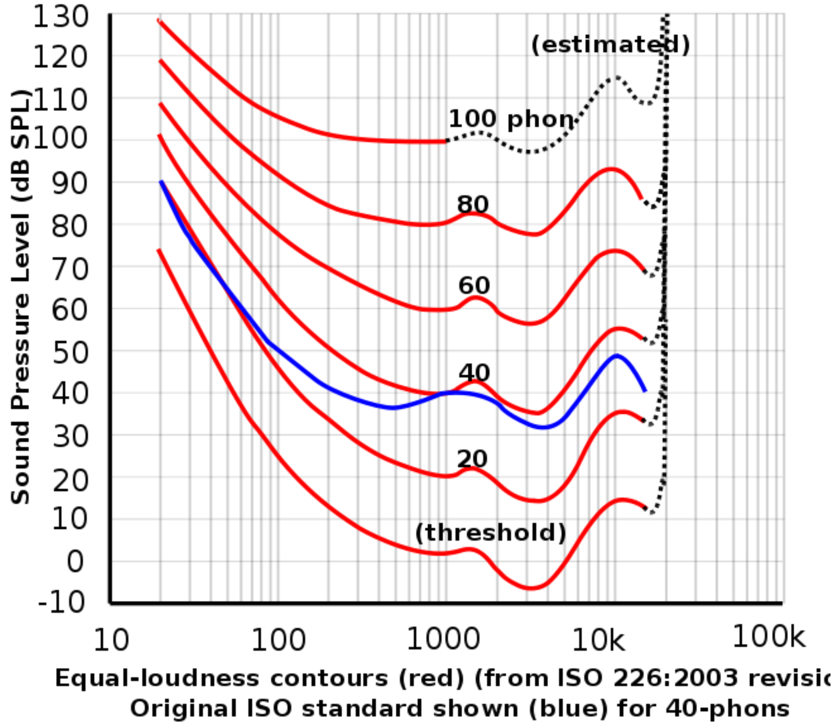

Figure 4.7 – Equal Loudness Contours furnished by ISO 226:2003

4.4. Common Signal Processing Issues

Two signal processing issues were faced when developing each of the transfer functions: information was measured or provided in (1) a decibel scale (dB) and (2) octave or one third octave bands. The decibel scale and octave or third octave bands are commonly used because they reflect human's perception to sound volume and frequency changes.

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However, when using the transfer function, the values must be known as a linear quantity (Pascal's rather than dB), and at individuals frequencies of the Fourier transform, and not the third octaves.

Measurement data from the calibration stimulus or Equal Loudness Contours can either be made in dB or Pascal. If the measurements are recorded in dB with a standard sound pressure level meter, a conversion to a linear scale Pascal is

$$P_{rms}^{\ 2} = P_o^{\ 2} * 10^{\frac{L}{10}} \qquad [4.15]$$

where L is the sound pressure level, and P_o is the reference pressure, $20*10^{-6}$ Pa. If a sound level meter is used, typically measurements can be taken at octaves, more advanced meters allow sound level measurements of the entire spectrum. If a digital data acquisition system is used sound level measurements can be taken at each frequency as well. The choice of instrumentation will influence the procedure required to implement the calibration process.

Data in octave or one third octave bands requires a relationship between the respective bands and individual frequencies, prior to a forward Fourier transform being performed. This is due to the distribution as a function of frequency is lost, with octave and one third octave data, since the one third octave band data gives a value over the frequency band. A conversion from the octave band to the individual frequency components can be accomplished by

$$V_{rms}^2(f)_{n^{th},band} = \int_{f_L}^{f_u} V^2(f)df \qquad [4.16]$$

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where f_L and f_u are the lower and upper bands limits for the n^{th} frequency band, V(f) is the voltage at the respective frequency f, and V_{rms} is the root mean square voltage. Similarly, one can assume a constant across an octave or one third octave band for data, due to the nature of the stimulus being used. Evaluating Equation 4.16 over a frequency band reveals

$$V_{rms}^{2}(f)_{n^{th},band} = V^{2}(f_u - f_L). \qquad [4.17]$$

Using equation 4.15 and 4.17, data provided in a Decibel scale, octave, or third octave can be converted for use in calibration procedures. As an example, consider the measurement of the transfer function for the computer, loudspeaker, and room,

$$P_{rms}^{2} = [A*Sc(f)][LS(f)][RA(f)][HH(f)][C(f)] \qquad [4.18]$$

where P_{rms}^2 is measured for a calibration input C(f). However, since C(f) is known at each frequency in the Fourier transform (e.g. white noise is assumed to be unity at each frequency and the output P_{oct}^2, is measured in one third octave bands, then equation 4.16 provides the transfer function as,

$$[A*Sc(f)][LS(f)][RA(f)][HH(f)] = \frac{[C(f)]}{[f_u - f_L]P_{rms}^{2}} \qquad [4.19]$$

this can be used to define the transfer function at each frequency. An example of the issues associated with the practical implementation of Equation 4.19 will be discussed in Chapter 5.

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CHAPTER 5. IMPLEMENTING THE CALIBRATION PROCEDURE

To calibrate is defined as the act of standardizing by determining the deviation from a standard so as to ascertain the proper correction factors (Merriam, 2008). Applying the process of a calibration to ADS ensures the auditory signal a subject is intended to hear is what is heard. If an auditory signal is un-calibrated it will be modified by the stereo equipment, room, and human hearing response enduring several alterations, which will create a discrepancy between what is intended for a subject to hear and what is heard.

The specific equipment used for this study and calibration was

1. Desktop Top Computer - Dell OptiPlex GX270 with OEM Sound Card,
2. Loudspeakers - MAudio AV 40
3. Sound Level Meter – Larson Model 800B

All measurements for this study were made in the Virtual Reality Application Center Usability Lab at Iowa State University, 2629 Howe Hall. A detailed drawing of the room is shown in Figure 5.1.

5.1. Room Environment

To begin the background sound level was measured with a sound level meter, recording the equivalent sound, L_{eq} (Hartmann, 1998). Measurements were taken at third octave bands, over a frequency range from 20Hz to 20 kHz, Figure 5.2 shows the measured background levels. One immediately notices that the lower the centre frequency corresponds to decreases in the sound level. This is typical in a new building such as Howe Hall, where

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low frequency rumble from the ventalation system is a significant noise source and ventalation flow noise is contolled through good ventalation system design.

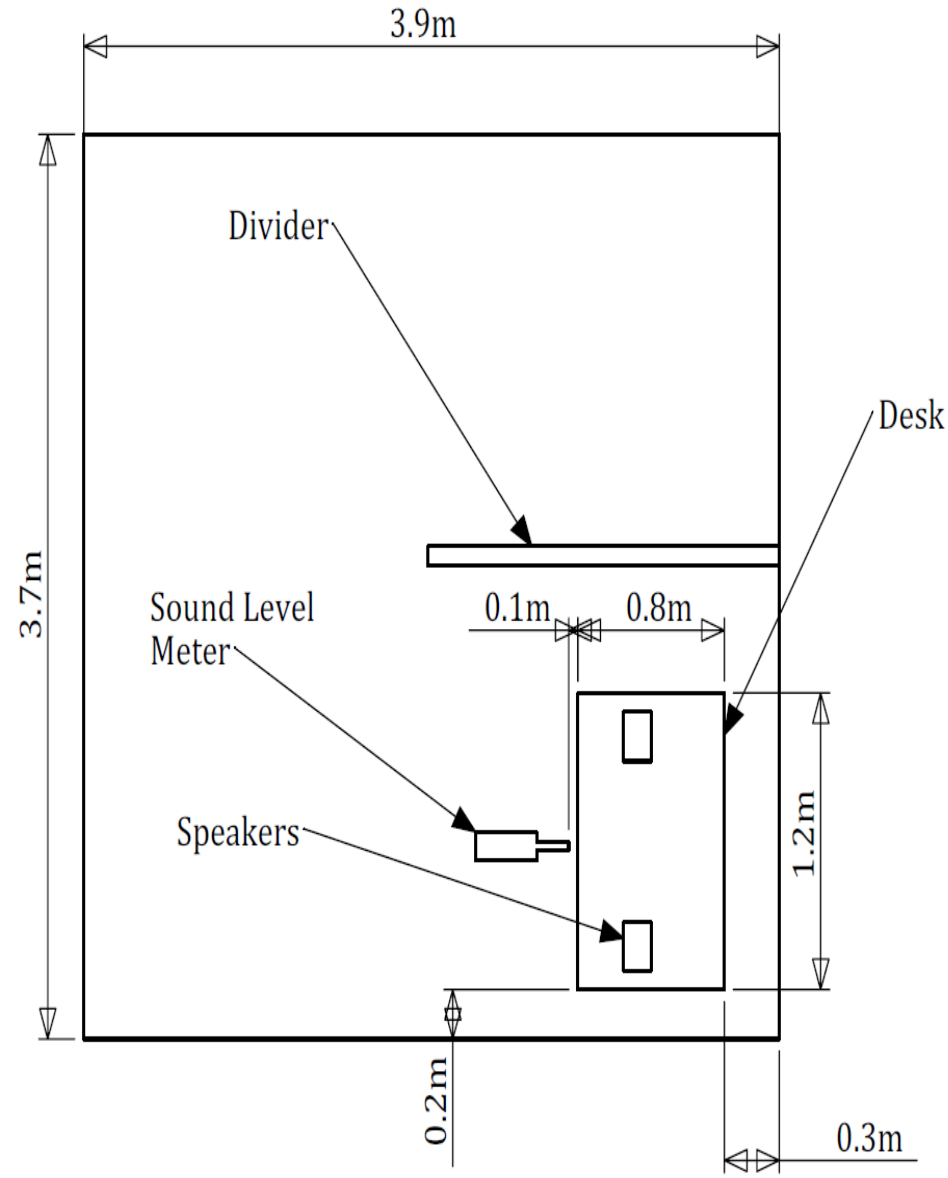

Figure 5.1– Scale drawing 2629 howe hall.

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It should be noted, that higher frequency ventalation flow noise is much more significant in a virtual reality environment, where there may be a large number of projectors which produces significant cooling fan noise.

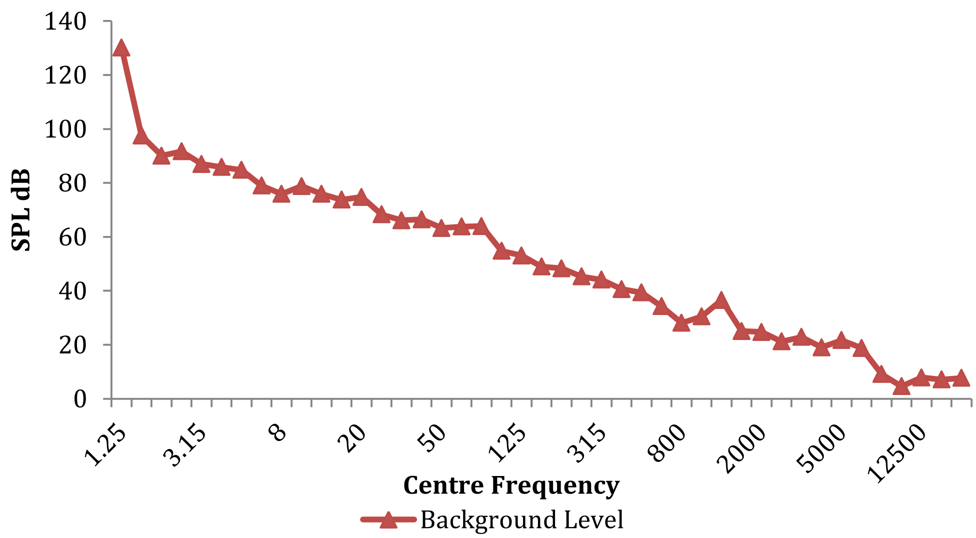

Figure 5.2 – Ambient background level measurement for usability lab located in 2629 howe hall.

The next step is to determine the best spatial location of the measurements; based on the criteria of the calibration stimulus in the direct or reverberant field. To determine the location of the direct and reverbent sound field, third octave band measurements at several bands, and at different locations from the source were measured as shown in Figure 5.3.

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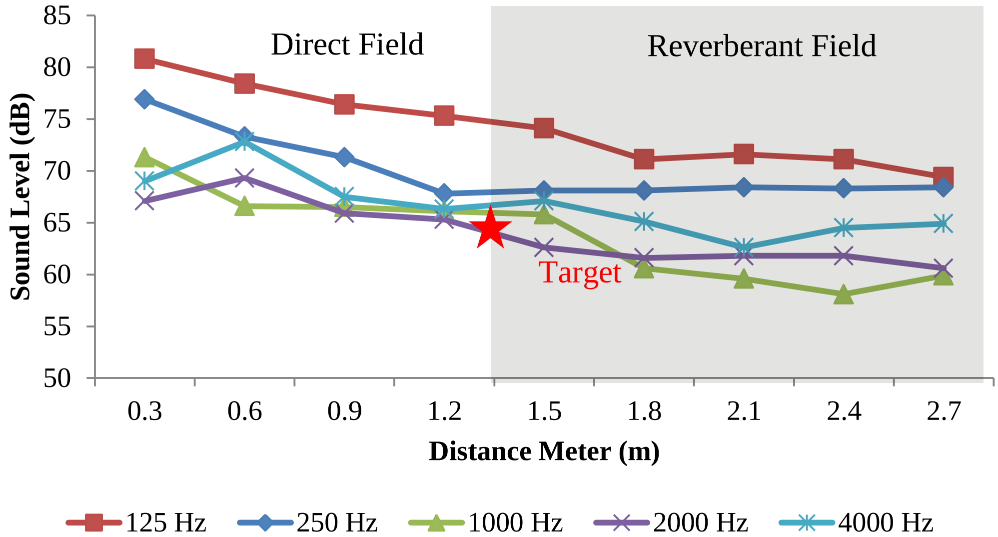

Figure 5.3 – Direct and reverberent sound field measurements at several octave bands.

The direct sound field is estimated to be at a location between .3 meters and 1.2 meters, the reverberent field is the all distances greater than 1.2 meters. An explanation for this choice is provided in section 5.1.1. Thus, the calibration stimulus measurements must be taken at a taken at a distance less than 1.2m, if the measurements are to be with the direct field, and a distance greater than 1.2m, if the measurements are in the reverberent field. The decision was made to perform all measurements in the reverberent field, in a practical application it is diffulcult to determine where a subject will be placed when monitoring an auditory display system, which will negatively influence what is heard at the ear due to the just noticeable difference criteria. In future work a more sophisiticated analysis should be performed to estimate the sound level changes, when a person moves in the direct field compared to typical flucuations in the reverberent field. Further, the trade off with masking by the reverberent field should be included in the analysis.

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White noise was the calibration stimulus selected for the characterization of the sound system. The white noise was generated using a random number generator function in matlab. Ideally, this stimulus should produce a curve with zero slope, which indicates the frequency response is identical at all frequencies. It was found after several measurements the low frequency content was not sufficient to overcome the background level. Violating the first criteria for a valid measurement, which states a signal must be a minimum of 6dB above the background level. To resolve this issue a second signal was constructed, one with a higher concentration of low frequency content.

Thus, two signals were used in the calibration process, a low frequency signal and a normal frequency frequency. Linear sound level measurements of the stimulus were recorded at the third octave bands. Both signals were measured twice to examine repeatability of the measurments. Measurements of the low frequeny signal were recorded at third octave bands from 1.25 Hz to 800 Hz. Figure 5.4 shows the two measured spectra along with the background level. There is a clear indications that the measured data is not valid for the calibration at all frequencies. To analyze the valid frequency range, (1) the difference between the two calibration measurements and the (2) difference between the two calibration measurements relative to the background level are shown in Figure 5.5 and 5.6. Each figure also has an accpetance criteria. Figure 5.5 shows that above 8Hz, the difference between the sound measurements is less than 3dB, leading to the conclusion that the data above 8Hz is considered valid.

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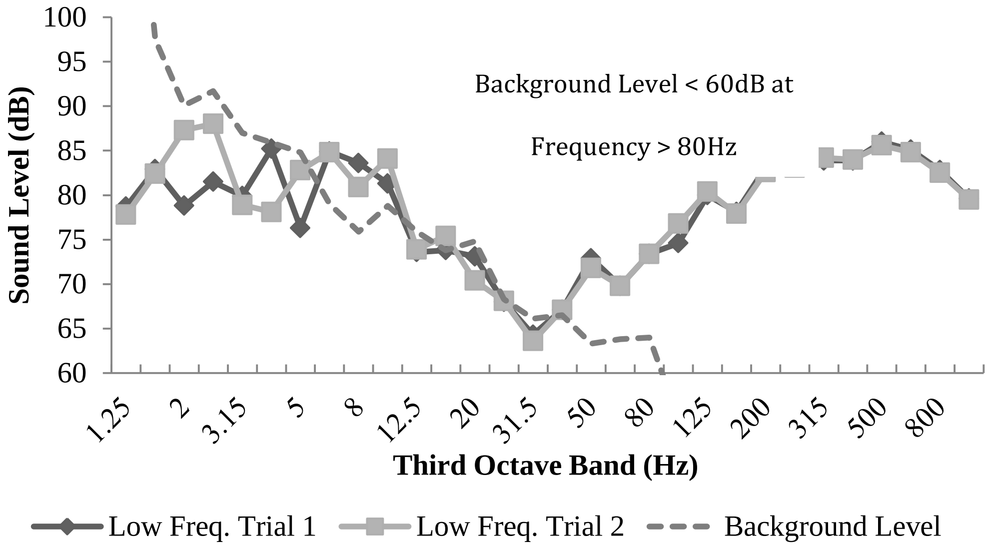

Figure 5.4 – Both low frequency signal measurements with the background level superimposed.

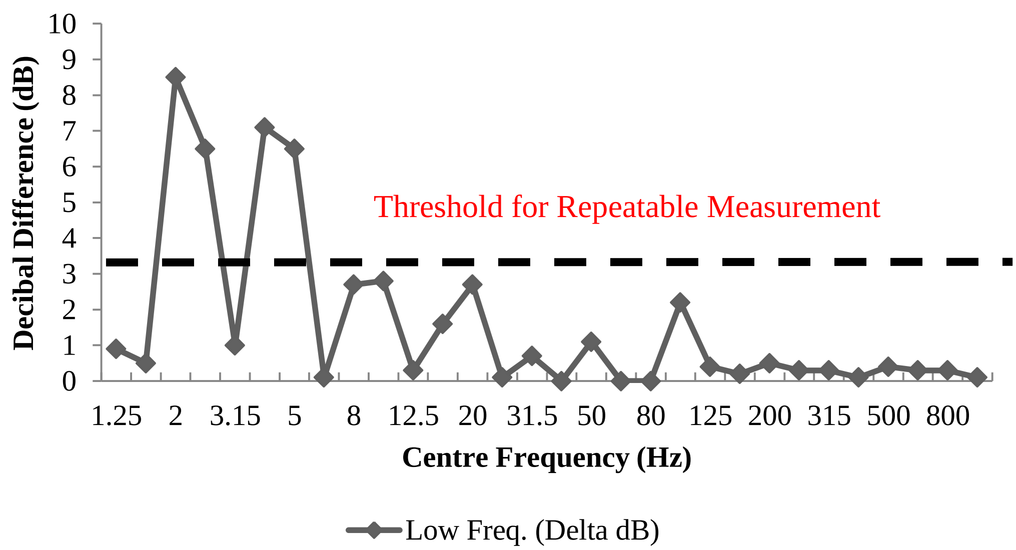

Figure 5.5 – The difference between both low frequency signal measurments, signals below the dash line are valid measurements.

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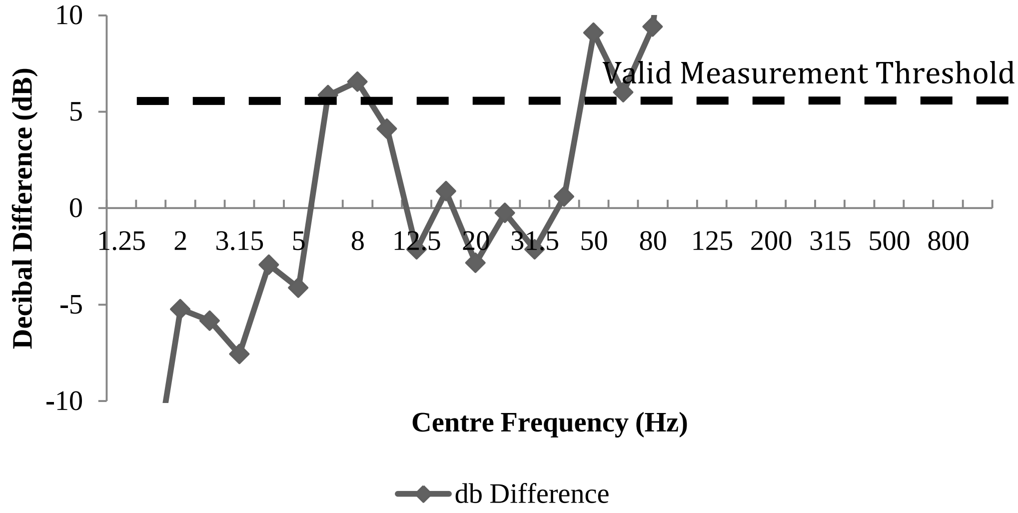

Figure 5.6 – Low frequency signal dB difference between the background level and average of two low frequency signals. All measurements above dashed line are valid, measurement above a centre frequency of 80Hz are greater than 10dB.

Figure 5.6 represents the average sound level of the two low freuqency measurements subtracted from the background level. Negative values indicates the background level is greater than the calibration stimulus, a positive value indicates the calibration stimulus is greater than the background level. The criteria for a valid measurement is the signal being 6dB above the back ground level. Figure 5.6 illustrates the low frequency signal data can be at centre frequencies above 50Hz. Thus, the measurement for the low frequency signal are valid only at a centre frequencies greater 50Hz. In performing the calibration the minimum centre frequency that can be calibrated is the 50Hz band. Applying the same assesment to the normal frequency signal is observed in Figures 5.7 to 5.9, thus data for this signal is valid above the 80Hz one third octave.

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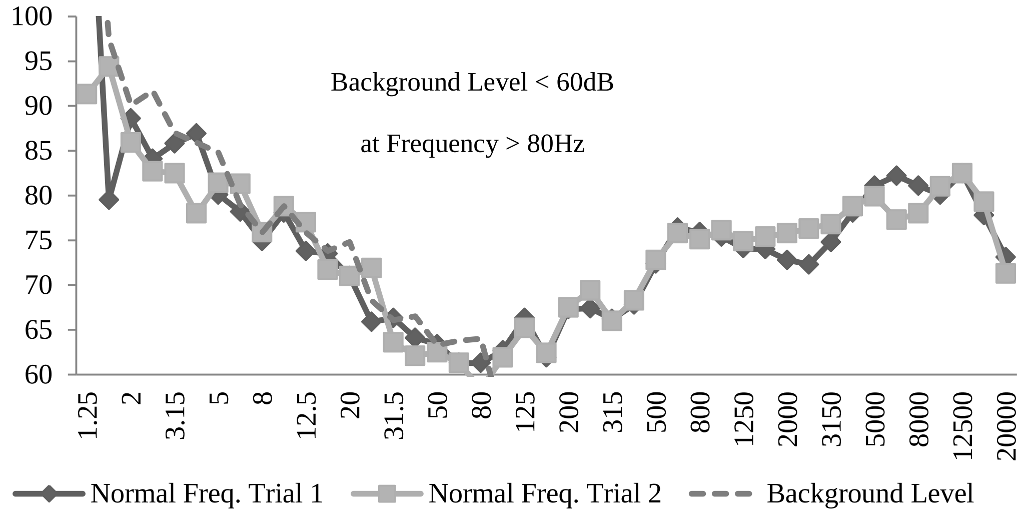

Figure 5.7 – Both normal frequency signal measurements with the background level superimposed.

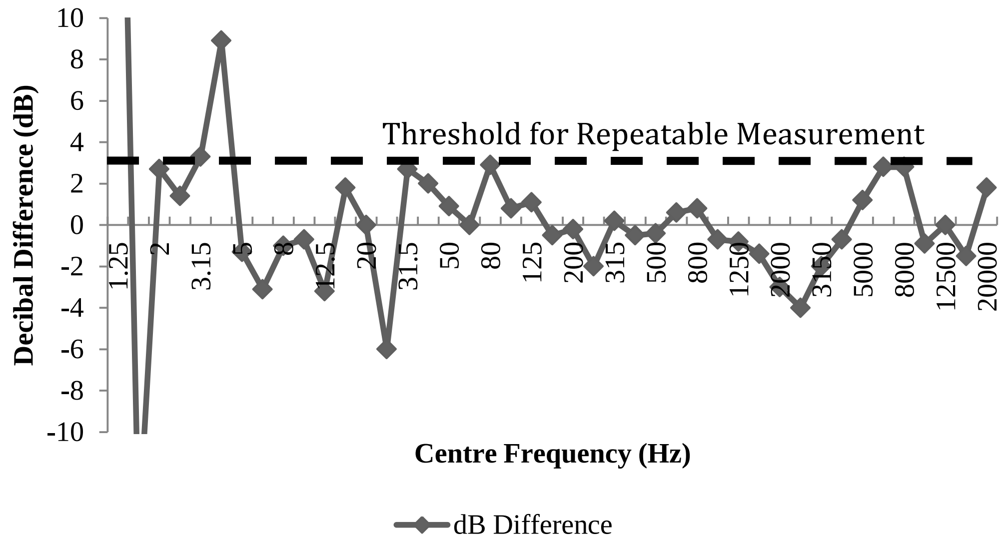

Figure 5.8 – The difference between both normal frequency signal measurments, signals below the dash line are valid measurements.

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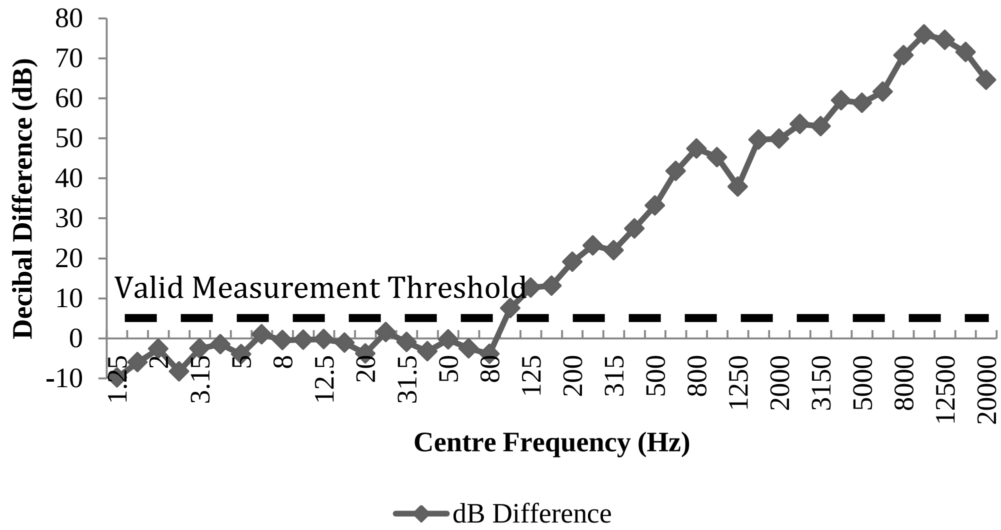

Figure 5.9– Normal frequency signal dB difference between the background level and average of two low frequency signals. All measurements below dashed line are valid, measurement above a centre frequency of 80Hz are greater than 10dB.

Combing the low and normal frequency signal provides a band from 50Hz to 20kHz, to perform the calibration. If the high frequency signal is the only signal used in the calibration, then the lower cutofff frequency would reduce to a centre frequency of 80Hz. Incorporating the low frequency signal into the calibration provides three additional low frequency bands in the calibration process. However, in order to use the calibration data to calculate a transfer function, the data from the low frequency and normal frequency signals must be combined. This is not simple, because the low frequency signal concentrated more sound in the low octave bands, and in order to have measured levels along with the background level, the low frequency signal was played at a louder volume.

To combine the low and normal frequency data sets, the sound level change created by the low frequency must be calculated. This was done by calculating the dB difference in

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the bands where the two data sets overlap from a frequency band from 80Hz to 500Hz. Figure 5.10, Table 5.1 gives the dB difference between the low frequency and normal frequency data in the overlap region shown in Figure 5.10, along with the average of the difference.

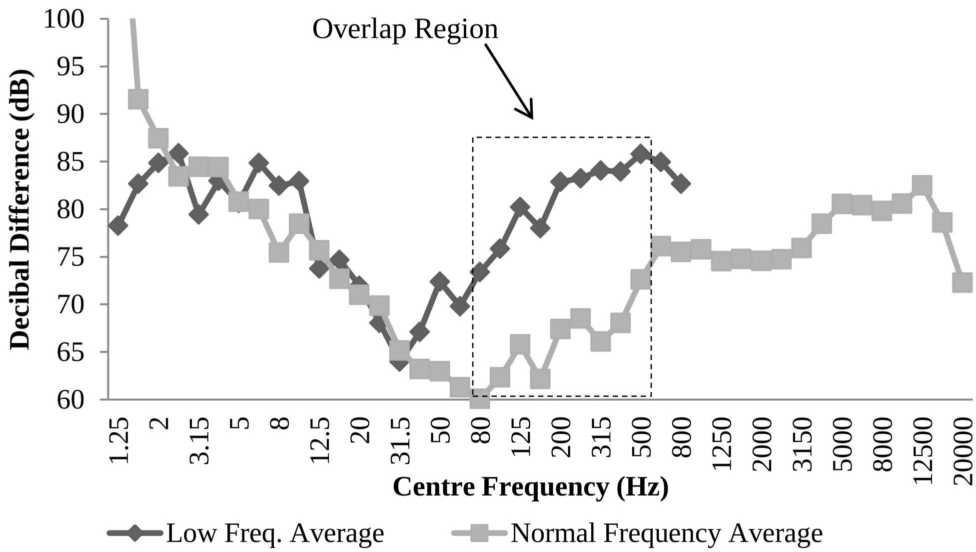

Figure 5.10– Average of low and normal frequency calibration stimulus, where overlap region is from the 80Hz to 500Hz band.

In order to construct the combined calibration curve, the average dB difference was subtracted for the low frequency data and then combined with the normal frequency data. This was done by subtracting 14.9 from the low frequency signal from the centre frequency bands 80Hz to 500Hz. Perfoming this amplitude correction resulted in Figure 5.11.

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Table 5.1 – Decibel difference between the low and normal frequency signals from the overlap region outlined in Figure 5.10.

Centre Frequency	dB Differnce
80	13.3
100	13.5
125	14.4
160	15.8
200	15.5
250	14.7
315	18.0
400	15.9
500	13.2
Average dB Difference	*14.9*

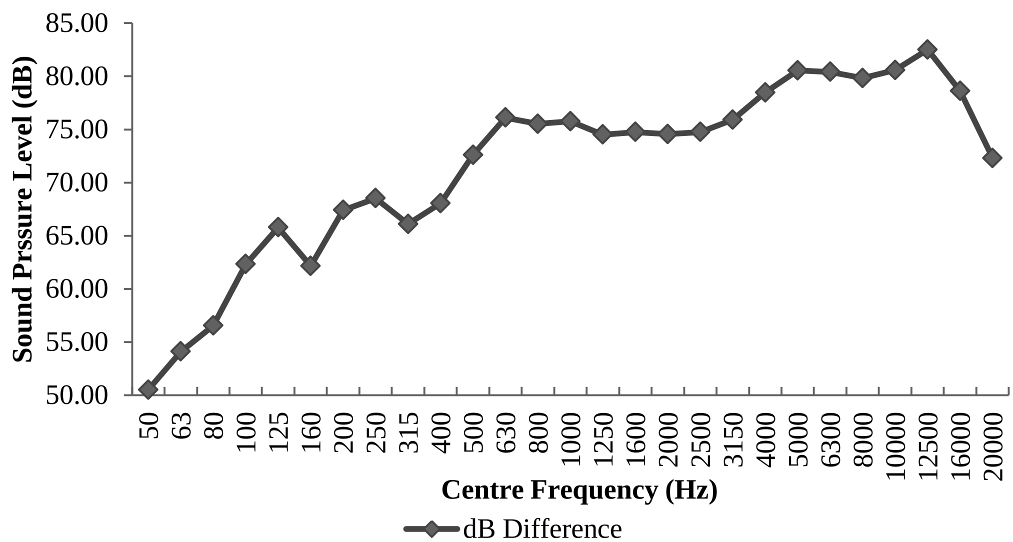

Figure 5.11 – Combined average of normal and low frequency signal, with the dB difference applied at a centre frequency from 80Hz to 500Hz in Table 5.1.

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The construction of the calibration curve is a preliminary step in the process of implmenting a calibration for the signal processing component. The next phase in the implementation, this requires shifting the measurements from the decibel scale, a logrithmic quantity, to the pressure scale, a linear quantity this can be accomplished by Equation 4.15 as seen in Figure 5.12.

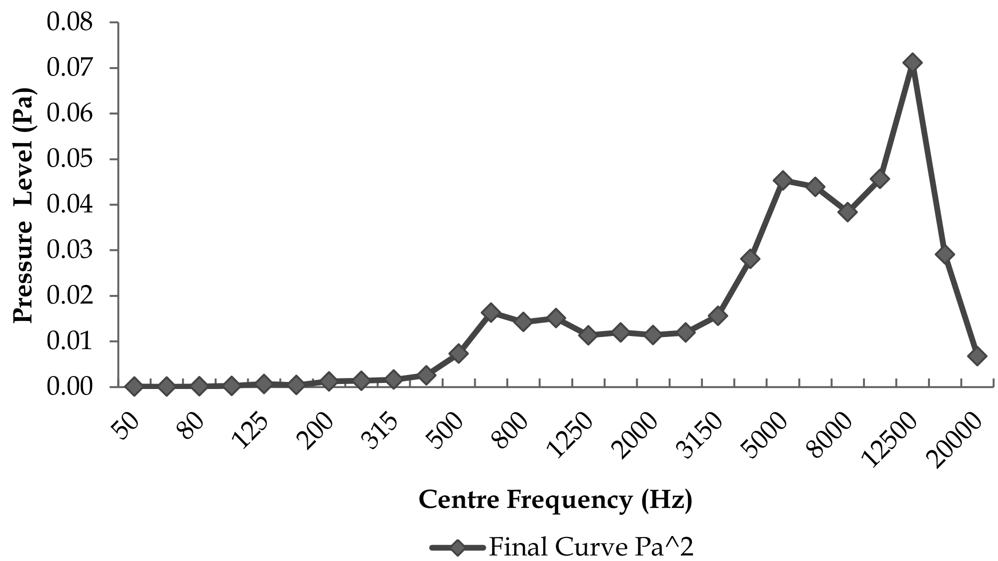

Figure 5.12 – Figure 5.11 converted from the Decibal scale to Pascals.

The next task is correlating the measuments from Figure 5.12, the pressure based values of sound level measurements, to the voltage amplitude of the calibration stimulus, since the average amplitude of the white noise is unity, as described in Chapter 4. Equation 4.17 can be employed to establish a relationship between the pressure of the calibration stimulus in the environment, and voltage of the calibration stimulus. This requires several key steps, (1) Figure 5.12, the pressure level of the calibration stimulus at each centre frequency, (2) The bandwith of each frequency band in the third octave, (3) The voltage or

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amplitude parameter for the calibration stimulus, (4) Steps 2 & 3 allow the implementation of equation 4.17. Once equation 4.17 is implemented, we now have the correction factor for the signal processing component as seen in Figure 5.13.

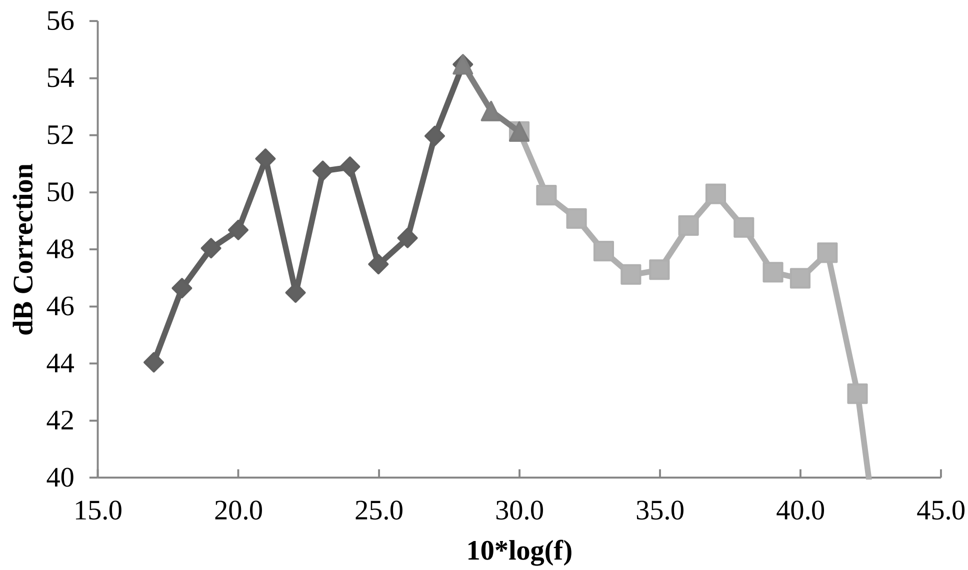

Figure 5.13 – Implementation of Equation 4.17 to yield to the dB correction for for the white stimulus and signal processing compnent from Figure 3.1.

This data in Figure 5.13 is the correction that should be applied at each frequency component of the treatment stimulus. The process to apply this correction is as followed.

1. Perform a Fourier Transform to take the treatment stimulus, T(t), from the time domain to the frequency domain, T(f) as desribed by Equation 4.1.
2. Apply the dB correction at each frequency as seen in Figure 5.13, this mitigates the effects of the signal processing component.

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3. Perform an inverse Fourier Transform taking the treatment stimulus from the frequency domain, T(f), back to the time domain $T_c(t)$, as described by Equation 4.2. Thus yielding a calibrated signal. As seen in Figure 5.14.

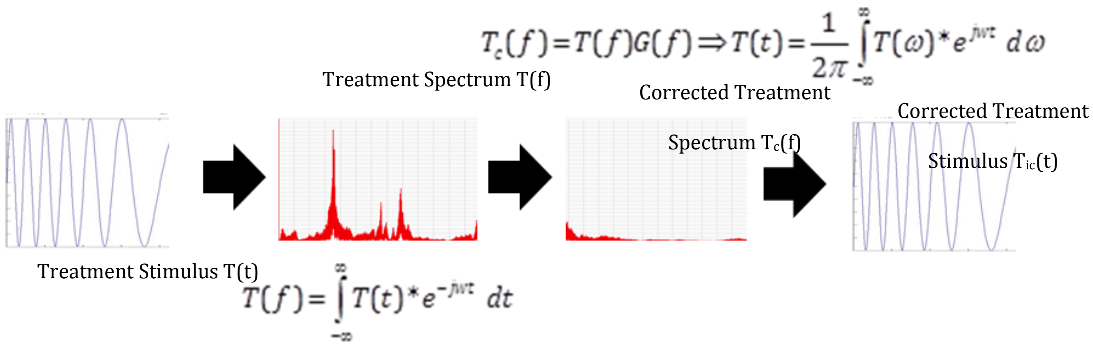

Figure 5.14 – Schematice representatation of the implemetation of Figure 5.13 for applying correction for the signal processing component from Figure 3.1.

The pratical implications of implementing this procedure will be discussed later in this chapter. The environmental and signal processing component from Figure 3.1, are confounded in the calibration procedure. The frequency response of the signal processing equipment will vary depending on the envrionment in which the signal is output. Thus, this calibration is unique to both the room and the frequency response of the system. To quantify the room acoustics the following procedure must followed.

5.1.1. Room Acoustics Construction Materials

The Sabine model used to quantify the acoustic characteristics of the usability lab, is done by defining the room absorbtion at the boundaries

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$$A_s(f) = \Sigma_i^n \alpha(f)_i * SA_i \qquad [5.1]$$

where $\alpha(f)_i$, is the material absorbtant coefficient of the ith surface, and SA_i, is the surface area of the ith suface. The area of each major surface in the room is show in Table 5.2 and Table 5.3 gives the published sound absorption characteristics for each of the surfaces. The average absorption coefficient is used in the Sabine equation to calculate the reverberation time of the room,

$$T_{60} = \frac{4V}{A_s(f)} \qquad [5.2]$$

where V is the room volume, (34.4m^2), and A_s(f) comes from using the data in Table 5.2 and 5.3 in Equation 5.1. The resulting reverberation time are shown in Figure 5.4. To accurately quantify the room acoustics, the parameters in equations 5.1 must be known. To obtain these parameters; construction materials, volume, and surface area. The costruction materials parameters for the usability lab are outlined in Table 5.2. The volume and surface area are summarized in Table 5.3.

Employing equation 5.1, with the measurments documented in Table 5.2 yields reverberation time outlined in Equation 5.2. The results are summarized in Table 5.3. The reverberation time is a frequency dependpent parameter, however on average is typically taken, which is.31 seconds for the usability lab.

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Table 5.2 – Surface area for i^{th} surface and construction materials for the usability lab in Figure 5.1.

Material	*North Wall*	*East Wall*	*West Wall*	*South Wall*	*Ceiling*	*Floor*
			Measurements in m^2			
Carpet	-	-	-	-	-	42.24
Painted Plaster Walls	14.6	31.2	34.2	24.8	-	-
Wood	-	0.1	0.1	6.4	-	-
Plastic	-	0.1	-	-	-	-
Plastic (White Board)	-	-	4.9	-	-	-
Painted Concrete	-	-	6.4	-	-	-
Glass Windows	13.6	-	-	-	-	-
Aluminum	2.0	-	-	-	-	-
Marble	0.1	-	-	-	-	-
Painted Aluminum	-	-	-	0.8	-	-
Acoustic Tile	-	-	-	-	41.8	-
Perforated Metal	-	-	-	-	1.1	-
Steel (Lights and Vents)	-	-	-	-	5.6	-
Total	*30.32*	*31.45*	*45.49*	*31.94*	*48.45*	*42.24*

Table 5.3 - Reverberation time for the usability lab from Figure 5.1 as a function of frequency.

Frequency	*125*	*250*	*500*	*1000*	*2000*	*4000*
Reverb Time (s)	0.28	0.40	0.40	0.29	0.25	0.26

The sound absobtion characteristics of the room can also be used to estimate the distance from a sound source where the direct reverberent field are equal. This is observed in Figure 5.3. This location is called the cricitical distance, d_c, and is defined by

$$d_c = .057 * \frac{V}{A_s(f)}. \qquad [5.3]$$

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The calculated critical distance for the usability lab is summarized in Table 5.7.

Table 5.4 – Critical distance calculation from Equation 5.3 for usability lab from Figure 5.1.

Frequency					
125	250	500	1000	2000	4000
d_c (m)					
.09	.14	.14	.10	.09	.09

Table 5.4 provides the basis to determine how far from the source measurements must be taken to be in the revberent field. Due to the approximations of the inputs and formulation of the critical distance, measured data was also used to identify the critical distance. This is done by measuring the sound pressure in the room at the several distances from the source Figure 5.15 to 5.19 shows the measured sound level at the same octave bands used in Table 5.5, as a function of distancefrom the source. On each an approximate locatation for the start of the reveberent field is indicated where the soundl level curve approaches zero.

Identifying the reverberation field in this manner is very subjective, and can vary by ones interpertation. The next approach is to model the room acoustics to determine the direct and reverberent field locations. Futher , verifying the critical distance model is correct or if the graphs from the raw data provides the best insight for determining the sound field.

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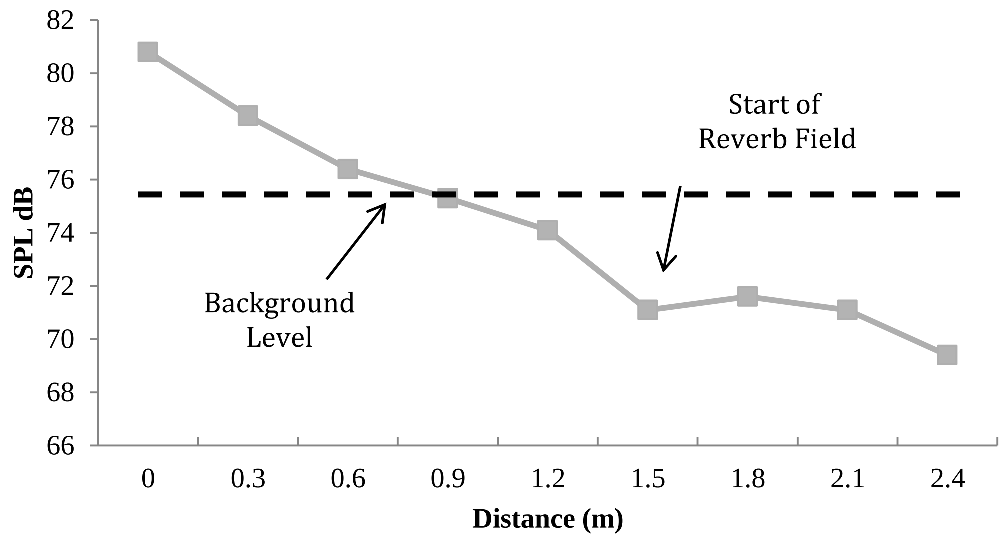

Figure 5.15 – Measured sound level as function of distance from the source. The start of the reverberent field at the 125Hz 3rd octave band.

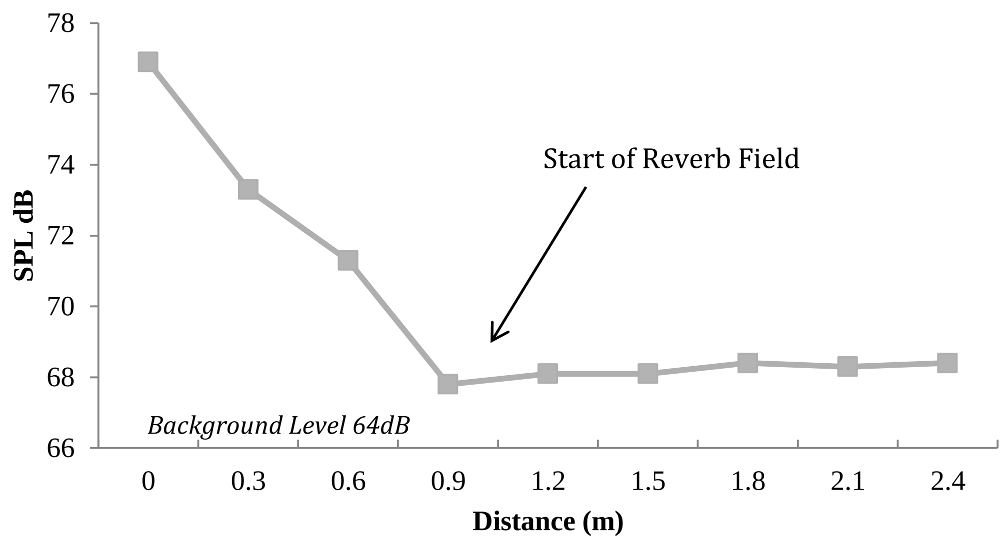

Figure 5.16 – Measured sound level as function of distance from the source. The start of the reverberent field at the 250 Hz 3rd octave band.

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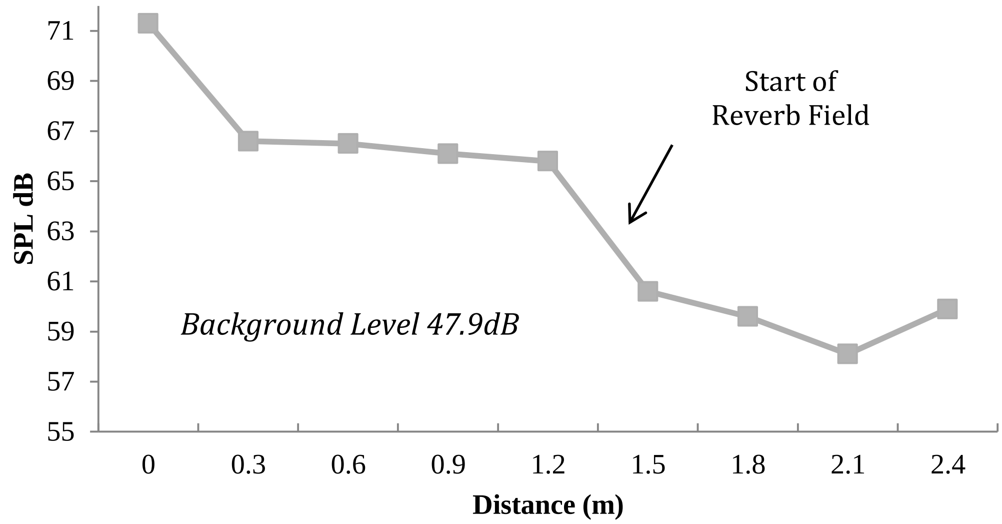

Figure 5.17 – Measured sound level as function of distance from the source. The start of the reverberent field at the 1000 Hz 3rd octave band.

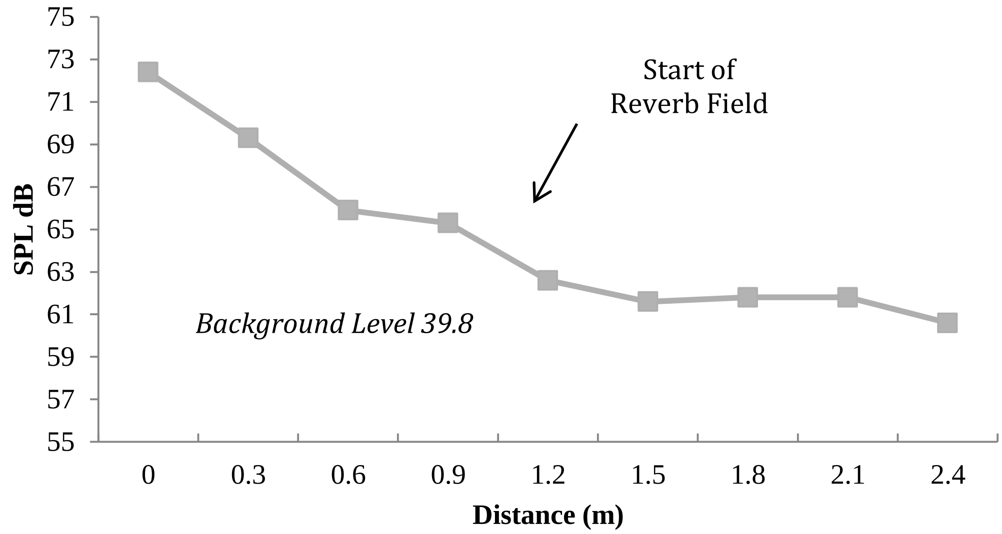

Figure 5.18– Measured sound level as function of distance from the source. The start of the reverberent field at the 2000 Hz 3rd octave band.

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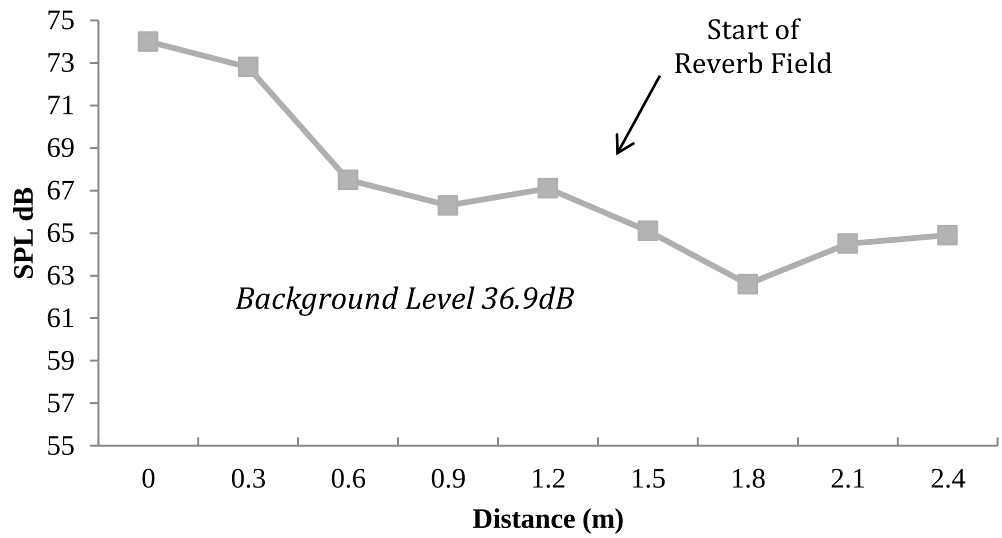

Figure 5.19 – Measured sound level as function of distance from the source. The start of the reverberent field at the 4000 Hz 3rd octave band.

From the Figures 5.14, there is an interaction at the 125Hz band between the background level and reverberation field, thus this measurement is not valid. Figure 5.15, the 250Hz band is not not 6dB abobue the background level, so the analysis will begin at Figure 5.17, the 1000Hz band.

Employing Equation 4.15 to determine the pressure level in the reverberent field. For example, at the 500Hz band, the reverberent field is 71Db, in the medium air. The pressure level is

$$P_{rms}^2 = \left(20 * 10^{-6}\right)^2 * 10^{\frac{71}{10}} = 5.03 * 10^{-3} Pa^2 \qquad [5.4]$$

This pressure level of the reverberent field is used to determine the source power of the stimulus, defined as

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$$\Pi = 9.7*10^{-5}*\frac{P_{rms}^{2}*V}{T} \qquad [5.5]$$

where, P_{rms}^2, is the pressure of the reverberent field, V is the room volume, and T is the reverberation time. Apply Equation 5.5 to the usability lab yields

$$\Pi = 9.7*10^{-5}*\frac{5.03*10^{-3}Pa^{2}*34.4m^{3}}{.40s} = 4.2*10^{-5}W \qquad [5.6]$$

The next step is to determime the the effective pressure, the effective pressure is described as

$$P_d^2 = \frac{Z\Pi}{4\pi r^2}. \qquad [5.7]$$

where, Z, is the medium impedence, Π, the source power, and r is distance from the source. The effective pressure is the location where the reverberent field and direct field pressure are equal. The effective pressure for 125Hz band can be obtained from equation 5.7

$$P_d^2 = \frac{409.9\frac{Nm}{s}*4.2*10^{-5}W}{4\pi*(.14m)^2} = .069Pa^2 \qquad [5.8]$$

If the effective pressure is compared to the effective pressure in the direct field calculated from equation 5.8, they align at a distance between .9m to 1.20m as seen in Table 5.5 . Following the steps outlined above the effective pressure the 1000Hz band is .19Pa^2,

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which corresponds to a distance between .6m to .9m. The 2000Hz band effective pressure is $.22Pa^2$, which corresponds to a distance between of .6m. The 4000Hz band effective pressure is $.22Pa^2$, which corresponds to a distance between of .6m.

Table 5.5 – Figure 5.17, 500Hz band direct sound field pressure calculation and comparison to critical distance to establish the reverberent field.

Distance (m)	Direct Field Pressure (Pa^2)
0.30	0.88
0.60	0.22
0.90	0.10
1.20	0.06
1.50	0.04
1.80	0.02
2.10	0.02
2.40	0.01

5.2. Developing Equations for the Transfer Function

There are practical issues associated with the implementation of both the calibration curve, and the quantifying the effects of the human hearing system through Equal Loudness Contours. An equation which describes the curve is needed for the implementation of the calibration factor. This requires curve fitting, but there are several issues associated with a curve fitting procedure.

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1. The curve must minimize the error associated with the fitted equation relative to the original curve. If the fit introduces additional error than the calibration fit will introduce error.
2. The x-axis must be shifted to a logarithmic scale in order to satisfy Item 1. This reduces the range of this axis, which allows for a greater fit for the polynomial equation. This is done due to the large range of frequencies in the third octave bands.
3. The curve must be broken into overlapping regions in order to satisfy Item 1. If the curve is not fit in sections, a higher degree polynomial must be fit which introduces error across regions of the frequency spectrum. The smaller the regions the better fit that can be achieved with a lower degree polynomial.
4. The curve fitting is a trial and error process requiring a number of iterations.
5. The number of significant digits obtained in a curve fit is critical is satisfying Item1.

When the curve fit was applied to Figure 5.20, each of the issues outlined above had to be considered. The calibration curve provides the correction at each frequency, but in order for the curve to be implemented, a curve fit was applied. This fit was done with polynomials; several polynomials were implemented over a different range, to minimize the error in the fit. For this sound system, the calibration curve was broken into three intervals.

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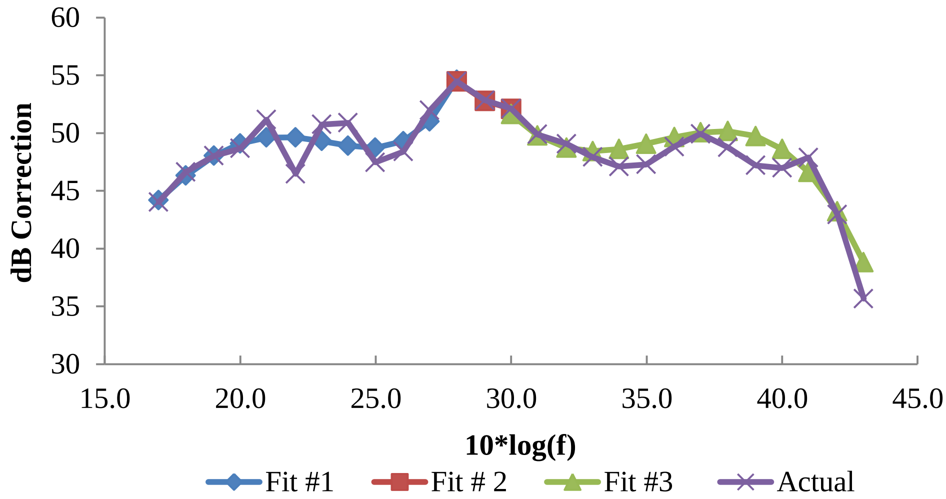

Figure 5.20 – Calibration fit for white noise with over plot of actual curve
The curve fit that was applied in are described by Equation 5.9.

The curves identified in Figure 5.20 are summarized by the following.

$$dB\ Correction = \begin{cases} 0.00381f^4 - 0.30039f^3 + 8.58577f^2 - 104.07348f + 489.77748 & f < 630Hz \\ 0.453f^2 - 27.452f + 467.95 & 630Hz < f < 1000Hz \\ -0.0336f^3 + 3.5697f^2 - 125.86f + 1521.9 & f > 1000Hz \end{cases} \quad [5.9]$$

The fit was an iterative procedure, influenced by the number of significant digits, x-axis log transformation, and interval selected. The same process can be followed for the Equal Loudness Contours, but for this study the ELC were implemented directly from the standard. A sample curve fit is observed in Figure 5.21 for the calibration curve fit.

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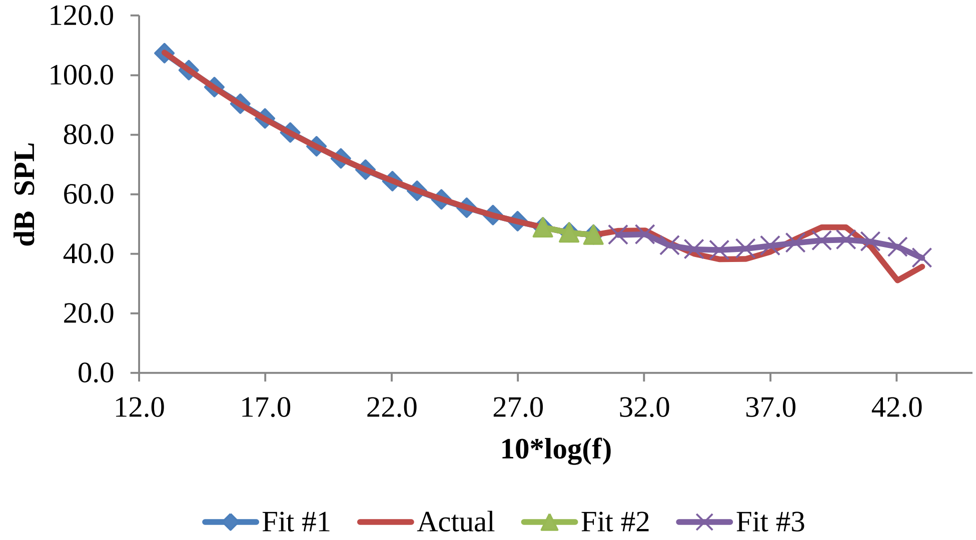

Figure 5.21 – Calibration fit for equal loudness contour with over plot of actual curve as described by equation 5.10.

$$dB\,Correction = \begin{cases} -0.05545 f^3 + 6.04124 f^2 - 218.34234 f + 2{,}660.60173 & f < 630Hz \\ 0.4651 f^2 - 28.218 f + 474.3 & 630Hz < f < 1000Hz \\ 0.14609 f^2 - 9.88293 f + 211.16938 & f > 1000Hz \end{cases} \qquad [5.10]$$

Attempting to fit a single curve to either Figure 5.20 or Figure 5.21, results in a fit that introduces additional error. The error is manifest as a dB difference that is greater than the dB of the original measurements.

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6. EXPERIMENTAL DESIGN AND TREATMENT SELECTION

The third task in constructing an appropriately designed auditory display is the design of the stimulus for the referent. The stimulus is the auditory signal used to communicate the situation via an auditory display. The stimulus for this experiment was an aircraft cabin recording, which is a broadband signal. This broadband signal was sampled from the interior cabin of an aircraft; this signal was selected due to its abstract nature.

An abstract representation was selected for the auditory display so the treatment applied to the stimulus would not confound the results. If a tangible representation was selected, interaction between the natural interpretation of the stimulus, and the treatment could communicate conflicting information, negatively impacting the results. The treatments applied to the broadband stimulus were amplitude modulation. Amplitude modulation varies the amplitude of the broadband signal at a constant frequency. This variation is periodic in nature, creating an amplitude envelope. These two factors are defined as the treatment combination for the interior cabin of the aircraft

The goal is to create treatment combinations to define varying levels of urgency, which can be resolved by a pilot. Under this methodology, it is assumed the resolution of the perceived urgency scale will be based on the number of sounds presented to a subject. The higher the range of sounds presented the larger the scale. Each sound will have two variables manipulated, its amplitude and frequency of modulation. The number of variables manipulated will correspond to physical changes in the characteristics of a sound. The total number of treatment combinations is nine, defined by variations in the amplitude and frequency of modulation, and applied to a sampled aircraft interior cabin broad band signal.

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The following criteria must be established to make some decisions on the number of treatment combinations. Primarily, determining a feasible experiment duration, the lack of a standard, contributes to the subjective nature in which this decision is made. In considering the duration, if an experiment is to long factors such as fatigue must be considered. The literature review provides a framework in which to construct the experimental design for this study.

In order to control the number of treatment combinations a tradeoff needs to be made between the number of intervals on the scale, and the variables manipulated. The basis for this decision will be made based on the previous literature in psychoacoustics. Sandrock and Marshall were able to rank annoyance of aircraft sounds as perceived by a user. Each desired to sort sounds in an interval ranking with semantic descriptors, to create an adequate list for the measure based on acoustical properties. The signal resolution experiment is similar in respect that it will rank auditory data that is similar in nature to a hierarchical ranking. In both experiments the sound were ranked on a seven point scale. The table below outlines the experimental disparities between the researchers as seen in Table 6.1.

Table 6.1 – Experimental design from Sandrock and Marshall experiments involving perception.

	Ordinal Scale	**# Treatment Combinations**	**Treatment Duration (s)**
Sandrock	7	31	15
Marshall	7	14	40

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This literature provided the justification for the experimental design needed for this experiment. Each of these items will be decomposed to identify the fundamental components of each method.

Table 6.2 – Experimental method breakdown for study conducted by Sandrock and Marshall.

	Origin
Ordinal Scale	One must consider the field of psychophysics which is the relationship between physical stimuli and their subjective correlation (Hellier et. el, 1999) The scientific study of the relation between stimulus and sensation. Psychometrics is used to measure these subjective evaluations (Hellier et. el, 2002) The Likert scale is a psychometric scale, and is widely used in research. Many psychometricians advocate using seven or nine levels; a recent empirical study (Parizet, 2005) "found that a 5- or 7- point scale may produce slightly higher mean scores relative to the highest possible attainable score, compared to those produced from a 10-point scale, and this difference was statistically significant." (Hellier et. el, 1999)
Treatment Duration	Treatment duration the reaction time based on sound needs consideration. Reaction time (RT), is the elapsed time between the presentation of a sensory stimulus and the subsequent behavioral response (Susied, 2008) Mean RT is approximately 180-200 milliseconds to detect an auditory stimulus (Suied, 2008). The time to interpret the sound varies based on the stimuli, so a reasonable estimation must be made.
No. of Treatment Combinations	The number of treatment combinations is composed of two factors the total number of sounds with scales, and the number of repetitions for each experiment.

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Based upon the information above a decision can be made for the number of sound presented, which based on psychometric recommendation consist of 9 sounds, with the amplitude modulation have 6 unique variables, and the frequency modulation having 2 unique variables. Based upon the reaction time to auditory stimuli and the literature reviews a reaction time of 15sec will be appropriate as a maximum duration. Defining the number of sounds, number of variables manipulated, and the maximum duration, requires an in depth discussion of the amplitude and frequency modulation terms and its interaction with its environment.

The amplitude modulation creates a well-defined envelope in which the time signal follows. This is observed by periodic changes in the sound level of the modulated signal. The frequency of modulation changes the periodicity of the amplitude modulation. The frequency of modulation of the amplitude modulated signal is only perceptible up to 300Hz. Varying the parameters amplitude and frequency modulation were evaluated in terms of perceived urgency.

The criteria and range for the amplitude modulation is the first step in constructing an ADS. The ranges of the treatments for amplitude modulation were between the values 0 to 1. A value of zero would not return any periodic signal, simply a constant value with amplitude unity. The value of 1 would return a non-modulated signal, thus the original signal. The ranges of treatments are for values which are greater than 0 and less than 1, which bounds the amplitude modulation. There were several objectives in constructing the treatments.

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Objective 1

Determine how the sense of urgency is perceived when there is no interaction between the amplitude modulated signal and the reverberant field.

Objective 2

Determine how the sense of urgency is perceived when there is an interaction between the amplitude modulated signal and the reverberant field.

Treatments were selected under two theoretical constructs. Objective 1 assumes a subject will interpret the signals purely on the effects of the modulation. The modulated signal SPL will remain above the reverberant field SPL. Objective 2 assumes masking will occur; the SPL at the minima of the modulation envelope will be partially masked by the reverberant field. Exploring both objectives will determine how the reverberant field interacts with amplitude modulation in terms of perceiving urgency.

The effects of amplitude modulation can be perceived up to modulation frequencies of 300Hz (Fastl, 2007). Frequencies above this threshold cannot be perceived, for the cabin signal stimulus, frequencies above the 40Hz threshold could not be perceived. This criteria was only evaluated by the individual performing the experiment, so there some subjectivity in regards to the selection of the frequency modulation stimulus. For the amplitude and frequency modulations the range of parameters are summarized in Table 6.3. These will be used as the guidelines for the selection of the treatments for the auditory display.

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Table 6.3 – Amplitude and frequency modulation bounds for cabin noise stimulus.

Modulation Type	Parameter
Amplitude Modulation	$0<A_m<1$
Frequency Modulation	$0<f_m<20$

The treatments were selected based on the criteria established in Table 6.3. Due to the exploratory nature of this research the entire design space was evaluated. Treatments were selected from A_m = .1 to Am = .6 for the amplitude modulation. For the frequency modulation the values were at an f_m = 10 and f_m = 20. There were nine treatments selected in the exploration of the design space for the evaluation of urgency.

The modulations parameters were then applied to the original cabin signal. Once each treatment was created, a measurement of each signal was taken to ensure there were no differences in the sound level. This was done to satisfy the just noticeable difference criteria, so no signal would evaluated solely due to sound level changes. Each signal was then normalized by applying an amplitude correction to the signal, this is amplification factor, A, from equation 4.7.

Exploring the manipulation of the amplitude and frequency modulation as a mechanism to communicate urgency can introduce additional factors in the construction of auditory displays. If varying the modulation parameters influence perceived urgency, then mapping a treatment stimulus to a referent can provide a new approach in the construction of auditory displays. In considering modulated signals in the design of auditory displays the signal processing component, room acoustics, and the effects of the human hearing system must be considered.

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The signal processing component is resolved with the calibration procedure described in Chapter 4. To design the modulated stimulus that is diffused in an open environment, the room acoustics must be considered, due to the nature of modulation. Modulation creates changes in the SPL, these changes corresponds to the degree of amplitude modulation applied to the signal. If the depth of modulation is such that it interferes with the reverberation field, masking can occur. Since, changes in SPL are negatively correlated to the modulation factor, the lower the modulation factor the greater the fluctuations between the maximum and minimum amplitude. This creates a ΔdB quantity, as described in Chapter 4.

The modulation depth is the lowest SPL of the broadband signal as seen in Figure 6.1. The blue signal represents the original time varying signal; the green signal represents the same signal with modulation applied. The difference between the minimum and maximum SPL of the modulated signal represents modulation depth of the signal.

The effects of the modulation can be seen in the differences between the original cabin signal and modulated cabin signal. From the Figure 6.1, the lower the modulation factor the greater the depth which can be interpreted as the larger the SPL fluctuations. If the modulation depth is large, such that the sound level is below the reverberant field, masking will occur. If the signal is masked, then the full depth of the modulation cannot be evaluated. The original signal modulates below the reverberation field, which indicates masking has occurred at the lower frequencies. To determine if a signal is masked, the ΔdB can be used as described in Chapter 4. Where a positive value indicates no masking has occurred, and a negative value indicates that masking has occurred. These results are summarized in Table 6.4.

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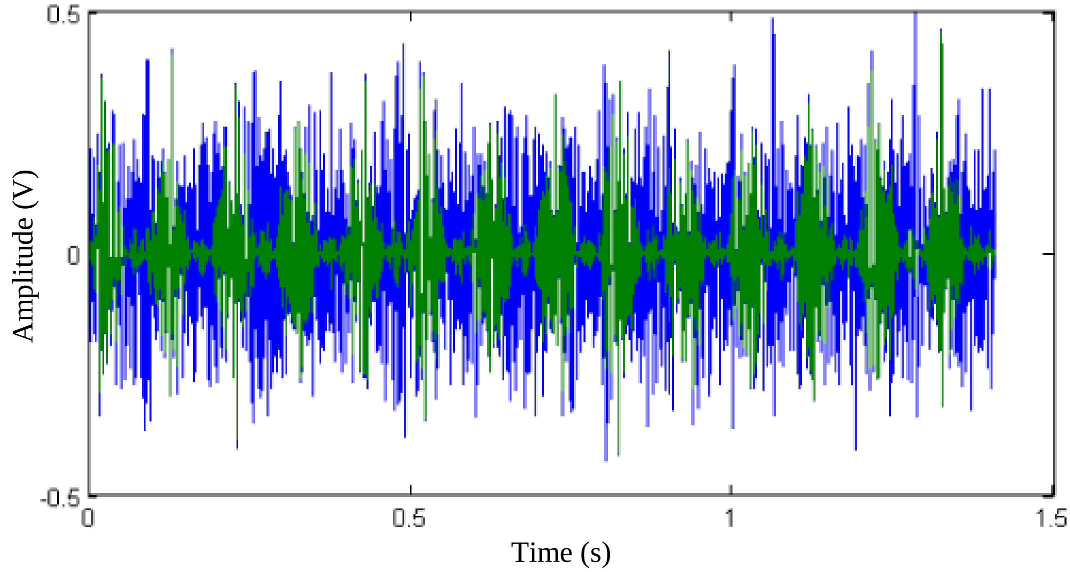

Figure 6.1. – Cabin broadband stimulus, represented by blue image, modulation applied to cabin stimulus, represented by green image.

Table 6.4 – ΔdB of treatments for modulated interior cabin noise, calculated values to supplement Figure 4.5 and 4.6

Treatment	A_m	f_m	ΔdB Difference
1	.1	10	3.51
2	.2	10	2.77
3	.3	10	1.59
4	.4	10	-0.75
5	.5	10	-54.95
6	.6	10	-62.61
7	.1	20	3.51
8	.2	20	2.77
9	.3	20	1.59

Implementing the reverberation model reveals the interaction between the modulated signal and reverberation sound field. At an A_m = .1, the modulated signal sound level is higher than the reverberation field. The sound level difference is large; there is no

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interaction between the reverberation field and modulated signal. This results in the modulated signal being heard as its intended, since no masking occurs as indicated in Figure 4.5. As the A_m increases the ΔSPL between the modulated signal and reverberation field converge. This is true for an A_m up to .3, at an A_m between .1 and .3, the modulated signal is unmasked by the reverberation field. At A_m greater than .3 the modulated signal becomes masked by the reverberation field, as indicated in Figure 4.6. This can be called is known as the threshold of masking. When the threshold of masking is exceed the signal is not heard as its intended.

When you are above the threshold of masking there is an interaction between the modulated signal and reverberation field occurs, as in Figure 4.6. At an Am $_=$ .4, the modulated signal becomes masked. The degree to which the masking occurs is only at the minima of the modulated signal. From the signal minima up to 1dB the modulated signal is masked by the reverberation field. As the A_m increases the interaction becomes greater, until the reverberation field completely masks the modulated signal. At an A_m = .5 the reverberation masks approximately 70% of the signal. This masking indicates the he signal is not heard as its intended.

Designing amplitude modulated signals that are both masked and unmasked by the reverberation field, can provide insight not only to how amplitude modulation influences urgency but the effects of masking. This allows one to determine if masking has any effect on perceived urgency. When masking occurs it is difficult to assess what a subject is evaluating. The interaction between the modulated signal and reverberation field fosters an uncertainty of distinguishing whether the signal is being perceived from the modulated signal, reverberation field, or the interaction between the two.

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The frequency of modulation for the amplitude modulated signal is the second variable that is manipulated in the construction of the auditory display. The lower the frequency of modulation the longer the signal remains in the ambient environment. At the f_m = 10Hz, the period for the signal is .1 sec, and at an f_m = 20Hz, the period for the signal is .05 sec. This period must be considered relative to the reverberation time of the environment. The usability lab has a reverberation time of .31sec.

If the reverberation time is large compared to the period of the signal, the reverberation field will dominate if masking is present. If masking is not present then the modulated signal will dominate. The degree to which it dominates depends on the reverberation time. This is critical when multiple signals are presents so there is no interference between multiple signals.

In selecting treatments for the construction of an auditory display, the reverberation model must be considered. The reverberation model is tailored for a specific environment; the reverberation time is unique to its environment. There are several parameters that are the products of the characteristics of the environment, while others are selected. So the results presented above are unique only to the parameters in Table 6.5. The variable that can be controlled is the SPL of the original signal; all other variables are the products of the environment.

In summary, as the A_m increases an interaction between the reverberation field and modulated signal occurs. This interaction results in the modulated signal being masked by the reverberation field. When masking occurs the signal is not heard as it was originally intended, but rather influenced by the reverberation field. The f_m must be considered relative to the reverberation time. If the reverberation time is high, the f_m should be low, resulting in

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a low periodicity of the modulated signal. If the reverberation time is shorter than the period, the modulated signal will decay before the next period.

Table 6.5 – Input variables for reverberation model parameters.

Input Variables	Value
ρ	1.18 $\frac{kg}{m^3}$
c	346.18 $\frac{m}{s}$
α	0.26
Surface Area	78.20 m^2
Volume	34.4 m^3
A_m	0.30
f_m	10Hz/20 *Hz*
Pressure	12.82 *Pa*
Input Work	$3.18 * 10^{-5}$ *J*
Background Level	39.80 *dB*
SPL	72.40 *dB*

This is only true when masking is not present. If the reverberation time is low, the f_m is high, resulting in high periodicity of the modulated signal. This results in the modulated signal decay before the next period, resulting in the original signal being heard as its intended. If masking is present then the reverberation field will dominate, and the relationship between the f_m and the reverberation time is irrelevant.

6.1. Experimental Design Selection

The major objective for this experiment is to determine the perception of perceived urgency in A_m and f_m broad band noise. Urgency can be defined as a force that constrains or compels (Merriam Webster, 2010). This experiment will examine how the sensation of

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urgency is perceived through sound by a subject. Urgency is defined by the quality of being important and calling for prompt action (Edworthy, 1991). The sensation of urgency experienced by the user will be correlated to the physical signal parameters (Edworthy, 1991). The manipulation of the signal parameters will create varying levels of urgency as perceived by a subject. These varying levels can be hierarchically ranked, and mapped to the sensation of urgency. This sequence of events listed above develops the conceptual framework of perceived urgency.

The studies deliverables are:

1. A subjective evaluation of sounds in which the signal parameters are manipulated and mapped to the sensation urgency.
2. A perceived urgency scale, minimum, maximum, and some degree of resolution.

The methodology for determining the sense of urgency will need to be developed and evaluated against competing theories and techniques. These methods were produced from literature reviews in experiments of a similar nature and application. The first factor under consideration is the methodology for creating the scale. In considering creating a scale there are several methods that can be employed to meet the experiment's deliverable of a perceived urgency scale. Each method has its unique advantages and disadvantages and carries certain assumptions. An evaluation of these methods and their assumptions will be evaluated below.

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The first method is providing a subject with a predefined minimum and maximum framework, setting the thresholds of the perceived urgency scale. In considering this option lays the assumption that the meaning of sounds is learned rather than a natural association (Hellier, 2002). Under this assumption the following are also assumed:

1. The presentation of the min and max will provide the framework of the perceived urgency scale.
2. Each sound has its own independent min and max.

The disadvantage of this method is that each subject will require training to successfully recognize the bound conditions. The second drawback is, training has the potential to provide a bias for the subject of predefined threshold, so any value outside these bounds cannot be categorized regardless of how they are perceived by a user. This can lead to a non-hierarchical ranking that will force the user to place a value within the min and max when it is perceived outside of these predefined bounds.

The second method is to provide no framework for the subject, requiring the subject to develop its own perceived urgency scale. This method carries the assumption that the min and max will be similar for each subject. This assumption is crucial in meeting the studies' deliverable. This method also has the potential to only provide insight into the range of perceived urgency amongst different subjects. Unlike the first method it offers the advantage, in that the data yielded determines the min and max, however the min and max may vary by user. This method has the potential to provide a highly individualized scale for each respective user and sound.

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The third method is to provide a single point framework for the subject in which perceived urgency will be scaled relative to reference sound. This method carries the assumption that the ranking will be made relative to the reference sound. The subject is allowed to create their own framework for perceived urgency. This method has the same disadvantage of method one. Each evaluation of each method was accessed with the following selection criteria, which will satisfy the project deliverables:

1. Degree to user input defines the model.
2. Theoretical Mathematical Model
3. Method Flexibility

A numerical weighting method can be applied to the selection criteria to select the appropriate method. Each of these criteria will be ranked according to importance. The perceived urgency is a subjective measurement the degree to which the user inputs into modeled is of high importance it will be weighted (.5). Preliminary consideration of the how the hierarchical ranking will need to be considered this will be weighted (.25). The method flexibility is also needs to considered and should be approached from the perspective that the perceived urgency scale can be universal or individualized this will be weight (.25). Each item will be evaluated on a 3, 5, and 7 point magnitude.

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Table 6.6 – Summary rankings for method of presentation stimulus to subjects.

	User Defined (.5)	Math Model (.25)	Method Flexibility (.25)	Total
Predefined Framework	5	5	3	2.25
No Framework	7	5	7	6.50
Single Point Framework	3	5	3	5.25

Using the criteria above the best choice is to provide no framework for the presentation of the stimulus. This allows each subject to create the concept of urgency based upon their own experiences. Once the method for creating a stimulus scale and the stimulus presentation is established the next item is to determine how the stimulus will be evaluated.

The treatments can be evaluated using several different methods such as: Taguchi Arrays, paired comparisons, and magnitude estimation techniques. The Taguchi array methodology estimates the entire design space, while only evaluating a few cases. This method was not suitable since urgency perception may not be aligned with the parameters of the treatment combinations. For example, an increase in a parameter may not align with an increase in the perception of urgency. Also due to the lack of literature in terms of sound urgency in which Taguchi arrays were implemented as an experimental method. This lead to the exploration of other techniques that align with existing methodology employed in the perception of urgency.

A paired comparison portion of the study was a suitable choice for this experiment, since it provides a relative assessment of urgency. For the paired comparison each treatment is randomly applied to another. In the paired comparison there is only one replication necessary, since it is a full factorial experiment. The perception of urgency will be evaluated

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without a weighting applied to the signal for the effects of human hearing, and an ELC weighting. The summary of the results for the paired comparison are seen in Table 6.7.

Table 6.7– Final selection for paired comparison and magnitude estimation experiment.

TRTS	Signal Processing	Room Effects	Human Hearing Effects
1-9	Calibrated	None	None
	Non-Calibrated		

This randomization was repeated for the non-calibrated experiment. The magnitude estimation was the second part of the experiment design. The experimental design for the magnitude estimation experiment is identical to Table 6.8, and the experimental design is seen in Table 6.9.

Table 6.8 – Full factorial experimental and randomization for calibrated and non-calibrated magnitude estimation experiment.

No.	TRT - A	No.	TRT - A
1	6	10	2
2	3	11	5
3	5	12	4
4	1	13	7
5	8	14	8
6	9	15	6
7	4	16	1
8	7	17	3
9	2	18	9

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Table 6.9 – Full factorial experimental and randomization for calibrated and non-calibrated paired comparison experiment.

Calibrated			Non-Calibrated		
Pair	TRT - A	TRT - B	Pair	TRT - A	TRT - B
1	7	2	37	2	4
2	8	7	38	5	6
3	3	5	39	3	6
4	7	5	40	6	8
5	1	3	41	9	1
6	5	4	42	7	9
7	6	9	43	9	4
8	2	4	44	7	6
9	5	6	45	3	2
10	3	6	46	4	6
11	6	8	47	2	9
12	9	1	48	7	2
13	7	9	49	8	7
14	8	9	50	3	5
15	3	2	51	7	5
16	4	6	52	1	3
17	1	2	53	5	4
18	5	8	54	6	9
19	7	4	55	8	9
20	7	3	56	7	3
21	9	4	57	2	5
22	4	1	58	6	1
23	7	6	59	5	8
24	2	5	60	9	5
25	3	9	61	7	1
26	6	1	62	4	1
27	4	8	63	6	2
28	9	5	64	4	8
29	1	5	65	1	2
30	4	3	66	3	8
31	6	2	67	8	2
32	3	8	68	1	8
33	1	8	69	1	5
34	2	9	70	7	4
35	7	1	71	4	3
36	8	2	72	3	9

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The magnitude estimation is replicated once the No. 1-9 are for the non-calibrated signal, while 10-18 are for the calibrated case. This was repeated for the ELC weighting factor. In selecting the treatments one must first investigate the acoustics of an environment through the reverberation model. Lastly, randomly assign the selected treatments in the appropriate design. Once the design is selected they need to be implemented in a GUI.

In summary, there were two experiments conducted to evaluate the studies objectives. A paired comparison experiment, and a magnitude estimation experiment. The paired comparison evaluation ascertained if a relative assessment of urgency could establish a scale of urgency. The magnitude estimation evaluation ascertained if an absolute assessment could establish a scale of urgency. Both treatments were evaluated in the both experiments, conducted in a single session. A full factorial paired comparison was conducted, in which every treatment was compared to every other treatment. Each sound was also evaluated in the magnitude estimation experiment. Both experiments were conducted using a graphical user interface (GUI) created in Python 2.6.

6.2. Experimental Session

The experiment session was scheduled for 50 minutes. The subject's heads were placed at a distance of 1.25m from the source, the sound level at that distance was 70 dBA, and the graphical user interface was presented on a 17" high-resolution LCD monitor. The objective was to evaluate the general population at Iowa State University, located in Ames, IA. The treatments in Table 6.8 and 6.9 were evaluated in two types of experiments: a paired comparison and magnitude estimation. The paired comparison is used to evaluate how one treatment is evaluated relative to another treatment in terms of perceived urgency. The

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second is magnitude estimation, where a subject will evaluate a perceived urgency on an absolute magnitude scale from 0 to 10. The treatments for this research consists of two variables, A_m and f_m , at several levels, as outlined in Table 6.4.

Nine different sound types presented to a subject. This is a reasonable number of treatment combinations, which was established through the existing literature. Each treatment was 5 seconds in duration, which was within the 15 second threshold. The entire experimental duration did not exceed 50 minutes. The experimental duration is total time the user is engaged in the actual presentation of the sound. Time was allocated for breaks to minimize the impact of fatigue on the experimental results. In conclusion the decisions above were based on controlling the overall experiment duration, and employing concrete methods of experimental design.

The paired comparison experiment was conducted on the graphical user interface in Figure 6.2, each subject was asked to provide a subjective evaluation of urgency. The subject controlled the dosage of the stimulus provide, but each subject was required to listen to the stimulus in its entirety at least. The duration for each stimulus was 5sec. Once the subject listened to the stimulus, an evaluation of the stimulus was performed on the following scale.

- *2 – A Much More urgent than B,*
- *4 – A More Urgent than B,*
- *6 – A Equally Urgent as B,*
- *8 – B More Urgent than A,*
- *10 – B Much More Urgent than A.*

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Each response recorded a respective value indicated above; this process was repeated 36 times for calibrated experiment, and 36 times for the non-calibrated experiment, by 28 subjects.

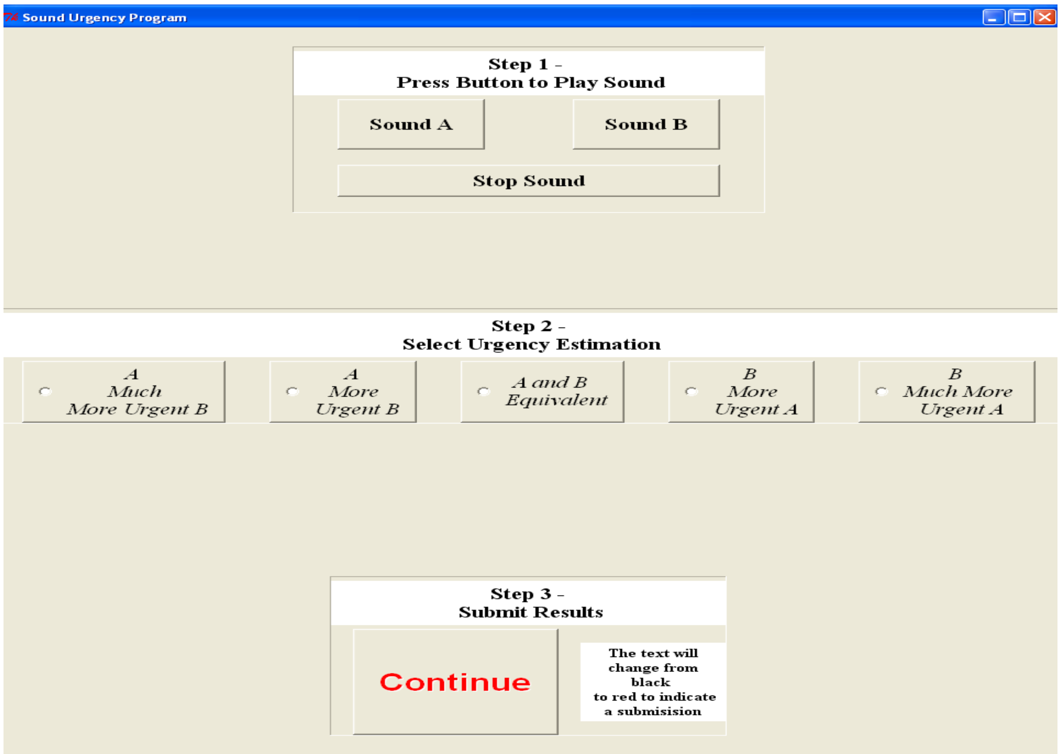

Figure 6.2 – Graphical interface for paired comparison experiment presented in the order outlined in Table 6.9.

The magnitude estimation experiment was conducted on a similar graphical user interface shown in Figure 6.3, each subject was asked to provide an absolute evaluation of urgency. The subject controlled the dosage of the stimulus provide, but each subject was required to listen to each stimulus in its entirety at least one time. The duration for each stimulus was 5sec. Once the subject listened to the stimulus, an evaluation of the stimulus was performed on the following scale

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- *0 – No Urgency*
- *10 – Very Urgency*

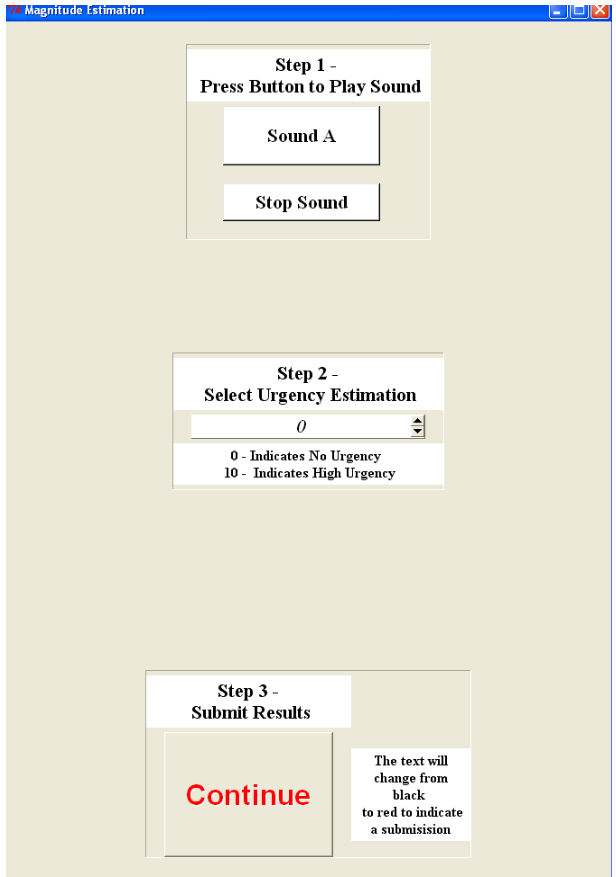

Figure 6.3 – Graphical interface for magnitude estimation experiment presented in the order outlined in Table 6.8.

Each response recorded a respective value indicated above; this process was repeated for all treatments for the calibrated and non-calibrated experiment. Each of the participants in this study performed the paired comparison experiment, followed by the magnitude estimation experiment. In order to qualify for the study, each participant was asked if they have normal hearing. The participants for this study were recruited from the Iowa State

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University Sona System, fliers, and word of mouth. Twenty eight participants were recruited from the Iowa State Community.

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7. STUDY RESULTS

The first objective of this study was the evaluation of the sensation perceived urgency, created by a modulated broadband signal. Modulation was applied to a broad band stimulus to create the sensation of urgency, amplitude and frequency of modulation created the stimulus. The second objective was to determine if a calibrated sound system influenced the perception of urgency. The calibrated system mitigates the influence of the sound system in the reproduction of an auditory signal. The third objective was to determine how the perception of urgency was perceived by humans. To address if urgency is universally perceived the same by all subjects or if perceived urgency differs by individual. The fourth and final objective, was to determine if a hierarchically arrangement of urgency could be established, and the resolution of such a scale.

The study consisted of two types treatments the first treatment featured the presentation of a calibrated stimulus, and the second was a non-calibrated stimulus. The calibrated stimulus implemented the procedures outlined in Chapter 4 and 5, where the signal processing and room acoustics were quantified, and the appropriate corrections were applied to its frequency spectrum. The non-calibrated stimulus did not incorporate any corrections in its frequency spectrum. This signal was confounded with the effects of the sound system and its treatment.

There were two experiments conducted to evaluate the experimental objectives. A paired comparison experiment, and a magnitude estimation experiment as described in Table 6.8 and 6.9. The paired comparison evaluation ascertained if a relative assessment of urgency could establish a scale. The magnitude estimation evaluation ascertained if an absolute assessment could establish a scale of urgency. Both treatments were evaluated in a

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study, conducted in a single session. A full factorial paired comparison was conducted, in which every treatment was compared to every other treatment. Each sound was also evaluated in the magnitude estimation experiment. Both experiments were conducted using a graphical user interface graphical user interface, created in Python 2.6 as seen in Figure 6.2 and 6.3.

7.1. Statistical Methods

Several approaches were employed to address the experimental objectives. In all the experiments, the response variable was the perception of urgency. The variables for this experiment are summarized in Table 7.1.

Table 7.1– Description of variables for experimental treatments and response.

Description Name	Units	Type
Amplitude Modulation	Volts	Amplitude
Frequency Modulation	Hz	Frequency
Urgency	None	Perception

The summary statistics communicates the largest amount of information as simply as possible. For the paired comparison experiment, the summary statistics are evaluated graphically through the merit score, and the number of circular triads, which are a measure of inconsistency in responses. This is due to the difficulty in differentiating between sounds, and is caused by subject's inattentiveness to stimulus and evaluation, and altering the assessment criteria of the response variable (Parizet, 2002). Circular triads, a measure of inconsistency can be defined as

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$$c = \frac{t}{24}(t^2 - 1) - \frac{1}{2}T \quad [7.1]$$

where

$$T = \Sigma_{i=1}^{t}(a_i - \bar{a}_i)^2 \quad [7.2]$$

and a_i being the score for the i^{th} noise, and $\bar{a}_i$ being the average of the scores $\bar{a}_i = \frac{(t-1)}{2}$, where t is the number of stimuli (David, 1988). Kendall and Babington define the coefficient of consistence as (David, 1988).

$$\xi = 1 - \frac{24c}{t\left(t^2 - 1\right)}, odd \quad [7.3]$$

If and only if $\xi = 1$, there are no inconsistencies in the preferences, thus the pairs constitute a ranking (David, 1988). The merit score is a measure of preference in which the paired comparisons were performed. The merit scores were computed from average of the cases, and computed linearly as (Parizet et. el, 2002)

$$S_i = \Sigma P_{ij,} \ i \neq j \quad [7.4]$$

where $\Sigma P_{ij,}$ is the preference probability of noise i versus noise j. In this study snine sounds were computed from Equation 7.4 and range from 0 to 8.

The next analysis technique was cluster analysis. The aim of cluster analysis is to group subjects according to their similarity on the urgency response variable. It is often

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called unsupervised classification, meaning that classification is the ultimate goal, but the groups are not known ahead of time. Hence the first task, in cluster analysis is to construct the class information. Hierarchical clustering algorithms sequentially fuse or split cases to make clusters, the process can be used using a dendrogram, and the vertical heights are used to decide how many clusters. The ward linkage method was employed which equates the smallest increase in the error sum of square after fusing two clusters. (Cook, 2010)

7.2. Results

The first step is to determine the pure count for the paired comparison experiment. The pure count is the total number of times a treatment was selected in a paired comparison evaluation. This experiment consisted of 36 pairs, where 28 subjects evaluated each pair, thus the total number of pure counts for the entire experiment across all treatments were 1,008. Each treatment was compared 8 times, by 28 subjects, thus the pure count for an individual treatment were 224. If a treatment was preferred 224 times in this study, then it was perceived as being the most urgent by every participant, thus the most urgent sound. From Figure 7.1, the pure count for the non-calibrated and calibrated experiment.

In addition to selecting either sound A or sound B, the subject was able to select if both sounds were perceived equally as urgent. In the calibrated experiment subjects perceived the treatments the same 208 out of 1008 times, which is 20.6% of the time. In the non-calibrated experiment the subjects perceived the treatments same 220 out of 1008 times, which is 21.8% of the time. This indicates that in nearly 20% of the cases, the sounds were not distinguishable from one another.

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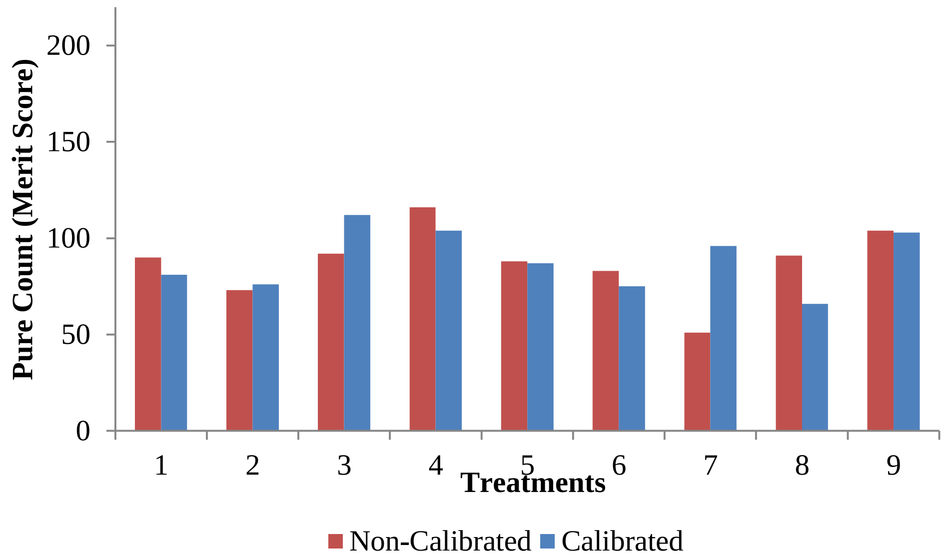

Figure 7.1 – Merit Score/Pure Count for the calibrated and non-calibrated paired comparison experiments for treatment in Table 6.4

For the non-calibrated experiment, the treatment selected the most frequently was treatment 4, selected 117 out of 224, 52.3% of the time. A treatment that is perceived accurately half the time, would ultimately lead to confusion and unreliability of the auditory display system. Imagine if you could only distinguish between a tornado warning, and tornado watch siren only half of the time. Each siren indicates two different levels of severity and would ineffectively communicate the situation through its auditory display.

Figure 7.1, reveals the distribution of the calibrated and non-calibrated system. The pure counts for treatment in the calibrated and non-calibrated experiment are almost identical. If treatment 9 is considered, the calibrated pure count is 87, and the non-calibrated count is 88. This indicates that calibration has little effect on the perception of urgency for this treatment. However, it does not reveal comparatively what treatment against it was selected. In general the distribution between the two experiments were different, indicating there is a difference in the perception of urgency between a calibrated and non-calibrated

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stimulus. This is most evident at treatment 7, where the calibrated experiment pure count is 96, and the non-calibrated case is 51. To gain more insight into the data, one must look at the individual cases as summarized in Figure 7.2, 7.3, 7.4 and 7.4. Perceived

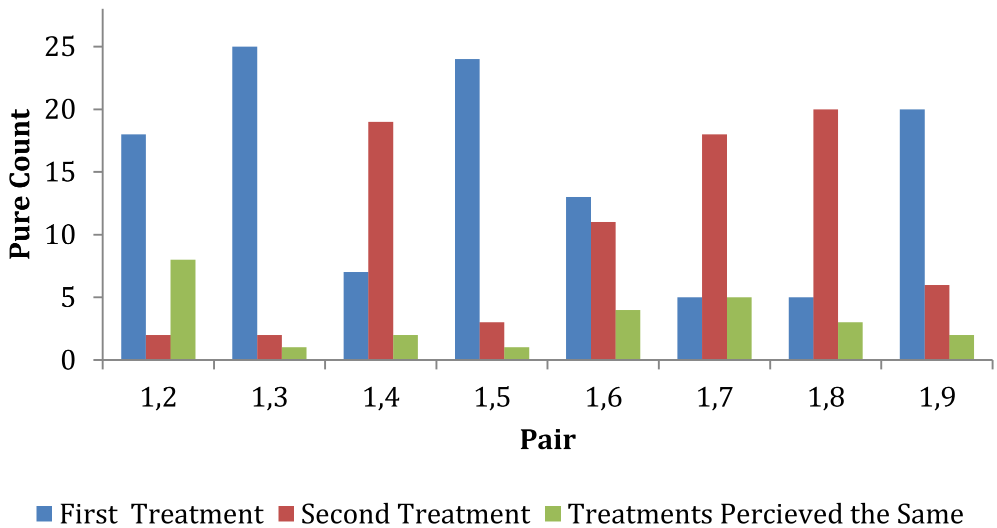

Figure 7.2 – Full factorial of Treatment 1 against all other treatments in the calibrated experiment.

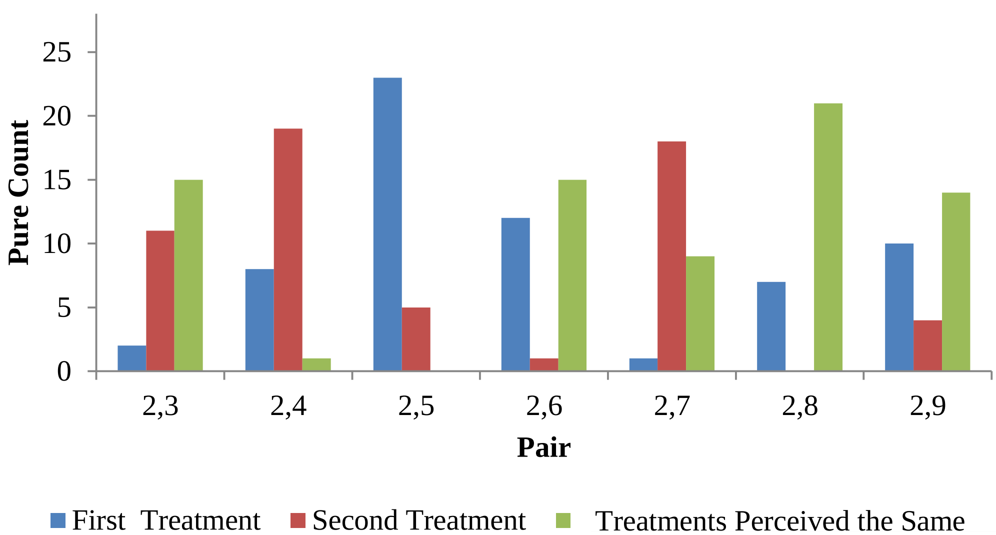

Figure 7.3 – Full factorial of Treatment 2 against all other treatments in the calibrated experiment.

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The pure count of each individual comparison made by all subjects, for treatments 1 & 2 are seen in Figure 7.2 and 7.3. The first treatment is the first element in the ordered pair, the second treatment is the second element in the ordered pair as seen in Table 6.9. The final selection was the perception of both treatments being perceived as the same urgency. The maximum pure count for each pair is 28, if an individual treatment approaches this maximum it indicates that there was a high degree of agreement amongst subjects for this pair. The total for the first, second, and same urgency will equal 28 as well.

If we consider the pair (1,2), Treatment 1, is selected with a greater frequency when compared to Treatment 2, with a pure count of 18. The selections with the second highest frequency for this pair are the treatments being perceived as the same, with a pure count of 8. The remainder is treatment 2, being perceived as more urgent, with a pure count of 2. Treatment 1 is selected with the most frequency when compared against Treatments 2, 3, 5, 6, and 9. Although, Treatment 1 is the most preferred in most cases, in order for it perceived as the most urgent in all cases it would have to be universally selected across all treatments. In addition to being preferred universally across all cases, each selection should be largely preferred against the other treatments. The overall pure count confirms this observation since there are no treatments that approach the maximum pure count for the individual treatment.

This is consistent across both the calibrated and non-calibrated cases. In each of the 36 pairs either a treatment is selected or both treatments are perceived as being the same.

The degree to which each treatment is preferred does not approach the maximum pure count of 28 in most cases. Any number less than 28, indicates an error exists in the evaluation of urgency across all subjects, since the treatment is not universally preferred. However, there is not a treatment that is universally preferred across all treatments; this

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indicates that there are inconsistencies in which the evaluations were made. These inconsistencies are measured by the number of circular triads. From the pure count, one cannot ascertain a discernable trend, the trends are treatment dependent. For the calibrated experiment from Figure 7.3, treatment 2 was perceived as being the same as treatments 3, 6, 8 and 9. Treatment 2, was only perceived as being more urgent when compared to treatment 5. This inconsistency is throughout, and is observed in both the calibrated and non-calibrated experiments. In order to identify any subtle trend in the data the parallel coordinate plots can provide the trend of the data set.

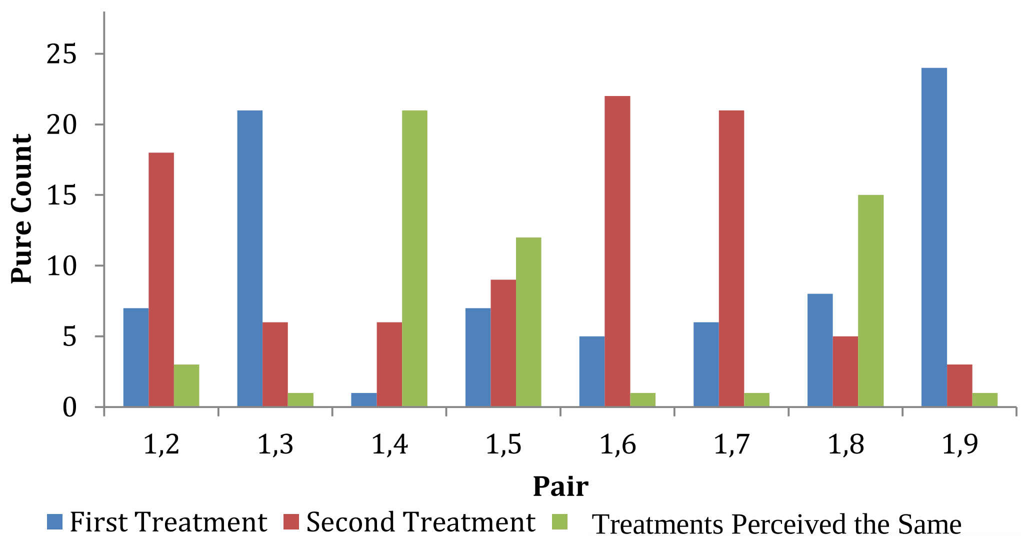

Figure 7.4 – Full factorial of Treatment 1 against all other treatments in the non-calibrated experiment.

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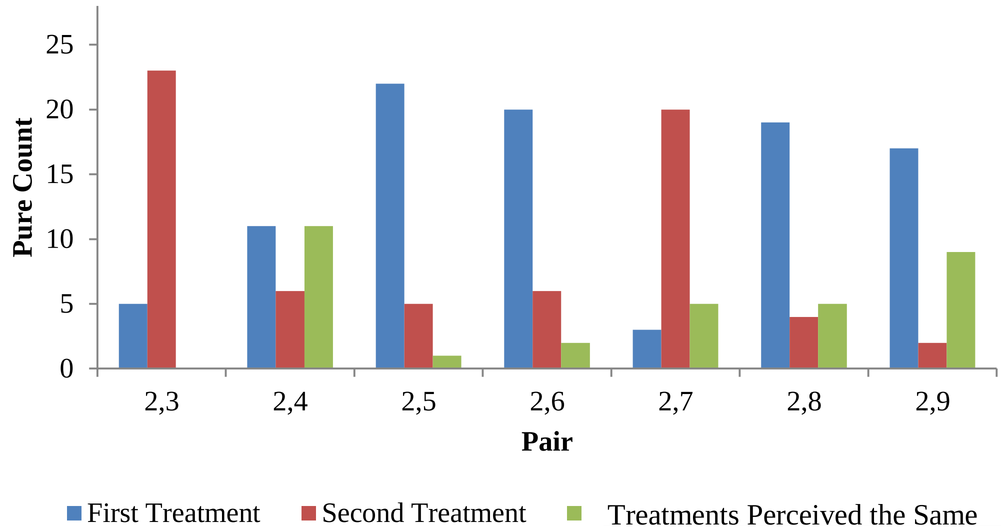

Figure 7.5 – Full factorial of Treatment 1 against all other treatments in the non-calibrated experiment.

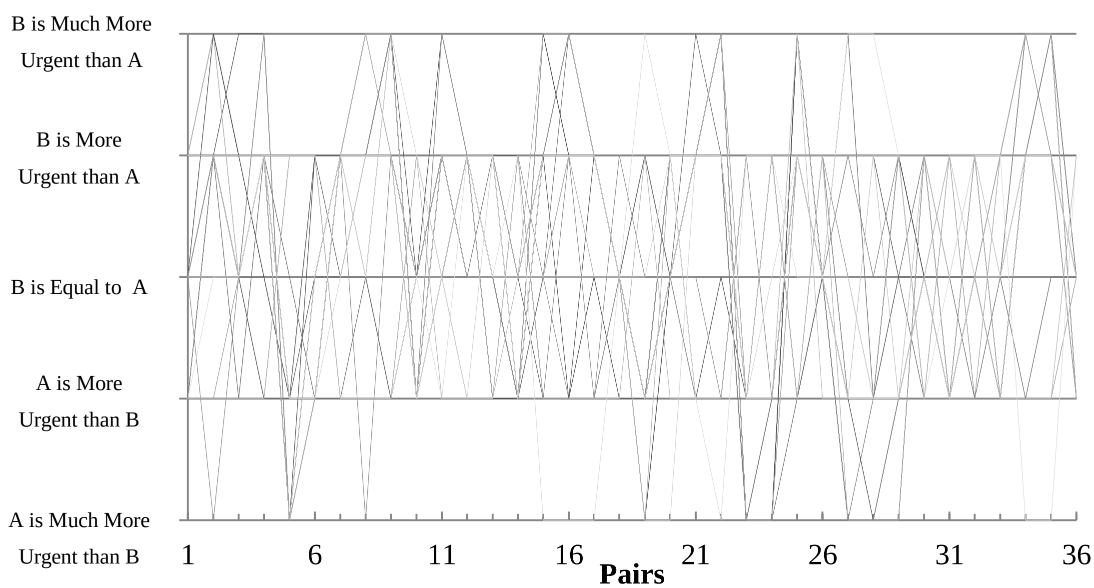

Figure 7.6 – Parallel coordinate plot for calibrated paired comparison experiment with no discernable trends.

The parallel coordinate plot is used to discern trends in the data set. The plot displays 28 subjects, across 36 pairs, clusters in the data reveals the similarity of responses across the

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subjects. There are no discernable trends from the data set, there are clusters for specific paired comparison such as pair 5, where a number of subjects cluster to, Sound A being Much More Urgent than B. From the parallel coordinates it is difficult to observe a trend throughout the data set, a trends would characterize how groups of individuals perceive sounds.

There are several pairs indicated, such as Treatment 5, where a group of subjects responded similarly. In general you see a large fluctuation of subjects that perceive, A is More Urgent than B, and B as More Urgent than A. There is also a groups of individuals that perceive, B is Much More Urgent than A, and A is Much More Urgent than B. These are the values that are the minima and maxima of the graphs. This indicates a cluster the subjects may provide some similarity in grouping. Analysis could provide more insight into the how each group of subjects perceives urgency.

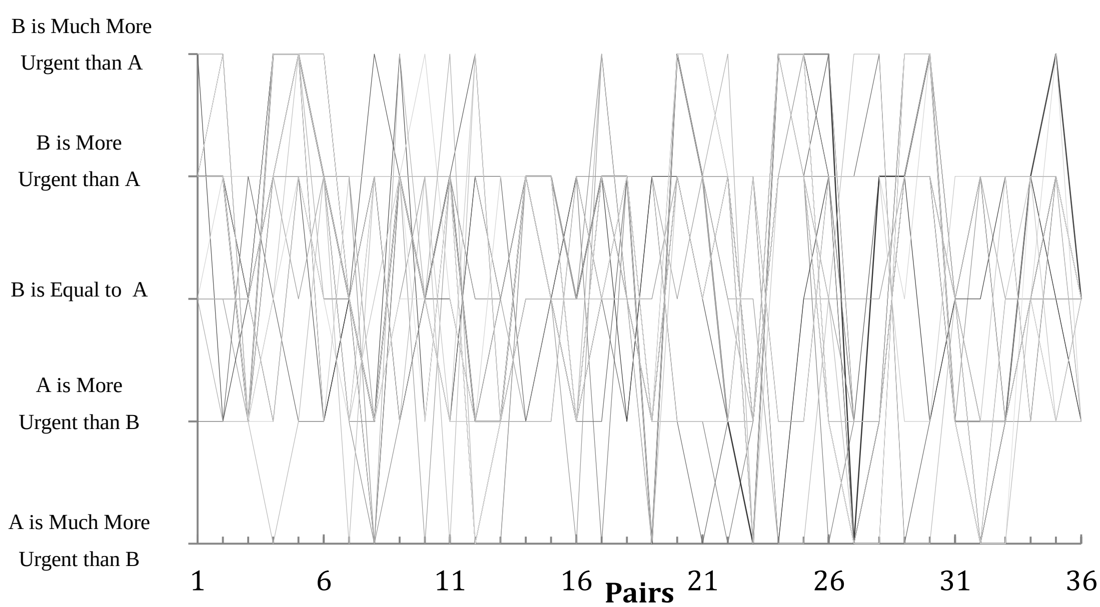

Figure 7.7 – Parallel coordinate plot for non-calibrated paired comparison experiment with no discernable trends.

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The lack of a trend indicates a high degree of inconsistency amongst the responses; this inconsistency can be quantified by the number of circular triads. The number of circular triads for the calibrated experiment is 631. The coefficient of consistency is .43 or 43%, that the preferences were consistent. The circular triads for the non-calibrated experiment were 962. The coefficient of consistency is .14 or 14%, that the preferences were consistent. In both cases there was a high rate of inconsistency, grouping the subjects into clusters may reduce the rate of inconsistency.

The cluster analysis groups subjects based on similarities in their responses of perceived urgency. Based on the dendrogram in Figure 7.8, the recommended number of clusters is four. This is indicated by the by line drawn in Figure 7.8 and 7.9, Group A, Group B , Group C, and Group D. The summary statistics for each group are summarized in Figure 7.6 and 7.7.

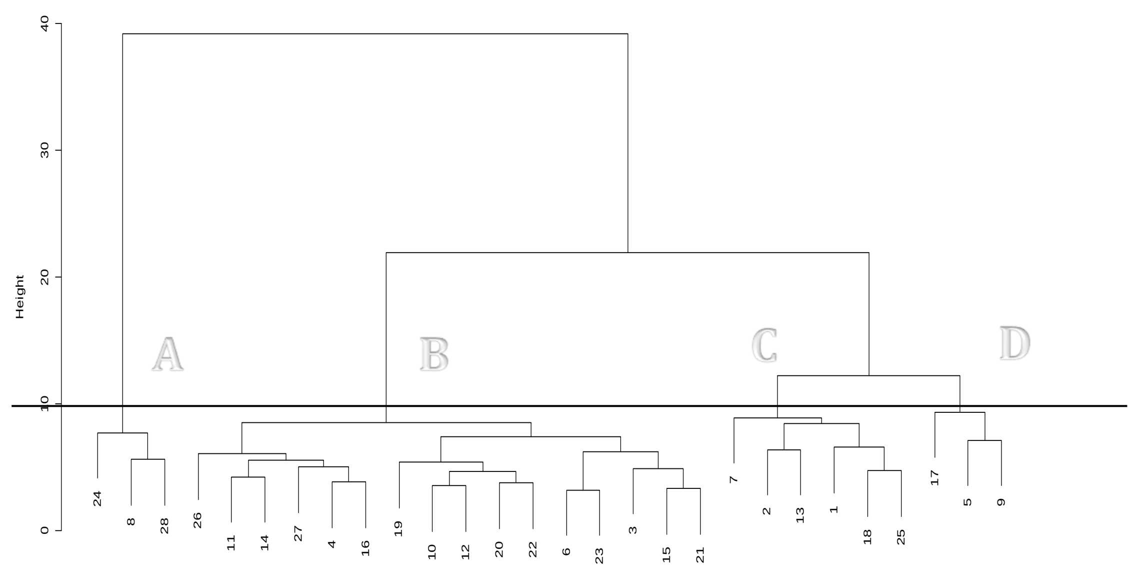

Figure 7.8 – Cluster dendrogram for calibrated experiment.

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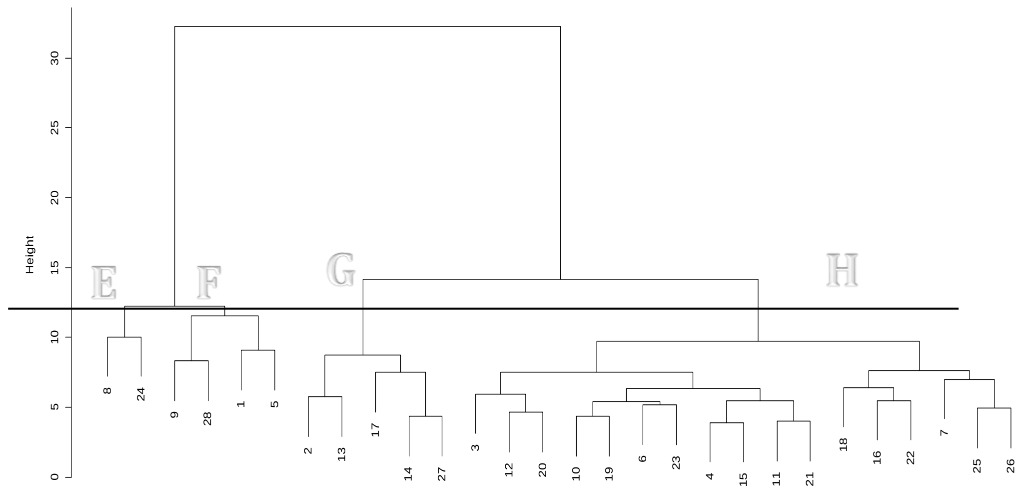

Figure 7.9 – Cluster dendrogram for non- calibrated experiment.

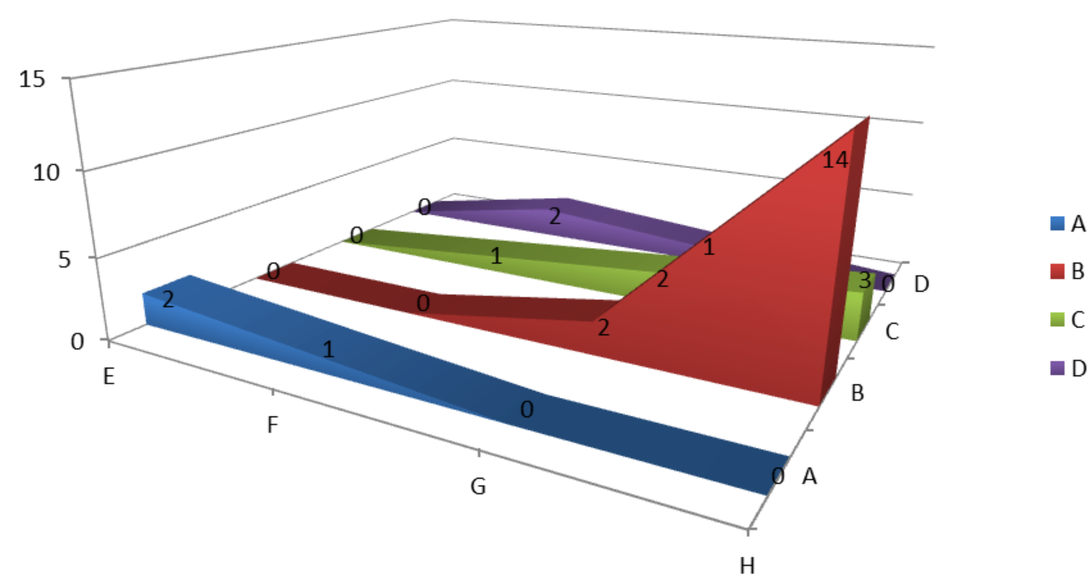

Figure 7.10 – Group alignment between Figure 7.8 and 7.9, for the calibrated and non-calibrated study.

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From Figure 7.10, the greatest alignment is between Group B and Group H, these are two largest groups for both the calibrated and non-calibrated experiment. The two individuals that did align for the non-calibrated case were clustered into Group G. Thus 82.3% of the subjects were similar amongst both groups; this similarity indicates there is consistency amongst the calibrated and non-calibrated experiments. The subjects in each of these groups evaluated the treatments in a similar manner. Thus, when a comparison is made amongst the groups these comparison are very similar due to the high degree of agreement amongst the subjects across the clusters.

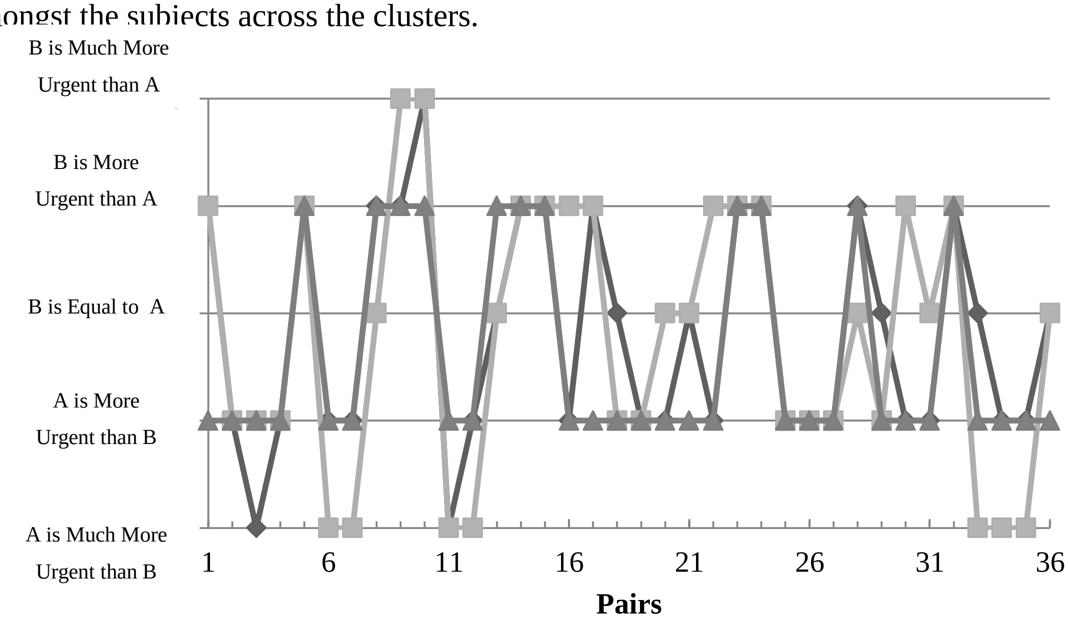

Figure 7.11 – Parallel coordinate plot for calibrated cluster analysis in group A.

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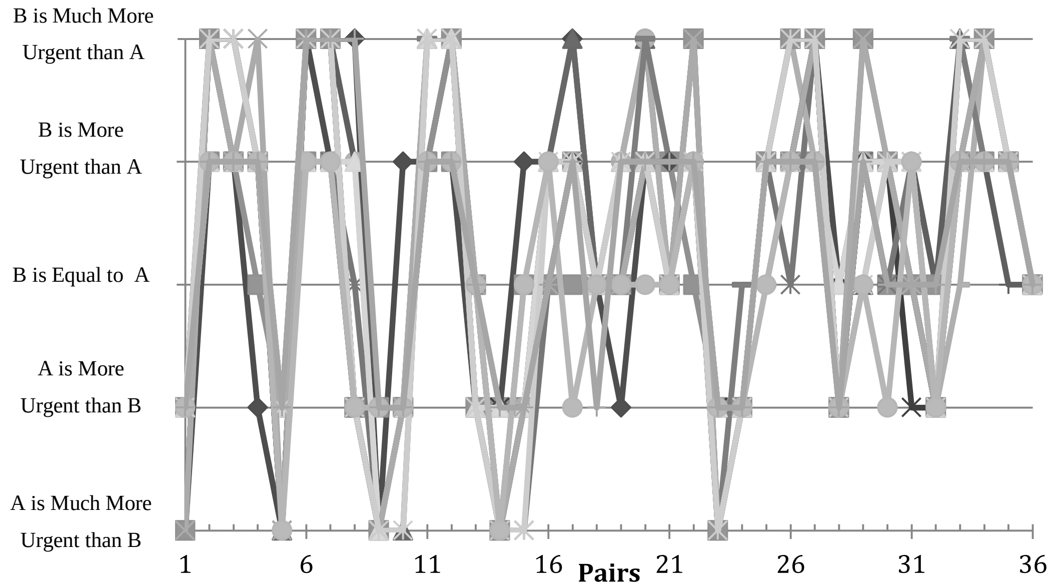

Figure 7.12 – Parallel coordinate plot for calibrated cluster analysis in group B.

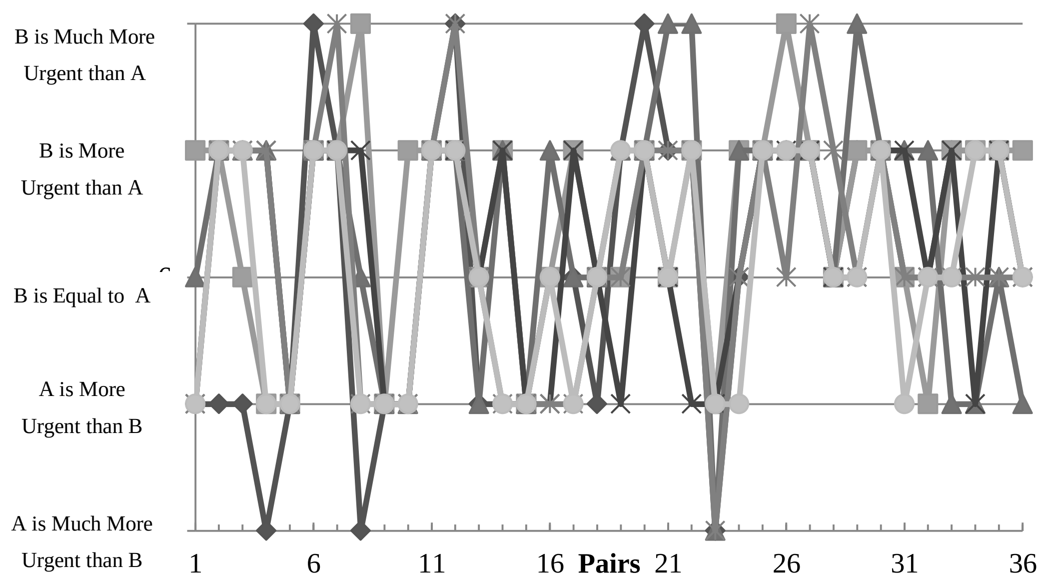

Figure 7.13 – Parallel coordinate plot for calibrated cluster analysis in group C.

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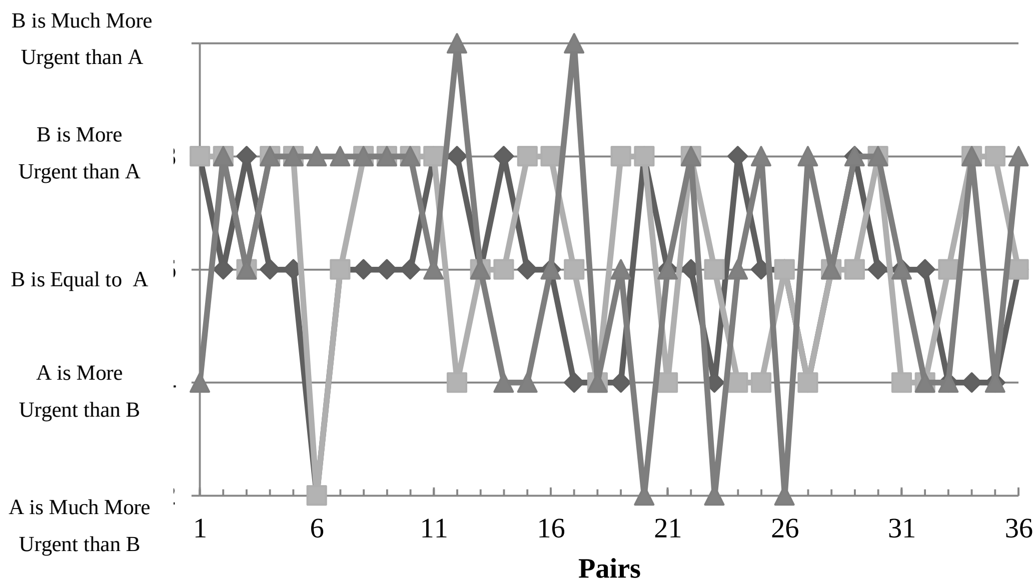

Figure 7.14– Parallel coordinate plot for calibrated cluster analysis in group D.

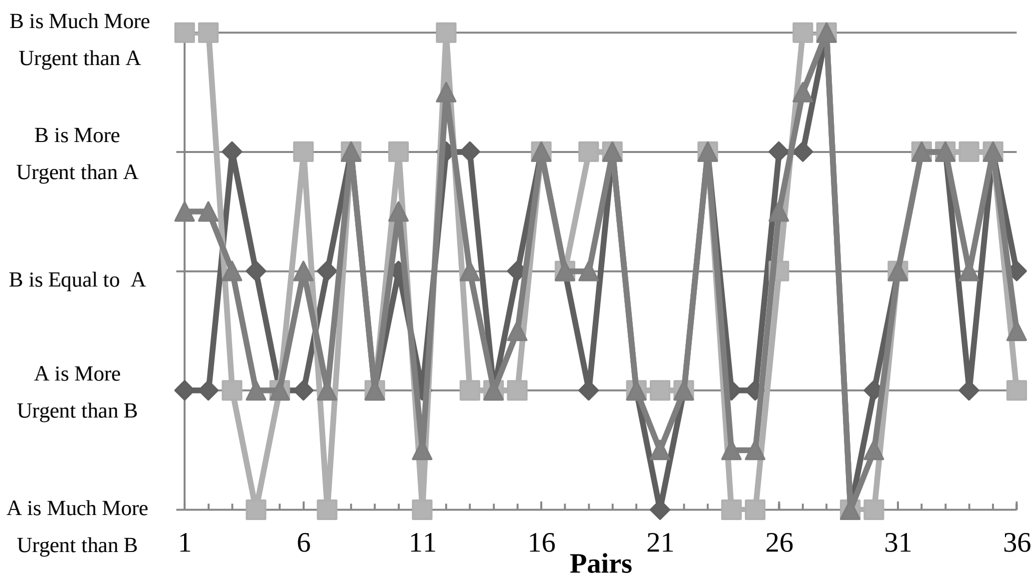

Figure 7.15 – Parallel coordinate plot for non- calibrated cluster analysis in group E.

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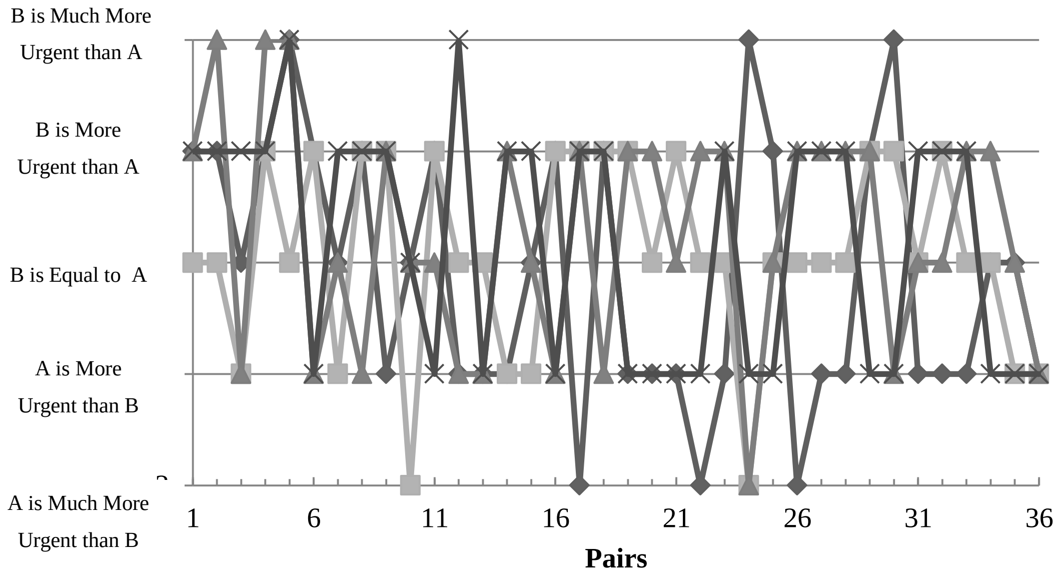

Figure 7.16– Parallel coordinate plot for non- calibrated cluster analysis in group F.

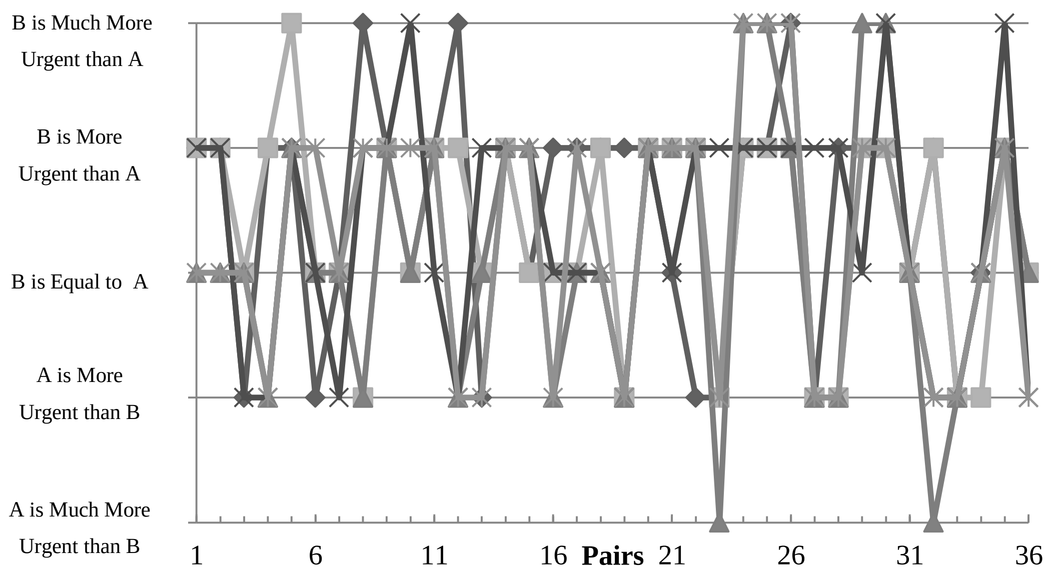

Figure 7.17 – Parallel coordinate plot for non- calibrated cluster analysis in group G.

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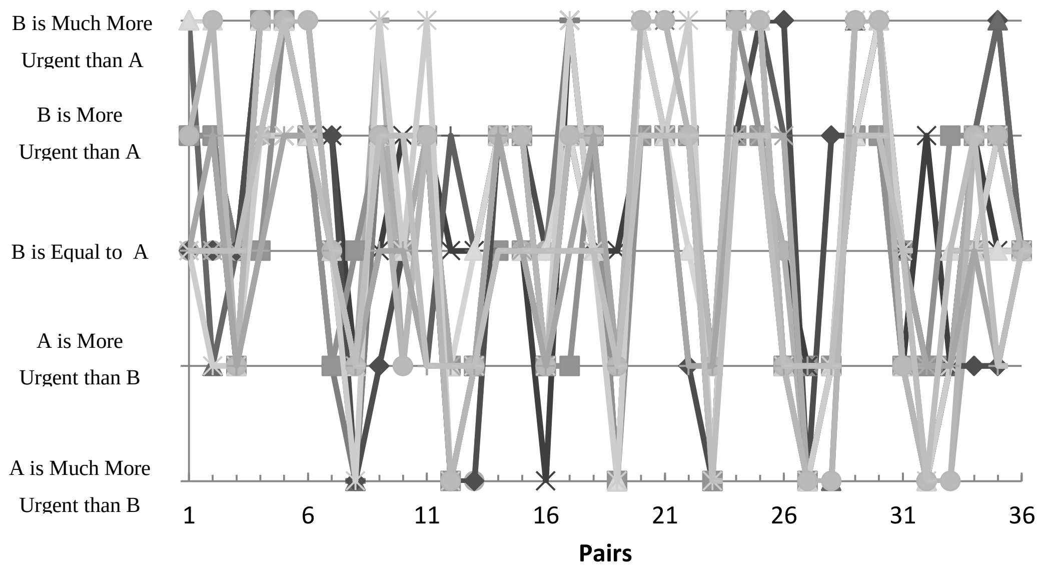

Figure 7.18 – Parallel coordinate plot for non- calibrated cluster analysis in group H.

The average of each group is plotted in Figure 7.19. The scatter plot in Figure 7.11 thru 7.18, reveal the trends of the data set for each group. One can see the difference when the responses are clustered, as compared to the whole population. The clustered groups in both experiments respond on average differently to each pair as seen in Figure 7.19. The manner, in which one group responds to a treatment, varies from the response of the other groups. If one considers Group B as compared to Group A. In general group B either selects, B is Much More Urgent, or A is Much More Urgent. There are few cases when both treatments in a pair are perceived the same. Considering Group A, a similar trend emerges, which is opposite of Group B. Group D on average perceives the sounds as being average; with a high degree of preference to, A is Much More Urgent. Group C has the most variation across the pairs, and falls within the bounds of all the other groups.

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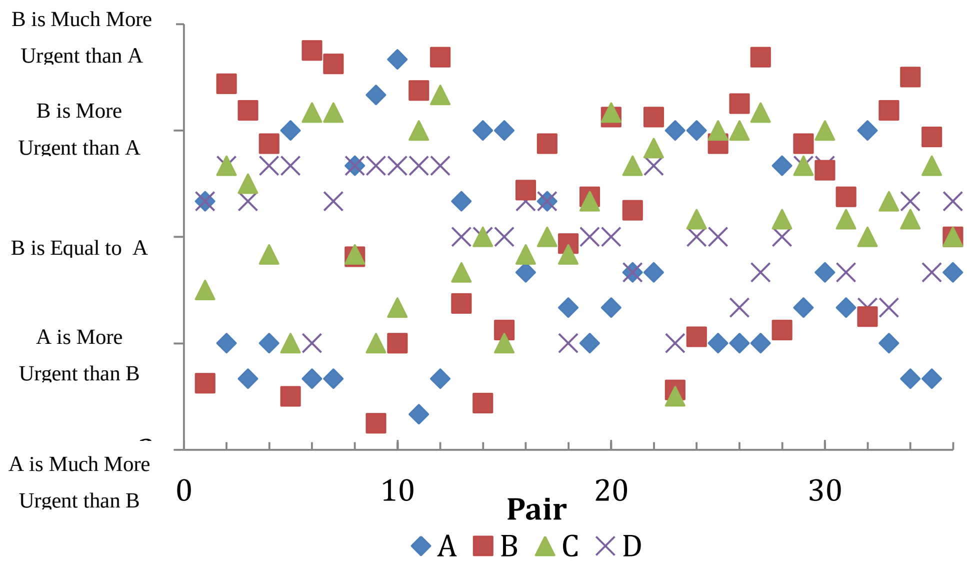

Figure 7.19 – Summary of Figure 7.11 thru 7.14 for calibrated experiment clustering.

A clearer representation of each group can be obtained by previewing the first ten cases of Figure 7.19 as seen in Figure 7.20, where the differences are much more pronounced. The different data point across each pair indicates the degree of variation across each group. This clarifies the overall trends amongst the groups for these cases. If the first pair is considered, group A and D are the same, group B prefers sound A, and group A prefers sound A. If we consider the second pair, Group B prefers sound B, Group C and D slightly prefers sound A, and Group A greatly prefers sound A. This rationale is the same rationale that was used to establish the clusters for the experiment.

There is little differentiation on average between each group in the calibrated experiment in the summary statistics. The general trend is when one group perceived sound A as being urgent; the other perceived sound B, while the remainder viewed them as being

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the same. The non-calibrated experiment may provide more insight into the relationship between perceived urgency and the study's subjects.

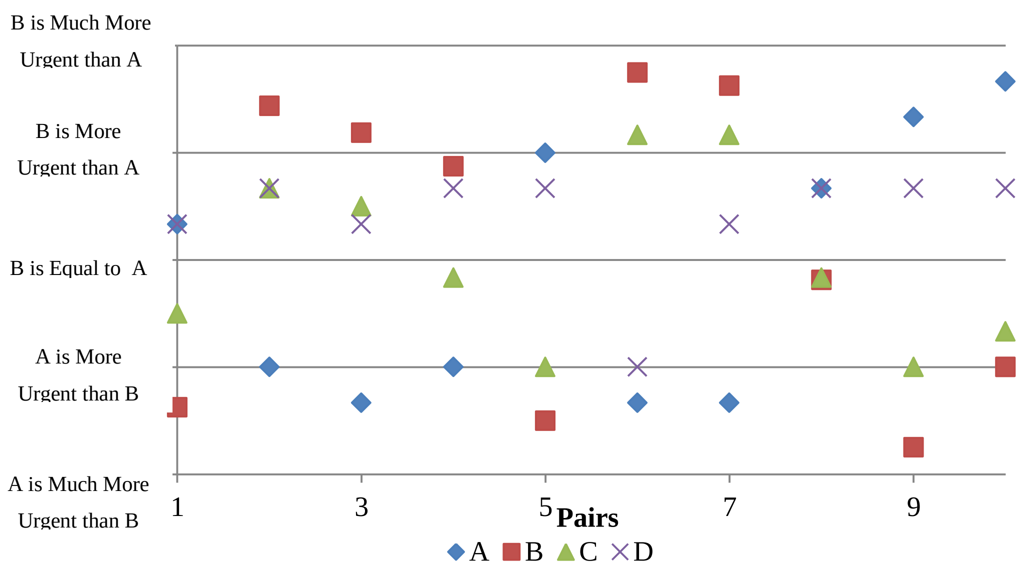

Figure 7.20 – First 10 treatments for Figure 7.19.

The average of each group is plotted in Figure 7.20 for the non-calibrated experiment. The scatter plots for the clustered cases reveal the trends of the data set for the groups. If one considers Group B & C, their average response is nearly identical, and in most pairs they are evaluated at B is Equal to A. Group A and Group D tracks as opposites, when Group A prefers Treatment A, then Group D prefers Treatment B.

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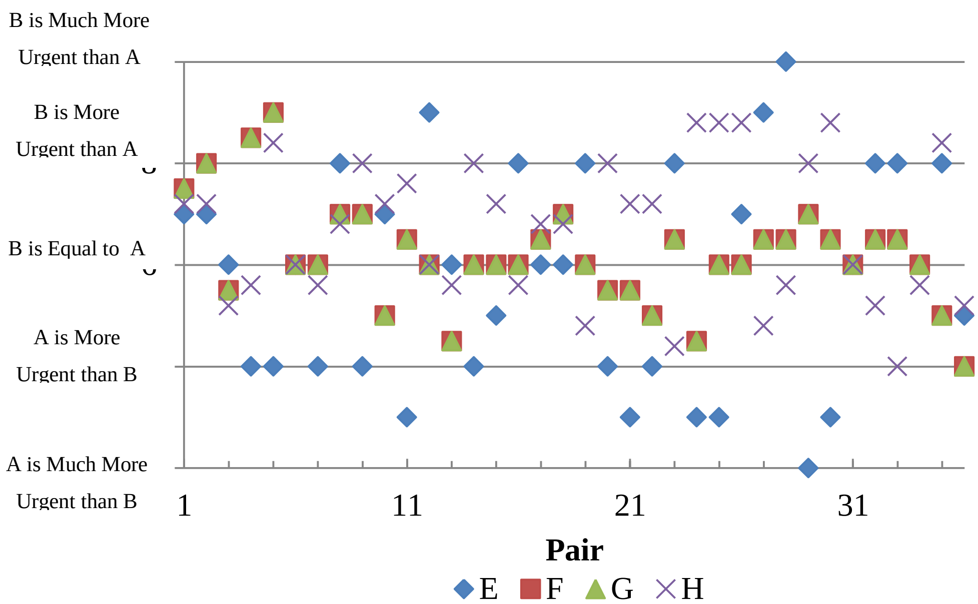

Figure 7.21 – Summary of Figure 7.15 thru 7.18 for calibrated experiment clustering.

Figure 7.21 reveals the pattern in which the subjects were grouped. Groups F and G responded very similarly in most cases. These two groups perceived the sounds as being equally urgent. In most other cases group E and group H responded in the opposite manner. If group E perceived sound A as being urgent, group H perceived sound B as being urgent.

The first ten cases of Figure 7.21 reveals the trends as seen in Figure 7.22. In pair 1,2, and 3 each group responded relatively the same, as we move to pair 4 we see an alignment between group B &C, and a disparity arises between group D and group A. This is the general trend that continues throughout the data set. A lack of consistency exists primarily between groups.

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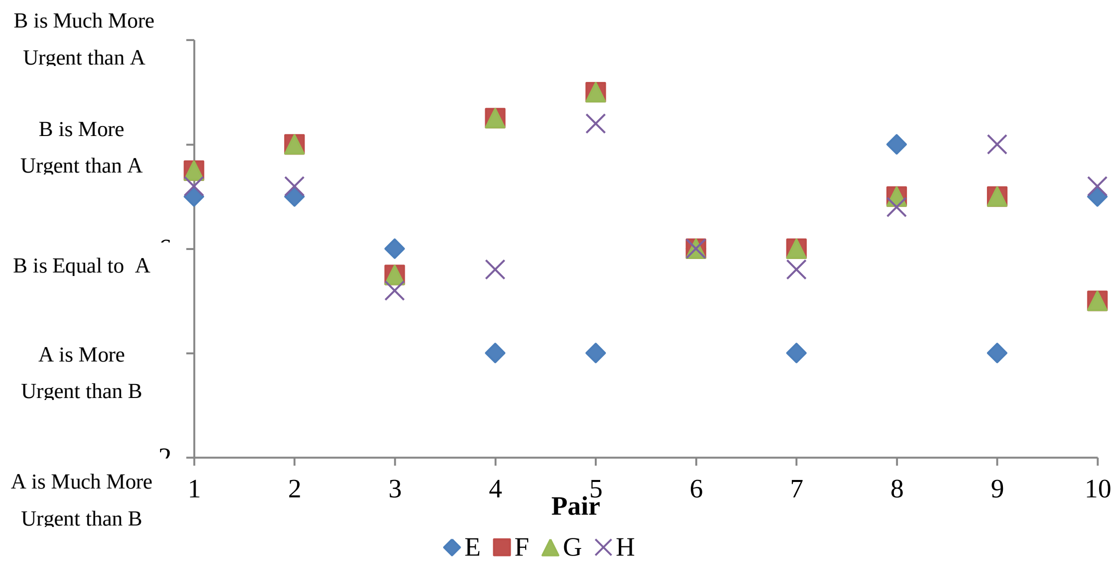

Figure 7.22 – First 10 treatments for Figure 7.19.

The final step in the paired comparison analysis is to determine the consistency within each group. The number of circular triads and coefficient of consistency are considered for Group B in the calibrated experiment and Group H in the non-calibrated experiment. Each group respectively represents 57% and 61% of the total population of this study. The total number of subjects for the calibrated experiments in group B is 17. From the figures one can see there are several groups, where there is universal agreement on the pairs for all the subjects. However, there is still inconsistency in the manner in which the treatments are made by the subjects. This trend continues for the non-calibrated experiment as well.

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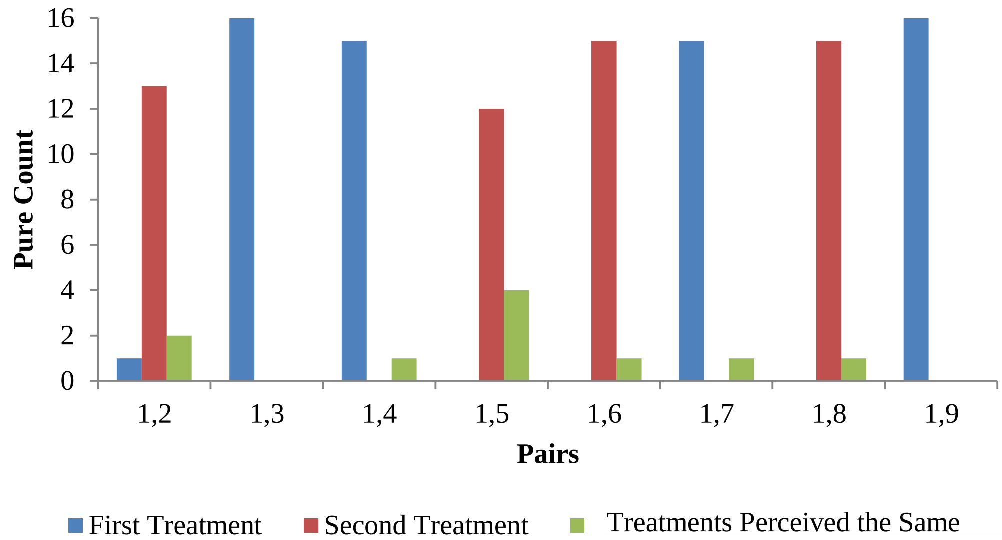

Figure 7.23 – Calibrated experiment pure count for Group B based on Treatment 1.

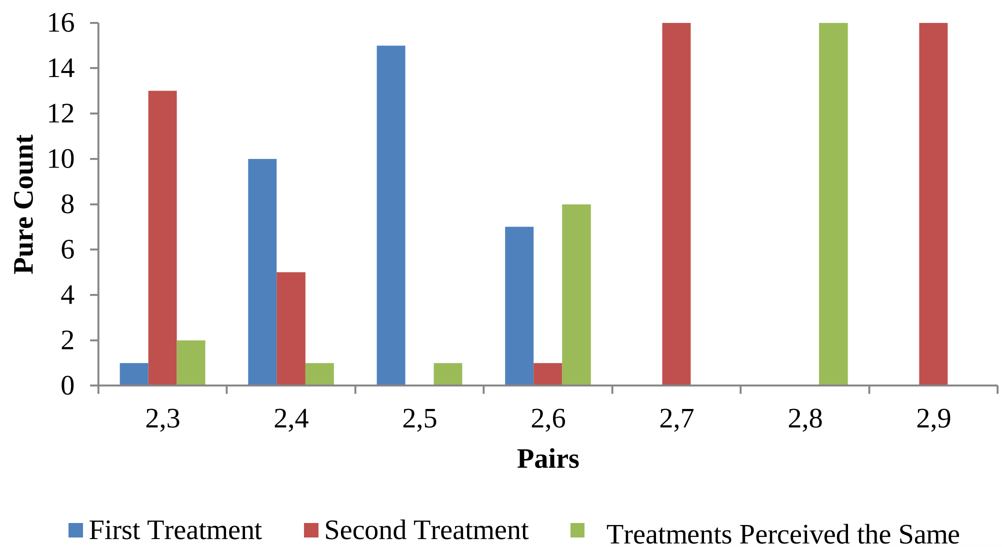

Figure 7.24 – Calibrated experiment pure count for Group B based on Treatment 2.

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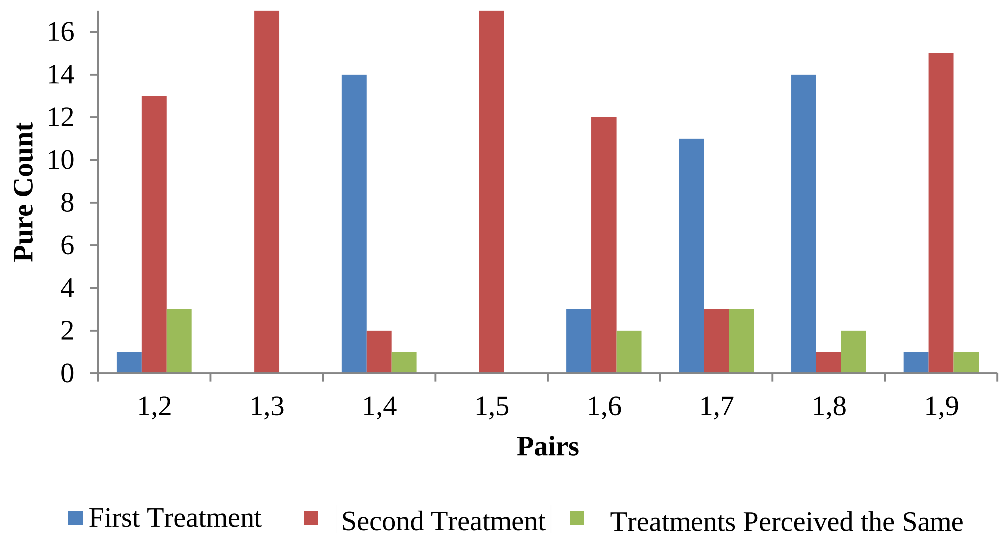

Figure 7.25 – Non-calibrated experiment pure count for Group D based on Treatment 1.

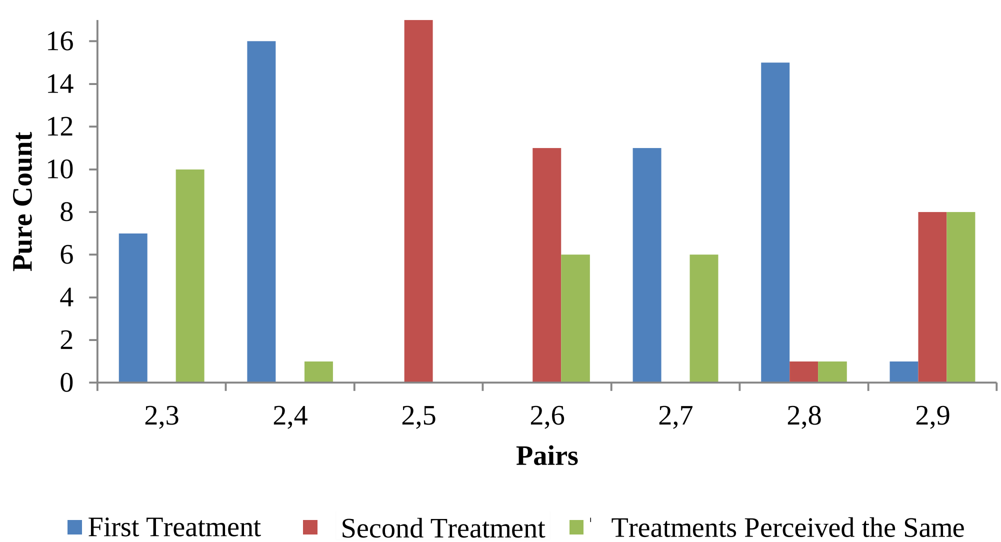

Figure 7.26 – Non-calibrated experiment pure count for Group D based on Treatment 2.

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The pure count for the calibrated group is Figure 7.24 and 7.25. In many cases this group had several universal measurements. If the maximum count is 17, then it is universal selection across all pairs. This consistency is only within groups. Although there is consistency within the respective groups and clusters, one needs to evaluate the consistency in which the paired comparisons were made.

The Merit Score in Figure 7.23 reveals each of the clusters response varied from the calibrated and non-calibrated cases. The merit score reveals signals were perceived the same 102 out of 476 or 20% of the cases. For the non-calibrated case the signals were perceived the same 106 out of 504 or 21% of the cases, roughly the same average across both cases. Thus, the clusters did not make a difference in how the subjects perceived the treatments the same.

This distinction is consistent with Figure 7.27, which only considers the clustered cases. Based on the distribution there is clearly a preference for treatments each subject preferred in the two cases. The cluster analysis revealed the nature in which different groups respond to stimulus in Figure 7.27. This shows a universal scale is not possible, although there are groups in both cases that make up a large percentage of the population the rating is group specific. These results are encouraging implying that a large group of subjects perceive urgency in the same manner. This result is expected due to the nature of how the human hearing system processes sound; a universal result is highly unlikely.

For the calibrated signal, the treatment evaluated as the most urgent is Treatment 1, with a pure count of 78 out of 136 or 57% of the time. The treatment evaluated as the least urgent treatment is treatment 6, with a pure count of 6 out of 136, or 4.41% of the time. In this case, the calibrated signal has a signal that is clearly perceived as the least urgent. For

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the non-calibrated signal, the most urgent treatment is treatment 5, with a pure count of 76 out of 128, or 59.4% of the time. The treatment evaluated as least urgent is treatment 8, with a pure count of 25 out of 128, or 19.5% of the time.

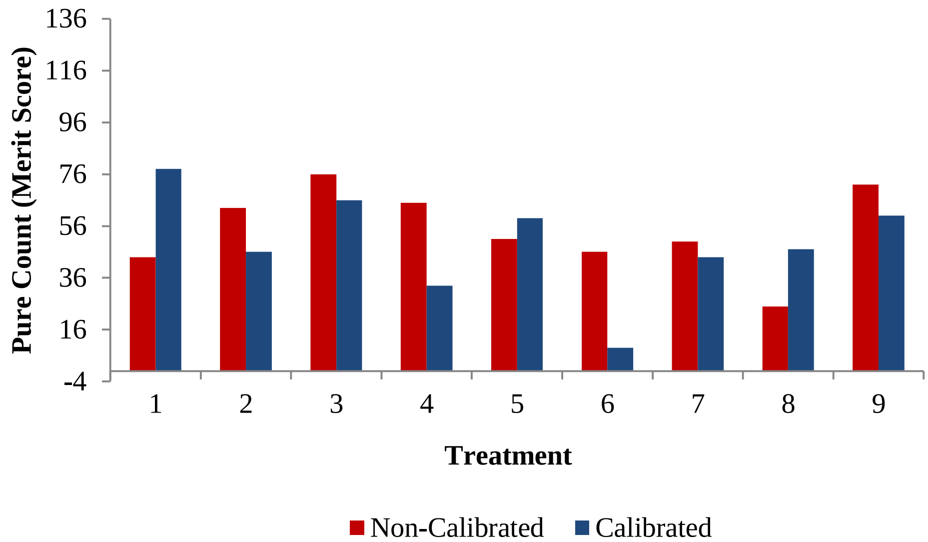

Figure 7.27 – Merit score for Group B calibrated experiment, and Group D non-calibrated experiment.

The cluster analysis reveals a decrease in the number of circular triads, and an increase in the coefficient of consistency. The number of circular triads for the calibrated experiment is 316, with a coefficient of consistency of .8372 or 84%. This is a much greater consistency compared to the 43%, when all the subjects were considered. The number of circular triads for the non-calibrated experiment is 776, with a coefficient of consistency of .60 or 60%, which is a sharp increase from the 14%. In clustering subjects the rate of consistency increases in the evaluation of perceived urgency. This provides sufficient information to perform a relative ranking of treatments.

For the calibrated case, starting at the highest urgency the treatments would be ranked as 3, 9, 4, 2, 5, 7, 6, 1, and 8, with this ranking being consistence 84% of the time within its

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cluster. For the non-calibrated case, starting at the highest urgency the treatments would be ranked 1, 3, 9, 5, 8, 2, 7, 4, and 6, with this ranking being consistent 60% of the time within the cluster. Calibration has a direct effect on the perception of urgency as seen by the ranking of the treatments. The ranking were established based on the merit score, the higher the pure count, the more times the sound was perceived as being more urgent. The paired comparison was able to provide a ranking scheme of relative urgency. In order to evaluate absolute urgency the magnitude estimation must be considered.

The next step is to determine how significant each of the treatments wer to determine if this ranking scheme is viable. In order to assess significance a t-test was performed on the magnitude estimation portion of the experiment. The first row in Table 7.3, show all 9 treatments, and the first column shows all 9 treatments, each treatment is compared relative to all the other treatments via a t-test comparison.

A full factorial t-test evaluated each treatment, a yes indicates significance when α=.05, and a no indicates no significance at the same confidence level. In order to create a ranking scheme each sound in the ranking must be statistically significant to all other sounds. The results of the t-test are summarized in Table 7.4 for the calibrated, and Table 7.6 for the non-calibrated signal.

Table 7.4 reveals five distinct levels of urgency, each scale with three levels. From the calibrated treatments one could create an urgency scale with high, medium, and low urgency that would be consistent across the entire subject population, indicated by its means to the right of the treatment. Table 7.6 reveals for the five distinct cases of urgency, ranging from three to four levels of urgency. This indicates the calibrated signal provide more scales

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than the non-calibrated case. The non-calibrated cases provide a greater resolution of the urgency scale.

Table 7.2 – Calibrated t-test significance table.

	Am_.1 Fm_10	*Am_.1 Fm_20*	*Am_.2 Fm_10*	*Am_.2 Fm_20*	*Am_.3 Fm_10*	*Am_.3 Fm_20*	*Am_.4 Fm_10* ☾	*Am_.5 Fm_10*	*Am_.6 Fm_10*
▲ ■ *Am_.1 Fm_10*		No	Yes ■	Yes ■	Yes	Yes	No	Yes ▲	Yes ▲
ϟ ★ *Am_.1 Fm_20*			Yes ★	Yes ★	Yes	Yes	No	Yes ϟ	Yes ϟ
Am_.2 Fm_10				Yes	Yes	No	Yes ☾	No	No
Am_.2 Fm_20					No	No	Yes ☾	No	No
Am_.3 Fm_10						Yes	Yes	Yes	No
Am_.3 Fm_20							Yes	No	No
Am_.4 Fm_10								Yes	Yes
Am_.5 Fm_10									No
Am_.5 Fm_10									

Table 7.3 – Summary calibrated t-test summary from Table 7.2.

■	▲	★	ϟ	☾
Am_.2__Fm_10 - 2.04	Am_.5__Fm_10 - 3.08	Am_.2__Fm_10 - 2.04	Am_.5__Fm_10 - 3.08	Am_.2__Fm_10 - 2.04
Am_.2__Fm_20 - 3.52	Am_.6__Fm_10 - 3.12	Am_.2__Fm_20 - 3.52	Am_.6__Fm_10 - 3.12	Am_.2__Fm_20 - 3.52
Am_.1__Fm_10 - 6.20	Am_.1__Fm_10 - 6.20	Am_.1__Fm_20 - 6.96	Am_.1__Fm_20 - 6.96	Am_.4__Fm_20 - 6.48

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Table 7.4 – Non-calibrated t-test significance table.

	Am_.1 Fm_10	*Am_.1 Fm_20*	*Am_.2 Fm_10*	*Am_.2 Fm_20*	*Am_.3 Fm_10*	*Am_.3 Fm_20*	▲ *Am_.4 Fm_10*	*Am_.5 Fm_10*	*Am_.6 Fm_10*
Am_.1 Fm_10		No	No	Yes	Yes	No	Yes ▲	No	No
Am_.1 Fm_20			Yes	Yes	Yes	No	Yes	Yes ⬮	No
Am_.2 Fm_10				No	No	Yes	Yes ▲	Yes ⬮	Yes ◆
Am_.2 Fm_20					No	Yes	Yes	Yes	No
Am_.3 Fm_10						Yes	No	Yes ⬮	Yes
Am_.3 Fm_20							No	Yes	Yes ◆
Am_.4 Fm_10								No	Yes ◆
Am_.5 Fm_10									Yes
Am_.6 Fm_10									

Table 7.5 – Summary calibrated t-test summary from Table 7.4

▲	⬮	◆
Am_.2__Fm_10 - 2.88	Am_.2__Fm_10 - 2.88	Am_.2__Fm_10 - 2.88
Am_.4__Fm_10 - 4.20	Am_.3__Fm_10 - 3.56	Am_.4__Fm_10 - 4.20
Am_.1__Fm_10 - 5.76	Am_.5__Fm_10 - 5.04	Am_.6__Fm_10 -6.68
	Am_.1__Fm_20 - 6.40	Am_.3__Fm_20 -6.80

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This study investigated the perception of perceived urgency from a relative urgency and absolute approach. In both cases a ranking of the stimulus was achieved, creating the foundation to build appropriately designed auditory displays. The relative urgency was evaluated in the paired comparison, where the results for the overall population displayed inconsistences, and a high number of circular triads. This can be contributed to human factors, stimulus, or physiological differences. Clustering the subjects in groups produced much more consistency, which indicates that without the appropriate training urgency is quasi-individualized scale. The consistency in which urgency is evaluated is highly dependent upon the population in which you reside.

The absolute evaluation of urgency was the magnitude estimation, which was able to produce an urgency scale that could be generalized across the entire population. Although, the resolution varied across the calibrated and non-calibrated experiments in both cases there were significant differences in the evaluation of urgency. In an absolute evaluation of urgency one of several universal scales could be adopted in the construction of an appropriately designed auditory display. In both the absolute and relative evaluation a hierarchical arrangement of urgency was achieved. Thus, amplitude and frequency of modulation applied to broadband stimulus does allow the construction of both a relative and absolute urgency scale.

The distribution for the calibrated and non-calibrated case varied in every aspect of the study. The paired comparison for each case produced significantly differences in preference amongst the treatments. When performing clustering, the groups for each cluster were not the same, although there was some degree of overlap across the groups, they still produced a distinct evaluation and ranking of urgency. The magnitude estimation of the

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experiment followed this trend in not only the number of urgency scales, but also its resolution. Thus, calibrating the broadband stimulus has an effect on the perception of urgency.

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8. CONCLUSION AND FUTURE WORK

This work only touches the surface of factors to be considered in the construction of an auditory display. Factors such cognitive load, sound quality parameters, context, and instrumentation were ignored. The study conducted in this thesis was from a general alarm design theory approach, where the goal was to determine if modulation is a viable treatment, which induces different responses amongst subjects. These results must be evaluated with supplemental studies to determine if the sound stimulus selected is the appropriate choice for the context of a UAV for Battle Space.

UAV pilots are required as the subjects, in addition the appropriate instrumentation, scenario, and events to quantify the context for the experimental setting. Contained within the events tasks with a representative cognitive load needs to be identified and designed within the scenario. Further, within an event there needs to be sub-events with act as a secondary cognitive load to evaluate how cognitive load influences the perception of the sound stimulus. If from this stud, the results are encouraging the sound quality parameters will need to quantified and mapped to the appropriate events and sub-events.

The existing literature in alarm design theory, sonification, and sound quality provides a unique opportunity to merge these domains in the construction of an auditory display. Alarm design theory and sonification has concluded robust guidelines and acoustics characteristics that can be modified to acquire the desired response. Sound quality has defined objectively the acoustics characteristics in terms of a sensation. The next step to progress this research is to quantify sounds in terms of a sensation as well as with the finding in the alarm design theory and sonifcation.

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The focus of this study was the evaluation of urgency, and the factors that influenced its perception. Amplitude modulation and the frequency in which the amplitude modulation is applied influenced the perception of urgency for the population in this study. This was observed in both the paired comparison and magnitude estimation study. The paired comparison revealed inconsistences exist in the evaluation of urgency across the studies' population. There was no pure sound that was unanimously perceived as being the most urgent. Each of 9 treatments were evaluated against all other treatments, 224 judgments of each treatment were made. Treatment 4, for the non-calibrated case was selected the most at 116 out of the 224 judgments, roughly 52% of the time. If one considers this stimulus being applied to an auditory display, 48% of time, the intended meaning of the stimulus will be lost due to its inconsistencies in judgments.

For the calibrated experiment the treatment selected the most frequently was treatment 4, selected 112 out of 224 times, exactly 50% of time it was selected as being more urgent than its respective pair. The converse is the remaining 50% of the time was either being perceived as less urgent or the same urgency in its respective pair. In a combat scenario 50% of the UAV pilots associating a situation with one level of urgency, while the remaining 50% associates the same sound with a different level of urgency, inciting confusion, which would ultimately lead to casualties.

In an auditory display, the stimulus would only be perceived distinguishable 80% of the time, which is not sufficient. Imagine if this stimulus was placed in a combat setting where 20% of the time the urgency of a situation could not be distinguished from other levels of urgency. There would be no distinguishable difference between UAV entering an

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ambush scenario, as compared to a no threat scenario. In an experimental setting these results are encouraging, but in a practical situation it merits further investigation.

There were two theories that can be used in the evaluation of auditory displays based on Gibson's theory of affordances (1977). This experiment assumed the sound stimulus has embedded a natural urgency. Each subject mentally constructed the concept of urgency, thus identifying the characteristics in the stimulus which align with their perspective. The only framework provided was the central theme of urgency. If another theme other than urgency was provided to the subjects, the results could be influenced. For example, if the theme of the experiment was the evaluation of pleasantness would the results be statistically significant between the two studies. This raises the question of whether the evaluation of the urgency is due to the inherent nature of the sound, or the theme presented to the subject. Urgency was selected as to provide a theme which supports a practical implementation.

If the decision was made to train the subjects, providing not only a theme, but also a minimum and maximum urgency context, a consistent framework would be supplied for each subject. This framework would define the mental boundaries in which a subject makes their subjective decisions. The decision not to use this approach may have negatively influenced the variability amongst the subjects. In constructing this experiment, every measure to control error should have been considered. The trade-off between determining if a sound has an inherent urgency and providing a framework for urgency, did not consider the error introduced by selecting the latter.

The criterion for the stimulus selection was based on the requirement of having an abstract representation. To satisfy this criteria broad band sampled noise was selected. Alarm design theory provides a theoretical framework for the construction of alarms,

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however there is concrete methodology for the selection of treatments when constructing auditory displays. This lack of foundation makes the selection for the stimulus appear being arbitrary or application specific. The arbitrary nature of the selection could induce an additional source of error. There needs to be extensive work to consider which types of stimulus are most suitable in the evaluation of urgency. A precursor to this study should have been a set of sounds presented to users to determine which sound has an inherent urgency, and evaluated with free verbalizations. The free verbalizations would allow users to construct responses based on descriptive indicators of the sound. This would ensure the selection of a stimulus with an inherent urgency based on the characteristics of the auditory signal.

This lack of foundation makes stimulus selection a difficult task, if the stimulus is not suitable than the treatment applied will not induce the optimal effect. The treatments of modulation applied to this stimulation, was an exploratory approach to determine if a new methodology could be explored in the construction of an auditory display. The range of parameters selected in this study was based upon differences perceived by the experimenter. These differences in perception were made solely on subjective evaluations. There were no stringent guidelines that were employed in the treatment parameter selection. For the amplitude modulation the range of values were limited to values from 0 to 1. To explore the design space and balance the size of the experiments, increments of an amplitude modulation of .1 were used. The frequency of modulation has a design space that ranges from 0 to 300Hz. A range was selected that provided a beat characteristic of a sound at frequencies less than 15Hz, and a roughness characteristics at frequencies greater than 15Hz (Zwicker, 2008)

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The study revealed both parameters in the results were comparable, either choices in amplitude or frequency modulation produced a difference in how it was perceived in the paired comparison experiment. There was no association in the magnitude of the modulation parameter and the degree of urgency. This is perplexing since one would assume the higher the modulation parameter the more urgent a sound would be perceived. The treatments revealed, the higher the modulation parameters the lower the urgency, and the lower the modulation parameter the higher the urgency. If one considers the treatments for the calibrated experiment, a treatment with an A m =.2 at a f m = 10, had a mean urgency of 3.29, when compared to an A m =.1 at a f m = 10, had a mean urgency of 6.64, when compared to an A m =.5 at a f m = 10, had a mean urgency of 3.04. In constructing a scale of urgency one would assume since the frequency of modulation is constant, the amplitude of modulation is what is producing the change. Ideally, a linear relationship between the amplitude of modulation and urgency would be desired for the construction of an auditory display. This inconsistency would make it difficult to construct an auditory display mapped to the natural world.

At amplitude modulation parameters greater than A m =.4, was influenced by the effects of masking as described in Chapter 3. These treatments on average were much higher than the unmasked treatments. There were exceptions in both the calibrated in non-calibrated cases, treatment 6 and 5 respectively. The cause for this discrepancy can be due to degree of masking that has taken place, if a signal is unable to modulate to its full depth, and then the urgency perception is not fully developed. This lack of development leads to the perception of an un-modulated signal, which inherently has a low urgency. This logic would violate the other cases in which the urgency was perceived to higher. Thus, an explanation

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for this phenomenon is unclear, and requires further investigation to determine the effects of masking in the perception of urgency.

There is a clear difference in the magnitudes between the calibrated and non-calibrated experiments. The non-calibrated experiment produced results that were on average much higher than the calibrated. This indicates the perception of urgency is influenced by the system used, a t-test was performed on the calibrated versus non-calibrate experiment.

This reveals only in the cases when masking is present are the results between the calibrated and non-calibrated experiments for its respective treatment not significant. In cases when the signal is being masked there is no way to know what each of the subjects is evaluating. The evaluation could be influenced by either the background level or the modulation of the stimulation.

The manner in which the treatments were presented can also contribute to the error in the experiment. The treatments were provided to an untrained group of subjects, in which they were not introduced to any stimulus prior to conducting the experiment. The experiment consisted of a paired comparison experiment in the first portion of the experiment followed by a magnitude estimation experiment. The calibrated experiment was performed first followed by the non-calibrated experiment. The paired comparison was used to orient the subject to all the treatments that would be heard during the study, so when the magnitude estimation portion was conducted a mental framework of all the possible choices could have been developed. This issue needs further investigation to determine if the order in which the experiment is conducted has an effect on the perception of urgency.

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Thus far, the inconsistencies from the experimental design, stimulus, experimental control, and general implementation have been considered. The largest source of variability in this experiment is due to the subjects. The average of all the subjects for the magnitude estimation was computed and an analysis of variance was conducted based on the averages. The dosages of treatments were applied to the subjects in the same manner. Performing an ANOVA on the experiment reveals the largest source of error in the experiment. This experiment results are for the paired comparison of the non-calibrated experiment

The sum of squares for the residuals is the largest which indicates the largest variation amongst subjects. The second source of variation is due to the Am. The Fm is the third source of variation, and the interaction between the two variables contributed the least to the variation in the experiment. If a subject was listening to loud music prior to performing the experiment, through head phones, in a car, or through constant exposure to loud noises it is conceivable they could have suffered from temporarily hearing loss prior to conducting the experiment. This could severely influence their perception of sound stimulus during this period. Additional care needs to be taken so the subjects have normal hearing at the time of the experiment, this can be accomplished with an audiogram. The differences were based upon the treatments rather than the effects of human hearing. This study offered encouraging results in the construction of an auditory display. There were several sources of error that were identified which could be remedied through an alternative approach to the experimental design. There are several different approaches that could be taken in future experiments and detailed explanation of the general methodology is outlined below.

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8.1 Future Experiments

This study revealed several opportunities that exist for future studies. (1) The selection of the stimulus used; the aircraft cabin noise was randomly selected. The selection of a different sound stimulus may have provided a better resolution of the urgency scale for the study. If a technique such as a free verbalization was employed, where subjects evaluated sound stimulus, by providing adjectives and nouns to describe the respective stimulus, the most appropriate stimulus may have been revealed. Further, performing a linguistic analysis would allow the clustering. This will provide a base set of sounds with the appropriate adjectives assigned to those sounds. For example, several distinct sound would be evaluated in terms of urgency, rather than a base sound with several treatments.

Once the sound stimulus is defined with the appropriate adjective, (2) mapping the appropriate context to the scenario is the next step. This requires knowledge of the events and scenarios for the respective domain. This will require the input from process experts familiar with the application to define these events. For example, the Battle Space project in mapping sounds to proximity, threat level, fuel level etc., will reveal sounds that are more intuitive to these quantities, this requires a large set of sound stimulus in step 1. The mapping procedure outlined by Edworthy from alarm design theory can serve as a suitable framework to accomplish this task. This issue arises during the implementation phase, when subjects for this study have the appropriate reference frame to select the sounds consistent with the target population.

If sounds are mapped to the general events, the next phase is to (3) quantify these sounds in terms of sound quality parameters. This creates a mapping which aligns with the nature in which humans hear. This is unique in that most auditory displays do not consider

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the sensation of sound to humans. Once the general sounds are quantified the, (4) the treatment applied to the general sounds needs to be selected. There are vast range of options that exist to manipulate sounds, such as the just noticeable difference criteria, filtering, alarm design theory findings, sonification guidelines, and modulation. Similarly, to step 1 an exploratory study may be required to determine which treatment is the most appropriate for the application. At this phase, the sounds have been appropriately mapped so treatment methodology can be explored.

Upon quantifying the treatment, (5) the sound quality parameters for each treatment needs to be calculated. Once the sound stimulus are quantified in terms of its sound quality, (6) mapping the parameters to the appropriate event is required. This allows discrete data points amongst where each point consists of its sound quality parameters and a respective event. From the discrete points a continuous function can be created be created to achieve a continuous scale. (7) The challenge at this phase is the creation of dynamic interface which modifies the sound quality parameters to the appropriate event, and intermediate events. This requires additional research on how to accomplish this objective. This system will need to be evaluated independently of cognitive load, to ensure the discrete data points are still distinguishable.

Lastly, factors such as (8) cognitive load needs to be considered, this introduces the task to be completed by a subject in a study. This allows the evaluation of the auditory as a communication mechanism. The sound stimulus should trigger the appropriate actions, which aligns with the respective events. This is a general methodology, which is suitable for the construction of an auditory display based the literature reviewed and limitations of the study described in this thesis.

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The existing literature in in aural communication does not provide a concrete methodology in the construction of auditory displays. The work of Patterson, (1982) and Edworthy, (1991) is extensive in providing the concept of a pulse, onset, offset, and the remaining parameter associated with alarm construction. However, none of the literature provides a firm mathematical model. In attempting to reconstruct this study, there were several methodologies that were available for the burst creation, but the lack of a mathematical model did not make it possible to extend the body of knowledge associated with alarm design theory.

In future work the alarm design community must not only presents their results of the experimental studies, but also the methodology in which the stimulus was created. This will create some commonality amongst all experimenters in the quest to construct appropriately designed auditory display. Researchers perform vast experiments and embark on the successful creation of an auditory display, but amongst fellow researchers there is no way to reproduce this stimulus due to lack of quantifying the sound stimulus.

To mitigate this issue the aural communication community shall define each sound in terms of its sound quality parameters. The sound quality parameters are proven measures, which provide sensations to the human hearing system. If sound stimulus is defined in these parameters, than a common basis on the characteristics of sounds can be established.

A sound stimulus is typically quantified by the approach taken to modify the sound, and the acoustics parameters used in the modifications. In this experiment amplitude and frequency of modulation were the parameters employed to create urgency. If an alternative approach was used, such as filtering was used the characteristics of filtering and modulation would not be directly comparable. The filtering and modulation parameters would quantify

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the sounds in each case. A method that is independent of modifying the sounds is employing sound quality parameters to quantify a sound stimulus.

Any complex continuous sound can be quantified by its loudness, roughness, sharpness, and fluctuation strength. Theoretically these parameters could be mapped to the perception of urgency. This approach would provide a common basis for the stimulus in which auditory displays to be compared. The existing approach used in the construction of auditory displays, makes this comparison difficult. For example, broadband noise is a general form of an auditory signal, but can vary in its methodology used in its construction and its frequency content. A description of a sound based on its sound quality parameters can provide a description not only in terms of a sensation but also its frequency content. This makes reproducibility of a sound stimulus possible. Sound quality parameter allows one to define a sound independent of the method employed to create the stimulus.

In order to implement the sound quality parameters a new methodology must developed to implement these parameters in a real time setting. This will require an in depth analysis on how to implement this approach from a theoretical perspective. The quantity ΔL will be catalyst that drives the auditory display. In order to derive this parameter the sound quality must first be defined. In order to define these quantities a user study must be conducted for each context, event, and scenario. The urgency for each of these factors must be appropriately mapped to achieving mapping as described in Chapter 2.

A sound perceived as most urgent must be defined by four or more sound quality parameters. Each level of urgency will have a different set of sound quality parameters. Once this quantity is defined in one domain it provides a basis for the implementation of an

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auditory display. This requires an extensive amount of work in order to construct an auditory display.

The quantity ΔL is defined over 24 critical bands as described in Chapter 2; this allows the creation the same sound quality parameters with different frequency content. ΔL is an amplitude quantity; theoretically shifting the content from one band to another will not have an influence on say the quantity loudness. It will be critical to not only to document the sound quality parameters, but the quantities in each critical band.

There are practical implications in implementing these parameters, such as the interface required to implement this solution. This requires performing the calculation in reverse, adjusting the excitation level across the frequency and temporal range of the signal. Once the excitation level is defined the middle ear, and outer ear transfer function needs to be applied to reconstruct the stimulus to a prescribed ΔL. The issues in implementing this solution are the number of variations that can be applied within a respective band, so the excitation level within a band must also be known. This requires a definition of ΔL, at each respective urgency, and at each band. This work requires further investigation in order to implement. First, ΔL needs to be fully defined, there is commercial software available to calculate these quantities, but they are not suitable for this application.

The goal is to develop the mathematical relationships between task completion (Q) and the sound quality parameters Q = f (L, R, T, S) where L-loudness, R-roughness, T-tonality, S-sharpness. Based on available empirical relationships published, the proposed research will need to determine the best functional form. To begin, a model by Fastl and Zwicker Q=L(1+(C11/T)2+C2(F,R)2)1/2), (Zwicker, 2008) where C1 and C2 are experimentally determined constants. During the research, additional models will be

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developed based on the data and theoretical frameworks that are found in further literature reviews.

To develop the empirical relationships, the sound introduced as the cue must be synthesized from a base sound or set of sounds. An initial experiment can determine if typical background sounds of an aircraft or other sounds are most effective to trigger an action on a task. The synthesis process on the selected sounds will begin with decomposing the sound into key spectral components used to control the Loudness, Roughness, Tonality, and Sharpness so that these quantities can be controlled. By exposing the subjects in a controlled experiment to the re-synthesized sound the human perception, which is manifested as task completion will be correlated to the spectral modifications (Marshall, 2006) . This re-synthesis will allow the extraction of the spectral components, which only contribute to the task completion . This can be achieved by a employing a short time Fourier transforms.

Once correlation of the spectral modifications to task completion is established, then a user effectiveness experiment will be performed. The scenario will initiate the task via a visual cue, only using the visual channel, serving as the experimental control. The second scenario will then be introduced to initiate a task, with sound serving as a cue, a dual channel scenario, serving as the experimental treatment. For each scenario, the response variable, task completion, will be measured, this measure will be correlated to the specific task, and the sound employed. The approach for this this experiment is a randomized block design, with a multivariate regression analysis of the sound quality parameters serving as the factors. As a final experiment, the relationships developed will be applied to a different scenario to provide an indication of the applicability of the specific relationships to other applications or if these relationships are application specific.

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Sound will be used to provide two pieces of information. The first is the health of the overall system; this status will be relayed to a user through sound. A change in the sound quality parameters will correlate to a change in the health of the overall system. Each parameter can be mapped variable of the virtual environment. In the Battle Space application for example loudness can be mapped to fuel level, roughness to proximity from enemy or threat level, etc. As these parameters dynamically change in the virtual environment, the sound quality parameters will change. These changes will represent the health of a system. The health of the system is used to make decisions regarding the planning of the UAV.

If this change is significant the health status will trigger a task, in which the sound is communicating that an action is necessary by the user to return to the healthy state. The goal is to operate the system under the alarm level, which for consideration is when a failure mode occurs. A failure mode is any error that exists which prevents the system from completing its intended mission. Reaching an alarm level indicates that either the user did not perform the appropriate action in time or the sound failed to trigger a task. A key objective is to ensure their distinguish-ability between the sounds in the continuous monitoring and task triggering.

A failure mode is any error that exists which prevents the system from completing its intended mission. Reaching an alarm level indicates that either the user did not perform the appropriate action in time or the sound failed to trigger a task. A key objective is to ensure their distinguish-ability between the sounds in the continuous monitoring and task triggering. The continuous monitoring system can be applied to the concept of urgency that has been discussed thus far.

The continuous monitoring is defined as the real time continuous feedback provided to a user that relays the status of the system of interest. The real time dynamics of the

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continuous system is analogous to the concept of urgency discussed thus far. The continuous monitoring sound will serve as the stimuli that when changed will be used to trigger the task. Triggering a task is considered when the threshold of urgency is exceeded from a monitoring passive task to an action. The continuous monitoring sound will always be present, and vary according to the input values of the signal. Once the continuous monitoring system has changed significantly the change will trigger the user to initiate a task.

In considering monitoring the auditory display, the workload associated with the task being performed, the distribution of this task across the modalities must be fully evaluated. Virtual environments contain a high degree of visual information; sound does not have enough dimensions to accurately offload all the pertinent visual data. It is not this researcher‘s opinion that the solely offloading visual information to sound will be effective if employed alone as virtual environments evolve. The best approach is to seek other modalities to exploit such as the sense of smell, taste, and touch. Development in haptic makes the sense of possible in virtual environment, but smell has yet been explored. To provide a balance in cognitive load additional modalities will need to be exploited.

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