DESIGN AND IMPLEMENTATION OF A LOW COST, ADAPTIVE HEARING AID

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Abstract: This paper presents the design and implementation of an adaptive hearing aid in the form of a software simulation and a hardware proof of concept. The hearing aid is able to apply compensatory amplification based off of an audiogram, and has a user steerable direction of focus. The simulation applies compensatory amplification to the full frequency range of speech (250 Hz to 8 kHz) with an average error of 1.41%, and has 19 steerable directions. The hardware solution is only able to process two frequency bands, 2.8-3.5 kHz and 5.6-7 kHz. Compensatory amplification can be applied with an error of 9.85%. The average error of the directionality feature is 32.88% within five steerable directions. The overall cost of the device was R1460. For future work, an integrated circuit chip should be created for the preprocessing of the audio signals.

Key words: compensatory amplification, directionality, hearing aid, low cost

1. INTRODUCTION

Hearing loss is a problem facing people in all parts of the world with numerous causes including disease, genetics and age [1]. To combat hearing loss, adaptive hearing aids can be worn [1]. However, hearing aids are very expensive, which results in them being inaccessible for the majority of South Africans [2]. As such, it is essential to develop a high functionality hearing aid with a low-cost to help combat hearing loss. This paper presents the design and implementation of an adaptive hearing aid, presented in the form of a software hearing-aid simulation and a hardware hearing aid proof of concept.

2. PROJECT BACKGROUND

An adaptive hearing aid is a device that is able to adapt to a person's individual hearing deficits. It is also able to adapt to different environments by allowing the user to select the direction in which they wish to listen, or to choose to listen to sounds omni-directionally.

2.1 Specifications

The project requires an investigation into the design of a low cost, adaptive hearing aid. This investigation takes the form of a software hearing aid simulation and a hardware proof of concept. Both solutions are required to implement all of the above mentioned functionality. However, the hardware solution is required to do so within a limited scope. The largest constraints within the project are budget and time. The project is required to be implemented within a budget of R1200. It also needs to be completed within a six week time frame.

2.2 Literature Review

In order for a hearing aid to adapt to a person's hearing deficits, the device must be able to apply compensatory amplification based on a person's audiogram [3]. The implication thereof is that different amplifications must be applied to different frequencies [4]. In order to do so, it is suggested by [3, 4] that a filter bank should be used in order to segment the audible frequency range (0-20 kHz) into a number of frequency bands. The ANSI S1.11 standard defines 43 frequency bands which cover the audible range [4, 5]. Each of the bands has a 1/3 octave bandwidth [4]. However, to reduce the computational complexity of the processing in the hearing aid, compensatory amplification can be applied exclusively to the frequency range of speech (250 Hz-8 kHz), implying the use of 16 frequency bands [3, 4]. According to [5], the use of an ANSI filter bank offers the best audiogram matching, making it ideal for use in a hearing aid.

The hearing aid also needs to be adaptable to different environments. This involves the ability to hear both directionally and omni-directionally [6]. In order to achieve directionality, a directional microphone or a combination of multiple omni-directional microphones can be used [6, 7]. Considerable research has been done on creating directional hearing aids that automatically focus in the direction of the largest signal to noise ratio using auto-correlation [6, 7]. However, there is minimal literature on the creation of a hearing aid that is able to focus on a direction that is tunable by the user. In order to achieve this functionality, delay-and-sum beamforming must be used with a microphone array consisting of three or more omni-directional microphones [8]. In delay-and-sum beamforming, the output from each microphone is time delayed and the resultant signals are added together [8]. The time delay per microphone is calculated according to the position of the microphone in the array and the desired direction of focus [8]. As a result, due to interference, signals originating in the same direction as the selected focus direction will be amplified, and others will be attenuated [8]. Thus by changing the time delay of the signal from each microphone, tunable directionality can be achieved.

3. SYSTEM OVERVIEW

The overall system block diagram of the hardware hearing aid solution is given in *Figure 1*. The solution makes use of analogue circuitry and embedded software which was implemented on an Arduino Due. In the figure, the numbers between the blocks indicate the number of signals being processed at each point.

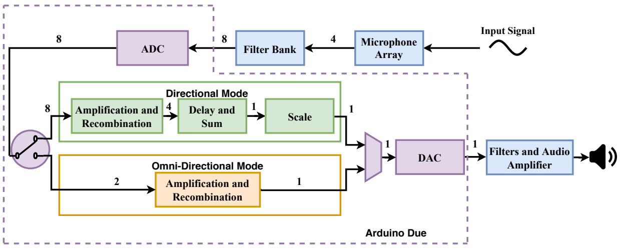

Figure 1: Block diagram of the adaptive hearing aid

In the designed system, audio signals are captured using an array of four omni-directional microphones. The signals are then passed through a filter bank, and the filtered signals are converted to digital by an Arduino Due's analog to digital converters (ADC). Thereafter, with a method depending on whether the user has selected the omni-directional or directional mode, compensatory gain and directionality are applied to the signals. The resulting signal is converted to analog by the Arduino's digital to analog converter (DAC) and then filtered and amplified using an audio amplifier. Finally, the audio signal is relayed to the user using a headphone jack.

The same process occurs in the simulation, except that a frame of pre-recorded audio is used. The simulation was created in the form of a graphical user interface (GUI) using MATLAB. In the simulation, an array of 10 omni-directional microphones was used to improve the system's beam steering ability.

4. SIMULATION

This section describes the components of the hearing aid simulation. The simulation was presented in the form of a GUI that was created using MATLAB.

4.1 1/3 Octave Filter Bank

In order to divide the frequency spectrum of speech into bands, a filter bank, consisting of bandpass filters, is required. To comply with the ANSI S1.11 standard, an octave filter bank was designed, consisting of 16 filters each with a bandwidth of 1/3 octave. The centre frequencies of the filters cover the frequency range 250 Hz to 8 kHz. 14$^{\text{th}}$ order filters were created. The order was selected as it allowed for a small number of coefficients, implying a shorter delay due to filtering, while still providing a steep roll-off for each filter band. FIR filters were used as they have a linear phase response which is essential when processing audio signals [9]. The filters were created using MATLAB's `octaveFilter` system object. *Figure D1* provides a bode plot of the filter bank. The yellow plot is the overall response of the filter bank. On an individual basis, each filter is compliant with ANSI S1.11 class 2, as they have a maximum ripple of 1 dB [10]. The selection of the filter order was essential in controlling the magnitude of the ripple on the overall response. Accordingly, the use of 14$^{\text{th}}$ order filters led to the lowest amount of resultant ripple (2.4 dB). Minimising the ripple is essential as large ripples may lead to unforeseen amplification of certain frequency components, which decreases the effectiveness of the device.

4.2 Compensatory Amplification

To apply compensatory gain to the audio signal, the appropriate gains need to be determined from an individual's audiogram. Since an audiogram only has seven readings between 0.25 kHz and 8 kHz, it is necessary to linearly interpolate the audiogram in order to calculate the required amplifications for all the frequency bands [11]. The audiogram was interpolated to 8000 points, with one point representing 1 Hz. As such, the gain corresponding to each filter's centre frequency was selected as the gain for that band. Once the appropriate amplifications had been selected, they were converted from decibels to magnitudes so that they could be multiplied with the audio signals.

When applying compensatory amplification to an audio signal, a frame of audio is loaded and passed through the filter bank. Thereafter, the output signal from each band is multiplied by its corresponding amplification. Finally, the compensated signals are added together to reproduce the audio signal.

4.3 Directionality

To implement the directionality, delay-and-sum beamforming was used. It was decided that in the simulation, the user would be able to steer the beam in 10° increments, from 0° to 180°. These increments were selected so as to allow the user a wide range of steerable directions, but to still ensure that the user could discern differences in the resultant audio when different steerable directions were selected. In order to implement the beamforming, time delays were calculated for each microphone per steerable angle. Since the processing was done in the discrete domain, the time delays were converted into sample delays, and were implemented as sample shifts. Reference [12] provides a description of the time delay calculations and of how the microphone array was simulated.

In the directionality feature, each frame of audio that is loaded is passed through the filter bank, and compensatory amplification is applied to each band. Thereafter, the 16 output signals from each microphone are delayed according the corresponding delay for that microphone at the chosen focus direction. Once all of the signals hve been delayed, the 16 signals from each microphone are added, and then the 10 microphone signals are added. To ensure that the resultant amplitude of the signal was correct, the final signal is scaled by the number of microphones.

4.4 GUI

The GUI was created in the form of a window with two tabs: *Compensatory Amplification* and *Directionality*. On each tab, the user has the ability to play, pause and restart the audio (and accordingly, the hearing aid functionality). On the GUI, a pre-recorded audio signal is played on a frame by frame basis. A frame size of 1500 samples was used to ensure that the time delay involved in the processing is minimal, while still ensuring that enough samples were loaded for the directionality feature to function correctly. *Figures E1* and *E2* graphically display the two tabs of the GUI such that the functionality can be seen.

On the first tab, the process of compensatory gain was illustrated through the use of Fast Fourier Transforms (FFTs). An FFT was provided for both the input and output signals from the hearing aid, such that the different amplifications could be easily discerned. The tab also contains functionality for the user to input their audiogram so that the compensatory gain can be tailored for a specific audiogram. When an audiogram is inserted, the interpolation process described in *Section 4.2* is applied and the appropriate amplifications are applied to each band.

The second tab illustrates the directionality feature of the hearing aid. On this tab, the user is able to select if they wish to hear in directional or omni-directional mode. In omni-directional mode, the signal collected by one microphone is processed and played. In directional mode, all of the processing described in *Section 4.3* is applied to the frame of audio. In order to make the GUI user friendly, a dial is present on the window. By changing the direction pointed to by the dial, the user is able to change their required direction of focus. As the value of the dial changes, the angle of the dial is extracted and rounded to the closest steerable angle. From this angle, the corresponding row is selected in the look up table, and the beam is steered accordingly. This data is then used to create and display polar plots, using the method described in [12]. Due to the frame based processing, there is a noticeable delay between when the dial was changed and when the amplitude of the audio signal changed. This is further discussed in *Section 7.1*.

5. HARDWARE

In the hardware solution, in order to implement all of the functionality contained in the simulation (compensatory gain and directionality), it was decided that the bandwidth of the device should be limited. As such, a solution was created which implements the necessary functionality, but only for two frequency bands.

The hearing aid proof of concept was created using an Arduino Due. The Due was selected since it has a DAC, which is required to output continuous signals. The ADC sampling frequency was set to 22.059 kHz, which is above the Nyquist sampling frequency of 16 kHz. The DAC write frequency was also set to 22.059 kHz. A timer interrupt that makes use of a 44.1 kHz clock was used to set the sampling frequency. Both the DAC and the ADC were set to a 12 bit resolution to improve the audio quality of the device. Since the directionality feature requires a frame of samples (to implement the sample shifts), a frame size of 100 was selected. This selection presents a tradeoff in the system, which is discussed in *Section 7.1*. In the Arduino code, processing is only applied to the signal once a full frame has been acquired. When performing the analog to digital conversions, a sample is taken from each channel and stored in a one dimensional vector where every eighth sample corresponded to one ADC channel. Furthermore, the voltage on the directionality dial is sampled once per interrupt.

5.1 Filter Bank

To avoid the introduction of a delay caused by the use of digital filtering, analog filters were created. The filters were second order, Butterworth, bandpass filters. On account of the limited number of ADC ports on the Arduino (12), a maximum of two bandpass filters could be used per microphone. The bandwidths of the two filters were 2.82-3.55 kHz and 5.62-7.08 kHz. Both of these filter bands were selected from those implemented in the simulation. 3.4 kHz corresponds to the upper frequency of the narrowband spectrum of speech, and as such it represents the smallest bandwidth (250 Hz to 3.4 kHz) of an audio system that can still allow for speech intelligibility [13]. Thus, as described by [12], the directionality feature was designed for this frequency. The first filter band was thus chosen as it contains this frequency. Due to the low order of the filters, two bands lying next to one another on the frequency spectrum could not be used, as the bands would interfere with one another. As such, the 5.62-7.08 kHz band was selected as it is not adjacent to the first band. The bode plots for the two filters are provided in *Figure D2*, and the circuit diagrams are given in *Figures D3* and *D4*.

5.2 Compensatory Gain

To apply compensatory gain, amplifications corresponding to the two frequency bands were manually programmed onto the Arduino. As such, the user does not have the ability to program their audiogram onto the device without having access to the source code. Once a frame of samples was stored, the samples corresponding to each filter band were multiplied by the appropriate gain, and the resultant signals were added.

5.3 Directionality

The hardware consists of a toggle switch which is used to select between omni-directional and directional mode, and a potentiometer used to select the focus angle of the device. The potentiometer was built to change the voltage across a voltage divider and the resultant divider voltage was used to decide which angle should be selected. This was done by creating a lookup table of ADC values and their corresponding angles, with the possible focus angles being 0°, 60°, 90°, 120°and 180°.

In omni-directional mode, the signal from a single microphone is processed. In doing so, the signals from the two filters were compensated and then added. In directional mode, the method used in the simulation was adapted for use with four microphones. The time delay (t_k) was calculated, and was converted to a sample delay (d_k) according to *Equation 1*, using a sampling frequency (F_s) of 22.059 kHz [8]. When non-integer shifts were calculated, the shift was rounded off to the nearest integer. To perform the beam steering, a resultant vector was created which was filled with zeros. The two signals from each microphone were added to the resultant vector, starting at the loop iteration corresponding to the calculated sample shift for that microphone. The number of samples to be shifted was controlled by the focus angle, which was determined by the potential difference across the voltage divider. Once the resultant vector had been calculated, it was scaled by the number of microphones to correct the signal amplitude.

$$d_k = F_s t_k \tag{1}$$

5.4 Construction

In the final hardware construction, low pass and high pass filters were used at the output of the DAC to improve sound quality. The cutoff frequency of the low pass filter was 12 kHz, and that of the high pass filter was 1 kHz. These frequencies were selected as they preserve the signal frequencies of interest, and attenuate those that are unwanted. The audio amplifier applied amplification to the final signal so that it could be comfortably heard through the earphones.

The hardware was constructed on veroboard to ensure that the positioning of the components was permanent. This was especially important for the microphones which required an equal spacing and an equal height off the board. The microphones required a 5 cm separation, which implied that the minimum size of the device was 15 cm [12]. The size of the final device was 20x10cm. To allow the device to be portable, it was powered by a 9 v battery. The veroboard, Arduino and the battery were all placed into a transparent box to improve the appearance of the device. An image of the hearing aid is given in *Figure G1.*

5.5 Cost Analysis

The cost analysis for the hearing aid is given in *Table 1.* The total cost of the device was R1462, which was slightly over the formal budget for the project. However, when considering that the objective of the project was the development of a low cost hearing aid, this cost is deemed to be acceptable as it is a fraction of the cost of a typical hearing aid in South Africa (R30 000 on average) [2].

Table 1: Cost analysis for the adaptive hearing aid

Component	**Cost (R)**
Arduino Due	569.00
MAX9814 Microphone Amplifier x4	546.00
LM385 x5	16.05
LM386	15.75
Headphone jack	16.31
9v Battery	70.00
Veroboard	49.70
Headphones	69.90
Casing	59.90
Miscellaneous	50.00
Total	**1462.61**

6. RESULTS

The results obtained when testing the two systems are given in the following sections.

6.1 Simulation

The errors present in the application of compensatory amplification in the simulation are given in *Table 2.* The errors were calculated based on the difference between the magnitudes of the input and output signals in *Figure F1.* The compensatory amplification was applied according to the audiogram in *Figure F2.* When calculating the errors, a frequency at the centre of each band was selected. From the table, it can be seen that the average error in the compensatory gain feature is 1.41%. The errors can be accounted for by considering that each filter has a maximum ripple of 1 dB, which for an amplification of 10 dB, corresponds to a maximum error of 10%.

Table 2: Error between expected gain and measured gain at various frequencies

Frequency (kHz)	**Error (%)**	**Frequency (kHz)**	**Error (%)**
0.24	0.23	1.60	1.02
0.32	2.21	2.00	0.57
0.40	0.15	2.50	0.92
0.50	1.15	3.14	1.10
0.64	1.35	4.00	1.10
0.80	6.57	5.00	1.68
1.00	0.92	6.30	2.56
1.24	1.02	8.00	0.03

Figures F3 to *F5* provide polar plots of the directionality feature of the simulation. The polar plots are given for a selected direction of 0°, 60° and 90° at 1 kHz, 3.15 kHz and 6.3 kHz. From the plots, it is evident that the most precise steering occurs at 3.15 kHz.

6.2 Hardware

In order to test the compensatory gain feature, two sinusoidal waves of different frequencies were played into the hearing aid device and the output signals were recorded. *Table 3* provides the errors in the measured amplification with respect to the expected amplification for the two frequency bands. The errors were calculated based on the difference between the magnitudes of the input and output signals given in *Figure G2*. In the table, for the first two measurements, a gain of 6 dB was applied to the 2.82-3.55 kHz band and a gain of 0 dB was applied to the 5.62-7.08 kHz band. For the third and fourth measurements, the opposite amplifications were applied. From the table, it is evident that amplification of one band causes signification errors in the other band. Due to the low filter orders, there are unwanted signal components present in the filtered signals. When gains are applied, the unwanted components are amplified, leading to errors.

Table 3: Error between expected gain and measured gain at various frequencies

Applied Frequency (kHz)	Frequency Band (kHz)	Error (%)
3.15	2.82-3.55	0.81
6.30	2.82-3.55	15.34
3.15	5.62-7.08	19.56
6.30	5.62-7.08	3.67

To test the directionality feature of the device, the hearing aid was placed on a rotating platform and rotated in 30° increments with a constant direction selected on the device. Sinusoidal signals with frequencies of 3.34 kHz and 6.00 kHz were played from a set direction 50 cm from the centre of the device and the output signals from the hearing aid were recorded. This procedure was repeated for each tunable direction. In order to calculate the error present in the directionality feature, a simulated polar plot was created for each tunable angle and the simulation was compared to the measured results. The results of the error are given in *Table 4*. The polar plots are given in *Figures G3* to *G5*.

From *Table 4*, it is clear that the smallest error in directionality is present at 90°. This is because at 90°, the signals from the microphones are not time delayed. At the other angles, the large errors are on account of the signals from the microphones being shifted by an integer number of samples, when in fact mathematically, they should have been shifted by fractional values. It is also clear that there is a large error in the omni-directional mode of the hearing aid. From *Figure G6*, it is evident that the microphone does not behave omni-directionally. As such, the error in the mode is due to the nonideality of the microphone. This non-ideal behaviour is also a contributor to the errors present in the steering of the beam.

Table 4: Error in between expected polar plots and measured polar plots for various focus directions

Dial Angle (°)	Average Error (%)
0	46.6
60	30.7
90	12.7
120	22.7
180	51.7
Omni-directional	42.7

7. DISCUSSION

7.1 Critical Analysis

The biggest strength of the designed system is its low cost. While the device is merely a prototype, it has been developed for under R1500. It is thus expected that when the bandwidth of the device is expanded upon to process the frequency range of speech, the low cost of the device will be maintained. A further strength is the large amount of functionality contained within the device. This functionality is of high quality as the average error in the application of compensatory gain is 9.85%, and that of the directionality feature is 32.88%.

The major weakness of the designed hardware solution is its inability to process the full frequency range of speech. As such, compensatory gain cannot be applied to all of the necessary frequency bands. Another result of this is that the audio that is obtained at the output of the system is highly distorted. A further weakness of the system is its large size. Due to the size of the device, it is infeasible for use as a hearing aid without a redesign of the hardware.

The most prominent tradeoff present in the hardware design was in the selection of the two frequency bands. By selecting two distinct bands that do not lie next to one another in the frequency spectrum, the output signal from the hearing aid sounds distorted. However, selecting bands that were separate allowed for the use of lower order bandpass filters. This helped to reduce the cost and complexity of the hardware. Another tradeoff was the decision to use two separate bands that do not cover the entire spectrum rather than dividing the spectrum into two large bands that covered the entire spectrum. By using two large bands, the output audio from the hearing aid would be less distorted, but the concept of compensatory amplification on a band-by-band basis would not have been proven. It was thus decided that proving that the functionality works (and could thus be expanded upon) was more important than creating a device with limited functionality that outputs clear audio.

A large tradeoff present in the simulation is the size of the frame used for processing. By selecting a small frame size, there is an unnoticeable delay present in the processing of the data. This means that when one of the buttons are pressed, or the directionality dial is turned, the result of the change is evident immediately. However, the increased ratio of the time to load a new frame (data acquisition period) to the processing time results in the introduction of perceptible lags to the output audio. As such, a decision was taken to improve the quality of the audio output at the expense of the responsiveness of the GUI.

7.2 Future Development

For future development of the hearing aid, an integrated circuit (IC) chip should be created for preprocessing the audio-signals. This preprocessing would include all the necessary filtering and amplification of the audio signals. By creating a dedicated IC, the entire frequency range of speech could be processed by the hearing aid. It would also allow for the use of higher order filters to reduce the interaction of neighbouring frequency bands and improve the accuracy of the compensatory gain feature. Another essential improvement that needs to be made is the use of a dedicate ADC module, rather than relying on a microcontroller for the conversion. This would ensure that the signals could be oversampled to reduce aliasing effects. Furthermore, high-quality omni-directional microphones should be used to ensure that an omnidirectional response is obtained from the microphones. Finally, the size of the device needs to be reduced for the device to be used as a hearing aid. In order to reduce the size of the device but still maintain the necessary spacing for the microphones, the microphones and circuitry could be embedded into the band of a set of headphones. For a more user friendly and responsive device, the number of steerable directions in the hardware solution should be increased.

8. CONCLUSION

The design and implementation of an adaptive hearing aid was presented. The adaptive hearing aid is able to apply compensatory amplification based on a person's audiogram and has a user tunable beam steering ability. The hearing aid was created in the form of a software simulation and a hardware proof of concept. The simulation was presented as a GUI, which is able to process signals with frequencies from 250 Hz to 8 kHz. It has 19 steerable directions and has an average error in applying compensatory gain of 1.41%. The hardware solution is only able to process two frequency bands, 2.8-3.5 kHz and 5.6-7 kHz, and only possesses five steerable directions. The average error in the compensatory amplification feature is 9.85% and that of the directionality feature is 32.88%. The hardware solution costs R1460 to implement. Future recommendations include the creation of an IC chip to perform the analog processing of the audio signals.

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Appendix

A GROUP WORK REFLECTION

Table A1: Work breakdown between the two group members

Subsystem		Kayla-Jade Butkow	Kelvin da Silva
Filter Banks			
	Digital filter bank design	x	
	Digital filter bank implementation	x	
	Analog Filter Bank Design	x	x
Hardware			
	Microphones, earphones, audio amplifier		x
	Preliminary circuit design		x
	Veroboard design and soldering	x	x
	Configuring the ADC and DAC		x
	Casing	x	x
Compensatory Amplification			
	Audiogram interpolation	x	
	Audiogram matching	x	
	MATLAB Code	x	
	Arduino Code	x	
Directionality			
	Simulation for optimal number of microphones		x
	Generation of weight tables		x
	MATLAB Code		x
	Arduino Code	x	
GUI			
	Design and Implementation	x	
	FFTs	x	
	Polar Plots		x

Table A1 provides the work breakdown between the two students. The work was divided into two main sections: the compensatory amplification feature and the directionality feature. Kayla-Jade primarily focused on the compensatory gain feature, while Kelvin mainly focused on the directionality feature. The work was divided in this way as it allowed for the students to work in parallel, since the features were not dependant on one another. Furthermore, the division of work in this way exploited both student's strengths, with Kayla-Jade's being software and Kelvin's being high frequency techniques and hardware design. By splitting the work in this way, both students were exposed to both the hardware and software aspects of the project. This was important as it provided both students with an exposure to all aspects of an Electrical Engineering project, and took both students out of their areas of expertise. Thus, the project presented as a great challenge and learning opportunity to both the students.

Overall, both students produced high calibre work and both students contributed equally to the project.

The students worked at the University of the Witwatersrand for the duration of the project. This was done to ensure that they were in a close vicinity to one another in case one of the partners needed help. The students worked from 8am until 5pm every day throughout the course of the project. Every morning, a stand-up meeting was held in which the partners discussed their plans for the day and any issues they encountered the day before. This process was essential in ensuring that the project progressed as expected, as any issue that was encountered was fixed immediately.

The students behaved professionally and ethically throughout the project. The students arrived on time for every meeting with their supervisor, and conducted themselves in a respectful manner.

Working in a group for an extended period of time had its challenges, but it was also an essential learning experience. The main challenge of working in a team is learning that a project of this magnitude cannot be completed by one person alone, and that one's partner needs to be trusted to produce high quality work. This process taught the team the importance of communication, as without proper communication the progression of the project would have been halted at the first problem. It also taught the students how to lead and how to follow. Both students had to lead the team on their areas of expertise and to follow the other member's lead on aspects they were not confident on. Furthermore, the students learnt the value of time management, as without proper time management the project would not have been completed within the six week time frame.

B PROJECT SPECIFICATION OUTLINE

School of Electrical and Information Engineering
University of the Witwatersrand, Johannesburg
ELEN4002/4012: Project Specification Outline

To be completed by supervisor
Assessment:
□ Unacceptable □ Poor
□ Acceptable □ Good □ Excellent

Project Title: Design of an Adaptive Hearing Aid

Group Number:	4	Supervisor Name:	Professor Rubin
Member 1 Name:	Kayla-Jade Butkow	Student number 1:	714227
Member 2 Name:	Kelvin da Silva	Student number 2:	835842

Project Specification:

Minimum Specifications:
The design, construction and testing of a pair of hearing aids is required. Each hearing aid is required to independently apply compensatory amplification based on the audiogram for that ear. Pre-recorded audiograms, supplied by the supervisor, will be utilized in implementing this amplification. The hearing aids also require an aspect of directionality, implying that hearing aids are required to be tuned to amplify sound in the direction that the user's head is facing. The user will not be able to choose a direction for sound amplification other than the one in which they are facing. The directionality feature must be able to be toggled on and off in order for the user to be able to hear in all directions if required. Achieving functionality in these areas would be regarded as meeting the minimum specifications for the project. All implemented features will undergo rigorous testing in order to determine the success rate of the device.

Target Specifications:
In the target specifications, the minimum requirements will be met and improved upon. These improvements will be in the form of enhancing the performance of the filters that are used to implement the features. The enhancements will improve sound quality and thus improve the overall performance of the system.

A large improvement to the directionality feature is that it will be tuneable by the user based on real time preferences. The user, by turning a dial, will be able to decide on the direction in which they wish to hear. The resulting direction of the dial will indicate the user's desired direction.

An additional improvement is the acquiring of audiograms from human participants, using equipment supplied by the supervisor. This will allow the group to determine a success rate based on user experience as well as from hardware testing procedures.

Testing Specifications:
The testing specifications will differ depending on the whether the minimum or target specifications are achieved. The minimum requirements for testing would involve the use of a signal generator. To test the compensatory amplification for a given audiogram, a sinusoidal signal consisting of a single harmonic will be applied to the elements of the system responsible for amplification and the resulting amplification will be analysed. The frequency of these sinusoidal signals will range from the lower to the upper bounds of the frequency range of human hearing. Additionally, background noise will be added to these signals to mimic a realistic environment whereby the user would require the functionality of a hearing aid.

Testing the directionality feature will involve producing single harmonic sound waves from multiple directions relative to the hearing aid. The resulting signals, after having passed through the hearing aid, will be plotted and compared. This will allow for the visualization of the degree of amplification and attenuation of signals from the desired and undesired directions respectively.

The target requirements for testing will involve achieving all minimum testing requirements as well as obtaining results from human participants. An audiogram will be obtained from each participant without a hearing aid. Thereafter, a second audiogram will be obtained with the individual making use of the developed hearing aid. These two audiograms can then be compared in order to assess the effectiveness of the device.

Milestones:

Date	Task
06/04/2018	Ethics clearance form final hand-in
25/06/2018	Project plan meeting with supervisor
26/06/2018 – 05/07/2018	Project research and preliminary simulations
06/07/2018 – 15/07/2018	Laboratory project plan
16/07/2018	Laboratory project plan submission
16/07/2018	Latest date for component arrival
06/07/2018 – 22/07/2018	Implementation of code for audiogram compensatory amplification
06/07/2018 – 22/07/2018	Analogue filter design and implementation
23/07/2018 – 27/07/2018	Full circuit design and implementation and ADC implementation
28/07/2018 – 08/08/2018	Implementation of directionality algorithms and hardware
09/08/2018 – 16/08/2018	Full system testing
17/08/2018	Latest date for laboratory project title changes
18/08/2018 – 26/08/2018	Report writing and poster design
27/08/2018	Staff inspection of laboratory projects
28/08/2018	Open day
29/08/2018 - 02/09/2018	Completion of reports
03/09/2018	Report submission
04/09/2018 - 12/09/2018	Conference preparation
13/09/2018	Laboratory conference

Preliminary Budget & Resources:

The resources discussed below are divided into two sections: resources that must be purchased, and those that are pre-owned or available for use.

Preliminary Budget:
The budget presented in this section provides cost estimates that are based on worst-case pricing per component. Once costs had been deduced for the microphones, headphone jack, miscellaneous components and import duties, all the remaining money was allocated to buying a high quality microprocessor. This is to ensure that exemplary signal processing can be performed. Miscellaneous costs include buying a PCB board and other analog components.

Resource	**Price Per Unit (R)**	**Quantity**	**Total Price (R)**
Microphone	50	4	200
Headphone jack	50	1	50
Miscellaneous	300		300
Microprocessor	400	1	400
Import duties	250	1	250
Total			1200

Pre-owned Resources:

Resource	Quantity
Speaker	1
Earphones	1

Available Resources:

Resource	Quantity	Obtained from
Signal Generator	1	University of the Witwatersrand
Audiogram generator	1	Emoyo

Risks / Mitigation:

The largest risk involved with achieving the target specifications is failing to obtain ethics clearance. This clearance is required for any aspect of the project requiring human participation. As such, without ethics clearance, user feedback as a method of obtaining results will not be possible. Furthermore, specific audiograms from individuals will not be obtainable. As a mitigation of this risk, if ethics clearance is not obtained, pre-recorded audiograms will be used to tune the hearing aid and testing will be performed using electrical equipment.

Since the equipment used to obtain audiograms will need to be obtained from an external company, the group is not in control of its availability. If the equipment is not available, pre-recorded audiograms will be utilized. Furthermore, signal generators will be required for producing single harmonic sinusoidal signals for testing. These devices may not always be available from the Faculty of Electrical and Information Engineering, and as such the device will be booked long in advance to ensure availability.

High quality microphones are not available at local electronics stores. Hence, they will have to be imported which will take an extended period of time. As mitigation, these components will have to be ordered well in advance to ensure that they arrive in time for the start of the laboratory project. In a worst case scenario, lower quality, locally available microphones will have to utilized.

The group has assumed that the earphones that are obtained with the purchase of a cell phone will be of a high enough quality to act as an effective sound conduit. If this is found not to be true, higher quality earphones will have to be purchased.

C PROJECT PLAN REPORT

PROJECT PLAN FOR THE DESIGN AND IMPLEMENTATION OF AN ADAPTIVE HEARING AID

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Abstract: This paper presents the project plan for the design and implementation of an adaptive hearing aid. The three main components of the hearing aid were found to be the compensatory amplification, directionality and full system testing. The developed hearing aid will provide compensatory amplification independently for each ear in accordance with an individual's audiogram. Additionally, the device will have a directionality feature that is manually tunable by the user. In order to fully test the device, the device will undergo both laboratory and clinical testing. All features of the hearing aid will be implemented using MATLAB and Simulink software and will make use of an Arduino Due microcontroller for analog to digital and digital to analog conversions. In managing the project over its full duration, it was found that it was best to divide the project into 11 phases. Certain phases will be completed separately by each group member while others will be performed together, as indicated on the included Gantt chart. The expected completion date of the three components of the project is 21 August 2018. The main risk in the project is that the resources required to test the device are not available, in which case clinical testing will not be possible. The project is expected to cost R1528.

Key words: Audiogram, Compensatory Amplification, Tunable Directionality

1. INTRODUCTION

Hearing loss is a prevalent problem in the world, which is caused by a number of factors including age, disease and trauma [1]. In order to compensate for hearing loss, a digital hearing aid can be used [1]. This report details the design of a digital hearing aid and provides the project plan for the implementation and testing of the system.

2. PROJECT SPECIFICATIONS

2.1 Problem Analysis

Conductive hearing loss is prevalent in the world and may result in members of society experiencing social stigmatization and isolation as well as a decrease in quality of life. Existing hearing aids are expensive and are therefore inaccessible to working class people. For these reasons, there is a need for the development of an affordable hearing aid device that is able to apply compensatory amplification in accordance with an individual's audiograms. Additionally, the hearing aid should have a directionality feature that is manually tunable by the user thus allowing the user to hear in specific directions in a noisy environment.

2.2 Requirements and Specifications

The design, construction and testing of a pair of hearing aids is required. The device to be developed must be able to apply individual-specific compensatory amplification for each ear according to an individual's audiograms.

The hearing aid device also requires an aspect of directionality. The user must be able to manually tune the device to choose a direction for sound amplification, thus allowing them to hear sounds in a chosen direction. The directionality feature must be able to be toggled on and off in order for the user to be able to hear in all directions when needed. Achieving functionality in these areas would be regarded as meeting the specifications for the project.

All implemented features must undergo rigorous testing in order to determine the success rate of the device. Laboratory tests must be conducted to obtain an objective basis for evaluation. Clinical tests will also be conducted for the purpose of obtaining a human opinion of how effective the device is at improving individual hearing.

2.3 Assumptions

A major assumption that has been made is that the KUDUwave audiometer will be accessible to the group. This device will be used to obtain audiograms of human participants without aid and while using the hearing aid device. Another assumption is that the hardware required for the project will be delivered within a reasonable time-frame. Finally, it is assumed that the ear phones that are acquired when purchasing a cell phone are of high enough quality to be used for the hearing aid.

2.4 Success Criteria

The design, implementation and testing of the hearing aid device will be seen to be successful if the following criteria are met:

- Time delay between actual and perceived sound of less than 10ms
- Passband ripple of 1dB for each filter band according to ANSI standards
- Stop band attenuation of -60dB for each filter band according to ANSI standards
- Polar pattern of the device matches the selected polar pattern with less than a 5% error
- Over 60% of participants indicate that their hearing has improved while wearing the hearing aid
- The audiograms obtained while wearing the hearing aid match the expected audiograms within a 5% error

2.5 Constraints

All project groups were allocated a budget of R 1200. While this will direct the group towards the development of a low cost device, it has offered a constraint on the development of the hearing aid since the group will be limited to purchasing hardware of a lower quality compared to that utilized in existing literature. This constraint is likely to have an effect on the overall quality of the hearing aid. An additional constraint that will effect the quality of the hearing aid is the limited time available for the completion of the project.

Audiograms offer an indication of an individuals capacity to hear monotone sounds across the audible frequency range. They do not offer any information on an individual's ability to hear sounds in a noisy environment. Therefore the compensatory amplification feature is constrained to the information provided by an audiogram.

3. PROJECT BACKGROUND

3.1 Literature Review

One of the main functions of a digital hearing aid is to apply gain to an input signal based on an individual's audiogram [2]. This means that signals with different frequencies are amplified by different values, thus allowing for frequencies that the participant has difficulty hearing to be amplified more than other frequencies [3]. In order to divide the frequency range to apply compensatory gain, it is suggested by [2–7] to use a filter bank. This will segment the frequency range into a number of bands allowing for a unique gain, matching the audiogram at that frequency, to be applied to each band [5]. According to the ANSI S1.11 standard, the frequency range 0-20 kHz (the audible range) should be divided into 43 bands each with a 1/3 octave bandwidth [2, 3]. Thus, in order to apply compensatory gain to the entire audible range, 43 filters are required. However, the use of 43 filters is computationally intensive and will result in the introduction of large delays in the system. Since it is stated by [8] and [9] that the group delay of the hearing aid must be less than 10 ms, the use of 43 filters is not a feasible solution. As such, [2, 3, 5] suggest that gain should only be applied to the frequency range of speech (250 Hz to 8 kHz), which implies the use of 18 filters.

A number of different filters are used for the implementation of the filter bank. FIR filters have a linear phase response and are stable and as such they are used by [6] and [9]. However, they are computationally expensive due to the large number of multiplications, and as such [3] proposes the use of IIR filters. [7] suggests the use of a gammatone filter bank which mimics the mechanism of the cochlea, thus allowing for more natural sounding auditory compensation.

When amplifying audio signals, it is necessary to apply dynamic compression to the signals. This is to ensure that the amplified signal never exceeds a person's comfort threshold [3, 7]. As such, each amplified signal must be compared to the comfort threshold and if it exceeds the threshold, the gain must be reduced [3].

Directional hearing aids were created as a method of improving the signal to noise ratio (SNR) in hearing aids [10]. By amplifying the signals in one direction and attenuating the signals in the other directions, the SNR is changed based on the spatial location of the required signals relative to the unwanted signals [10]. This can assist in improving a user's speech recognition in a noisy environment [10].

In order to implement directionality, two different types of microphones can be used: omni-directional and directional [10, 11]. To achieve directionality with omni-directional microphones, two must be used, with one placed at the back of the hearing aid and one at the front [10, 11]. By using software created time delays, a software directional microphone is created [12]. If a directional microphone is used, one microphone is required [10]. Most hearing aids allow the user to switch between a directional and an omni-directional mode [10]. If a directional microphone is used, an omni-directional microphone is also required for when the user selects omni-directional mode [10].

There have been numerous implementations of the directionality of a hearing aid. An adaptive directionality system was implemented by [11] and [13]. With an adaptive system, the hearing aids focus automatically in the direction of the sound with the greatest signal to noise ratio [11]. By making use of two omni-directional microphones, adaptive directionality is implemented by cross-correlating the signals from the front and back microphones. The cross-correlated signals are then used to determine an adaptive weight for altering the position of the nulls (the angles of greatest attenuation) [11]. However, research has not been done into changing the directionality of the device to a direction selected by the user.

In order to achieve directionality, different delays between the microphones are used. By changing the delays, different attenuation patterns can be acquired [10, 12]. The three most commonly used patterns are cardioid, hyper-cardioid and bi-directional [10].

4. SYSTEM DESIGN

A block diagram of the full hearing-aid system is given in *Figure 1.*

The microprocessor block consists of all of the signal processing required for the functioning of the hearing aid. The required signal processing is expanded upon in *Figure 2.*

The software components of the system will be programmed using MATLAB and Simulink and thereafter burned onto the Arduino Due microcontroller. By using Simulink, the Arduino can be programmed to function autonomously without needing to be connected to a laptop.

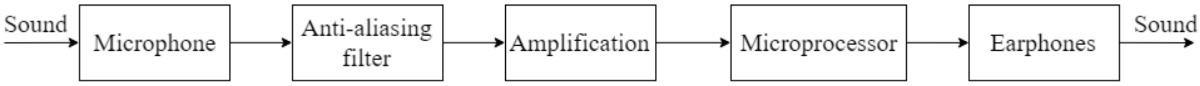


Figure 1: Full system block diagram

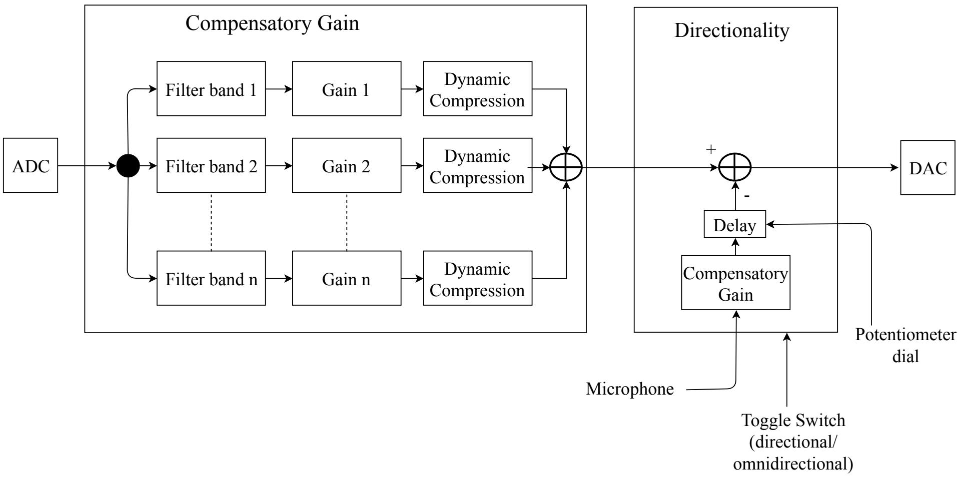


Figure 2: Block diagram of the software components of the hearing aid

4.1 Compensatory Amplification

In performing the auditory compensation, amplification will only be applied to the frequency range of speech. As such, 18 filters will be used, each with a bandwidth of 1/3 of an octave. Since phase relationships are crucial in implementing directionality, FIR filters will be used for their linear phase response. In complying with the ANSI standards for filters, the filters will be designed such that the stop band attenuation is -60 dB. Since the bandwidth (in Hz) of each filter is different, the order of the filters will vary. The designed filters will be bandpass filters which attenuate all but a selected range of frequencies.

The input signal from each microphone (in directional mode) and from the front microphone (in omnidirectional mode) will be passed through the filter bank. This will result in the 18 bands of the signal being isolated from one another. Thereafter, a gain matching the audiogram for that frequency range will be applied to each band. In doing so, each signal in the band is multiplied by the gain. Thereafter, each signal is compared to the comfort threshold, and if it exceeds the threshold, its amplitude is reduced until it equals the threshold. Finally, the resultant signals will be added together to produce the compensated signal.

In order to calculate the required gains for each frequency band, the audiogram requires linear interpolation to increase the resolution. This is necessary since an audiogram only consists of 12 points. In order to ensure that the shape of the audiogram is maintained, a large number of points are required (greater than 1000). The gain for each frequency band will then be selected as the gain on the interpolated audiogram corresponding to the nominal centre frequency of the band.

4.2 Directionality

In order to achieve tunable directionality, a software directional microphone will be used. In the software directional microphone, the signals from two omni-directional microphones are captured and combined in order to create a directional pattern [12]. The two microphones will be situated on the front and back of the hearing aid device. The desired directional pattern will be acquired by delaying the signal acquired by the rear microphone and subtracting it from the signal acquired from the front microphone. This configuration is known as a differential microphone array [12].

The directionality feature will be designed such that the user will be able to specify the desired direction of amplification using a potentiometer dial. The analog output voltage of the dial will be read by the ADC of the microcontroller and converted to a time delay. This time delay will be applied to the rear microphone signal and then the delayed signal will be subtracted from the front microphone signal. Thereafter, the direction of maximum gain will be in the same direction that the potentiometer dial is pointing. A block diagram showing this process can be found in *Figure 3*.

In order to create a software directional microphone, certain physical configurations need to be adhered to. One of these considerations is that of the inter-microphone distance *d*. This distance must be equal to half of the wave length of the frequency of interest [12]. Since the frequency range of interest is that of speech, the half wavelength of the centre of the speech range is chosen as the separation distance between the two microphones. Therefore this distance will be 42.875 mm assuming that the speed of sound, c, is 343 $m.s^{-1}$.

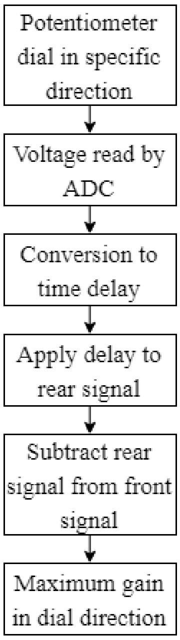


Figure 3: Block diagram of user tunable directionality feature

The hyper-cardioid directional pattern is expected to be sufficient to amplify sounds coming from the the direction of interest to the user while attenuating noise from other directions. If it is assumed that the front of the hearing aid device is 0° , the time delay is required to equal $\frac{d}{c}$ to ensure that maximum amplification in the 0° direction and maximum attenuation in the 180° direction are obtained [12]. The polar pattern plot for this case can be found in *Figure 4*. The direction of the polar pattern maximum will be manipulated by varying the time delay using the potentiometer dial.

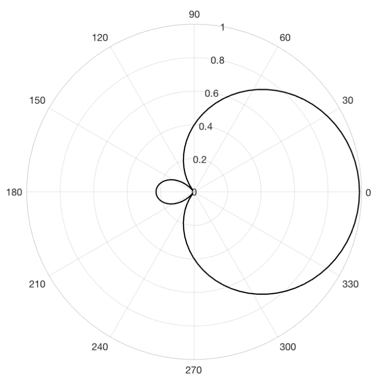


Figure 4: A hyper-cardiod polar plot

4.3 Hardware

Analog circuitry is required to acquire the sound signal, condition it and convert it to digital. To acquire the signal, two omni-directional microphones are used. The front microphone is the primary microphone which is used alone when the omni-directional mode is selected. The rear microphone is used along with the front microphone when directional mode is selected. The signals from the microphones are amplified (using the MAX9814 audio amplifier) and then passed through an eighth order switched capacitor anti-aliasing filter chip (MAX292). The filter has a cutoff frequency of 25 kHz, which was selected to ensure that the entire audible range is left unattenuated, while still attenuating the unwanted signal components. Once the signals have been filtered, they are converted to digital by the analog to digital converter (ADC) of an Arduino Due microcontroller. Thereafter, the signals undergo signal processing, as discussed above, before being converted back to analog signals by the Arduino's digital to analog converters (DAC). The analog signal corresponding to each ear is then relayed to the appropriate ear using earphones which interact with a headphone jack. The earphones to be used are those that are acquired when purchasing a cell phone. These were selected as they are readily available, and are thus inexpensive.

In order to tune the directionality of the hearing aid, a potentiometer, which acts as a dial, will be used. The signal from the potentiometer is also converted to digital by the ADC for signal processing. The hearing aid will include a power button to turn the device on or off, and a toggle switch to switch between directional and omnidirectional amplification modes. The entire system will be powered using rechargeable batteries.

A circuit diagram of the hardware components of the system is given in *Figure 5*.

4.4 Laboratory Full System Testing

In order to obtain objective results that are free from human bias, the system will be tested in a laboratory using a signal generator to generate pure tones. By generating pure tones of specific frequencies and feeding these tones to the hearing aid, the ability of the device to apply compensatory amplification, and to apply amplification to the correct frequencies, can be quantified. The frequency of the sinusoidal pure tone signals will range from the lower to the upper bounds of the frequency range of speech. Additionally, high frequency noise will be added to the signals such that the performance of the device can be evaluated in the presence of noise.

In order to test the directionality of the device, one speaker will be placed in a fixed position. The hearing aid (without the earphones) will be placed in the centre of a rotating platform with the dial facing towards the speaker. A pure tone signal is played from the speaker and the resulting output signal from the hearing aid is saved. The platform is then rotated by 5° and the procedure is repeated. This is done for a 360° range. Once signals have been acquired corresponding to each 5° increment, a polar plot of the signals will be created. In the polar plot, the angle of the signals is taken as the angle of the hearing aid relative the dial (the 0° reference point). This will allow for an analysis of the degree of

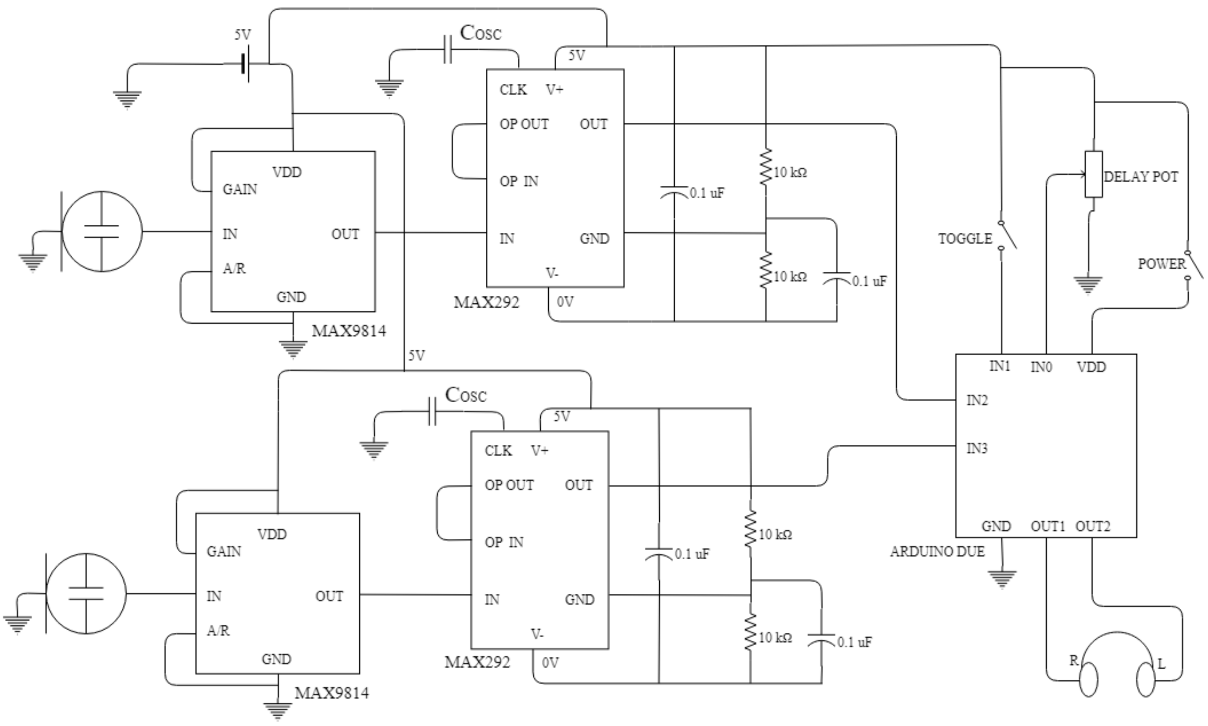


Figure 5: Circuit diagram for the system

amplification and attenuation of signals in all directions with the directionality being fixed in one direction. This procedure is illustrated by *Figure 6*.

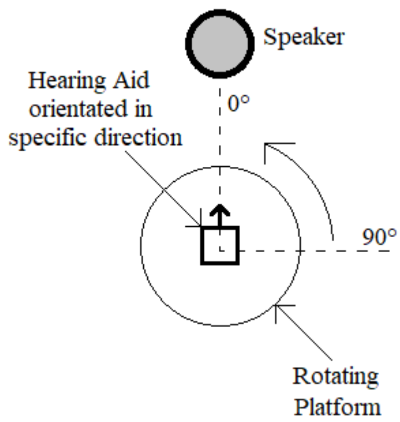


Figure 6: Illustration of the procedure for testing directionality in a laboratory setting

4.5 Clinical Full System Testing

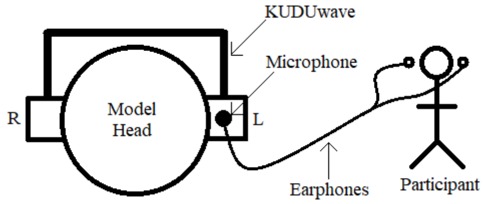
In order to verify the performance of the hearing aid, clinical tests are required. This will allow for a quantification of the value that the hearing aid provides to the participant.

The method of the clinical testing is as follows. First, staff from the School of Electrical and Information Engineering will be asked to participate in the clinical testing. Prior to performing the tests, an audiogram will be collected from each participant using the KUDUwave headset. A hearing aid will then be customised for each participant. Thereafter, a second audiogram will be obtained with the participant wearing the hearing aid. Since the KUDUwave cannot be used on top of earphones, the microphone will be mounted on a model head (made out of polystyrene) and the KUDUwave headset will be placed on top of the model head. The participant will then wear the earphones which receive the processed signals and a second audiogram will be obtained. This configuration is illustrated by *Figure 7*. The two audiograms will then be compared in order to assess the effectiveness of the compensatory amplification in the device.

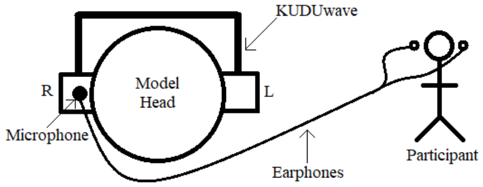
In order to test the directionality of the device, eight speakers will be set up in a circle, separated by 45° angles. The participant will be asked to sit in a chair which faces 0° while wearing the hearing aid. Pure tone sounds will be played from a single speaker at a time and the participant will be asked to tune the dial controlling the directionality until the sound is at its loudest. The angle of the dial will then be recorded and compared to the angle of the speaker, thus allowing for a quantification of the error present in the directionality.

5. PROJECT MANAGEMENT

This section details the project management to be used in the project, including the project timeline and the management of team work in the project.



(a) Testing the left ear



(b) Testing the right ear

Figure 7: Illustration of the procedure for testing compensatory amplification

5.1 Project Timeline and Work Breakdown

The project has been divided into 11 development phases, which encapsulate the entire project, from ethics clearance to the presentation. The first three phases occurred prior to the submission of this document. The phases are detailed in the following sections.

The project Gantt chart is given in *Figure 8*. In the chart, KJB refers to Kayla-Jade Butkow and KDS refers to Kelvin da Silva.

Phase 1: Obtain Ethics Clearance

This phase involved obtaining ethics clearance for the project. This clearance is necessary for clinical testing of the hearing aid, using human input. The procedure for obtaining clearance involves setting out the testing procedure, creating consent forms and creating the questionnaires for the participants to complete. This phase was completed between the 29$^{\text{th}}$ March and the 6$^{\text{th}}$ April. Ethics clearance was granted on the 11$^{\text{th}}$ May. Both partners were involved in the completion of this phase.

Phase 2: Research

This phase involved performing the research for the project. This included research into compensatory gain, directionality, filtering and the conversion of digital signals to sound signals. This phase occurred from the 25$^{\text{th}}$ June to the 1$^{\text{st}}$ July. However, it must be noted that research will be an ongoing process throughout the entire duration of the project. Kayla-Jade performed research into gain and filtering and Kelvin investigated directionality and conversions. Thereafter, the research was shared and both partners familiarised themselves with both sets of research.

Phase 3: Hardware Design

In this phase, the circuit diagrams required in the project were designed. The overall system circuit diagram is given in *Figure 5*. This phase was completed between the 2$^{\text{nd}}$ July and the 6$^{\text{th}}$ July. The two partners performed this phase together.

Phase 4: Construction of Hardware

This phase involves the construction of the circuit that was designed in Phase 3. The circuit will be prototyped on a breadboard before being finalised on a PCB board. The phase will begin on the 9$^{\text{th}}$ of July and will end on the 16$^{\text{th}}$. Kelvin will perform the construction of the hardware.

Phase 5: Implementation of Compensatory Amplification

In this phase, the compensatory amplification aspect of the hearing aid will be implemented. This phase has numerous sub-phases:

- Sub-phase 1: Interpolation of audiograms and selection of the gains for each filter based on the audiogram
- Sub-phase 2: Development of the filter bank
- Sub-phase 3: Dynamic compression
- Sub-phase 4: Code for ADC and DAC
- Sub-phase 5: Optimisation of code from the previous stages to allow for real-time signal processing

The first two sub-phases will be implemented by Kayla-Jade. Kelvin will implement sub-phases 3 and 4. The two partners will implement sub-phase 5 together. The phase will be completed between the 9$^{\text{th}}$ of July and the 27$^{\text{th}}$ of July.

Phase 6: Implementation of the Software Required for Directionality

In this phase, the software required to implement directionality will be implemented. This software will interface with the dial built into the hardware to tune the direction in which the sound must be amplified. The two partners will implement this phase together. This phase will be completed between the 28$^{\text{th}}$ of July and the 5$^{\text{th}}$ of August.

Phase 7: Laboratory testing of the full system

Following the implementation of the entire system, full system testing is required, as described in *Section 4.4*. Both partners will perform the laboratory testing from the 6$^{\text{th}}$ until the 10$^{\text{th}}$ of August.

Phase 8: Clinical testing of the full system

Phase 8 involves the testing of the audiogram in a clinical setting, using human participants. The procedure to be implemented in this phase is that described in *Sec-*

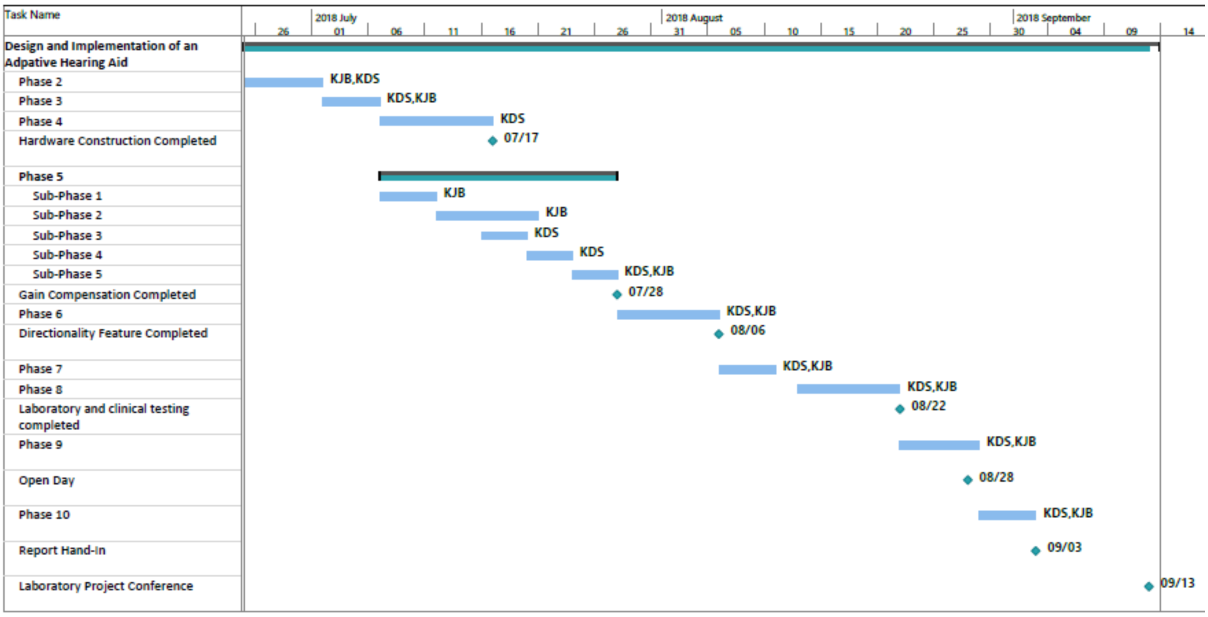


Figure 8: Gantt chart for the project

tion 4.5. Both partners will perform the clinical testing. The phase will occur between the 13$^{\text{th}}$ and the 21$^{\text{st}}$ of August.

Phase 9: Demonstration

This phase involves the preparation for open day. This includes creating a poster and a video explaining the project. It also includes preparing the demonstration for open day. Both partners will be involved in all aspects of the phase. Phase 9 will be completed between the 22$^{\text{nd}}$ and the 28$^{\text{th}}$ of August.

Phase 10: Documentation

This phase involves the writing of the final project report. The report will be worked on throughout the entire project, but in this phase, it will be the sole focus. The partners will each work on their reports individually. Phase 10 will be completed from the 29$^{\text{th}}$ of August to the 2$^{\text{nd}}$ of September.

Phase 11: Presentation Phase 11 is the final phase of the project which involves the preparation of the presentation for the laboratory project conference. This will be done by the partners together. The phase will occur between the 3$^{\text{rd}}$ and the 12$^{\text{th}}$ of September.

5.2 Team-work and Work Allocation

The team will be loosely following Agile methodologies in the implementation of the project. In doing so, there will be a daily stand-up meeting where the work that each member did the previous day will be discussed. This will allow for any potential problems to be fixed as soon as they occur. It will also help to ensure that the project schedule is being followed. If the project falls behind schedule, the stand-up meetings will allow for the schedule to be re-evaluated before the backlog gets too severe.

The project management application *Trello* will be used to assign tasks to each member of the team. The team will also use the application to record any problems they encounter and to indicate which tasks they have completed.

While each member has their own tasks that they will work on, the other team member will be available to offer assistance at any time.

6. PROJECT RISK ANALYSIS

Since the equipment used to obtain an audiogram (the KUDUwave) will need to be obtained from Emoyo, the group is not in control of its availability. If the equipment is not available, pre-recorded audiograms will be utilized. This will imply that clinical testing will not be possible, since audiograms are needed for each participant in the study.

Furthermore, signal generators are required for producing sinusoidal signals for laboratory testing. These devices will be obtained from the School of Electrical and Information engineering, and as such, might not be available during the week when testing is scheduled to occur. To mitigate this risk, the required equipment will be booked well in advance to ensure its availability.

The earphones to be used with the hearing aid are earphones that are obtained with the purchase of a cell phone. An assumption is being made that these earphones are of a high enough quality to effectively deliver the sound. If this is found to be an incorrect assumption, high quality earphones will need to be purchased, which poses the risk of exceeding the group's budget.

A large risk in the project is that of not having enough time to complete all the phases. In this situation, the phase that would be most severely compromised would be the clinical testing. However, since laboratory testing would have already been performed, the device would have been thoroughly tested. As such, the omission of this phase is not detrimental to the success of the project.

7. PROJECT COST ANALYSIS

A breakdown of the components and resources required for the implementation and testing of the hearing aid are given in *Table 1*. The table includes the cost of each component and the total cost of the system. The system is estimated to cost R1528, with the cost of components to be purchased being R853.

8. CONCLUSION

The design and project plan for the implementation of an adaptive hearing aid has been presented. The main components of the project are the implementation of compensatory amplification, directionality and the development of the required hardware. The final essential component is the testing of the device, consisting of laboratory and clinical testing. The project will be implemented using MATLAB and Simulink and will make use of an Arduino Due microcontroller. The overall cost of the project is expected to be R1528. The main risk to the project is the lack of availability of the resources needed to test the device.

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Table 1: Cost breakdown for the digital hearing aid

Purchased Components:

Component	Quantity	Cost (R)
MAX9814	2	240
Headphone Jack	1	17
MAX292	2	296
Miscellaneous	-	300
	Subtotal:	853

Pre-owned Resources:

Component	Quantity	Cost (R)
Arduino Due	1	675
Earphones	1	-
Speakers	8	-
Signal Generator	1	-
KUDUwave	1	-
	Subtotal:	853

Total cost: **R1528**

D FILTER BANKS

This section contains bode plots for the simulated and hardware filter banks, as well as circuit diagrams for the hardware filters.

Figure D1 provides the bode plot for the simulated filter bank. It is clear from the figure that there are 16 bandpass filters each with a 1/3 octave bandwidth. Each of the filters is 14$^{\text{th}}$ order.

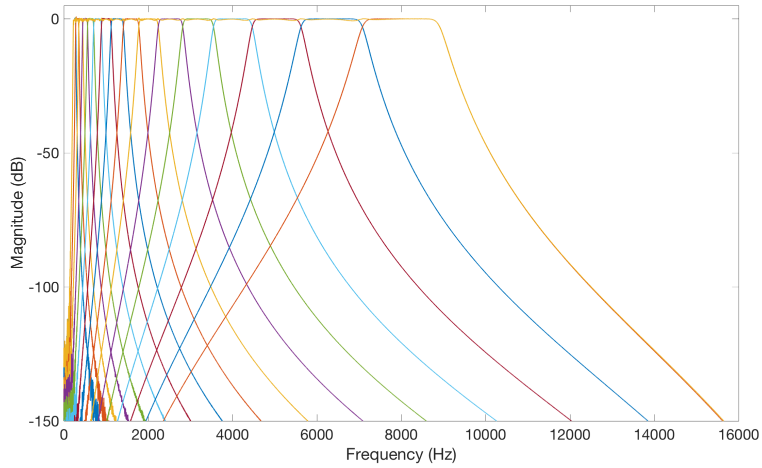

Figure D1: Bode plot of the full 1/3 octave filter bank implemented in the simulation

The bode plots for the analog filters are given in *Figure D2*. The filters are 2$^{\text{nd}}$ order, Butterworth bandpass filters. The circuit diagrams for the two filters are provided in *Figures D3* and *D4*.

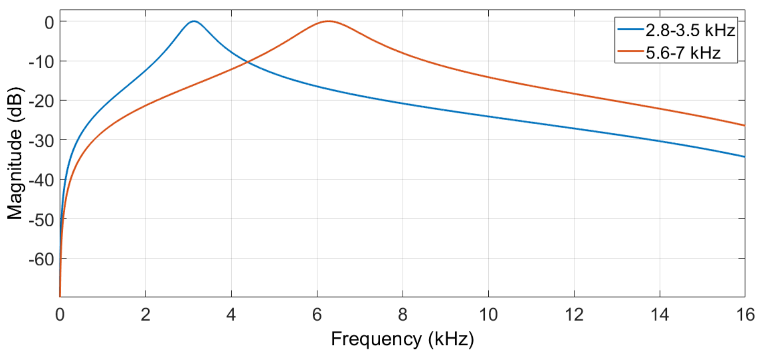

Figure D2: Bode plot of the partial 1/3 octave filter bank implemented in the hardware

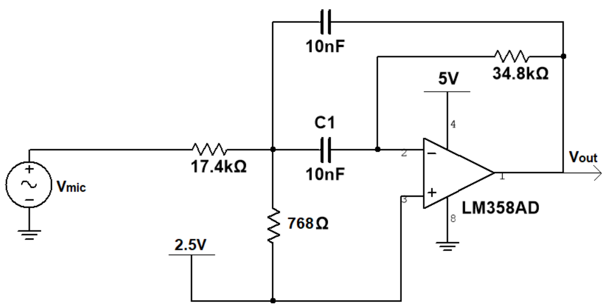

Figure D3: Circuit diagram for the 2.8-3.5 kHz bandpass filter

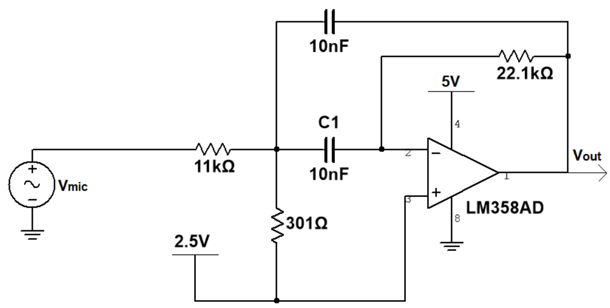

Figure D4: Circuit diagram for the 5.6-7 kHz bandpass filter

E GRAPHICAL USER INTERFACE

This section provides images of the two tabs in the GUI. The Compensatory Amplification tab, as seen in *Figure E1*, contains a fast fourier transform (FFT) of the input frame to the hearing aid, and it's output signal. The two plots allow the user to visualise the effects of the compensatory gain, whilst simultaneously hearing the effects.

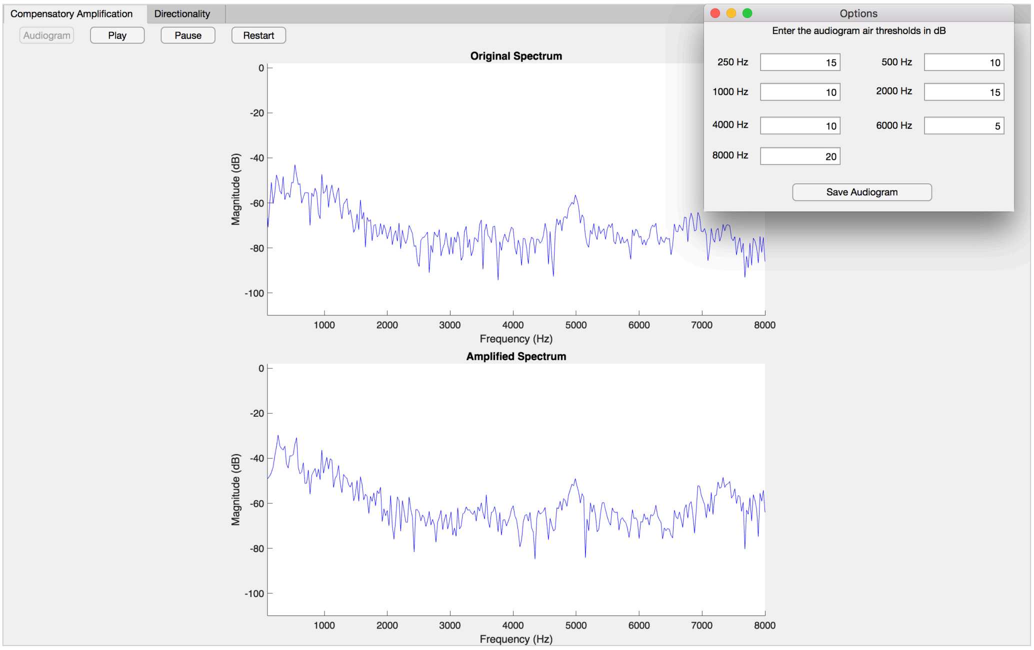

Figure E1: Compensatory amplification tab of the GUI

The Directionality tab in *Figure E2* allows the user to tune the direction in which they wish to hear and to see the resultant polar plot. In the figure, the user has tuned the dial to 0°.

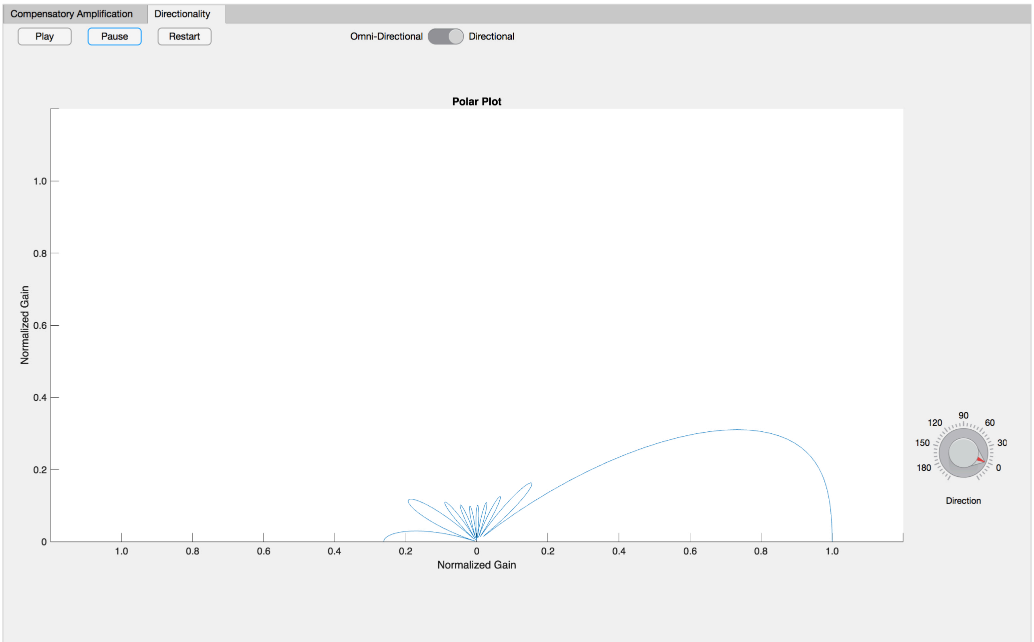

Figure E2: Directionality tab of the GUI

F SIMULATION RESULTS

This section provides the results obtained from the hearing aid simulation.

Figure F1 provides an FFT of a frame of an input signal to the hearing aid, and the compensated output signal from the device. In the figure, the vertical grey lines represent the -3 dB points of the filter bank. The lines were placed on the figure so that the effects of the different amplifications to the different frequency bands could easily be discerned.

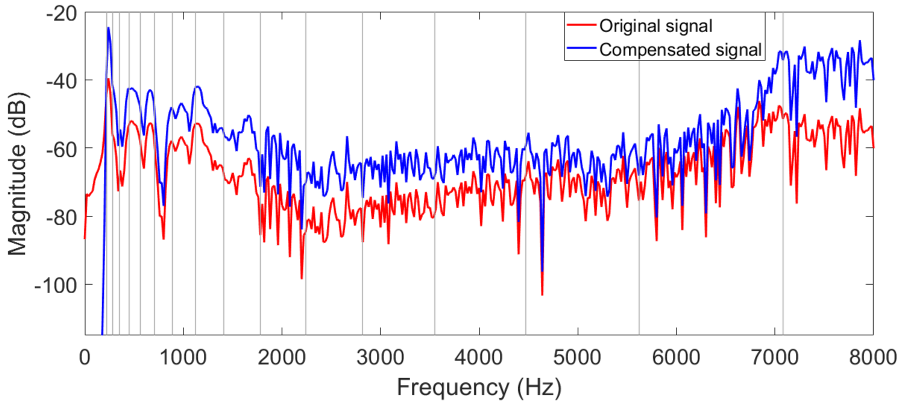

Figure F1: FFT of the input signal and the output signal from the hearing aid with compensatory gain applied

The amplifications seen in *Figure F1* were applied based on the audiogram in *Figure F2*.

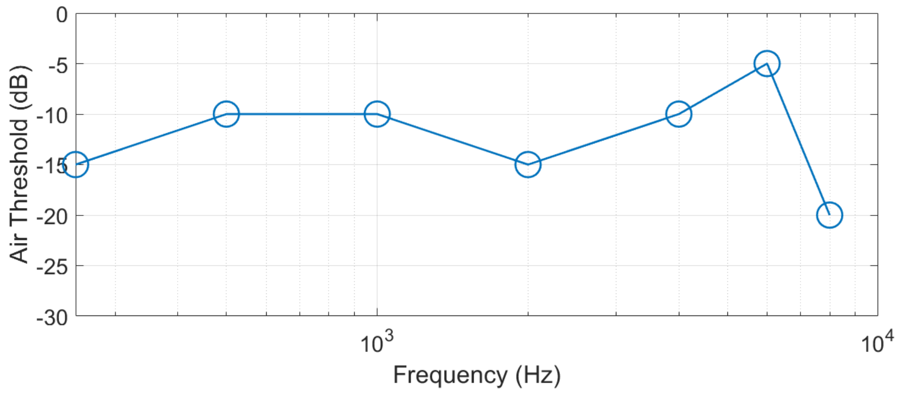

Figure F2: Audiogram for which compensatory amplification is applied

Polar plots of the directionality feature are given in *Figures F3* to *F5*. In each figure, polar plots are provided for input signals of 1 kHz, 3.15 kHz and 6.3 kHz. This was done such that the frequency dependence of the directionality feature could be seen. It is clear in all of the figures that the highest precision, in terms of angle and breadth of the beam, is achieved for input signals with a frequency of 3.15 kHz. In *Figure F3*, the user has steered the beam to 0°. Likewise, in *Figures F4* and *F5*, the beam has been steered to 60° and 90° respectively.

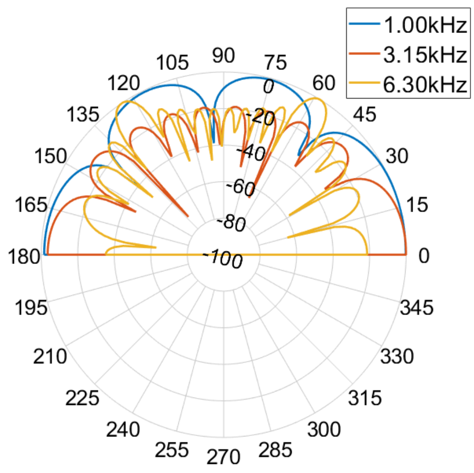

Figure F3: Polar plots of the simulated hearing aid with the direction selected to be 0°

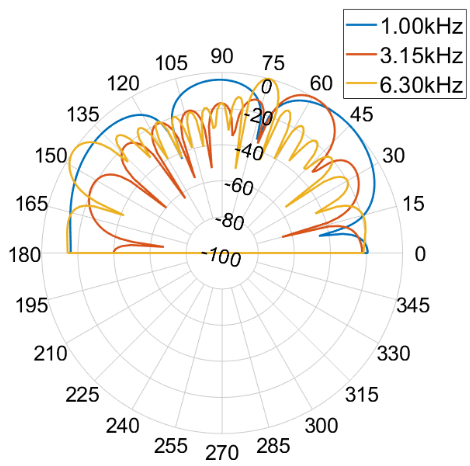

Figure F4: Polar plots of the simulated hearing aid with the direction selected to be 60°

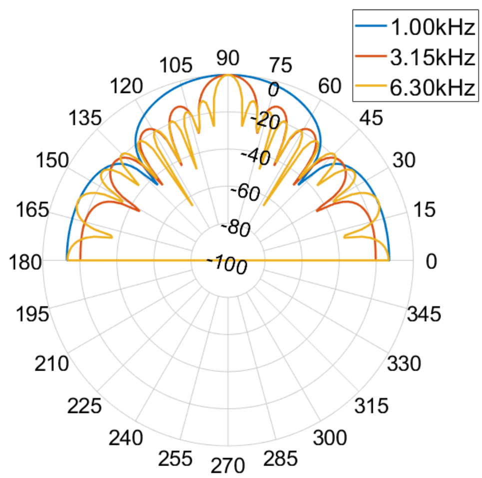

Figure F5: Polar plots of the simulated hearing aid with the direction selected to be 90°

G HARDWARE RESULTS

The results of the hardware testing are provided in this section.

Figure G1 provides a photograph of the final device. The device is 20x10cm, and is battery powered. The audio is relayed to the user using headphones.

Figure G1: Photograph of the final device

FFTs of the input signal to the hearing aid and the output signal from the hearing aid are given in *Figure G2*. The input signal to the hearing aid was a 3.15 kHz sinusoidal wave. In the image, the red line represents the output from the hearing aid when no compensatory gain is applied. The black line represents the output from the hearing aid when a gain of 6 dB was applied to the 2.8-3.5 kHz band, and the blue line when a gain of 6 dB was applied to the 5.6-7 kHz band.

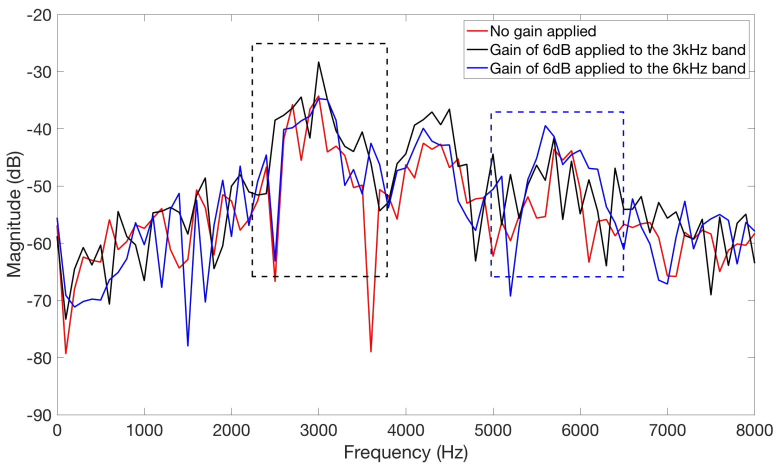

Figure G2: FFT of the output signal from the hearing aid with varying amplifications

Polar plots of the directionality feature are given in *Figures G3* to *G6*. In order to quantify the error present in the feature, simulated polar plots were created using four ideal omni-directional microphones. In the figures, the red plot is the simulated plot and the blue plot is the measured result. From the figures, it is evident that the measured results closely resemble the simulations, but that the steep attenuations exhibited by the simulation are absent in the measured results.

In *Figure G3*, the user has steered the beam to 0°.

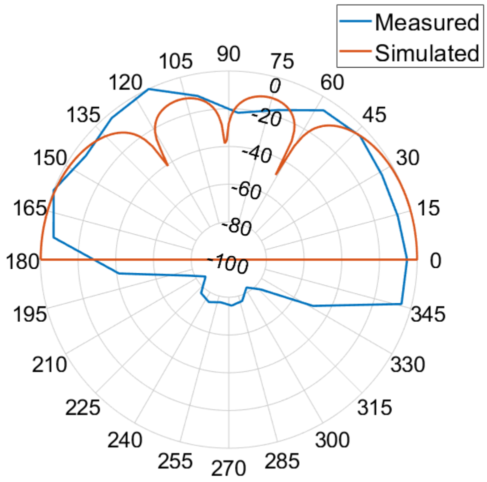

Figure G3: Polar plot of the hearing aid with the direction selected to be 0°

In *Figures G4* and *G5*, the beam has been steered to 60° and 90° respectively.

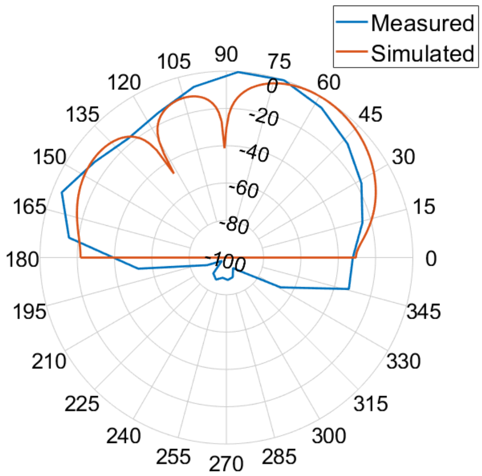

Figure G4: Polar plot of the hearing aid with the direction selected to be 60°

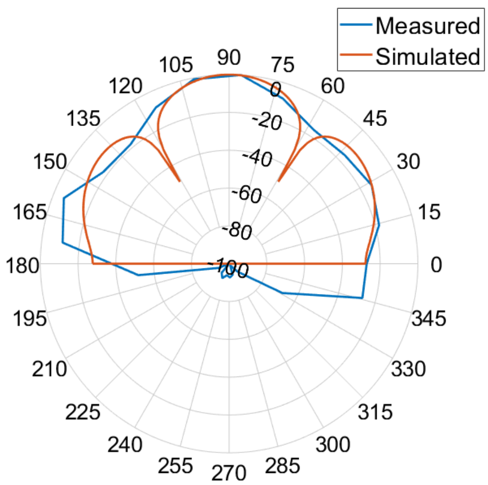

Figure G5: Polar plot of the hearing aid with the direction selected to be 90°

Figure G6 provides a polar plot of the output from a single microphone and that of the hearing aid when the device is in omni-directional mode. From the figure, it is clear that the microphone does not behave like an ideal omni-directional microphone, and that the behaviour of the hearing aid closely mimics that of the microphone. The non-ideal behaviour exhibited by the microphones results in the introduction of errors into the system.

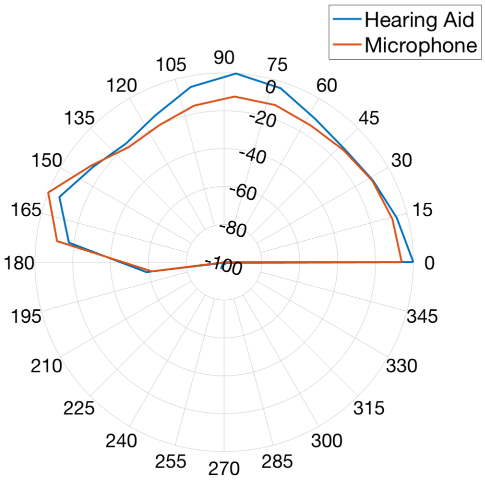

Figure G6: Polar plot of the hearing aid in omni-directional mode

H MINUTES

MEETING 1

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

Obtain an overview of the requirements for the project outline specification

Minutes of meeting:

Start date: 09/03/2018
Start time: 13:00

Requirements for the document:

1. Specifications of the project
2. Milestones: Using Gantt Chart or timeline
3. Preliminary budget and resources
4. Risks or Mitigation: What do you do if you can't get a piece of equipment (what do you fall back on)

The minimum specifications for the project are as follows:

- There must be two hearing aids (one per ear)
- Each hearing aid must correct for the audiogram for that ear - To do this, you tweak the filter for each ear
- Directionality feature comes in mainly in processing - It is filtering of a signal based on direction (can possibly be done using cross correlation). This filtering changes the signal to noise ratio
- Ideas for directionality: The user changes the direction using a potentiometer or the user always hears best in the direction they are facing
- You must be able to turn off directionality
- Testing the device using humans is not specified in the brief of the project, but it can be done

Testing the device:

- Can use a signal generator to produce pure tones (single harmonic) and then examine the resulting signal after processing
- To test directionality, the sound source can be moved and the SNR examined
- It is also necessary to produce signals in the presence of noise and test using them

Audiogram:

- An audiogram is the response of each ear to different frequencies
- The magnitude in dB is with reference to some standard
- The layout of an audiogram is given in *Figure 1*

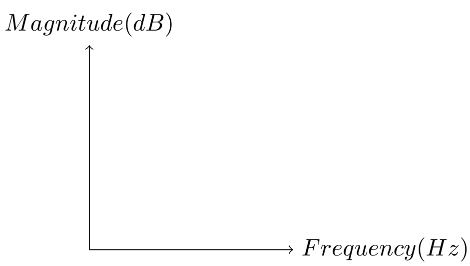

Figure 1 : Audiogram

Budget:

- Is a huge constraint in the project

- It doesn't include any things you have in the house such as headphones

Ethics Clearance:

- For open day, you don't need ethics clearance
- We need to decide how many people we would like to use to test the device
- We need to specify who the people are (colleagues and lecturers)
- Provide a protocol for testing

MEETING 2

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

Review ethics clearance application

Minutes of meeting:

Start date: 03/04/2018
Start time: 13:00

The ethics form was reviewed and appropriate changes were made

MEETING 3

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

Obtain an overview of the requirements for the project planning document

Minutes of meeting:

Start date: 20/06/2018
Start time: 13:00

Project Planning Document:

- Needs to Gantt chart which shows how the work will be split
- The document is a plan of the project assuming that everything is going to go exactly according to plan
- Needs to detail the full project implementation - can possibly include circuit diagrams and flow charts or algorithms

Engineering notebook:

- Use it as a diary - include dates and times for all entries
- Write down anything that you've done on that day - eg. sketch of a circuit, phone number of a supplier
- Put summaries of your minutes in the engineering notebook

The aim of the project is to develop a low cost hearing aid that is very simple to use. This means that the device only has a few buttons and is not fiddly. If we can achieve this, we can possibly work with Emoyo to develop a new low cost hearing aid solution.

The optimal directionality for the project would be that you are able to tune the direction using a dial, able to hear in the direction you are facing and able to turn of the directionality. This could be implemented using a combination of directional and non-directional microphones.

People to contact for the project:

- James Braid - Contacts in speech and hearing if necessary
- Keegan Malan - Ask for use of Kuduwave to test device and to obtain audiograms

To Do:

- Remind Prof. Rubin to email us audiograms
- Remind Prof. Rubin to enquire about a trial version of MATLAB for the project

Going forward, meetings are to be held once a week. Prof. Rubin will be away from the 29/06/2018 until 22/07/2018 and during this time, skype meetings will be held.

MEETING 4

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Dr. Derek Nitch

Agenda:

Consult regarding plans and intentions for the hearing aid directionality using array theory.

Minutes of meeting:

Start date: 06/07/2018
Start time: 10:00

Microphone Array:

- The acoustic beam of the microphone array will not be able to be steered if the array only has two microphones.
- The array must have a minimum of four microphones to enable beam steering. Having more microphones allows for a more directive beam and improved resolution when steering.
- It was recommended that incoming sounds be constrained to the horizontal plane when performing simulations.
- Beam steering ability should be restricted to 180° in front of the user.
- Inter-microphone spacing, d, must be such that the condition $d < \frac{c}{2f_{max}}$ is met. c is the propagation speed of sound (343 $m.s^{-1}$) and f_{max} is the maximum frequency of interest.

Simulating array configurations:

- Circular array with 4 or 6 microphones: There appears to be insignificant differences between each case. Azimuth polar pattern is acceptable but there is always two dominant beams in opposing directions.
- Semi-circle array with posteriorly placed plate: This configuration did not prove to be suitable as there were always more than two dominant beams. Posterior plate appears to be a good idea to incident sound waves from behind the array.
- Uniform linear array with posteriorly placed plate: This appears to be a promising configuration. There is a single main beam that remains dominant when steering it in a range of directions of $0° - 180°$. The polar pattern obtained is acceptable when four microphones are used.

The minimum frequency range required to detect speech intelligibly must be determined to allow for optimal inter-microphone spacing. This will allow for a more consistent acoustic beam throughout all steering angles and a constant inter-microphone spacing.

To do:

- Find the minimum frequency range required to detect speech intelligibly and use the upper bound of this range for the spacing of the microphones.
- Construct a weight table that indicates the phasing of each microphone required to focus the main acoustic beam in a specific direction. This should be done for the centre frequencies of each filter band.
- Obtain polar plots to verify the relevant weightings.

MEETING 5

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

- Discuss Arduino Due RAM limitations
- Discuss limiting frequencies of interest to speech frequency range and limiting directionality steering range
- Work station allocation

Minutes of meeting:

Start date: 18/07/2018
Start time: 10:00

Limiting frequency range and directionality steering range:

- It is acceptable to limit the frequency range of interest too 0 - 8000 Hz and to limit directionality steering to the 180° plane in front of the user.

RAM limitations:

- It is expected that there will be a shortage of RAM if the designed filter bank is implemented digitally.
- Implementing the complete filter bank is seen to be important because it is an ANSI standard.
- It was concluded that the implementation of the hearing aid would require a dedicated IC. Additionally, the project time line is too short and the budget is too small to implement a completely modular design.
- It has therefore been decided that a complete, non-real time software design must be implemented. Thereafter a limited electronic/hardware version using simpler electronics will be implemented.

MATLAB software system design:

- Full implementation of ANSI filter bank.
- Utilize pre-recorded audio files.
- Simulate detecting sound from different directions and show that directionality has been achieved.
- GUI for directionality: interactive dial to specify direction of desired hearing and polar plots of directivity.
- GUI for compensatory gain: Frequency spectrum of original and gained audio signals.

Hardware/electronic system design:

- Pick three of the filter bands of the frequency bank and implement theme as analog filters.
- Gain relevant signals using the micro-controller.
- Focus on utilizing the processing capabilities of the micro-controller on directionality.

To do: It has been recommended that the entire software solution must be completed first. Thereafter, the compensatory gain portion of the hardware version must be completed. The directionally feature of the hardware version should be implemented last.

MEETING 6

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Dr. Derek Nitch

Agenda:

Obtain advice on how to address the poor quality audio output when an audio sample is simulated from different directions in MATLAB.

Minutes of meeting:

Start date: 24/07/2018
Start time: 10:00

Simulated sound collection by array:

- Discussed how the collection has already been implemented using the *phased.WidebandCollector* MATLAB function.
- Check to see if there was any distortion of original audio file immediately after the microphone array. There was no distortion.

Filter bank:

- Demonstrated the performance of the filter bank evaluating distortion of the audio before and after the filter bank. There was a clicking noise that was present in the audio.
- Clicking noise was improved by passing the entire audio stream through the filter bank at once compared to when passed through in frames.

Audio with applied weights passing through filter bank:

- Through demonstration, the audio output is severely distorted even when the collected audio is multiplied by 1.
- It is found that each filter needs to be reset/released before passing consecutive frames through the filter.
- Weights were being applied to collected signals incorrectly as frequency domain weightings were being applied to time domain signals
- Weights (currently imaginary numbers) representing phase shifts have to be converted to time delays. It was suggested that by doing this and adding delayed signals of all filters bands appropriately, the distorted output audio would be improved.

Analog filter bank: When we pick the filter bands, put them as far apart as possible. This will mean that the order of the filters does not have a large effect on their performance.

To do:

- Convert microphone weights for all angles and centre frequencies to time delays.
- Implement time delays as array element shifts and add all arrays together appropriately.
- Vectorise all the for loops in order to speed up the execution time of the code

MEETING 7

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

- Weekly progress report
- Polar plot frequencies for the GUI
- How do we calculate the required gain for the compensatory amplification?
- How closely does the software have to match the hardware?

Minutes of meeting:

Start date: 25/07/2018
Start time: 12:00

Polar Plot: Pick a few frequencies (5) and plot them to show the polar patterns at different frequencies

Compensatory gain:

- The required gains are just the gains on the audiogram

- We have implemented the compensatory gain correctly

Software vs hardware:

- We can use any number of microphones in software - the hardware and software number of microphones do not have to match
- We don't have to implement 3 of the software filters in hardware
- However, it is not ideal to use 3 bands that span the entire frequency range of speech as this will not accurately show how the compensatory amplification works
- It is fine to use three bands that are very far apart
- We should expand as much as we can in the software to create a full hearing aid simulation

Report:

- It is essential that we talk about the cost and time limitations
- We must have a discussion about where the project needs to go to be implemented in real life (in the future recommendations section)

To do: At the end of the project, remind Prof. Rubin to thank Kirsten at Optinum for the MATLAB licences.

MEETING 8

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

- Dynamic compression
- Title changes
- Planning report feedback
- Compensatory amplification for both ears vs one ear

Minutes of meeting:

Start date: 01/08/2018
Start time: 10:00

Planning report:

- Report was good in terms of style and content
- Content is not really relevant since the re-scope

Title changes:

- Option 1: "Towards the design of an adaptive hearing aid"
- Option 2: "An investigational study into the design of a low-cost adaptive hearing aid"
- Meaning of adaptive in this context: Adapts to the user's audiogram and to their environment (directional/omni-directional)
- Email Hugh Hunt for the title change procedure
- Option 2 is the best option

Dynamic Compression:

- Talk to Prof Hanrahan

Compensatory amplification:

- If we have time, it would be ideal to apply separate amplification for both ears and then to output two analog signals
- If not, do one ear and say that it proves the concept that the amplification is working

Advice from Prof Rubin:

- For the audio amplifier, the inbuilt controllable gain might have less noise than a voltage divider
- For the GUI, we must make a recording of our voices

Polar plots:

- It is not ideal to plot the polar plot on a rectangular axis
- Try to plot the line plot as a 2D scatter plot according to Equations 1 and 2. Then join the points together
- So that the calculations don't add extra delays into the system, pre-calculate all the points and store them into a 2D array. Then just pick the correct row/column to plot

$$P_{new} = \frac{P_{old} - min}{max - min} \tag{1}$$

$$(x, y) = (P_{new} \cos\theta, P_{new} \sin\theta) \tag{2}$$

MEETING 9

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

- Weekly progress report
- Arduino sampling frequency
- Project title changes

Minutes of meeting:

Start date: 08/08/2018
Start time: 10:00

Arduino sampling frequency:

- Monotone sounds that are being sampled are distorted due to a low sampling frequency.
- The sampling frequency gets lower as more lines of code get added.
- The obvious solution is to remove one frequency band per microphone to keep sampling frequency as high as possible (Plan B).
- Another proposed solution is the use of two Arduinos that serially communicate with one another.
- Have each Arduino deal with separate frequency bands and synchronize them to avoid significant undesirable delays.
- MAke use of an external ADC chip for reading potentiometer values to ensure that sampling frequency of Arduino is maximised. Do not waste a channel on simply reading a potentiometer voltage

Advice on report content:

- State that a major limitation of the hardware design is the use of the Arduino Due.
- State the reasoning for use of two Arduinos: Increasing sampling frequency and preventing interference between each frequency band.

Title changes:

- Action required for title changes only involves sending the new title to Mr Hugh Hunt.

Arduino interrupts:

- To determine the sampling frequency, it is recommended that a timer is started at the end of program set-up and ended once the sample counter is 50 (end of interrupt).

To do:

- Investigate the use of timer-based interrupts to improve sampling frequency.
- Only move onto the approach using two Arduinos once the system functions as expected when dealing with a single frequency band.

MEETING 10

Attendees:

Kayla-Jade Butkow, Kelvin da Silva, Prof. David Rubin

Agenda:

- Weekly progress report
- Directionality feature issues
- Poster details
- Open-Day demonstration

Minutes of meeting:

Start date: 15/08/2018
Start time: 10:00

Sampling frequency:

- The issue was rectified by making use of a timer based interrupt.

Directionality:

- The converted MATLAB code for directionality only functions correctly for a single direction (90°).
- If directionality is cannot be achieved in other directions, reasoning must be provided in the report.
- It is suggested that a logic analyser should be used to monitor the behaviour of each channel of the Arduino for the purpose of determining the reason why the program malfunctions.

Open-Day:

- Hardware demonstration: Connect an oscilloscope to the output of the device and move a sound source around to demonstrate a change in signal magnitude when in directional mode.
- Hardware demonstration: A similar procedure is to be taken in omnidirectional mode.
- Software demonstration: Make use of the developed GUI.
- Permission granted for use of Professor Rubin's digital oscilloscope for testing and on Open-Day.

Testing of software:

- Test the software solution in exactly the same way that the hardware is tested.
- Prove that the gain applied to a signal (in accordance with a given audiogram) manipulates the signal appropriately.
- Test for a range of frequencies and gain.
- Test directionality by obtaining polar plots when the acoustic beam is facing different directions and show the effect on magnitude of the applied signal.

Poster:

- The poster must mainly consist of pictures and as few words as possible.
- Make use of bullet points, large font and include a high level block diagram of the developed system.
- An abstract should be included whereby the reader is informed about what the project entails.
- Polar plots and frequency spectrum plots will be useful aids for a results section.
- Present the error in directionality as well as compensatory gain.

To do:

- Look at technical conference posters for guidance.
- Design a casing for the hardware solution to be housed in.