WAVELET BASED SPECTRUM HEARING TEST AND EQUALIZING HEARING AID

An Undergraduate Research Scholars Thesis

by

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RESEARCH COMPLIANCE CERTIFICATION

Research activities involving the use of human subjects, vertebrate animals, and/or biohazards must be reviewed and approved by the appropriate Texas A&M University regulatory research committee (i.e., IRB, IACUC, IBC) before the activity can commence. This requirement applies to activities conducted at Texas A&M and to activities conducted at non-Texas A&M facilities or institutions. In both cases, students are responsible for working with the relevant Texas A&M research compliance program to ensure and document that all Texas A&M compliance obligations are met before the study begins.

I, Lauren Williams, certify that all research compliance requirements related to this Undergraduate Research Scholars thesis have been addressed with my Research Faculty Advisor prior to the collection of any data used in this final thesis submission.

This project did not require approval from the Texas A&M University Research Compliance & Biosafety office.

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ABSTRACT

Wavelet Based Spectrum Hearing Test and Equalizing Hearing Aid

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This project consists of two parts: the design and programming of a test to better map frequency response of a user's hearing; and the design of a device to process the results of that test into equalized soundwaves for the user. We have designed the test using wavelets as opposed to pulses or sinusoids, under the hypothesis that wavelets (which generate fewer high frequency characteristics) will allow for a more accurate hearing test. The test is ultimately intended to automatically electronically program a device, which will also be designed in the scope of this project, that will equalize sound signals. The equalizer of the hearing aid is the only part of the hearing aid that will be designed in the scope of the project (other parts of the hearing aid model will be acquired and assembled for the purpose of demonstration, but not substantially modified by us). Furthermore, the hearing aid model will not attempt to be made ear-sized, but we will only use components that (if a higher budget or Application Specific Integrated Circuits were available) could be a small enough size. The equalizer, composed of a bank of filters, will be kept analog because analog technology is faster and smaller than digital technology for these

purposes. It will use programmable potentiometer integrated circuits to adjust the sound levels of different frequency ranges. The hearing test (which uses Morlet wavelets) has been validated according to: its spectral outputs; its comparison to existing literature; and its results when compared to other hearing tests. We were unable to use volunteers to validate the hearing test, so the only results used are my own. Due to this, the wavelet hypothesis cannot be fully researched or proven, but we expect that using wavelets in the hearing test instead of sinusoids would reduce a source of confusion among the test takers and produce moderately more accurate results, especially among test takers who do not have a trained ear. This would be because someone with musical training listening for a sinusoid tone at a particular frequency, is less likely to be distracted or confused by the tone's higher frequency characteristics and should therefore be expected to respond similarly to the sinusoid- and wavelet- based tests. The significance of this research is twofold: the wavelet-based test, if found to be effective, should replace sinusoids or pulses as the standard hearing test; and the self-tuning hearing aid would make an improvement in quality of life, should this inspire a manufacturable and marketable product. The test can potentially be programmed into an app, such that the intended user could themselves assess the frequency response of their hearing and update the tuning of their hearing aids without frequent and expensive trips to an audiologist. Such technology does not exist on the market but could radically improve quality of life for those who cannot afford uniquely tuned hearing aids.

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NOMENCLATURE

ASHA	American Speech-Language-Hearing Association
Audiology	The branch of science and medicine that concerns hearing
Audiologist	Healthcare professional who identifies and manages hearing disorders. This is typically the person who would tune a hearing aid to fit a person's particular hearing needs.
dB	Decibel, a measure of a ratio of pressure (such as sound pressure)
dBA	A-weighted decibel measurement, which describes loudness as perceived by the human ear
dB SPL	$20*\log_{10}$(RMS of sound pressure/reference sound pressure) where the reference sound pressure in air is 2 x 10-5 N/m^2, or 0.00002 Pa)
Digipot	Digital potentiometer
Filter	A type of circuit which passes signals of some frequencies and attenuates signals of other frequencies
Filter Bank	A collection of filters. In my case, when I use this term, I am referring to a collection of bandpass filters with variable gains which, when combined, will span the hearing spectrum of about 20 Hz to 20 kHz.
I2C	A serial data protocol
IC	Integrated Circuit
Masking	A reduction in the ability to detect, discriminate, or recognize one sound (the signal or target) due to the presence of another sound (the masker), measured as an increase in the detection threshold caused by the masker

PCB	Printed Circuit Board
Q value	A measurement of a filter which contributes to the steepness of its slope
Threshold in Quiet	The quietest sound level that is still audible; varies person to person as a function of frequency
Wavelet	A type of waveform which has an amplitude beginning at zero, increases, and then decreases back to zero

1. INTRODUCTION

1.1 Introduction to the project

Hearing aids are necessary medical equipment for many people, especially the elderly as our natural hearing systems take on a lot of damage over time. [1, Fig 2.1] shows a map of the hearing area (the space between threshold of pain and threshold in quiet), as sound pressure over frequency. The threshold in quiet represents the quietest noise that can still just barely be heard, and is heavily dependent on the frequency of that noise. The solid curve near the bottom represents the threshold in quiet of healthy hearing among young subjects. The dotted line shows a deviation of the threshold in quiet for young people who self-reported frequent listening to loud

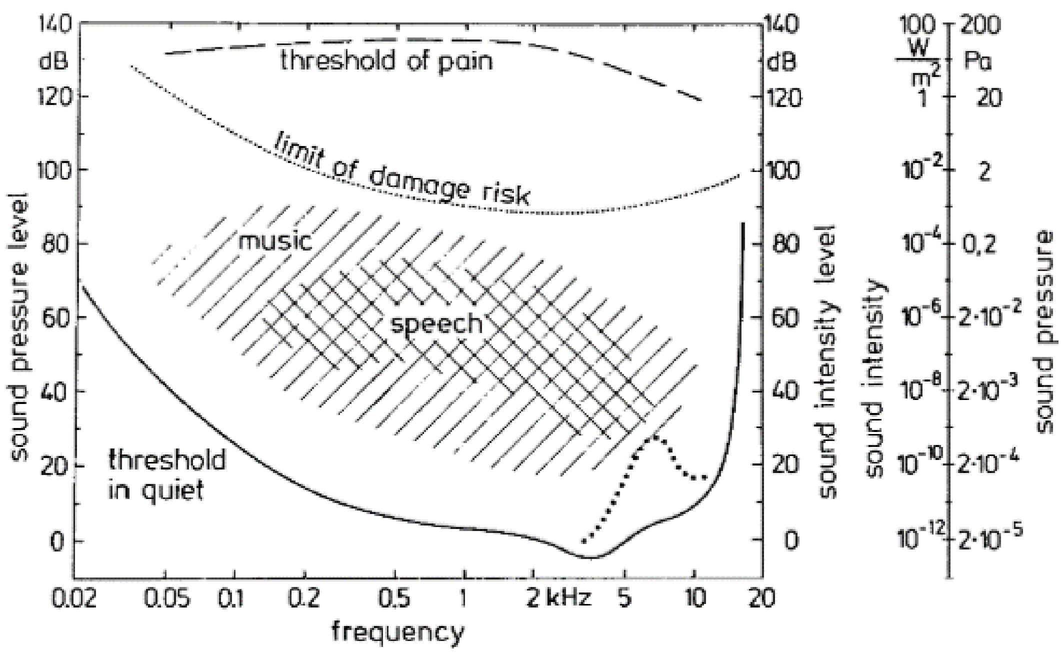

Fig 1.1: Hearing Area
Source:[1]

music. This is an excellent example of high frequency hearing loss, which is the most common form of hearing loss. Exposure to individual loud noises, as well as loud ambient noise over time, can both be damaging to our hearing. Hearing damage can vary dramatically as a function of frequency, which is why it is important to have hearing aids that correct differently across different frequency ranges (such hearing aids would be referred to as equalizing) rather than simply amplifying all sounds.

An individual's hearing area is far from constant over time. In fact, it changes frequently. And even if an individual's loss of hearing isn't significant enough to necessitate hearing aids for normal conversation, there is still aesthetic value in intaking audio (such as music) at its intended equalized values across the frequency spectrum. Currently, people who have their hearing aids tuned specifically to match their hearing area are expected to go in to a professional audiologist when they first get the hearing aids. And usually, those hearing aids will not be tuned again due to the inconvenience and expense of getting them professionally retuned. On the other hand, a few hearing aids exist on the market (such as from Bose) that are advertised as self-fitting. These supposedly allow the consumer to skip out on some audiologist appointments, but the hearing aids don't actually assess the frequency response of the user's ear and tune themselves accordingly. Rather, they have a manually controlled adjustment bar for treble and for bass. The user has only those two frequency ranges to work with, and has to decide themselves how much gain it should need (without having a reference for what equalized or normal should be.) My project proposes an improvement to this situation by giving the user the power to assess the frequency response of their own ears and having hearing aids that will automatically adjust to them. That is, self-tuning hearing aids as opposed to the commercial product that is "self-fitting".

This has the benefit of an audiologist's tuning without the cost and inconvenience of an in-person audiologist.

There is also some room for improvement in the hearing test itself. Currently the standard for hearing tests is to use pure (sinusoidal) tones. A problem with hearing tests has to do with the high frequency noise associated with an abrupt on and off transition of a tone. The person whose hearing is being tested can be tricked into thinking they hear a signal because they can notice that high frequency noise, even if the fundamental frequency of interest is in fact below their threshold of quiet. This false positive is something that a wavelet tone should correct, as the smooth transition from off to on in the time domain should eradicate the high frequency characteristics in the spectrum. This hypothesis will be explored over the course of this project.

1.2 Background Information

The basics of pure tone audiometry can be found in any clinical audiology handbook or reference. Some of the ones I've referred to include [1], [2], [3], and [4].

A pure tone hearing test uses pure tones (sinusoids) at set frequencies (usually 1 kHz, 2 kHz, 4 kHz, 6 kHz, 8 kHz, 500 Hz, and 250 Hz) to determine the lowest sound pressure required for the user to register it as sound at each of the tested frequencies (threshold in quiet, sometimes just referred to as the threshold). A tone is first played at a loudness that should be clearly audible (this is called familiarization), and from there the tone is decreased by 10 dB every time the tone is determined audible, or increased by 5 dB every time the tone is determined inaudible. The threshold is defined as "the lowest decibel hearing level at which responses occur in at least one half of a series of ascending trials" [4, p.7].

An audiogram is a typical way to present the results of a pure tone hearing test. An example of an audiogram is shown below, which is taken from [2]. The "O"s and "X"s represent different ears. This is a sound level (dB) versus frequency plot, just like Fig 1.1. The differences between this audiogram in Fig 1.2 and the hearing map in Fig 1.1 are: the hearing map features a continuous line to approximate the threshold in quiet, while the audiogram only shows a few discrete, measured points; and the y axis is inverted – higher decibel values are at the bottom of this audiogram to emphasize that hearing loss is shown by a higher threshold dB value. Fig 1.2 also labels the decibel ranges at which hearing loss is (roughly) denoted in ranges. The higher the threshold in quiet, the more severe the hearing loss. Note that increasing decibel levels are depicted in this plot going downward, so that worse hearing, though associated with a higher decibel level, is shown lower in the plot.

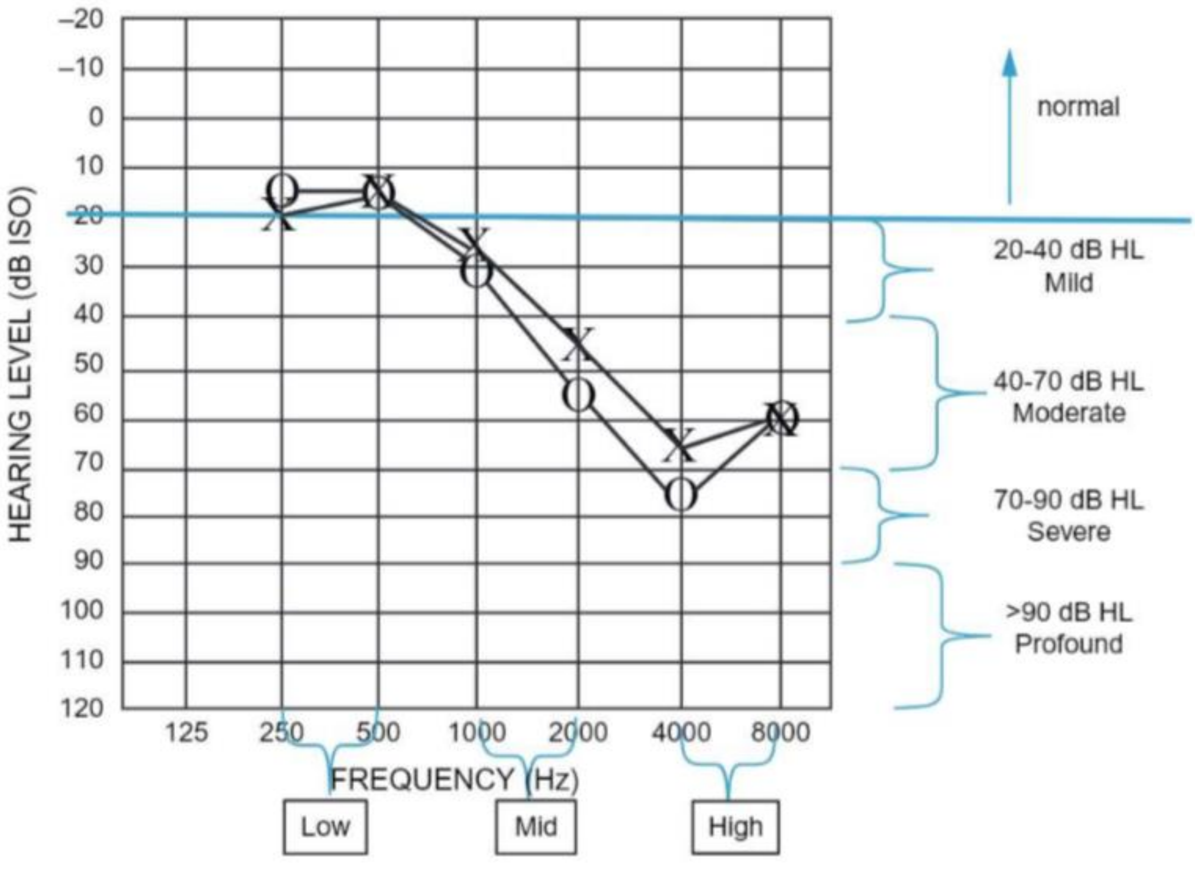

Fig 1.2: Audiogram Labelled with Degree of Hearing Loss
Source: [2]

Mild to moderate high frequency (at and above 4 kHz) hearing loss is very common, even among young people who tend to have less damaged hearing compared to older people. The equalizer I design in the course of this project focuses on correcting high frequencies, so I'd like to take this paragraph to explain the importance of that range. One aspect of high frequencies to discuss is that they play a pronounced role in characterizing instrument tones and timbres. The fundamental frequencies of musical instruments usually fall in the low to mid frequency ranges, but the reason a piano playing 440 Hz sounds different from a violin playing 440 Hz has to do with their higher frequency harmonics. Because listening to loud music is one of the most common causes of hearing damage in young people, many of the individuals who tend to suffer from hearing damage also tend to appreciate listening to music at its intended (equalized) levels. For many of these users, even if they don't need or desire the burden of every-day hearing aids, they could still derive much benefit from having an equalizer (tuned to their specific hearing's frequency response) modifying their headphones. Another, perhaps more important, aspect to consider regards speech and conversation. High frequencies are well above the fundamental frequencies of speech, and yet they are still critical in understanding speech. This is because while our vowels might exist at frequencies around 100 to 300 Hz, our consonants produce frequencies between 500 and 8000 Hz [5]. English is an example of a non-tonal language, meaning the majority of information in the language is conveyed in the consonants, as opposed to vowels. It can be seen in Fig 1.3 ([5] using data from [6]) how important different frequency ranges – especially mid to high frequencies – are in the understandability of speech in non-tonal language. Hearing aid priorities are undeniably focused on increasing the user's understanding of conversation. One of the most compelling reasons to use hearing aids is to keep mental acuity sharp and ward off dementia, which is done by engaging in conversation. Much research has been done on this particular subject, and one example is by Frank Lin, M.D., Ph.D., who conducted a study which found that mild hearing loss doubles the risk of dementia, moderate hearing loss triples the risk of dementia, and severe hearing loss increases the risk of dementia by five times [7].

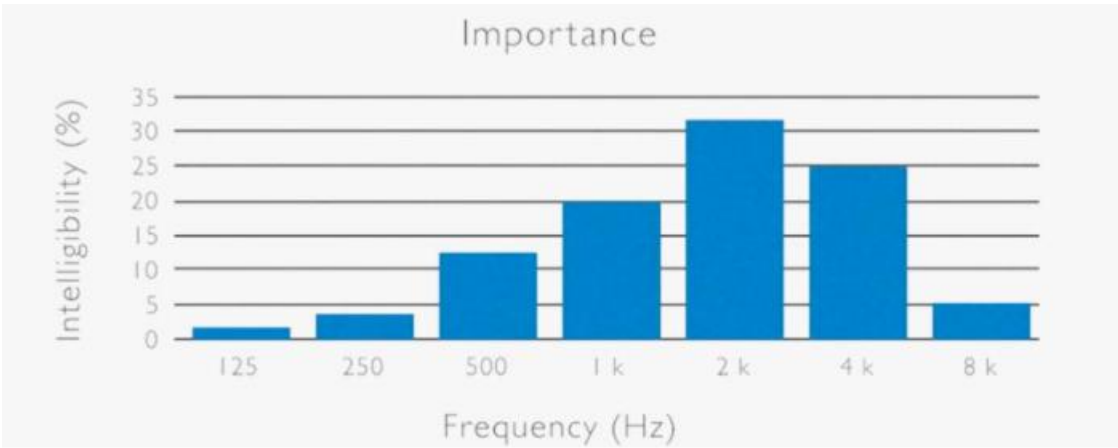

Fig 1.3: Importance of Frequency Ranges in Speech Intelligibility
Source:[5] using data from [6]

In the context of hearing aids, “equalizing” and “tuning” can almost be used interchangeably. But here I will define and explain these two concepts. Equalizing refers to the process of modifying sound waves such that frequency ranges with hearing damage are amplified in proportion to the scale of the damage. The result is referred to as an equalized sound wave. For example: if the subject has no hearing loss at 1 kHz, mild hearing loss at 2 kHz, and moderate hearing loss at 3 kHz, then the equalized sounds would not alter those around 1 kHz, would amplify those around 2 kHz by 20 - 40 dB, and would amplify those around 3 kHz by 40 - 70 dB. “Tuning” refers to the act of altering the equalizer so that it will work for a particular scenario. In most cases, this involves changing the resistance value of variable resistors in order to affect the magnitude of filters.

A filter is a type of circuit that isolates a frequency range (its bandwidth) and modifies its amplification or attenuation (gain magnitude). Equalizers are built primarily from bandpass filters, which attenuate low and high frequencies but pass a middle range of frequencies. Once each bandpass filter has isolated and modified their respective bandwidths, a summing amplifier is used to combine all of the signals to create a single signal that spans the entire frequency range. The Q value (quality factor) is a way to describe the steepness of the slope between a passed frequency and a not passed frequency. The lowest magnitude that is considered “passed” is usually minus 3 dB from the desired gain. Fig 1.4, taken from [8], shows a simple bandpass filter where the gain is 0 dB, the bandwidth is f2 - f1, the center frequency is fc, and Q = (f2-f1)/fc. Using multiple bandpass circuits to create one bandpass of a higher order is the only way to get a very high Q value.

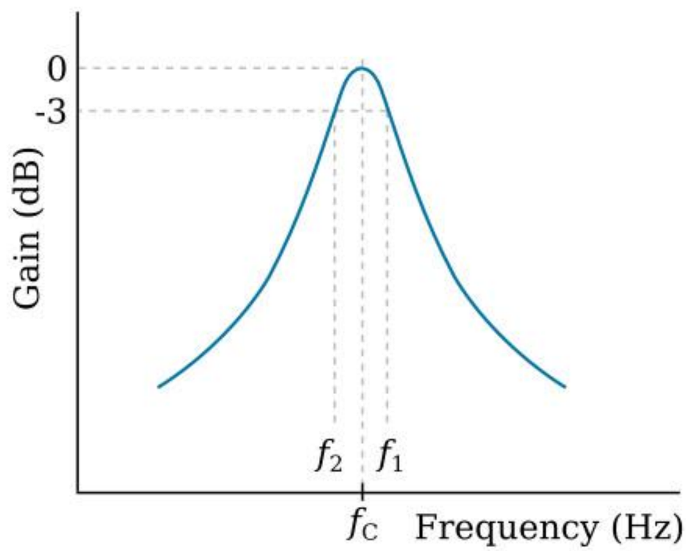

Fig 1.4: Example of a Basic Bandpass Filter
Source:[8]

1.3 Literature Review

This research paper will propose the use of wavelet tones in place of pure tones in psychoacoustical/behavioral audiometric tests (in which the subject is asked to signal when they hear a tone). Physically measuring electrical signals in the brain is an alternative often proposed (and used) by many in cases where behavioral testing is impractical or impossible. Auditory brainstem response (ABR) and auditory steady state response (ASSR) electric potentials that are already somewhat established and effective. [9] proposes Acoustic Change Complex (ACC) – a type of physical measurement with electrodes temporarily glued to skin – as an alternative to psychoacoustical/behavioral audiometric tests. Measuring and quantifying Difference Limen for Intensity (DLI) was a main priority for the researchers of [9]. Difference Limen for Intensity refers to the "just-notable difference of intensity" [10, p. 292]. In other words, DLI is the difference in sound level required for two sound levels to be noticeably difference from each other. [10] found that this is a complicated function of frequency. The generally accepted minimum for DLI is 2 to 3 dB (of a healthy-hearing, untrained listener, found repeatedly using the trusted behavioral response standard). [9] states that the minimum intensity change required to evoke ACC to be, at a minimum, +2 dB or -3 dB. The remarkably close nature of this ACC measurement to the accepted DLI value is very promising for ACC as an alternative to psychoacoustic/behavioral measurement. ACC is valuable to the field of audiology because it reflects a physical and objective quantity, while much of the field of audiology depends on psychoacoustic (subjective) qualities. It may not make a good replacement for normal hearing tests because it is more difficult to set up, monitor, and interpret. But for infants, toddlers, mentally impaired, or physically paralyzed (those who would have a difficult or perhaps impossible time with the typical test) this could be a valuable alternative "provided the individuals of the clinical population fulfill the prerequisite of the presence of Auditory Long Latency Responses" [9, p.375] (which are bioelectric responses of the thalamus and cortex activity). [11] also uses a measurable electrical signal in the brain – cortical auditory evoked potentials (CAEPs) – to determine auditory threshold in quiet. The researchers of [11] determined that "CAEP thresholds were within 10 dB HL of behavioral thresholds," [11, p.28]. This is admirable accuracy considering the required accuracy to improve quality of life for the hearing paired is minimal, and there are possible (and unmeasurable) inaccuracies present in the behavioral threshold collection. When research compares their techniques to behavioral thresholds, this is done as a comparison to the established norm, not as a comparison to necessarily perfect, true data.

It is best to conduct hearing tests in locations that are as quiet as possible in order to avoid conflict with ambient noise and problems with masking that would skew hearing test results. Normal guidelines for how quiet the testing environment should be are outlined in [12]. But in many cases (such as conducting the hearing test in a regular home instead of a very quiet room) such standards cannot be practically followed. [13] looks at the implications of conducting hearing tests in situations where the ambient noise is not negligible: on the order of 55 to 65 dB. The researchers in [13] based their study on rural villages in India where there is no easily accessible environment with ambient noise less than those levels. The hearing test frequencies that were also heavily present in the ambient noise spectrum were definitely skewed in the hearing test results, showing a higher (worse) threshold for those frequencies. [13] found that low frequencies (about 500 Hz) were the majority of ambient noise, and that the low frequency in the hearing test was the only frequency result that was significantly impacted by the noisy environment (comparing test takers in ambient noise versus quiet, ANSI standard-compliant settings). This means that, for the sake of my research, I will not worry about finding an ANSI standard-compliant quiet room, as I can account for the impact of ambient noise at low frequencies. It is also hopeful that a prospective future user of an at-home hearing test should not need to stress about finding an impossibly quiet environment, particularly in cases where the user is more concerned about mid to high frequency hearing loss.

This research will propose the use of wavelets on the basis of eradicating high frequency characteristics present in hearing tests. But for some cases, there is research to recommend the use of pulsed tones. Rather than minimize high frequency characteristics, as we intend to do, pulsing tones purposely produces frequencies other than the intended fundamental tones. As presented in [14], to test takers with tinnitus, the pulsed tone is useful to make the fundamental frequency stand out compared to the usual frequency they hear due to tinnitus. The goal of the researchers in [14] is to make the frequency under test the most obvious possible. The way they found to do this necessitates widening the frequency spectrum present during the test. While I argue that there is a marked possibility of false tone recognition in these tests due to the high frequency characteristics, the recommendations of [14] to use pulsed tones for subjects with tinnitus has become an industry standard and is probably a necessary consequence of making the hearing test process easier for those with tinnitus. This research suggests that the hearing test designed using wavelets is not practical to apply to users with abnormal hearing like tinnitus. An interesting conclusion of [15] that this paper should attempt to refute, is the recommendation of pulsed tones in hearing tests in general audiometry, including persons with normal hearing and no tinnitus. Their primary reasoning is that 67% of their subjects indicated a preference for the pulsed tones during test taking (particularly for low-level or high frequency tones), and that their results did not display a large difference in thresholds between the pulsed hearing test and the continuous sinusoid hearing test, or a larger number of false positives. Their method of identifying false positives, however, would not be expected to identify the kind of false positive I suggest would be more likely to take place. The test administrator of [15] noted a false positive if the subject indicated hearing a signal either when no signal was being played, or if the signal had already been playing for longer than a second or so. The kind of false positive I suggest (hearing the high frequency characteristic and not the fundamental tone), however, would overlap exactly with the presentation of the signal, so it would not be noticeable using this method of identifying false positives. Additionally, the test subjects in [15] were exclusively healthy of hearing. This means that the kind of false positive I suggest is not nearly as likely to happen, because high frequency characteristics are going to be lower in magnitude compared to the fundamental.

Implementing an at-home version of a hearing test has been done several times before, to varying degrees of professionalism, and will not be unique to this project. One research project with this goal in mind is [16], on which I have heavily based my own implementation. In particular, their time-efficient modified Hughson Westlake ascending method of finding thresholds was very useful.

1.4 Scope of Project

The purpose of this project is to provide research for the use of wavelets in hearing tests for audiometric diagnosis. The current standard is to use pure sinusoidal tones, which have high frequency characteristics when the tone is initially turned on. A wavelet, however, would introduce a smooth transition in the time domain when going from off to on that can hopefully reduce confusion when a user is taking a hearing test (a person who cannot hear the fundamental frequency under test can still often tell that a tone has been played because of the high frequencies present). The second, product-oriented purpose of this project is to design and produce a model of an equalizing hearing aid. The portion of a hearing aid I will design is the equalizing filter bank, which will divide the entire hearing spectrum into different frequency ranges (via bandpass filters) with electronically programmable gains (via digital potentiometers, or digipots). The relationship between these two parts is that the hearing test could be made accessible to everyone in the form of an app, and the equalizing hearing aid shall be able to tune itself (i.e., without an audiometrist's intervention) when given access to the output of the hearing test. So, the scope of this project contains two products: a hearing test program based on wavelets that shall characterize the user's hearing spectrum; and an equalizing hearing aid technology that shall be electronically programmable by the results of the aforementioned hearing test.

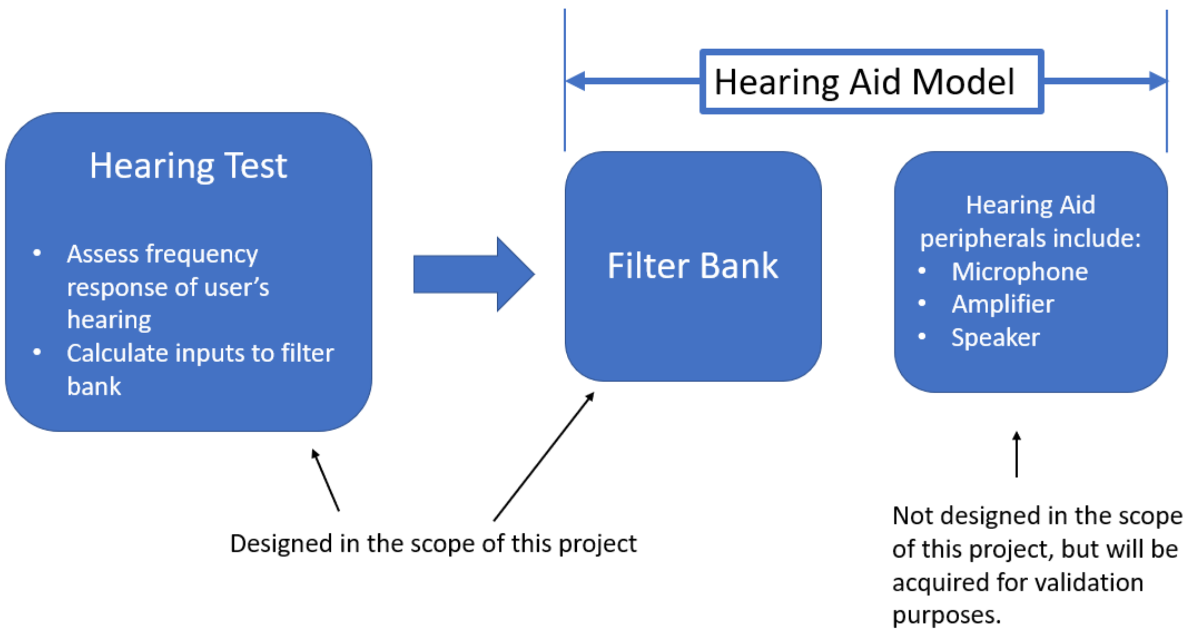

Fig 1.5: Scope of Project Diagram

The hearing test shall produce wavelet tones to present to the user of varying frequencies and volumes in order to identify audiological thresholds. The test shall accept as input binary responses from the user ("yes" and "no" in response to "have you heard a tone?"). The test will be programmed such that it could theoretically be reproduced as an app, but will not be actually made into an app for the purpose of this research project. The hearing test shall characterize the user's hearing spectrum and output digipot step values necessary for tuning the filter bank of the hearing aid system.

The filter bank shall be able to interpret the results of the hearing test in order to equalize the user's hearing spectrum by raising or lowering gain of the frequency channels. The individual bandpass filters should have Q values that are not too high (to avoid lack of coverage across some of the spectrum) but not so low as to cause noticeable conflict between channels (more than 2 to 3 dB of error would be unacceptable, as this is the required difference in sound level to be noticeable to a human ear). This filter bank may have as few as two frequency channels for demonstration purposes, but given the planned copy-and-paste nature of the filters, this could be trivial to expand. The electronic programmability of the filter bank needs to be very small in size and also very fast. It shall be shown that with a larger budget and specially tailored integrated circuits, the filter bank circuit could be small enough and fast enough to be effective in a hearing aid. This should be accomplished with exclusively analog technology, without utilizing a (large and slow) microcontroller for equalization operation. A microcontroller may be used to program the digipots, but shall be able to disconnect from the filter bank circuitry after the tuning has completed for normal operation.

It is important to mention here that this product will not include the construction of a hearing aid, but rather, the construction of what I am calling a "model of" a hearing aid. With a budget of $100, there will not even be an attempt to produce something remotely ear-sized. Instead, there will be an attempt to choose components, technologies, and concepts of operation that can be reproduced in small enough sizes, especially if Application Specific Integrated Circuits were available. This model of a hearing aid is not a hearing aid, but rather a proof of concept. It will be a tool in order to demonstrate the equalization and self-tuning functions developed in the scope of this project.

2. METHODS

Because this project consists largely of two subsystems, this section has been divided into two subsections: one for the wavelet-based hearing test research, and one for the hearing aid model design and construction.

2.1 Wavelet-Based Hearing Test

This subsystem is coded in MATLAB because of the easy manipulation of wavelets in this language. It is designed to find the threshold of the user's hearing in order to determine hearing loss. The subsystem consists of the main file, a function file, and a large library of audio files which are played over the course of the test. This subsystem has been tested for bugs, duration, and accuracy.

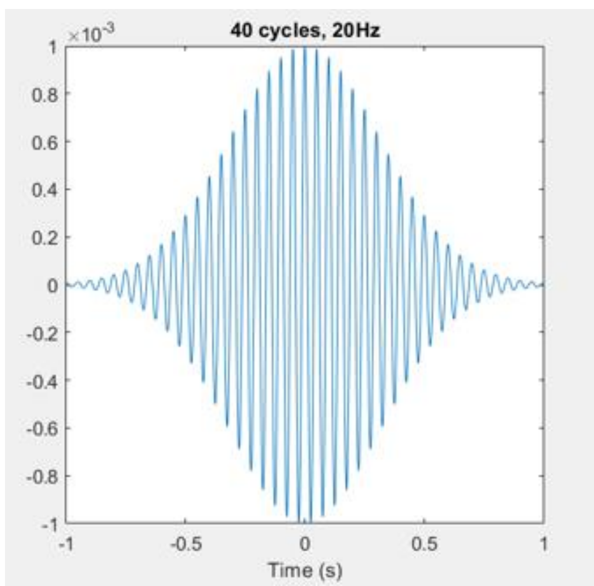

Fig 2.1: Morlet Wavelet

First, I built a collection of wavelet tone audio files of varying sound levels. The Morlet wavelet was researched and selected for this purpose. A Morlet wavelet is defined as a sinusoidal wave modified by a Gaussian. It has the shape shown in Fig 2.1. The reasoning behind selecting this type of wavelet is that it is the simplest possible modification of a standard sine wave to turn it into a wavelet. The frequency of the sine wave being modified by the Gaussian curve is equivalent to the frequency of the overall wavelet tone. This makes for simple construction of the hearing test frequency control. The frequency featured in Fig 2.1 is a very low 20 Hz, which is never actually used in the hearing test. This frequency was selected for the figure because at higher frequencies the plot begins to look like a filled in solid bounded by the Gaussian bell curve, and I wished to demonstrate how the sine wave exists inside the Gaussian. The wavelet files in the hearing test's library assume a certain speaker and a certain volume setting because this is where the calibration occurs. This means that the validation for accuracy that I have done only applies to my headset at a specific volume setting. For possible general home use (should this be rewritten as an app in the future), several methods currently exist to ensure calibration. One of these methods has the user input their model of phone, laptop, or headset so that if there is data collected on the frequency response of those items, the test can look up and factor that into the modification of the tones. This data is not always available, though, and another method I've seen involves having someone with known good hearing take the test, and comparing that to the test results of the original user. This is not a very accurate method, but to its credit, is easier than this more accurate alternative: having a sound level meter (a physical device or as a phone app) measure sound levels across the frequency spectrum output by the speaker of choice. This data could then be used to adjust volumes of tones presented by the hearing test. On the other hand, if this hearing test process is directly incorporated with the equalizing device, the calibration process can be skipped entirely. By this, I mean if the user intends to use this test in combination with the self-tuning equalizer designed in this project, then during the test the user should simply use the headphones or hearing aids they wish to have equalized for their unique hearing.

I also wrote a function focused on simply playing those tones. It is called many times by the hearing test program. This file is called playFile and it is featured below:

```
function p = playFile(freq,level,side)
   %import signal
   try %in case the file doesn't exist
      [monoSig,Fs]= audioread(strcat(freq,'_',level,'db.wav'));
   catch
      waitfor(msgbox(sprintf('Audio file for %s at %g db has not been created yet.',freq, level)))
      return
   end
      %create array of zeros
   monoZeros=zeros(size(monoSig));
   %put together the stereo signal, only left ear playing sound
   if (side=='left')
      stereoSig=[monoSig,monoZeros];
   else %meaning 'right'
      stereoSig=[monoZeros,monoSig];
   end
   sound(stereoSig, Fs) %play sound
end
```

Fig 2.2: MATLAB Code to Play File

This function first finds the desired file according to frequency and sound level. It then feeds the audio file (.wav) into either the left or right ear. I created an error catching functionality so that I would be notified when I needed to make a new file for the tone library, instead of just crashing the test.

The hearing test code is included in Appendix A. Its outline is as follows:

- Message box to the user, explaining the instructions of the hearing test.
- In the order of Left ear: 1k, 2k, 3k, 4k, 6k, 8, 1k, 500, 250 Hz and then right ear: 1k, 2k, 3k, 4k, 6k, 8, 1k, 500, 250 Hz

- Familiarization (play a clearly audible tone… increase by 20dB until acknowledged)
- Threshold Search
  - Decrease 10 dB if heard, raise 5dB if not heard
  - Threshold = heard about half of the time it has been presented
    - That has been implemented as: heard 2 out of 3 presentations or 3 out of 5 presentations
    - Alternatively, if the tone has been presented 5 times already, marked as heard all 5 times, and the tone 5 dB quieter has only been marked as not heard, then the heard tone will be declared the threshold. This is to save time and reduce frustration, and it also serves to mimic the flexibility that an in-person audiologist would have.
  - Write threshold to file
- Report Thresholds

To validate the wavelet hearing test, my options were very limited by the lack of an IRB. I used results of a spectrum analyzer when playing the wavelet tones compared to the sine waves to defend my argument, as well as my own results of the hearing tests. I compared my wavelet-based hearing test results to my results from an existing normal hearing test in order to establish that the wavelet-based results are generally accurate and not entirely off-base.

2.2 Hearing Aid Model, Design and Construction

The audio signal propagation will travel linearly through the hearing aid model, as shown in Fig 2.3. This image labels the positioning of the filter bank, the component I have designed, amongst the other peripheral components in blue boxes that I am acquiring for demonstration purposes. Not featured are the preamplifiers between microphone and filter bank and between filter bank and power amplifier. These preamplifiers are crude and simple op-amp circuits that boost the signal such that they will propagate effectively; they are not designed with sound quality in mind. These preamps are not strictly necessary, as sound will propagate without them, but they serve to increase the signal to noise ratio by amplifying the signal of interest as much as possible.

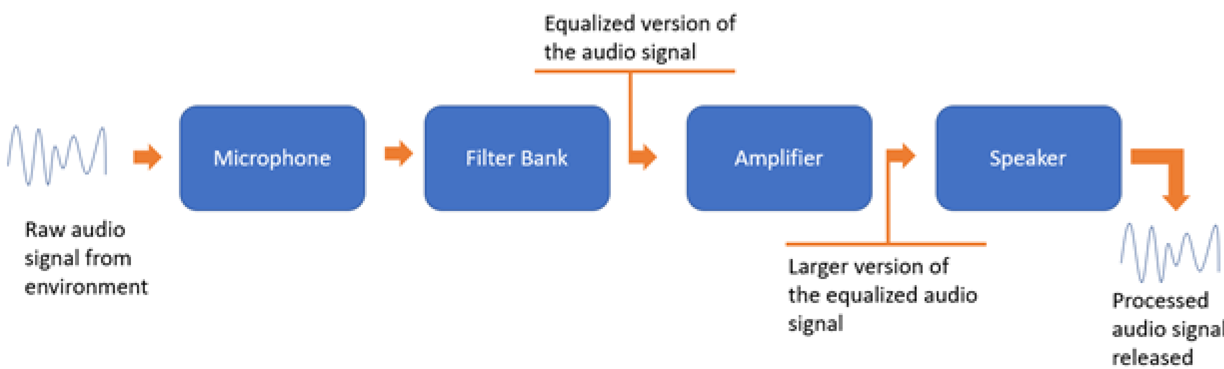

Fig 2.3: Audio Signal Propagation Diagram

The equalization of the hearing aid will be accomplished within the filter bank subsystem. By equalization, it is meant that frequency ranges an individual cannot hear as well as "normal" will be amplified appropriately in order to compensate. Every good hearing aid exhibits an equalizing functionality, allowing it to be a medically prescribed and tailored device rather than just turning the volume nob up on everything. A nice equalizing hearing aid can expect to have 20 channels or more. Due to practical construction and cost constraints, I am simply planning to demonstrate the act of equalization, for which 2 channels is sufficient. The goal of the validation of this item is to show that it equalizes sound. Because a normal human ear in a normal environment cannot discern the difference between 2 - 3 decibel sounds (at mid frequencies; higher frequencies will require higher intensity differences in order to have noticeably different loudnesses), I have allowed this much as an acceptable margin of error.

The primary goal of the filter bank was to correct for high frequency hearing loss, the most common type of hearing loss (described by Fig 1.1). Initially, I designed a 5-channel filter bank with each of the five, two-stage biquad filters dispersed between 2 kHz and 11 kHz in a manner that my research suggested best suited the curves of average high frequency hearing loss. In the following discussion, please note that the simplest bandpass filter is 1 stage and second order. The next highest complexity is 2 stages, making the bandpass filter 4^{th} order. I considered 5 channels each of 2 stages (totaling the equivalent circuitry of 10 filters) to be the max extent of my construction efforts. Unfortunately, it surpassed the extent of my budget. And after running extensive validation simulations, I found that 5 two-stage filters were no more effective at equalizing this specific curve than a single two-stage filter (equivalent circuitry of 2 filters). Now, 5 eight-stage filters (equivalent circuitry of 40 total filters) would do a fantastic job, but that is essentially what it would have taken in order to see a significant increase in accuracy of the equalization. With all of this in mind, I determined it economical to opt for a single two stage filter to equalize the high frequency hearing loss curve, along with a second, nearly identical two stage filter at lower frequencies so that my device is capable of demonstrating equalization at multiple frequency ranges.

In order to maximize efficiency of the hardware that is implemented, and considering that hearing loss is less common at low frequencies, the filter bank will pass all signals at unity gain and simply add the other amplified signals on top (via a summing amplifier). But adding the filtered signals on top of the original posed an interesting conflict of phase and some deconstructive interference presented at the border of the two signals. My solution was to run

Vin first through an all-pass before sending it on to the adding amplifier. What this does is change the shape of the summed phase around 600 Hz so that it is no longer deconstructive there. What this all-pass does not do, however, is significantly affect the relative shape of the phase anywhere else. This means that after the bandwidth of the 6 kHz filter, there is some deconstructive interference that affectively attenuates frequencies above 20kHz. This is desired! Humans cannot hear sounds above 20kHz, but those sounds can still be annoying and harmful. Fig 2.4 shows the schematics of the 3k and 6k filters (left), the all-pass (top right), as well as the summing amplifier (bottom right). Please note that the resistors shown at the input of the summing amplifier (R130 and R131) both represent 100k digital potentiometers (though they will be used as two-terminal programmable rheostats). Plots of the phase I am discussing are in Fig 2.5 (uncorrected phase) and Fig 2.6 (corrected phase). In both of those simulated plots, the two filter gains are set to 20 dB.

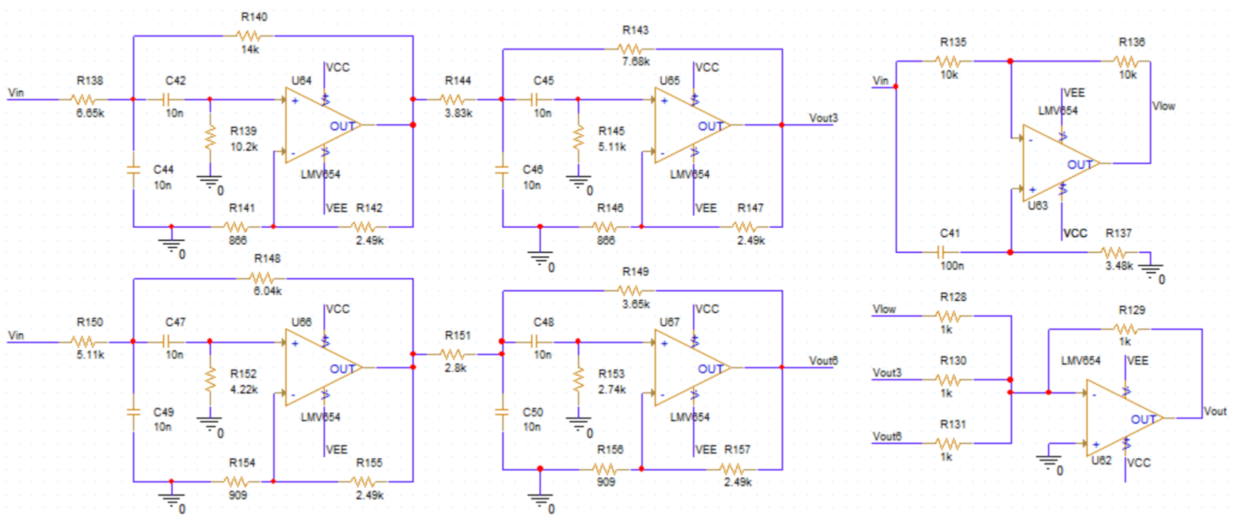

Fig 2.4: Filter Bank Schematic

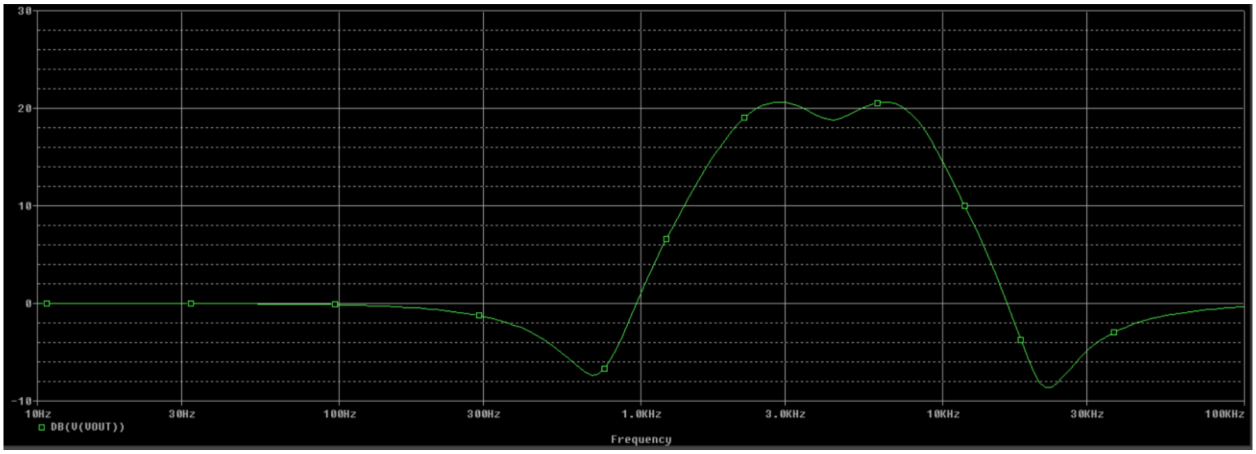

Fig 2.5: Uncorrected Phase Simulations, Channels at 20 dB

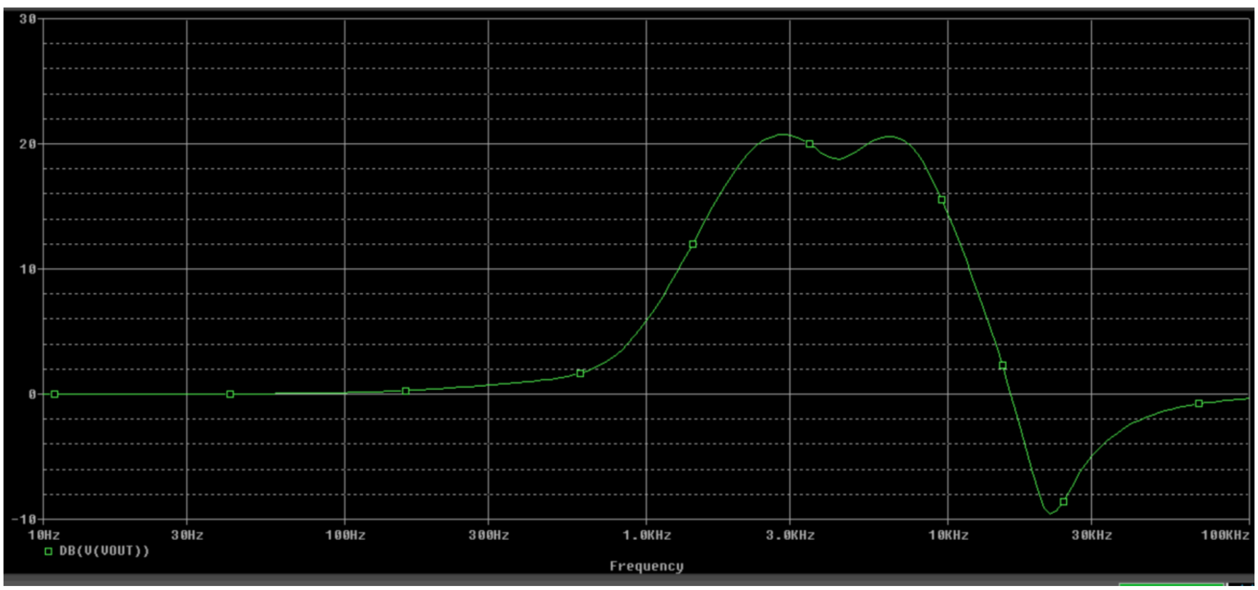

Fig 2.6: Corrected Phase Simulations, Channels at 20 dB

Fig 2.6 shows the simulated filter bank when both channels are at max gain of 20 dB. It would be trivial to raise this maximum gain (a matter of changing out 4 resistors: R138, R144, R150, and R151), but 20 dB was chosen in order to optimize the filter bank for correcting the type of hearing loss shown in Fig 1.1. The max error in the equalized frequency response is at the small dip between the two filters, which is 1.2 dB lower than the target 20 dB at 4.3 kHz. Again, a 2 to 3 dB error has been deemed acceptable since this deviation is not noticeable to a normal human ear, so a max error of 1.2 dB is reasonable. Fig 2.7 shows the simulated filter bank when both channels are off (meaning neither frequency range should be altered and we are expecting a gain of 0 dB across the spectrum). The max error is 0.758 dB at 3.2 kHz.

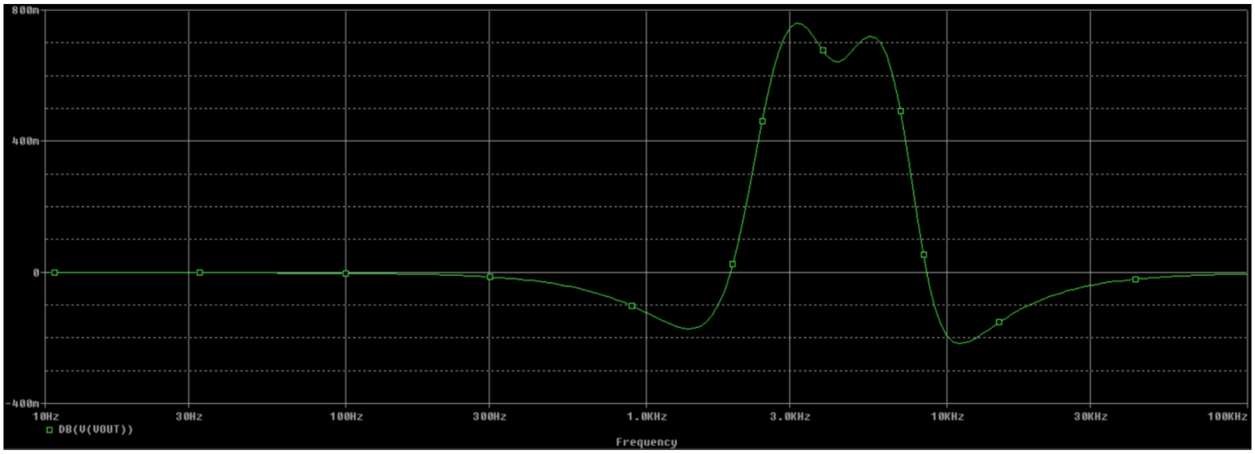

Fig 2.7: Simulated Filter Bank, Channels Off

Fig 2.8 shows the simulation for the filter bank when it has been configured to correct for the hearing loss demonstrated in Fig 1.1. It does this exceedingly well, with a max error of 3 dB at 3 kHz. Given that once the center frequency is set, nothing else about the filter can change, this error is more than acceptable. To substantially improve this error, there would need to be several more filters of a much higher order.

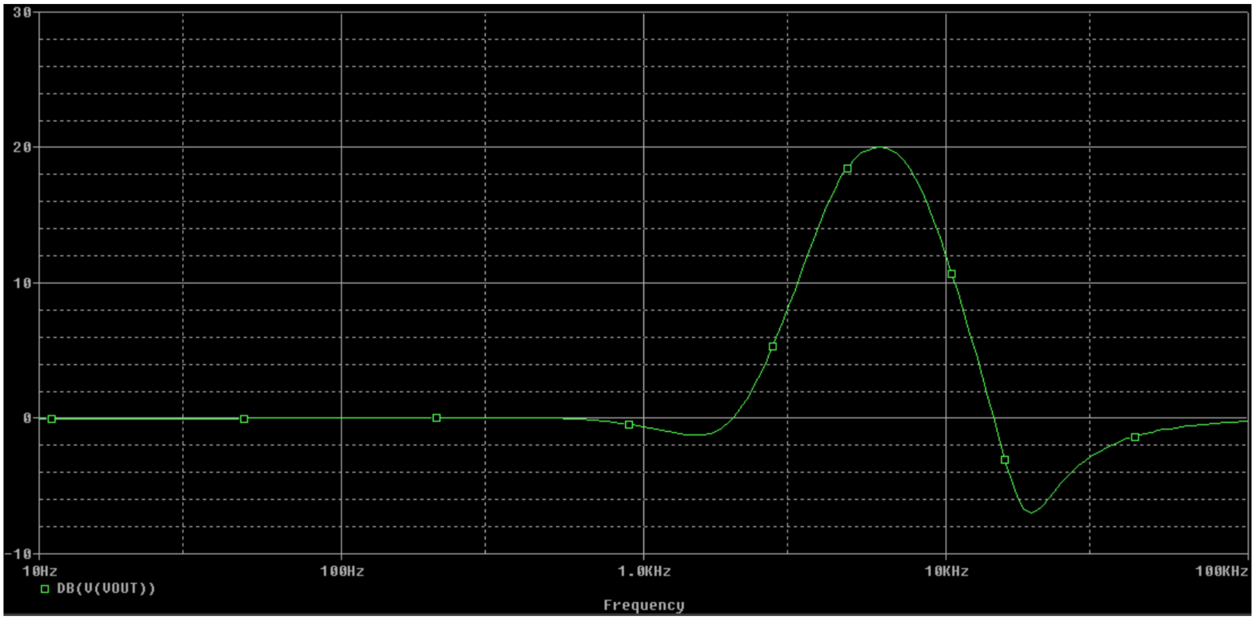

Fig 2.8: Simulated Filter Bank, Applied to real equalizing scenario

PCBs have been designed and constructed for the filter bank and digipots. The layout is shown in Fig 2.9, where U1 and U2 are quad op-amp ICs and U3 is the double potentiometer IC. All the passives are very spread out, which may not be the best PCB design practices, but it is easy to solder which is why I have laid it out like this.

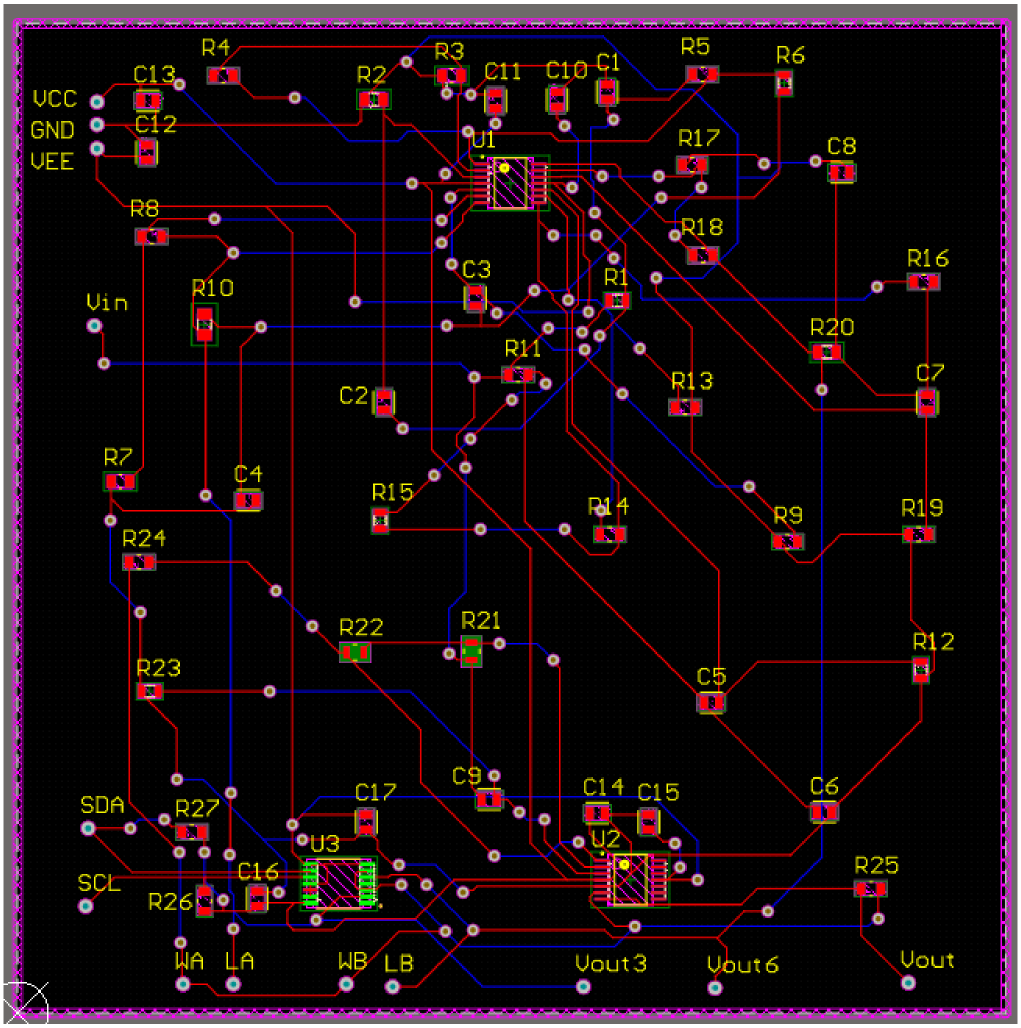

Fig 2.9: PCB Layout of the Filter Bank

The self-tuning feature will be implemented with TPL0102 digital potentiometers, being used as two-terminal variable resistors. They will be programmed with I2C serial communication and will connect to a computer command via an Arduino nano. A simple MATLAB program was written to interpret the results of the hearing test program and calculate the necessary filter gains and digipots settings. These settings can then be exported to the digipots with an Arduino code and connection. The Arduino was selected because of the ease at which one can write I2C commands using the Wire Library. The Arduino was also a cheap option to consider because I already own several. Another convenient feature of the Arduino is that it provides an easy access pin to the 5 V supply from my laptop via a USB connection. My filter bank runs on +/- 2.5 V, so with a simple circuit to split 5 V into positive and negative rails, my demonstration is good to go without needing to use batteries or a bench power supply.

3. RESULTS

3.1 Wavelet-Based Hearing Test Validation and Results

Use of a spectrogram shows visibly reduced high frequency characteristics of wavelets compared to sine waves. This is a significant item to clearly show as this is the mechanism by which my hypothesis suggests wavelets to be an effective alternative to pure tones. Please see the following figure, which uses a spectrogram comparing the wavelet tones used by the hearing test I have designed to the sine waves used in the control group hearing test. The leftmost image shows the spectrogram of the background noise of the room in which this test was conducted. It is there so that the noise present in the room can be visibly factored out of consideration when looking at the spectrograms of the tones of interest. The lighter the color, the louder the sound. Yellow is the loudest, and represents the fundamental frequency of the tone being played. In the middle image, which depicts a sinusoid at 1 kHz being turned on and off, the start and the end can be clearly noticed if looking at the whole spectrum's light red appearance, even if the yellow fundamental tone were to be ignored. The Morlet wavelet (rightmost image), however, lacks this distinct high frequency marking, even though its max fundamental intensity matches that of the sinusoid.

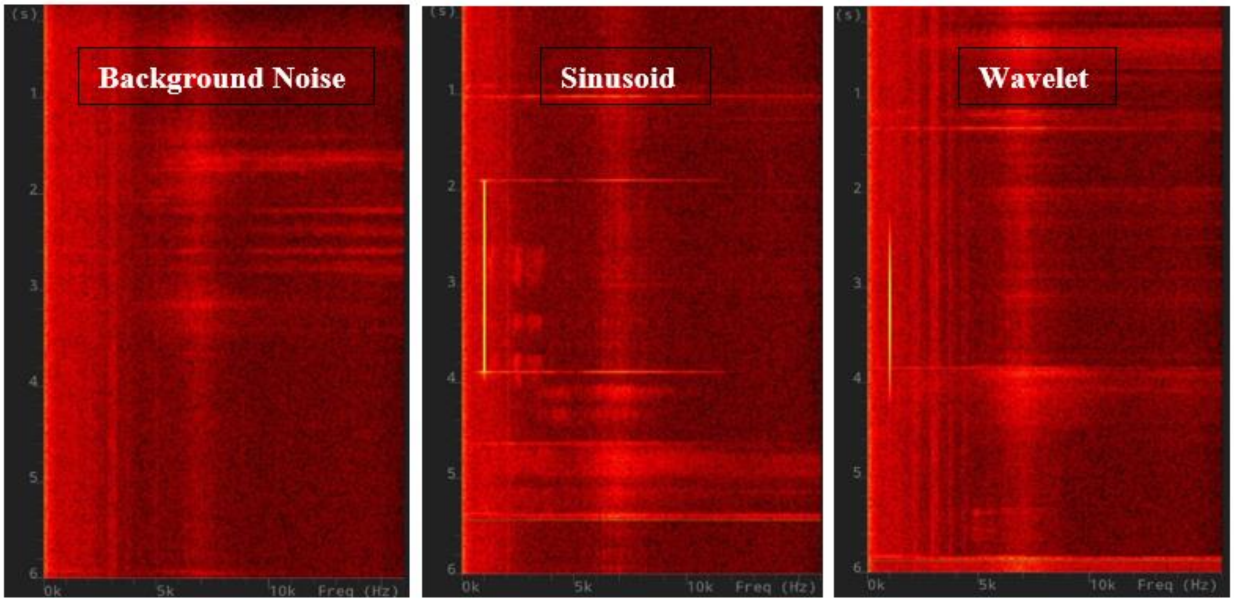

Fig. 3.1: Spectrogram of 1000 Hz Tones

For the hearing test data, I have taken a test that is available online for free (budget constraints meant that I could not easily acquire a professionally administered test) from https://hearingtest.online. It uses pulsed tones instead of sinusoids, but I don't believe this fact could have influence my results. This is because the user was asked to manually click on the file to play it, so it was very obvious when a tone was being played, whether or not a tone was audible. Additionally, I have a trained ear and can distinguish whether or not I've heard a noise at 1 kHz or 2 kHz, so I can actively avoid being tricked into a false positive due to the high frequency characteristics. The calibration strategy implemented by this online test is for the user to adjust the computer's volume level until the sound played on the website of hands being rubbed together matches the user's real life sound of rubbing their hands together in front of their nose. I made sure to take this test in the same environment at the same time as the wavelet-based test, to ensure very similar levels of background noise (since I could not feasibly eliminate background noise). I also used the same laptop and the same headphones to run both tests. The audiogram that describes my hearing from this online source is shown in Fig 3.2. Red circles indicate right ear results and blue 'X's indicate left ear results. The purpose of this test is to confirm that the test I wrote produces trustworthy results. The test reported that my hearing exhibits mild low frequency hearing loss, but I believe this to instead suggest that my very loud air conditioning is contributing large amounts of noise at those low frequencies, masking those tones much more significantly than the mid to high frequencies. Because this test and the wavelet-based test were taken in the same approximate background noise setting, I would expect both tests to report mild hearing loss in this frequency range despite my confidence that I have healthy hearing.

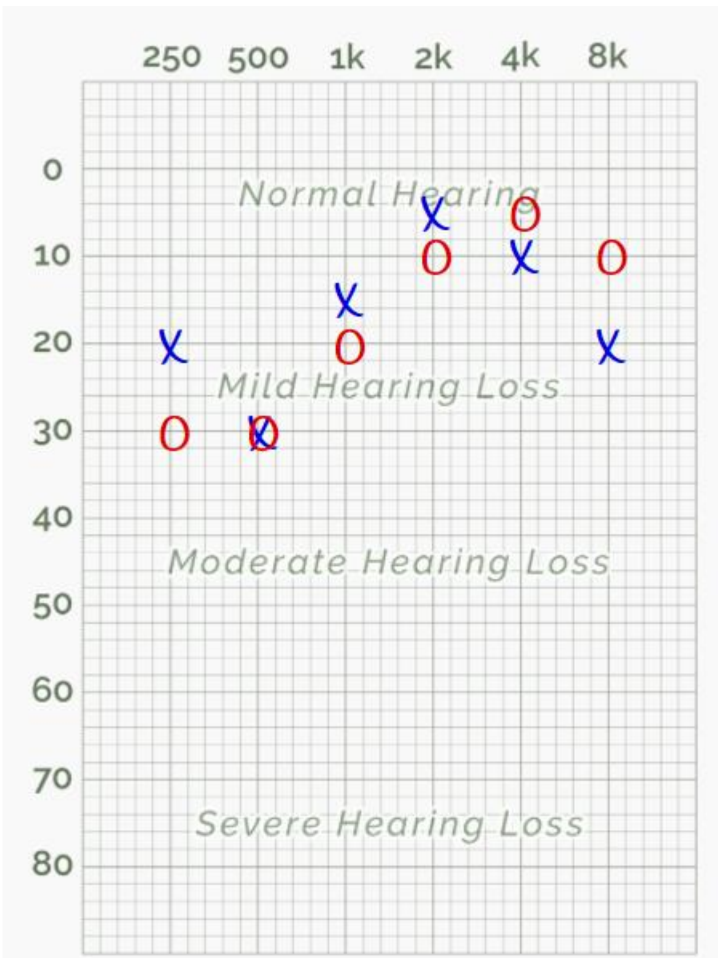

Fig 3.2: Audiogram Results From Normal Hearing Test

My personal results from running the wavelet-based hearing test I wrote in MATLAB are shown in Table 3.1.

Table 3.1: Wavelet-Based Hearing Test Results

	Left Ear Threshold (dB)	Right Ear Threshold (dB)
250 Hz	20	30
500 Hz	25	25
1 kHz	15	25
2 kHz	10	15
3 kHz	5	5
4 kHz	15	5
6 kHz	10	10
8 kHz	25	10

I also used the formatting of my wavelet-based test to give myself a test using pure tone signals. I found that the testing was slightly faster, but the results did not differ significantly. For my trained, healthy hearing, this is unsurprising.

To determine the accuracy of my hearing test I will allow for a deviation of +/- 5 dB at each tested frequency. This is because 5 dB increments are the smallest increments allowed by either hearing test, and because this is fundamentally a subjective test, and because of the margin for error in the calibration method of the online test. Table 3.2 shows the differences between the thresholds found by both tests, where the value in each entry is wavelet-based minus online test.

Table 3.2: Wavelet-Based Minus Pulsed Tone Test Results

Frequency	Left Ear Threshold Error	Right Ear Threshold Error
250 Hz	0	0
500 Hz	-5	-5
1 kHz	0	5
2 kHz	5	5
4 kHz	0	-5
8 kHz	5	0

Because the results' differences shown in Table 3.2 appear somewhat random, I am confident that there is no systematic error left to be eliminated. These results enforce the idea that the wavelet-based hearing test designed in this project is capable of producing accurate threshold findings, particularly in individuals similar to myself.

Because these tests are subjective in nature, I will now give my subjective opinions on taking these tests. It is widely reported in the field that pulsed tests reduce confusion. This is because the pulsed tone makes it very obvious that there is a tone being played. My experiences reflect this report, as I found it a lot faster to take the pulsed test. I spent overall less time wondering if I had actually heard something when taking the pulsed test. Something I implemented in the wavelet-based hearing test to combat this is that the user is required to answer "yes" or "no". Because no "maybe" answer is allowed, the user is forced to give a gut response. This will make the experience seem more difficult to the user. But the nature of the threshold, according to its definition, is that it is only identified as being heard about half the time. This allows for, and perhaps requires, some amount of uncertainty or doubt. The pulsed tone produces a very bright spectrogram across the entire frequency range, while the wavelet tone, objectively, does not. There will be cases, such as my own, where the pulsed tone in a hearing test will produce very similar results to the wavelet tones. This data is incapable of either proving or disproving the prediction that there may be cases, particularly those with low frequency hearing loss and an untrained ear, where the high frequency characteristics of a sinusoid or pulsed tone will be more noticeable to the user than the fundamental frequency under test.

3.2 Hearing Aid Model Validation and Results

The physical circuitry for the filter bank was soldered together and is shown below. There was an error in the PCB layout of the first filter bank prototype as concerned the digipots (U3) so a second PCB was made with this single error corrected. Because the rest of the filter bank works as expected, the digipots have been soldered onto the second corrected board and the resistances are connected onto this board (into WA, LA, WB, LB) via jumpers.

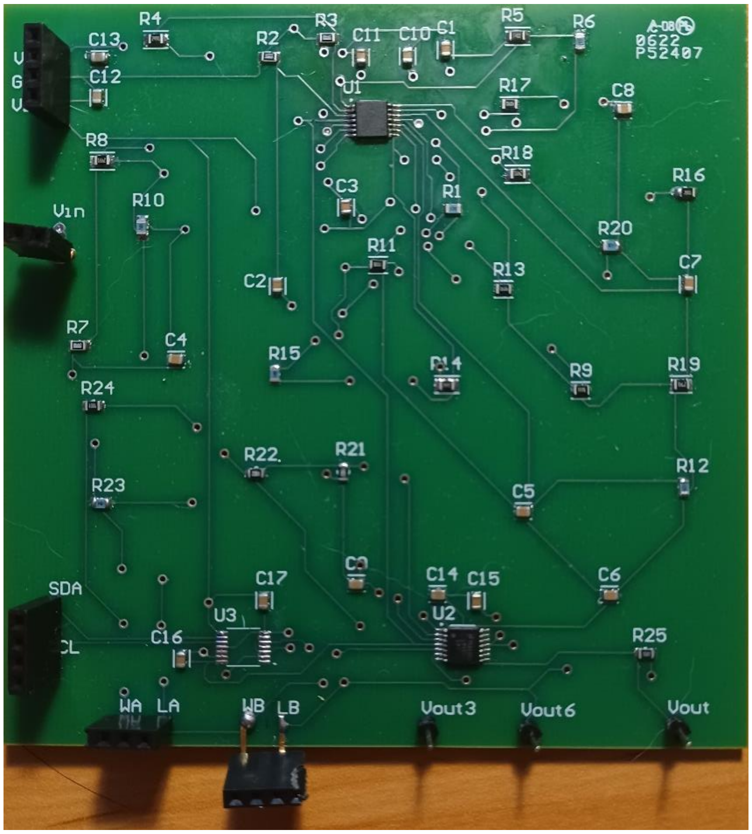

Fig 3.3: Soldered Filter Bank PCB

In general, the physical circuitry is a very good match for all simulated expectations. I have found no significant discrepancies between simulations and measurements.

What follows are the AC response plots of the individual filters collected with the Analog Discovery 2.

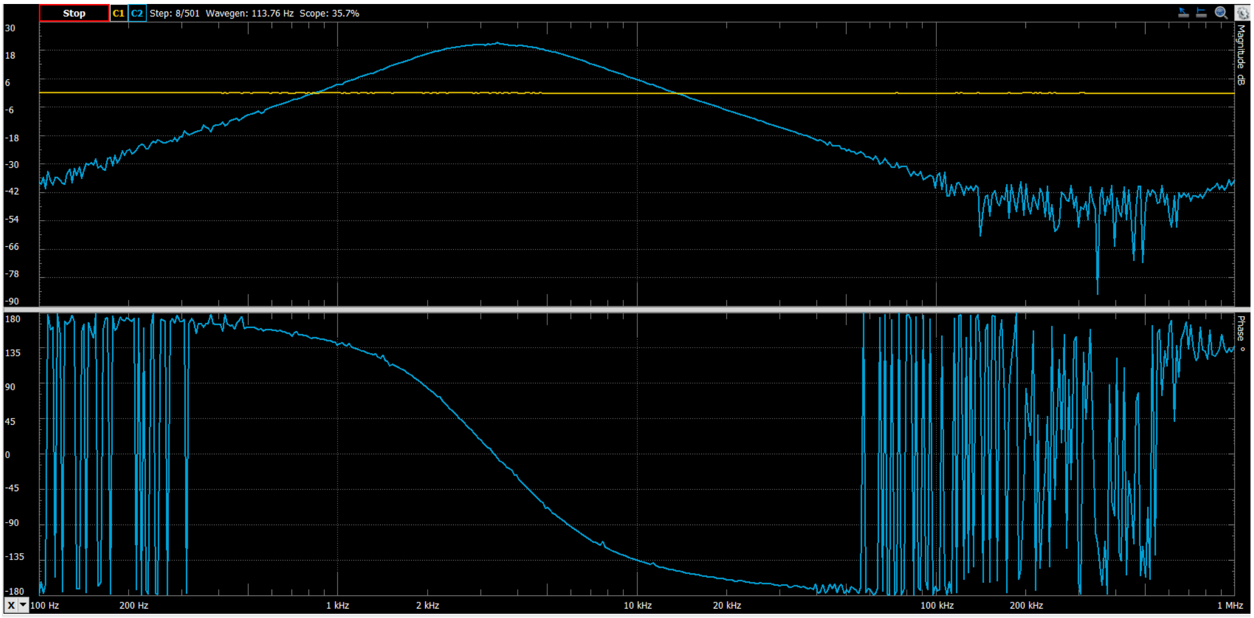

Fig 3.4: 3.2 kHz Filter AC Response

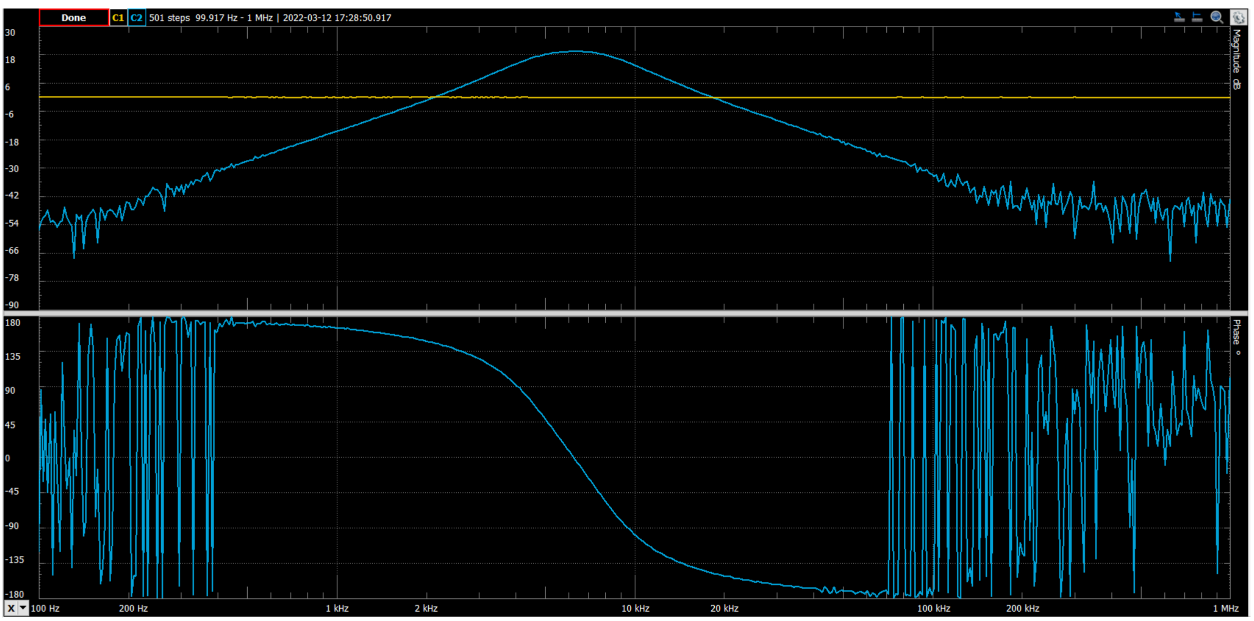

Fig 3.5: 6 kHz Filter AC Response

Table 3.3 describes the numerical measurements compared to the design parameters. Overall, the filters as built reflect the filters as designed to a good degree. Due to the log nature of the frequency axis during simulation and design, a higher nominal error is expected and observed in the higher frequency filter.

Table 3.3: Filter Parameters

	3.2 kHz Filter Design	3.2 kHz Filter Measured	6 kHz Filter Design	6 kHz Filter Measured
Center Frequency	3.2 kHz	3.17 kHz	6 kHz	6.29 kHz
Bandwidth	3 kHz	3.14 kHz	4 kHz	3.7 kHz

The all-pass filter can be visually confirmed as accurate, so it's simulation versus measured magnitude and phase plots are shown below.

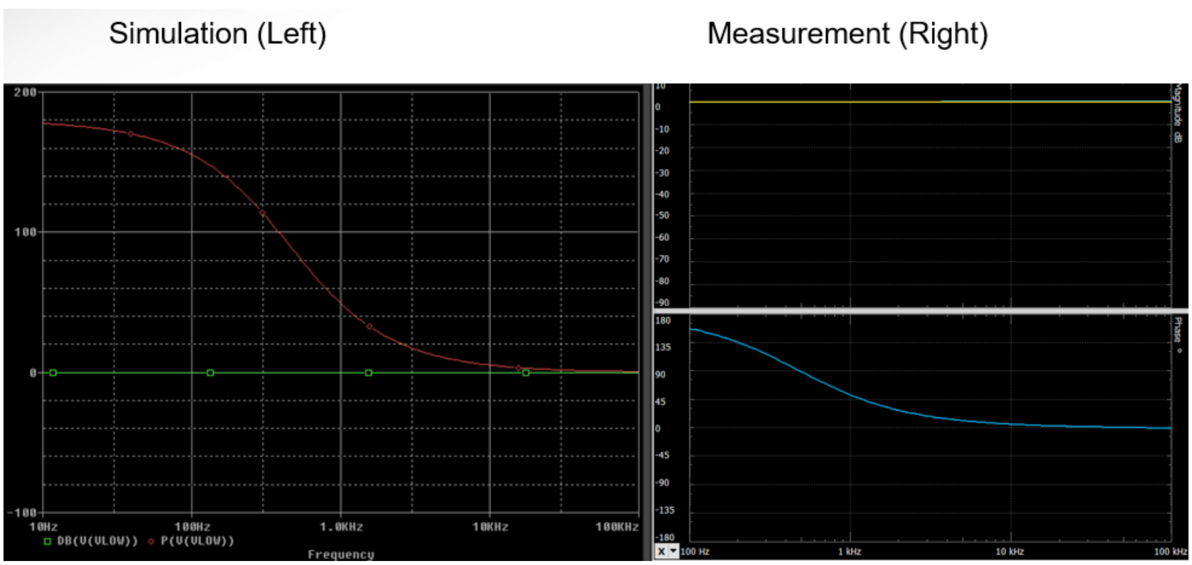

Fig 3.6: All Pass Filter AC Response

Using the digipots, it was not possible to set the potentiometers (functioning as rheostats) controlling gain modulation to exactly 1 kΩ. 860 Ω and 1.2 kΩ were the closest options, putting the filter's gain at either 21 dB or 19 dB, rather than the target 20 dB. Some examples of the filter bank correctly equalizing input signals, using the digipots, are shown next. Fig 3.7 shows the AC response at Vout when both filters are set to 21 dB. The max error is well below 2dB at any point of interest.

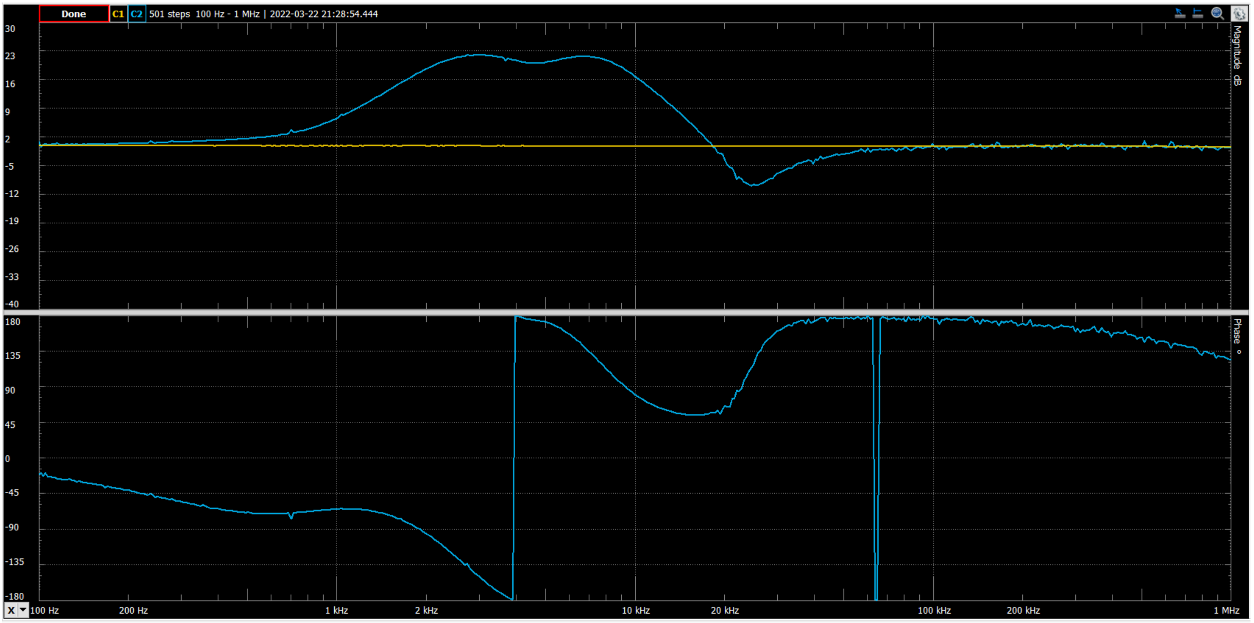

Fig 3.7: AC Response for Both Filters On

When only the 3.2 kHz filter is on, the AC response looks like Fig 3.8.

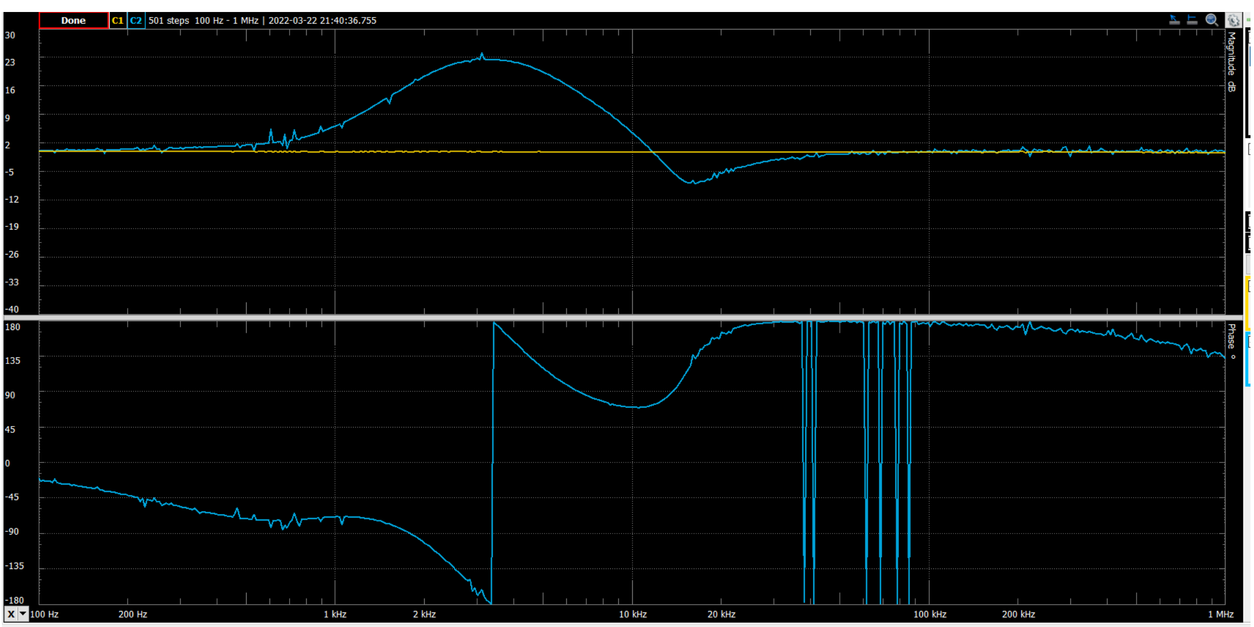

Fig 3.8: AC Response for 3.2 kHz Filter On Only

When only the 6kHz filter is on, the filter bank is designed to equalize the common form of high frequency hearing loss depicted in Fig 1.1. Fig 3.9 shows the measured equalization curve for this scenario, which is nearly exactly as designed and simulated. The max error is approximately 3 dB at 3 kHz, as it was in simulations. This is not an error in the circuitry, but is merely a shortcoming in the design limitations as a steeper slope is impossible without giving the bandpasses an extra stage.

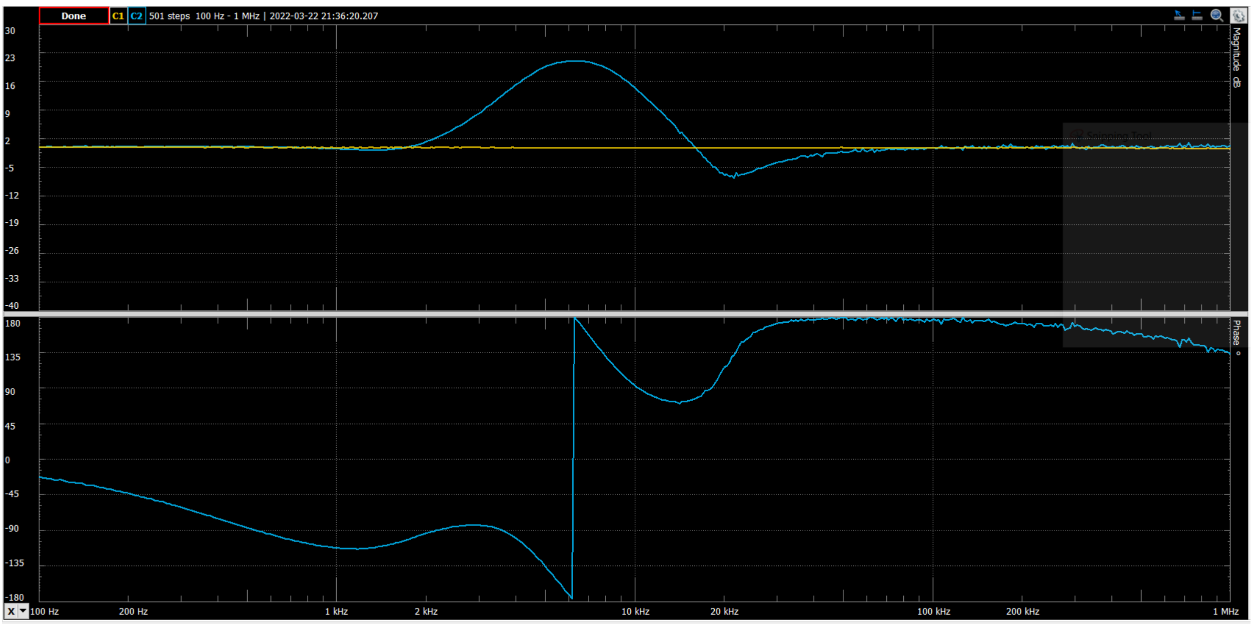

Fig 3.9: AC Response for 6 kHz Filter On Only

To take another equalization example, see Fig 3.10, which shows the scenario of the 6 kHz filter at a gain of 16 dB but the 3.2 kHz filter at only 10 dB. These two curves can be adjusted up or down to achieve a large variety of equalization scenarios.

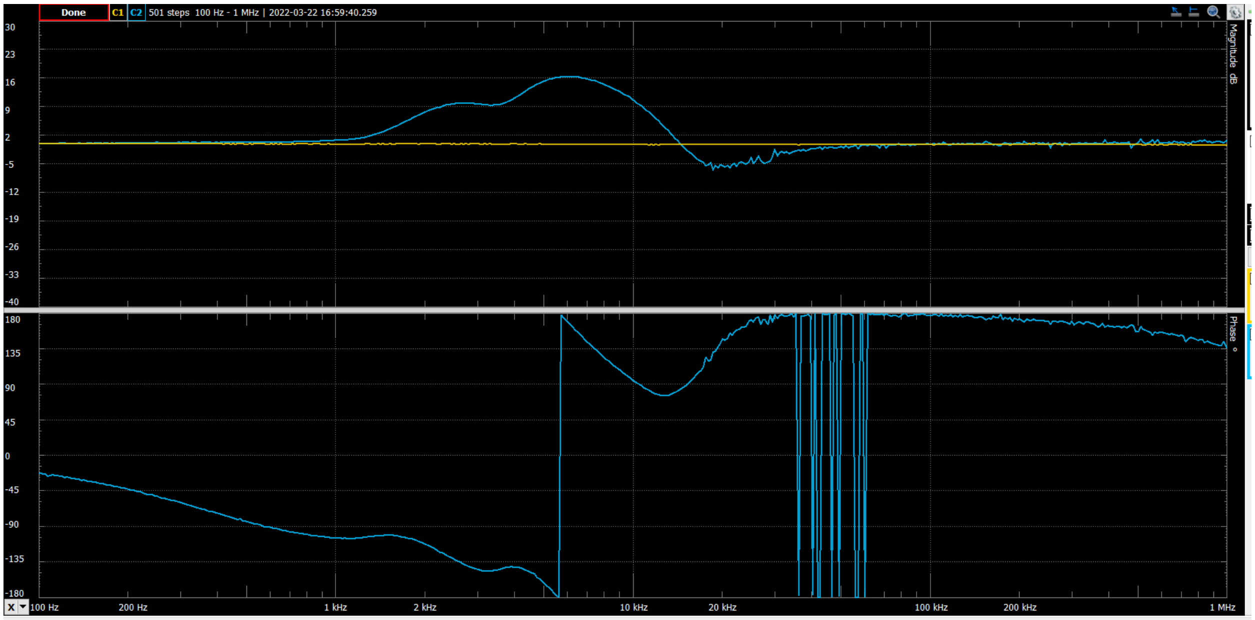

Fig 3.10: Equalization Example

The measured responses very closely match the designed and simulated ones, when accounting for the differences in digipots increments. The one case where simulations differ from my measured results are in the case that both filters need to be off. In this case, it is desired that all signals are passed through with a gain of 1. In simulations, when both potentiometers are turned to 100 kΩ, it does this with a max error of 0.758 dB. But the actual potentiometers I have acquired, though labeled as 100 kΩ, max out at 37.5 kΩ. In most cases, this makes no discernable difference because at least one potentiometer is usually going to be set somewhat low, and 37.5 kΩ is high enough in ratio that it might as well be infinite. But in this case, when no gain is desired, the measured simulation response encounters a max error of nearly 2 dB. This is still barely within my allowed error margin, but it is a notable discrepancy between simulations and measurements. Different programmable digipots with non-volatile memory could not be acquired due to the chip shortage, so this shortcoming of the digipot IC is unavoidable.

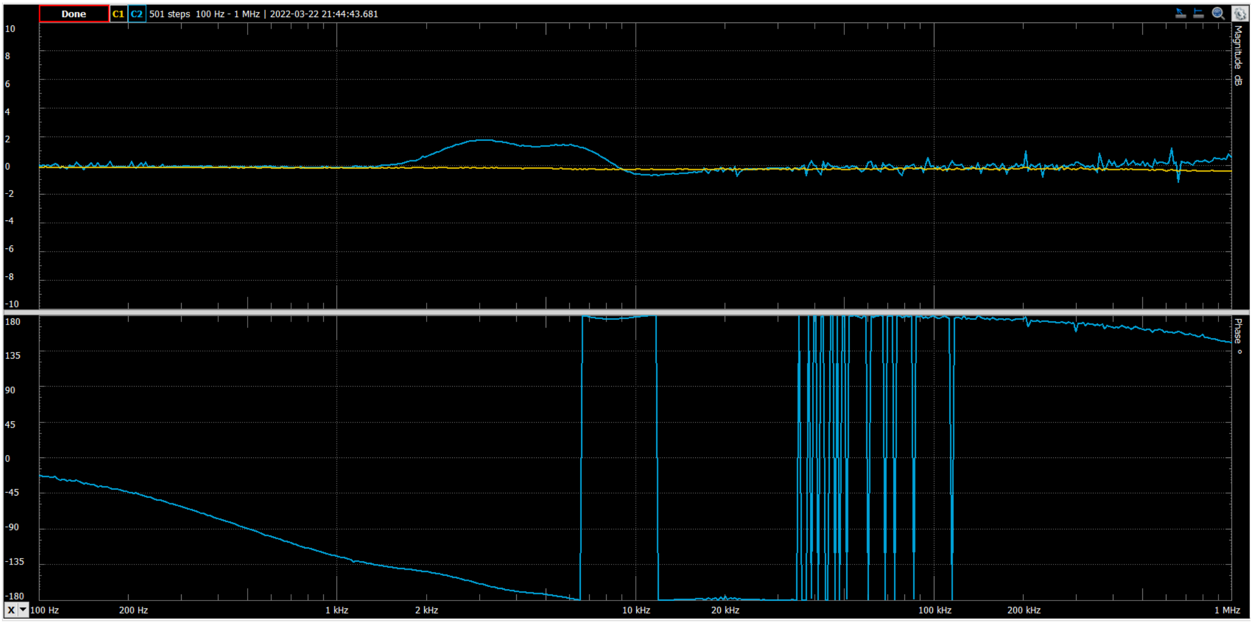

Fig 3.11: AC Response for Both Filters Off

The hearing aid model was set up using this PCB as the equalizer. The circuit used to split +5 V into +/- 2.5 V is shown in Fig 3.12. It connects directly to the power supply and then provides power to the rest of the circuit. Components that required DC +5 V simply connect to the red and blue lines, while components that need +/- 2.5 V connect to the red, blue, and black lines. It is essentially a voltage divider circuit, but allows for a very consistent voltage level because this single circuit draws significantly more current than any other component.

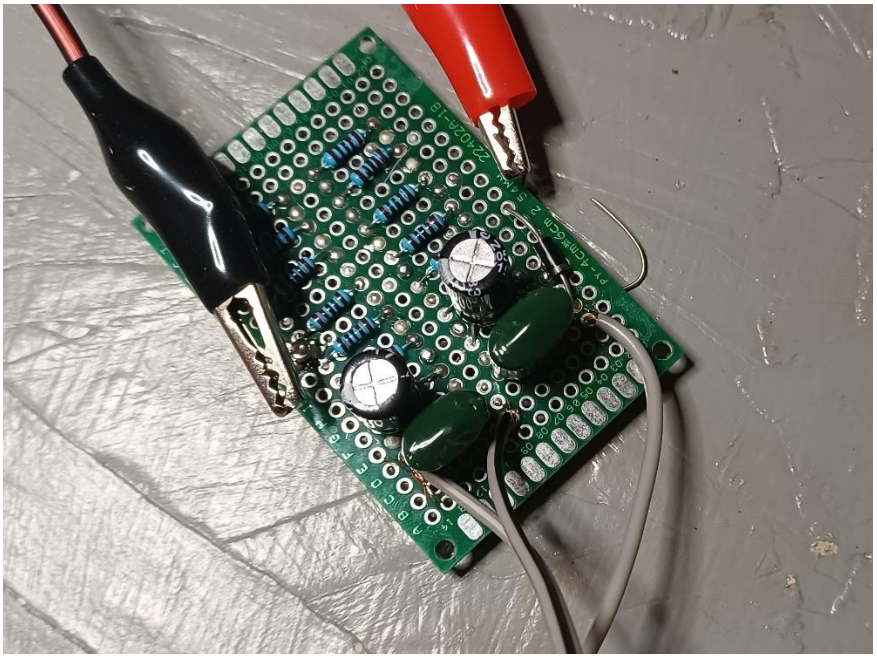

Fig 3.12: Power Splitting Circuit

A unidirectional condenser microphone with a 3.5 mm TRRS jack hookup was used. It is being provided with 5 V plug-in power. Its output is fed through a large 470 µF capacitor to center it at 0 V. The plug-in power, the 2.2 kΩ resistor that power is connected through, and the capacitor are all currently breadboard connections due to time constraints. This is undoubtedly a major source of noise in the circuit, and will be turned into soldered connections before the hearing aid model is demonstrated. In the meantime, these breadboard connections still allowed for reasonable measurements to be taken and basic function to be shown.

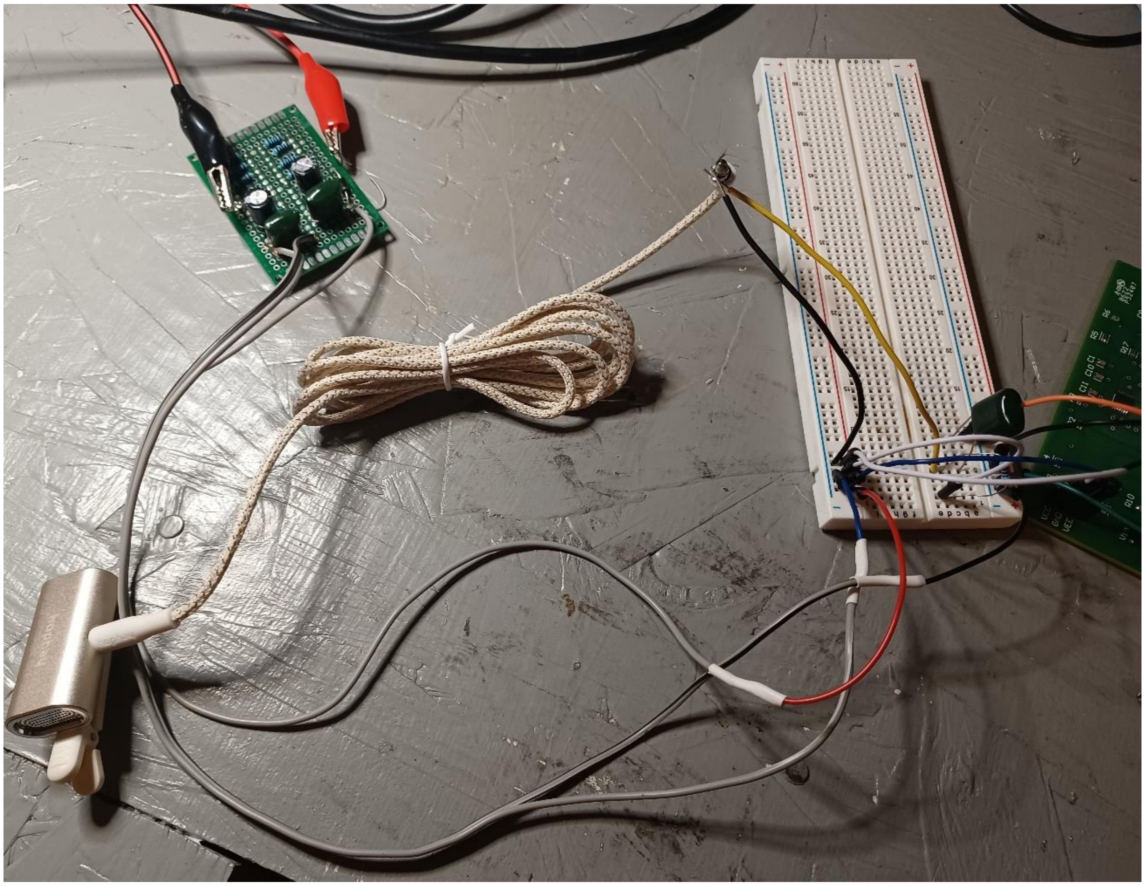

Fig 3.13: Microphone Connections

The following image shows the filter bank on the left, the potentiometers on the PCB on the right, and the Arduino set up to program them. The PCB on the right has the capacity to completely replace the PCB on the left, once the necessary components are soldered on. Because the circuits currently function as they are, this additional soldering step was considered purely cosmetic and has been delayed.

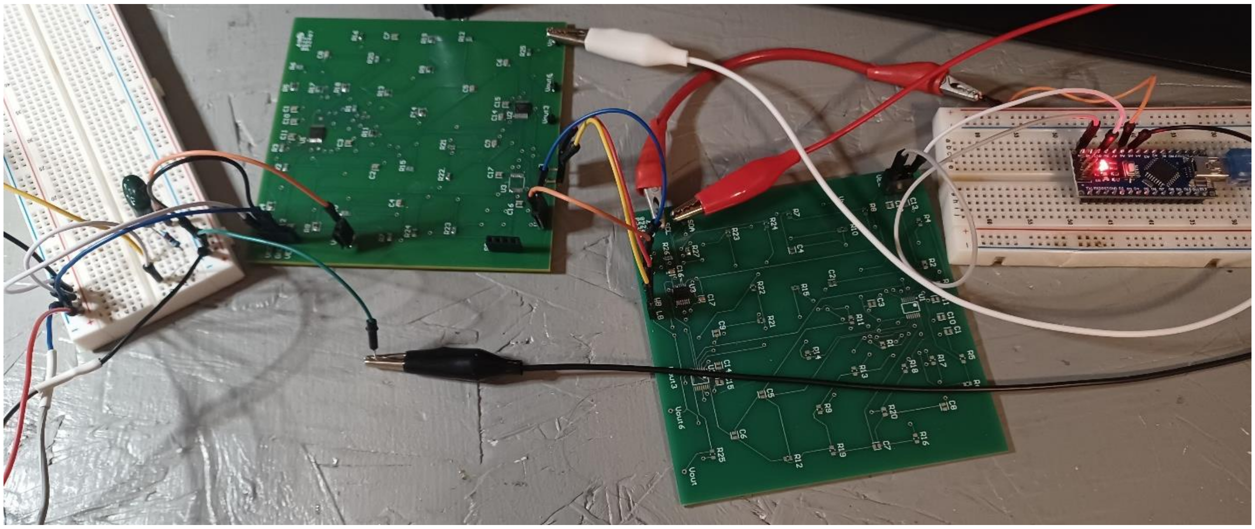

Fig 3.14: Filter Bank and Digital Potentiometer Setup

The amplifier has a coaxial connection to the output of the filter bank. This amplifier is old, humongous, but of good quality and came free to me with a matched speaker. The electrical lines run between the amp and its speaker were made very long so that I could measure output sound with minimal influence from the input sound.

Fig 3.15: Amplifier

Fig 3.16 shows how a sound level meter was used to monitor the output of the speaker.

Fig 3.16: Speaker and Sound Level Meter Setup

Before measurements of the hearing aid model are discussed, Fig 3.17 shows a spectrum analysis of the background noise present.

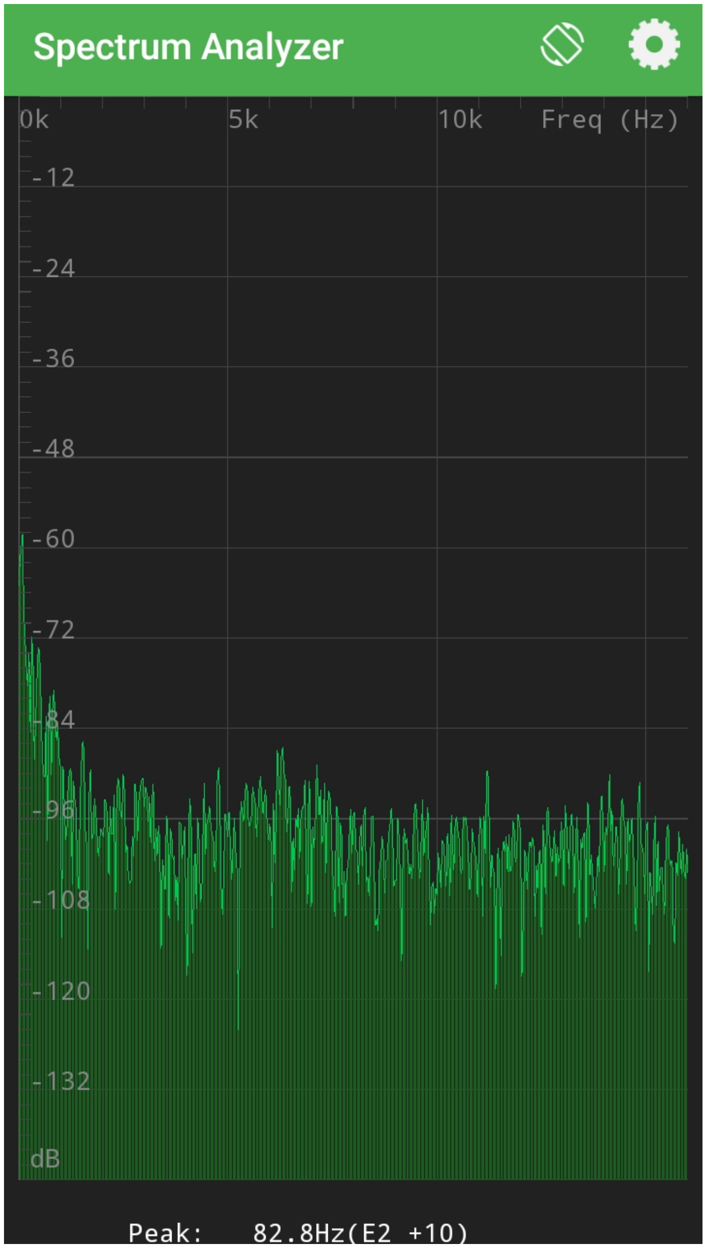

Fig 3.17: Spectrum Analysis of Background Noise

The next spectrum analysis was taken at the output of the speaker when the system (including the microphone) was turned on, but no particular signal was being purposely given to the microphone. A small amount of noise, including a quiet popping sound, was present at the speaker. This noise will be corrected and further minimized in the near future, but did not (for the sake of the following measurements) substantially affect the qualitative function of the hearing aid model.

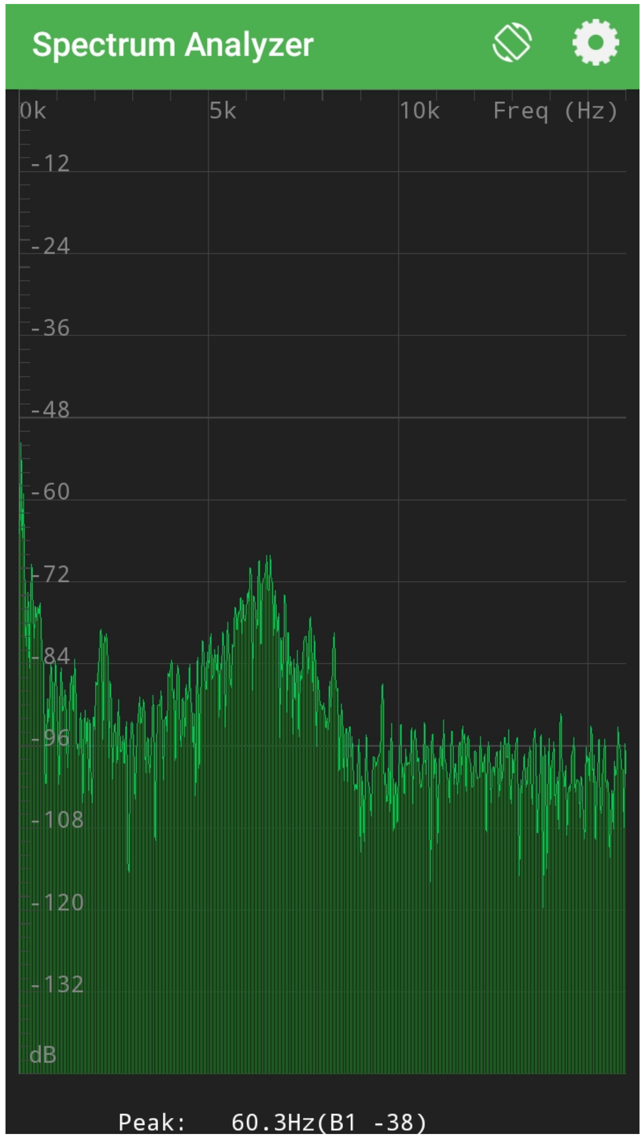

Fig 3.18: Spectrum Analysis of System Noise

The basic function tests performed were very simple in nature. In general, sine waves were played at the microphone and measured at the speaker. When the 3.2 kHz and 6 kHz filters were set to equal levels, the 3.2 kHz and 6 kHz sine waves at the speaker were expected to be of very close loudnesses. When one filter was set to be quieter than the other, its corresponding sine wave was expected to be quieter at the output. This behavior was confirmed with a sound level meter. The specific numerical measurements are not particularly trustworthy because this took place in a room of much unavoidable background noise, with a sound level meter that is not officially calibrated, and some amount of noise in the circuitry that appeared at the speaker to varying degrees. Furthermore, the meter measures in dbA, which is not very easy to convert to dB SPL. But the sound level meter can still qualitatively (if not quantitatively accurately) confirm the functionality of the system.

Table 3.4: Sound Level Meter Measurements

		Filter Settings	
		3 kHz filter max, 6 kHz filter max	3 kHz filter off, 6 kHz filter max
Sound Wave Input/Output Frequency	None (background noise)	64 dBA	64 dBA
	3 kHz	73 dBA	66 dBA
	6 kHz	73 dBA	73 dBA

The measurements above show that when the 3 kHz wave and the 6 kHz wave were supposed to be amplified equally, they were amplified approximately equally. When the 6 kHz wave was supposed to be amplified by 20 dB SPL compared to the 3 kHz wave, the 6 kHz wave was shown to be amplified significantly louder (7 dBA) than the 3 kHz wave as expected.

4. CONCUSION

4.1 Wavelet-Based Hearing Test

The wavelet-based hearing test has been designed and constructed. It encounters no errors when running and takes between 8 and 12 minutes to complete. The Morlet wavelet as the testing tone is capable of producing results as accurate as a sinusoid or a pulsed tone. Spectrogram measurements support the idea that the wavelet produces far fewer high frequency characteristics than a sine wave or a pulse. Whether or not these high frequency characteristics commonly produce inaccurate hearing test results cannot be determined by the limited data set of this research project. But if it is discovered that for some people, these high frequency characteristics cause people to respond that they hear a tone when in fact they cannot hear the fundamental tone in question, then the wavelet will fix this problem. It is expected that wavelets as a replacement to sinusoids will not be helpful in the case of users with tinnitus, since it will be much harder to distinguish the usual tone they hear from the tone they are being asked about. It is also expected that users with a trained ear who can identify particular frequencies will not give very different results to the test when using wavelets compared to sinusoids, because if asked to identify a 500 Hz tone they will be able to listen specifically for a 500 Hz tone instead of just responding that they could tell a tone was played. Because most hearing tests are formatted with a question like “raise your hand when you hear a tone”, it could be easy for normal people to simply raise their hand when they can tell a tone has been played. Sometimes there is a noticeable pop when the tone is turned on and off, exacerbating this source of possible error. However, because the subjective nature of the hearing test is already rather imprecise, this source of error may not be as significant as we expect. Further research with human subjects will be required to establish this.

4.2 Equalizing Hearing Aid Model

The design of the filter bank was necessarily simple due to cost and labor constraints, but it works as intended and successfully equalizes signals with gains of 0 to 20 dB. This max gain would be trivial to increase as it would only require a change of 4 resistors, but because the targeted type of hearing loss only required a gain of 20 dB this is a sensible design parameter for this particular model. The digipots selected for this design are not perfect for several reasons: because they have a linear taper instead of logarithmic, higher filter gains cannot be as precisely set; also, the 100 kΩ potentiometers do not actually reach 100 kΩ. But, due to the supply shortages, I have managed to find some of the last programmable digital potentiometers in stock and am therefore satisfied to have them at all. Neglecting the imperfections of the digipots, the filter bank PCB works exactly as designed. Including the imperfections of the digipots, the only significant errors occur when both filters are set to 0 dB or when one filter is set to 0 dB and the other to 5 dB. These errors are still within the acceptable margin of +/- 2 dB. The equalization and self-tuning functions are both well validated. Self-tuning describes the process by which the filter bank is equalized via the digipots according to the results of the hearing test simply by pressing start on a MATLAB code and an Arduino code. It is possible with simple software only because of the simple filter design. Because this design only sets two filter gains, no serious algorithms are required. A more complicated filter bank (20 channels instead of 2 is common) would require a more complicated algorithm. These algorithms to calculate necessary gains already exist and are widely used by audiologists today, so it was not seriously considered in the scope of this project.

4.3 Applications

The wavelet-based hearing test should be looked into as a replacement to the standard sinusoid-based hearing tests in many general audiological testing purposes. It could also potentially be integrated into an app such that the user could themselves assess the frequency response of their left and right ears. This app could potentially connect to a marketable version of the self-tuning filter bank designed in this project. This would enable the user to have audio products (hearing aids, stereos, headsets, etc.) tailored specifically to their unique hearing by incorporating the filter bank into these products either internally or externally. The benefits of such a system could radically improve the quality of life for elderly people. According to the MarkeTrak 2018 findings in [17], only about a third of the people who would benefit from wearing hearing aids actually use them. And of those that do, many are cheap versions that are not tailored to the user’s hearing damage. And of those that do get their hearing aids professionally tuned, few will ever go back to get them adjusted, despite our hearing levels changing frequently. The expense of accurately adjusted hearing aids, in terms of time, money, and effort, is rather steep even though the elderly population tends to be less equipped with money and mobility. The market and our society could benefit from a self-tuning option that allows for free and frequent hearing tests.

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APPENDIX A: HEARING TEST MATLAB CODE

```
waitfor(msgbox('Welcome! This program is going to test your hearing by playing tones of
varying frequency and loudness. You will be asked whether or not you hear certain tones. If
you have heard the tone, please respond "yes" no matter how faint that tone is. You must
respond either "yes" or "no". The test will begin when you click "OK" on this message box.',
'Instructions'));

frequencyOrder = {'1k', '2k', '3k' ,'4k', '6k', '8k' ,'1k', '500', '250'};

%Array called thresholds to store the results of this test
thresholds = zeros(9,2); %9 frequency tests, 2 ears

%left ear is up first, meaning third variable in funcion playFile is 'left'
for i=1:length(frequencyOrder)
    freq=frequencyOrder{i}
    level = 40 %starting level at every frequency
    %start familiarization
    playFile(freq,string(level),'left')
    pause(5)
    answer = questdlg('Have you heard a tone?');
    if (answer=="No")
        level = level+20
        playFile(freq,string(level),'left')
        pause(5)
        answer = questdlg('Have you heard a tone?');
        if (answer=="No")
            msgbox("You may have severe hearing loss. Please exit this test and seek
professional testing.")
        end
    end %end of familiarization

    %initialize list; will be used to track which sound levels have been played
    %1==heard, 0==not heard
    levelHistory=[level;1];

    %start threshold search

    level = level-10 %because we already know user can hear the most recent tone
    count=2;%need counter for storing data in an array for level history

    wantToBreakOuterLoop=0;
    while(true) %will manually break when threshold is found
        if wantToBreakOuterLoop==1
            break
        end
        playFile(freq,string(level),'left');
        pause(5)
        answer = questdlg('Have you heard a tone?');
        switch answer
            case 'Yes'
                nextLevel=level-10;
                levelHistory(2,count)=1; %memory marking this level as heard
            case 'No'
                nextLevel=level+5;
                levelHistory(2,count)=0; %memory marking this level as not heard
        end
```

```
        levelHistory(1,count)=level; %store this sound level in memory

        %now need to check whether we've found a threshold yet
        %threshold is defined as being heard 2 out of 3 or 3 out of 5 presentations

        %count how many presentations of current level we've had
        numPresentations = 0;
        heardOrNotList = [];
        lowerLevelHeardOrNotList = [];
        for iter=1:length(levelHistory)
            if levelHistory(1,iter)==level
                numPresentations = numPresentations + 1;
                heardOrNotList(end+1) = levelHistory(2,iter);
            end
            if levelHistory(1,iter) == level - 5
                lowerLevelHeardOrNotList(end+1) = levelHistory(2,iter);
            end
        end
        %handle 2 out of 3 case
        if numPresentations==3
            for iter=1:3
                if(sum(heardOrNotList, 'all') == 2)
                    thresholds(i,1) = level %found threshold
                    wantToBreakOuterLoop = 1;
                    break
                end
            end
        end

        %handle 3 out of 5 case
        %but also... what if this level can be heard every (or nearly every
        %time) but the one below it cannot? I'll also consider that a
        %threshold because an in-person audiologist would have a flexibility
        %to handle that which this test cannot.
        if numPresentations==5
            if(sum(heardOrNotList, 'all') == 3)
                thresholds(i,1) = level %found threshold
                break
            end
            if (sum(heardOrNotList, 'all') >3 && sum(lowerLevelHeardOrNotList,'all') < 2)
                thresholds(i,1) = level %found threshold
                break
            end
        end

        %otherwise, just continue chugging through this while loop adding
        %to level history
        count=count+1; % count refers to how many levels played
        level = nextLevel %increases or decreases volume for the next test
    end
end

%right ear tests now
%what follows is nearly 100% copy and paste of the above code, with only a
%few changes, but this is less confusing than making an additional loop

for i=1:length(frequencyOrder)
    freq=frequencyOrder{i};
    level = 40; %starting level at every frequency
    %start familiarization
    playFile(freq,string(level),'rite') %left changed to right, but right is 4 chars
```

```
    pause(5)
    answer = questdlg('Have you heard a tone?');
    if (answer=="No")
        level = level+20;
        playFile(freq,string(level),'left')
        pause(5)
        answer = questdlg('Have you heard a tone?');
        if (answer=="No")
            msgbox("You may have severe hearing loss. Please exit this test and seek
professional testing.")
        end
    end %end of familiarization

    %initialize list; will be used to track which sound levels have been played
    %1==heard, 0==not heard
    levelHistory=[level;1];

    %start threshold search

    level = level-10; %because we already know user can hear the most recent tone
    count=2;%need counter for storing data in an array
    while(true) %will manually break when threshold is found
        playFile(freq,string(level),'rite'); %changed left to right again
        pause(5)
        answer = questdlg('Have you heard a tone?');
        switch answer
            case 'Yes'
                nextLevel=level-10;
                levelHistory(2,count)=1; %memory marking this level as heard
            case 'No'
                nextLevel=level+5;
                levelHistory(2,count)=0; %memory marking this level as not heard
        end
        levelHistory(1,count)=level; %store this sound level in memory

        %now need to check whether we've found a threshold yet
        %threshold is defined as being heard 2 out of 3 or 3 out of 4 presentations

        %count how many presentations of current level we've had
        numPresentations = 0;
        heardOrNotList = [];
        lowerLevelHeardOrNotList = [];
        for iter=1:length(levelHistory)
            if levelHistory(1,iter)==level
                numPresentations = numPresentations + 1;
                heardOrNotList(end+1) = levelHistory(2,iter);
            end
            if levelHistory(1,iter) == level - 5
                lowerLevelHeardOrNotList(end+1) = levelHistory(2,iter);
            end
        end
        %handle 2 out of 3 case
        if numPresentations==3
            for iter=1:3
                if(sum(heardOrNotList, 'all') == 2)
                    thresholds(i,1) = level %found threshold
                    break
                end
            end
        end
```

```
        %handle 3 out of 5 case
        %but also... what if this level can be heard every (or nearly every
        %time) but the one below it cannot? I'll also consider that a
        %threshold because an in-person audiologist would have a flexibility
        %to handle that which this test cannot.
        if numPresentations==5
            if(sum(heardOrNotList, 'all') == 3)
                thresholds(i,1) = level %found threshold
                break
            end
            if (sum(heardOrNotList, 'all') >3 && sum(lowerLevelHeardOrNotList,'all') < 2)
                thresholds(i,1) = level %found threshold
                break
            end
        end

        %otherwise, just continue chugging through this while loop
        count=count+1; % count refers to how many levels played
        level = nextLevel; %increases or decreases volume for the next test
    end
end

%need to export results of test
%CSV will be convenient since we're dealing with a 2 by 9 array
writematrix(thresholds);
```