TOWARDS MEASURING AND IMPROVING HUMAN SOUND LOCALIZATION AND PHYSICAL RESPONSE TO PERCEIVED SOUND THROUGH AUDITORY CONDITIONING

Melvin Luis D. Mendoza
Ateneo de Manila University
Katipunan Ave., Loyola Heights,
Quezon City, Philippines
(+632) 4266071
melvin.mendoza@obf.ateneo.edu

ABSTRACT
This study aims to develop a tool that measures how accurate humans can localize sounds as well as how fast they can visually locate sound sources (which this paper will refer to as "response speed"). It also aims to provide a training segment along with the tool that could help improve reaction times and localization accuracy. Accuracy and speed baselines will be measured by the tool, which features automated and manual production of sounds from virtually different sound sources through a graphical user interface. These sounds will then be delivered to the users through a pair of headphones running in virtual surround sound, while the visuals will be provided through a head-mounted display (HMD). Test subjects will then be asked to go through training provided by the tool that should improve their localization accuracy as well as reaction speed. Another set of measurements will then be taken after training, which will be compared to the baselines to see whether significant changes in the users' speed and accuracy in terms of localization are present or not.

CCS Concepts
• **Applied Computing →Life and Medical Sciences→Consumer Health**

Keywords
Localization; Response Time and Accuracy; Audio-visual cues

1. INTRODUCTION
1.1. Context of the Study
Sound has been an invaluable aspect in the evolution of humans. Several hunter-gatherer tribes from Africa used clicking noises made by the tongue and the roof of the mouth to communicate with each other for successful hunts 8. Animals also communicate through vocalization for many tasks, including mating rituals, warning calls, social learning, and others. In many species, males perform calls during mating rituals to express dominance against competition and to signal females 17. Sound is also essential for day to day communication, as most people rely on sound to communicate themselves and be understood by others.

The ability to accurately determine the location of sound sources, also known as sound localization, is equally critical to the survival of many species. Sound localization aids in everyday tasks such as responding to calls, driving, or even simple tasks such as focusing on a specific sound or voice in a noisy area, known as the "cocktail party effect" 7. Children, for example, need this skill in order to listen to their teachers as well as socialize with their peers, which is critical to their development. Children who cannot localize sound are at higher risk for academic, speech-language, and social-emotional difficulties than their normal-hearing peers 11.

Being able to respond to these sounds and visually locating their sources is equally important, especially in situations where quick reflexes are critical. For example, a driver must hear, localize, and see an incoming car in order to avoid it. This is also essential for unexpected circumstances in places like construction sites or zoos, as well as for people who deal with safety such as policemen and firemen – they can only respond to these critical situations if they are first able to hear, localize, and see the possible dangers in these types of situations.

1.2 Research Objectives
The primary objective of this study is to provide a tool that can measure how accurate humans can localize sound as well as their response speed. The tool should be able to improve these abilities through auditory and visual conditioning through a virtual simulation.

The tool has been developed and is at its final state after testing on numerous test subjects. The ability of the program to measure and improve accuracy and speed, however, has yet to be tested, as participants for the experiment have not been selected yet.

1.3 Research Questions

This study aims to answer the following questions:

1. How can sound localization accuracy be measured? How about the speed at which humans visually locate sound sources?
2. How can a real-world environment where perceivable sounds can come from anywhere be accurately simulated?
3. Which specific tools should be used for the auditory aspect of the study? What about the visual aspect?
4. What can be done to improve human sound localization and reflex?

1.4 Scope and Limitations

This study will focus on the ability of humans in localizing sound sources as well as their response speed. Only participants with normal hearing and normal vision are to be invited as test subjects, as these two factors directly affect sound localization. Blind people are more accurate in localizing sound compared to sighted people in most cases 5, and people with hearing impairments are at a handicap compared to people with normal hearing. People who suffer from spatial hearing loss, which is the inability to track conversations, focus on specific sounds, and tell where sounds come from, especially in noisy environments 2, and unilateral hearing loss, where there is normal hearing in one ear and impaired hearing in the other, have less accurate sound localization abilities compared to people without hearing impairments 1. It should also be noted that having normal hearing does not equate to having accurate sound localization, as proven by Menezes et al 12. Participants of the study will also be selected by age. According to a study conducted by Dobreva et al, younger people have better sound localization than older people in most sound frequency ranges 4. The study will also utilize a specific range of sound frequencies for the program. Some sounds are more easily perceived by humans than others, and using the sound frequencies that are easiest to hear will be helpful in producing more accurate results.

To make the environment more realistic, 25 sound sources will be utilized by the tool, which will be described in detail in the methodology.

Visual elements shown through the Oculus Rift will also be simplified so that they do not introduce any distractions or elements that could affect the users' ability to detect and localize sounds. This will also be discussed further in the methodology.

The program will provide the following core functions:

1. Simulate a 3D environment with visual stimuli shown through an HMD and auditory stimuli through a pair of headphones running in virtual surround sound
2. Measure participants' sound localization accuracy
3. Measure participants' response speed
4. Improve the users' localization abilities along with their visual reflexes through auditory and visual conditioning
5. Provide automated and manual sound-production from virtually different sound sources

1.5 Significance of the Study

This study will be beneficial to people who have impaired reflexes to perceived sounds, and to people who have normal hearing but have difficulty in localizing sounds and/or visually locating sound sources. This skill is critical to people whose jobs require accurate sound localization such as policemen, construction workers, firemen, and many others. This is also useful for the average person, as sound localization plays an important role in our everyday lives – in driving, keeping track of conversations, crossing the road, etc.

2. REVIEW OF RELATED LITERATURE

2.1 Sound – Human Perception and Localization

The human ear has three major parts, namely the outer ear, the middle ear, and the inner ear, all of which play a distinct role in the human hearing process.

Sound is the reception of vibrations, also known as sound waves, which propagate through a medium such as air or water and their perception by the brain. Sound waves enter the ear canal, causing the eardrum to vibrate. These vibrations then transfer to a chain of bones called the ossicles located in the middle ear. The last bone in this chain then knocks on the membrane window of the cochlea, causing the fluids inside of it move, which then triggers a response in the hearing nerve 9.

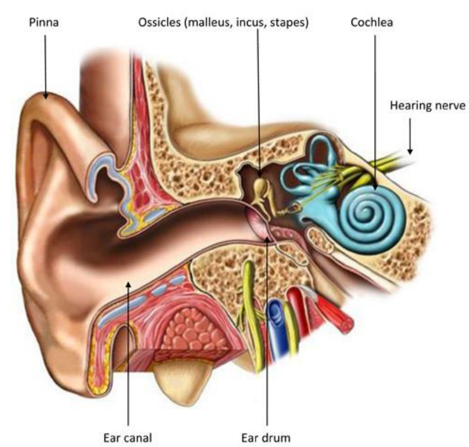

Figure 2.1 Cross-section of the Human Ear 9.

Unless the sound source is equidistant to both of the ears of the listener's head (i.e. directly in front of or behind the listener), sound arrives earlier to the ear that is closer to the sound source, which causes the *interaural time difference* (ITD). The sound also arrives with greater intensity to the ear nearer to the sound source than the other ear because that ear is slightly "shadowed" by the head, preventing some of the sound energy from reaching that ear. This causes the *interaural intensity difference* (IID). These differences in time and intensity are critical in determining sound sources, as they provide physical information that the brain interprets to localize the sound sources 18.

Sounds come in varying frequencies which are measured in Hertz, the higher frequency sounds being more "high-pitched" 14. Although humans are known to hear frequencies ranging from 20 Hz to 20 KHz, with the upper limit decreasing as age increases

13, human ears are most sensitive to frequencies between 2 kHz and 5 kHz 6, and that narrowband sounds are more difficult to localize 15. This means that the tool should only use frequencies within this range, since the study is focused on measuring and training localization abilities, not analyzing localization in different frequencies. This would produce the best-case scenario for the testing environment and eliminate another confounding variable.

Sounds can also come in different intensities which are measured in decibels (dB). The following table shows the dB ranges that categorize the human hearing thresholds in relation to hearing loss. The study will only involve participants with normal hearing, and this information will help determine whether participants are qualified or not.

Table 2.1 Hearing Thresholds in Humans 10

Hearing Threshold	Interpretation
0-25 dB	Hearing within normal limits
26-50 dB – Mild Hearing Loss	Has difficulty with soft sounds, background noise, and when at a distance from the sound source
51-70 dB – Moderate Hearing Loss	Has significant difficulties with normal conversational level speech and relies on visual cues
71-90 dB – Severe Hearing Loss	Cannot hear conversational speech and misses all speech sounds Can hear environmental sounds, such as dogs barking and loud music
91+ dB – Profound Hearing Loss	Hears only loud environmental sounds, such as jackhammers, airplane engines, and firecrackers

Given this information, the tool will only use sounds which are neither damaging to the ear nor too soft or nearly inaudible.

2.2 Processes that Help Improve Human Sound Localization

MED-EL, a company focused on human hearing solutions such as hearing aids and cochlear implants, provided six procedures to improve sound localization in humans even without the use of their products. This study will utilize the techniques provided by the company which will all be simulated through the speaker array and the HMD.

1. Recognize and localize a known sound that occurs in a known location (like the telephone at one's house, where it normally sits)
2. Recognize and localize a known sound from an unknown location (like the telephone in one's house, except moved somewhere new)
3. Recognize and localize an unknown sound from an unknown location (have someone else pick out what, and where, the noise comes from)
4. Recognize and localize a known sound from a known location with background noise
5. Recognize and localize an unknown sound from an unknown spot with background noise
6. Track sounds as it moves between different locations

While the author mentioned that some people may take months to make even the slightest progress with everyday practice, the improvement all depends on their hearing capabilities. These localization improvement techniques will be used in the tool that will be developed for the study.

2.3 Sound Localization Training

A similar study by Philbert Bangayan, V. Sundareswaran, Kenneth Wang, Clement Tam, and Pavel Zahorik was conducted to evaluate the efficacy of a sound localization training procedure.

Their study utilized studio-grade stereo headphones (Sennheiser HD265) paired with a 3D sound card (Turtle Beach Montego II A3D, with Aureal Vortex2 chipset) in an Intel-based computer for their audio setup. This was then combined with an HMD (Sony Glasstron PLM-A35) with a field of view (FOV) of approximately 30^o along the horizontal plane and 23^o along the vertical plane. Head movement was then tracked using an ultra-sonic 6-degree-of-freedom position/orientation sensor (Logitech, Inc.), which was accurate to 0.1^o in orientation angle. The virtual space was spherical with the participant's head as the center and a 1.5m radius. The partial sphere included a full 360^o of azimuth (horizontal space) and ±40^o of elevation relative to the ear level.

Three types of visual stimuli were used, and all were presented through the HMD on a uniform black background. The first stimulus was a "crosshair" that was present to the listener at all times. This was used as a marker for the vector that pointed straight ahead from the listener's head, and as a result, was not updated with respect to changes in the listener's head orientation. The second visual stimulus provided the listener feedback as to the correct sound-source location. This stimulus was a small point of light with high contrast, also presented via the HMD. This was always paired with the auditory stimulus, repeated at a rate of 1 Hz. This stimulus was updated with respect to the listener's head orientation. The third stimulus indicated a reference location directly in front of the listener (0^o azimuth and 0^o elevation). This stimulus was also a point of light with high contrast presented through the HMD and was updated to reflect changes in listeners' head orientation. Unlike the second stimulus, however, this point of light was illuminated at all times. The second and third stimuli were rendered in green and blue respectively. Due to the compact size of the HMD, additional spatial reference information may have allowed listeners a partial (peripheral) view of the testing environment, which remained illuminated at all times.

The participants, separated into control and test groups, were asked to go through 30-minute sessions of training in which they were tasked to locate sound sources through sounds produced through the headphones and the visual feedback from the HMD. Measurement of accuracy was done prior to and after training for two different data sets to be compared.

The researchers proved that the training procedure used in their study improved the participants' accuracy. The number of front-back reversals the participants made during testing were reduced, and in turn, improving the participants' localization, especially when compared to the control group. According to the researchers, this improvement even lasted for up to four months after training 19.

2.4 Effects of Occupational Noise on Sound Localization

Another study by Menezes et al. was conducted to determine the effects of occupational noise on sound localization in different spatial planes and sound frequencies among firefighters with normal hearing.

29 adults with pure-tone hearing thresholds below 25 dB (no hearing loss 10) took part in the study, and were divided into two groups – 19 firefighters, 15 being male, exposed to occupational noise and a control group of 10 adults who were not exposed to such noise.

The setup consisted of 13 speakers representing 13 different sound sources from the coronal, sagittal, and transverse anatomical planes. All of the speakers were 1m away from the subject being tested. All subjects were tasked to indicate the origin of the sounds sent to them using push buttons that corresponded to each of the 13 speakers presented by the researcher in a 3m x 3m reverberating room. The sounds used were square waves with fundamental frequencies of 500Hz, 2000Hz, and 4000Hz presented at 70dB and were randomly repeated three times from each speaker (sound source) for a total of 117 stimuli.

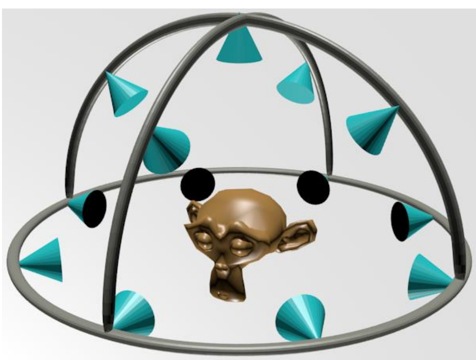

Figure 2.2: An Illustration of the Research Setup 12

After gathering results, the researchers concluded that the control group performed better than the firefighters in both the test in relation to the anatomical planes and the test in relation to different frequencies. They found that localization in the horizontal plane (transverse) was most efficient and accurate, followed by the frontal and sagittal planes, where the subjects performed equally well. The researchers, however, indicated that although performance on the sagittal and coronal planes were similar, the results were not statistically identical to each other after an ANOVA coupled with a Tukey's test, and were only equal with regards to the mean. The study also found that although the subjects were most accurate in localizing the 2 kHz sounds, this was not statistically significant after the same ANOVA and Tukey's test.

The study concluded that a) occupational noise, although not affecting hearing thresholds or causing hearing impairment, can affect the ability of a person to localize sound, b) localization is most efficient in the transverse plane, and although localization in the coronal and sagittal planes were similar, they are not necessarily identical, and c) there was a similarity among accuracies in the .5 kHz, 2 kHz, and 4 kHz tests, frequencies that humans hear the easiest 6.

2.5 Sound Localization and Gender

In 2011, Ida Zündorf from the Center of Neurology at Tübingen University, together with Prof. Hans-Otto Karnath and Dr. Jörg Lewald examined the audio-spatial abilities in men and women through a sound localization task. Participants of the experiment were asked to listen to sounds and determine their source either by pointing to it or naming the exact position (e.g. 45 degrees to the left). All of the participants, none of whom suffered from any hearing disorder, accurately determined all of the sound sources in this test.

The participants were then asked to do the same task, but this time, there was noticeable ambient noise. This is known as the "cocktail party effect," the human ability to localize and focus on sounds in a noisy environment 7. Women found this task more difficult compared to men who were able to locate sources more accurately. There were even cases where some women participants thought the sounds were coming from the opposite direction.

These results indicate that men are better than women in auditory-spatial tasks, albeit only in the cocktail party situation (i.e. women performed equally well in the first test where there was no ambient noise). According to the study, these audio-spatial abilities among men have developed as the result of natural and sexual selection through human evolution 20.

3. METHODOLOGY

The goal of this study is to provide a tool that will measure how accurate humans can localize sound along with their response speed. This will be done through the use of an HMD and a pair of headphones. The tool should also be able to improve these abilities through a short training program. The study will begin by selecting qualified test subjects, followed by a pretest to establish a baseline of their localization abilities (both speed and accuracy). Participants will then go through the training program. A posttest will then be conducted to see if there are significant changes in the participants' performance.

3.1 Participant Selection

If possible, 30 participants will be selected for testing in order to make the study statistically significant. Otherwise, at least five participants will be required to test the usability of the program. Prior to the pretest, Rinne and Weber hearing tests will be administered by the researchers to determine whether participants have significant hearing impairments. Participants of the study should satisfy all of the following criteria:

1. The participants should have normal hearing and no hearing impairments. This will be determined by the hearing test conducted before the pretest.
2. The participants should be of young age (college students will do) and have not been exposed to harmfully loud noises for extended periods of time in the past.
3. The participants should not be legally blind, and are visually capable.

Rinne and Weber hearing tests will be used to simplify the selection of participants without compromising the reliability of the study. The test requires only a 512 Hz tuning fork, and determine whether a person has sensorineural or conductive hearing loss. These tests are also non-invasive, cause no pain, and have no risks or side-effects involved.

The Rinne test compares air-conduction hearing, which is hearing through the air near the ears, and bone-conduction hearing, which is hearing through vibrations from the tuning fork placed near the back of the ear. These two types of hearing will be timed and compared. If air-conduction time is twice as long as the bone-conduction, the person taking the Rinne test has normal hearing. If the bone-conduction, is longer than or equal to the air-conduction, the patient has conductive hearing loss, which means sound waves have difficulty travelling from the outer ear to the inner ear. On the other hand, if boneconduction is shorter-than-half of air-conduction, the patient has sensorineural hearing loss, which means there is damage to either the inner ear or the nerves from the inner ear to the brain (both, in some cases).

The Weber test will then be administered to confirm the findings from the Rinne test. If sound is heard in both ears, the person taking the test has normal hearing (the tuning fork will be placed on the middle of the forehead). If not, the person has either conductive hearing loss (sound is perceived through the poor ear determined by the Rinne test) or sensorineural hearing loss (sound is perceived through the non-impaired ear) [10].

3.2 Research Setup

3.2.1 Software

The virtual environment will consist of a camera that represents the test subjects' eyes in the program. Images that this camera captures will be shown through the HMD in real-time. A blue reticle will be shown through the HMD to aid the users in pointing out sound sources and to avoid ambiguity in keeping track of their answers during the test. This reticle will not be updated with head movement, and will always be fixed at the center of the participants' view.

There will be a 1.5m-radius sphere with its center being the camera which serves as the participants' eyes. This sphere will also be shown through the HMD. The sphere will be divided into 25 sections, each of which will serve as one sound source. Starting from the position directly in front of the user, rotating clockwise, each 45^{o} position will contain a sound source, for a total of eight sources. These will be duplicated above and below the horizontal, following the curvature of the sphere, exactly 45^{o} from the center of the sphere, for a total of 24 sound sources. Another sound source will then be positioned directly above the user as the 25th sound source (figure 3.1, 3.2). No sound source will come from the bottom of the sphere, as this will supposedly be "obstructed" by the participants' physical bodies. This sphere will be fixed in the virtual environment, and will be updated to accommodate head movement. These panels will not be visible to the user, and will only be used by the program to differentiate areas in the sphere.

Each panel will contain a marker at its center which will be visible to the user. These will serve as sound sources. The panels will also be divided accordingly. Unlike the panels, these virtual speakers will be shown to the participants. They will be marked with an image, and will be shown through the HMD. These will also be updated to accommodate head-movement, as they are fixed in the virtual world. It should be noted that the horizontal array of speakers (teal-colored in figure 3.1) will always be as high as the participants' ears in the virtual environment. In figure 3.3, a section of the setup (in development) is shown, as seen on the HMD. The red marker indicates the vector directly in front of the user, who is facing approximately 45^{o} to the left in the photo. The blue reticle is also shown in the image.

The virtual environment will be created using Blender v2.75a and Unity3d v5.2.0f3. The program will run on the Oculus Runtime v0.4.4, downloaded from the Oculus website.

Figure 3.1: The virtual environment with and without divisions

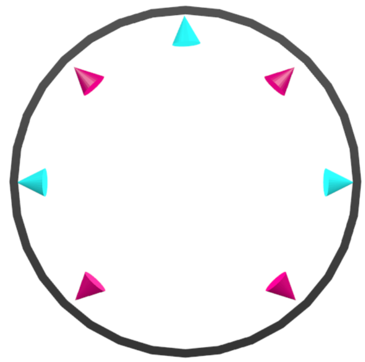

Figure 3.2: A meridian section of the setup

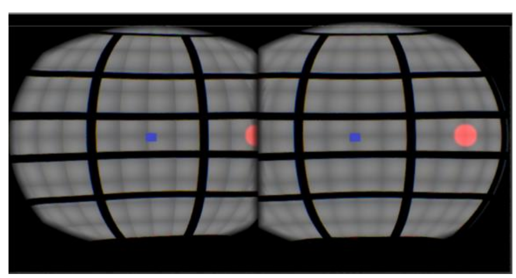

Figure 3.3: A section of the setup (in development)

After assuming all prerequisites for measurement (discussed in the hardware section), the testing program will then be executed. The software will wait for input from the tester to signal the commencement of the measuring process, which will be discussed in a section below, along with the functions implemented by the program.

3.2.2 Hardware

Participants will be asked to sit in front of a computer running the program. For visual and auditory stimuli, participants will be provided with an HMD (Oculus Rift Development Kit 1) and a pair of headphones in virtual surround sound (Beyerdynamic DT990). The headphones will be connected to an amplifier (O2 Amplifier by JDSLabs). The program will run on a desktop computer. The test will be done inside a library-quiet room without sound-proofing, with ambient noises not exceeding 40dB. Participants will be given a wireless mouse using the 2.4 GHz band to record their input during testing. They will all be asked to keep their heads at level with the horizontal for calibration of the HMD.

In summary, the setup will consist of an HMD and a pair of headphones in virtual surround sound plugged into a laptop or a desktop running the program. For visual elements, a blue "crosshair" will be fixed in the HMD to indicate the area directly in front of the user. A sphere with the participants' head as a center will be divided into 25 sections, each of which representing a sound source. A wireless mouse will also be provided to the participants and the researchers for input during the measurement.

3.3 Measurement Process – Pretest

Response speed and accuracy measurement will be done through the program and will be divided into two phases with generally similar procedures, the first being critical to the study (automated) and the second being optional (manual). At execution, the program will immediately present the virtual environment that will be used throughout the study (discussed in section 3.2.1). It will then wait for input from the researcher which signals the commencement of the measurement (that is, starting the timer and playing the first sound stimulus for the participant to detect).

A single sound stimulus will be used throughout the test. It will be an 100ms-audio clip of a sine wave at 3KHz frequency repeated every 300ms, presented at a reasonable sound level calibrated for each test subject before the test begins. This sound is in the range that easiest to hear for humans 6. No other sounds will be produced by the program aside from the auditory stimuli.

Sound stimuli will be played from all of the 25 sound sources. Each sound source will only be used once in the automated phase of the program, and will all be chosen at random. The automated phase of the program will end when all of the sound sources have been used, and the program will automatically proceed to the manual testing phase.

The participants will be asked to localize sounds produced by the program by first aligning the crosshair with the source they perceive the sound is coming from then pressing the left button on the wireless mouse on the keyboard provided to record their answer. They will only be given one chance to answer per auditory stimuli. They will not be told that each sound source will only be used once in the first phase of the test.

The wireless mouse will serve three main purposes. The first one is to record which sound source the crosshair is currently aligned with and record this response. The second is to stop the timer, record the time it took for participants to locate a particular sound, and reset the timer for the next sound source. This will be done regardless of the participant's answer, since, technically, this function is only measuring reaction time. The third function will be to play a distinct sound to let the user know that their response has been recorded, and should now reset to their neutral position (i.e. the position at the beginning of the test) so the next sound stimulus can be played.

Overall accuracy will be calculated by dividing the participants' number of correct answers by the total number of stimuli. This will be calculated and updated for each sound source during the test. The same formula will be used when computing for individual accuracy. Accuracy will also be calculated per sound source to determine whether users have trouble or excel at localizing particular sound sources.

When all of the 25 sound sources are used up, the program will enter manual mode, which can be skipped altogether at the researchers' discretion. Manual mode enables the researchers to take additional measurements should they deem it necessary. Manual mode allows the researchers to choose specific sound sources to play sounds from instead of being chosen randomly. A separate accuracy measurement will be used in manual mode.

If time permits, a second measurement test will be administered. The participants will be asked to accomplish the same task, but this time, white noise will be played in the background to imitate ambient noise in the real world. Other sounds, such as car horns and bicycle bells, can also be used in this test in lieu of the beeping sound to simulate a scenario closer to the real world.

3.4 Training and Improvement

This segment of the study will utilize the same setup and the same auditory stimuli, only this time, the sound will be moving smoothly from one sound source to another, meaning there will be no pauses during transition, and the sounds will only move to adjacent sound sources. The participants will be asked to follow the sounds with their ears (i.e. identify whether the sound is moving from left to right, bottom to top).

Sound will come from the speaker directly in front of the participant and will rotate in a clockwise direction when viewed from above. This will be repeated on the speakers below and above the participant, as shown in figure 3.4. Participants will be asked to track the sound as it travels around their heads. Visual feedback as to where the sound currently is will also be provided to the participants to help them track the sound. The participants will be asked to keep their heads fixed as much as possible, but they may do so if they find tracking the sounds difficult. The same process will be repeated in the vertical planes, also shown in figure 3.4.

After training, another set of measurements will be taken and compared with the baseline to see if there are significant changes in the participants' performance. If possible, this part of the program will also be done with and without background noise.

For the last part of the training program, participants will be asked to localize sounds, either automatically generated by the tool or manually produced by the observers. This time, the actual sound source will be shown to the participants to let them know where the actual sound sources are. This is to condition their localization, and hopefully, improve their accuracy and speed in localizing these sounds. The same sound stimulus will be used for this segment of the program, although other sounds, such as dog barks and car horns, are valid options.

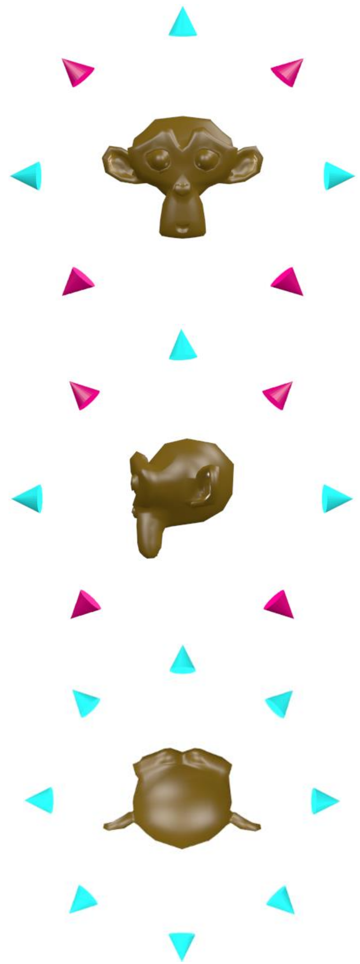

Figure 3.4: Localization Training Setup

3.5 Measuring Process – Posttest

After the training segment, another set of measurements will be taken. These will be compared with the baseline to see if there are significant changes in each participants' performance. If possible, this part of the program will also be done with and without white noise.

4. PRELIMINARY RESULTS

4.1 Program Development

After numerous revisions, the program is now almost at its final state. Further testing will be done to make any necessary changes to the program or the methodology.

The program was able to execute the complete the whole experiment, which includes taking pre- and post-measurements (automatically and manually) and saving results to a file as well as training (auditory conditioning). Visual and auditory feedback were implemented according to the specifications mentioned in the methodology.

Aside from a desktop computer, the program was also able to run flawlessly – no crashes, bugs, or sudden stops – on a laptop with a dual-core processor and an entry-level discrete graphics card, although running it on a desktop is still preferred.

4.2 Preliminary Testing

Tests have been done to determine any changes needed to be done for the program. Five different people participated in only the measurement segment, while one person was able to test until the training segment (post-measurements were not taken, as this was only preliminary testing). Proper hardware, software, and environment according to the methodology was observed during testing.

Of the five people, four were male and one was female. Of the male participants, three were college students and one was a graduate. The female participant was in graduate school. As this was only preliminary testing with the main focus being debugging and improvement rather than results, their ages did not matter just yet. The 6^{th} tester was a female high school student.

All of the testers were instructed to sit in front of the screen and use a provided keyboard for input. They were asked to look at the circle where they think the sound stimuli were coming from and press the space bar (to record their answer and time it took for them to answer) as fast as they could.

All of the testers found sitting to be difficult. One of them suggested that standing and using a wireless peripheral instead of a keyboard for reading input could have been easier and more realistic rather than being limited by sitting on a chair (which led to the use of a wireless mouse in the methodology).

Initial results reveal that participants had difficulty differentiating sound sources that were aligned vertically, which could also be present in actual testing. None of the initial participants reported any difficulty in going through the test, aside from sitting instead of standing. Accuracy of the six initial participants ranged from 30% to 50%, with most of their errors coming from vertically aligned sound sources.

5. FUTURE WORK

Training and actual testing has yet to be done. Although the tool was able to measure response time and accuracy, the question of whether it actually helps improve localization or not has not been answered yet. This requires further testing on more people, as well as data analysis.

More testing should also be done to ensure that the testing process will be easily replicable while not compromising on the quality of the data gathered.

6. ACKNOWLEDGEMENTS

I would like to thank my advisers, Dr. Ma. Mercedes Rodrigo and Dr. Andrei Coronel, and my panelist Mr. Walfrido David Diy, for guiding me and keeping me sane throughout the process of making this study. I would also like to thank my friend Alyanna for lending me her EBSCO Host account, Kyle for helping me out and listening to my frustrations, Ms. Jessica Sugay for helping me with formatting, and Floriane, for being very supportive and helping me look for the resources I needed for this study.

7. REFERENCES

1. ABEL, S. AND LAM, K., 2008. Impact of unilateral hearing loss on sound localization. Applied Acoustics, 69(9), pp.804-811.
2. CAMERON, S. AND DILLON, H., 2008. The Listening in Spatialized Noise-Sentences Test (LISN-S): Comparison to The Prototype Lisn and Results From Children With Either a Suspected (Central) Auditory Processing Disorder or a Confirmed Language Disorder. Journal of the American Academy of Audiology, 19(5), pp.377-391.
3. Deafwebsites.com, 2015. Spatial Hearing Loss.
4. DOBREVA, M., O'NEILL, W. AND PAIGE, G., 2011. Influence of aging on human sound localization. Journal of Neurophysiology, 105(5), pp.2471-2486.
5. DOUCET, M., GUILLEMOT, J., LASSONDE, M., GAGN, J., LECLERC, C. AND LEPORE, F., 2004. Blind subjects process auditory spectral cues more efficiently than sighted individuals. Exp Brain Res, 160(2), pp.194-202.
6. GELFAND, S., Essentials of audiology,
7. GROTHE, B., PECKA, M. AND MCALPINE, D., 2010. Mechanisms of Sound Localization in Mammals. Physiological Reviews, 90(3), pp.983-1012.
8. GU̇LDEMANN, T., 2007. Clicks, genetics, and "proto-world" from a linguistic perspective, Leipzig: Institut für Afrikanistik, Universität Leipzig.
9. Hearinglink.org, 2015. The Ear, The Human Ear, How Do We Hear, How Does The Ear Work, About Your Ear.
10. LALWANI, A., 2012. Current diagnosis & treatment in otolaryngology, New York: McGraw Hill Medical.
11. MCKAY, S., GRAVEL, J. AND THARPE, A., 2008. Amplification Considerations for Children With Minimal or Mild Bilateral Hearing Loss and Unilateral Hearing Loss. Trends in Amplification, 12(1), pp.43-54.
12. MENEZES, P., ANDRADE, K., CARNAUBA, A., CABRAL, F., LEAL, M. AND PEREIRA, L., 2014. Sound localization and occupational noise. Clinics, 69(2), pp.83-86.
13. OLSON, H., 1967. Music, physics and engineering. 2. ed, New York,N.Y.: Dover.
14. PILHOFER, M. AND DAY, H., 2007. Music theory for dummies, Hoboken, NJ: Wiley.
15. RAKERD, B., HARTMANN, W. AND MCCASKEY, T., 1999. Identification and localization of sound sources in the median sagittal plane. J. Acoust. Soc. Am., 106(5), p.2812.
16. SCHLEICH, P., 2014. 6 Ways to Practice Your Sound Localization | The MED-EL Blog. The MED-EL Blog.
17. SLABBEKOORN, H. AND SMITH, T., 2002. Bird song, ecology and speciation. Philosophical Transactions of the Royal Society B: Biological Sciences, 357(1420), pp.493-503.
18. WANG, D. AND BROWN, G., 2006. Computational auditory scene analysis, Hoboken, N.J.: Wiley interscience.
19. ZAHORIK, P., BANGAYAN, P., SUNDARESWARAN, V., WANG, K. AND TAM, C., 2006. Perceptual recalibration in human sound localization: Learning to remediate front-back reversals. J. Acoust. Soc. Am., 120(1), p.343.
20. ZÜNDORF, I., KARNATH, H. AND LEWALD, J., 2011. Male advantage in sound localization at cocktail parties. Cortex, 47(6), pp.741-749.