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ELEN4011 - Design Project
Biomedical Engineering - Design of an Adaptive Hearing Aid
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Contents

Abstract

This paper presents the design of an adaptive hearing aid. The hearing aid is divided into two subsystem, the directionality and adaptive filtering subsystems, the latter of which is the focus of this paper. Various filter bank designs are simulated using `MATLAB` and `Simulink`. The ANSI S1.11 octave filter yielded the best matching error, but did not meet the group delay requirements. The filter bank was optimised using iterative relaxation techniques yielding a matching error of $0.83dB$ and a group delay of $8.88ms$. A dynamic range compression algorithm was implemented to keep the loudness output of the filter bank within a comfortable auditory range. Further optimisations can be applied to this design to minimise the matching error and group delay. The cost of the hearing aid is $R4300$ per ear.

1. Introduction

Hearing loss is a common problem in society. In *2016*, the University of the Witwatersrand released a statement saying that in South Africa, approximately 20% of the population suffers from a form of hearing loss [1]. As a result of hearing loss, an individual can suffer from isolation, dependence, depression and frustration which are factors that may affect quality of life [2]. The amount of people with an impaired quality of life provides motivation for the necessity of devices to improve an individuals hearing, namely a hearing aid. These devices aim to improve the patients speech intelligibility.

A hearing aid is adaptive in the sense that it has the capability of being set to match the specific needs of a patient. This paper presents a comparison, through simulation, between various adaptive filter bank designs to determine the best performing design. The chosen design is further optimised. Dynamic range compression is also considered to ensure that the hearing operates at comfortable loudness levels. A hardware system is realised in addition to the design, such that this system could be implemented physically.

2. Design Research

Many factors contribute to hearing loss. A common type is conductive hearing loss which is caused by older age, excessive noise or various diseases [3],[4]. This is a permanent form of hearing loss which can be treated with hearing assisting devices [4]. This section presents an analysis of existing solutions to this problem (Section 2.1) and a literature review (Section 2.2).

2.1 Existing Solutions

Conductive hearing loss can be treated surgically or with the use of a hearing aid [5]. When surgery is performed, the damage caused to the ear cannot always be reversed. A hearing aid however, has very positive benefits for those suffering from conductive hearing loss [5]. Therefore, this design will make use of hearing aid technology.

There are two main types of hearing aids, analog and digital. Analog hearing aids amplify all sounds, including noise [6]. Ricketts [7] states that the largest problem for the hearing impaired is listening in noisy situations. Digital hearing aids however, contain a signal processing element which gives them additional functionality such as filtering noise, and isolating sounds in specific directions [7]. Thus, these hearing aids improve the patient's speech intelligibility which can positively influence their social experience.

2.2 Literature Review

Sebastian *et al.* [8] define a hearing aid, as an electro-acoustic device with the purpose of making speech intelligible. This purpose, together with the patients quality of life, must be at the center of each design decision.

Audiograms, performed by audiologists, are used to detect hearing impairments of a patient [9]. An audiogram illustrates the hearing threshold of a person at different pitches and frequencies [8]. These graphs highlight the frequencies at which the patient battles to hear. Frequency response matching is a technique utilised by hearing aids to correct the audiogram of the patient. Kakol *et al.* mention that in many cases, patients experience hearing difficulties at different frequency ranges [10]. Therefore, by correcting the audiogram at the required frequencies, the patients hearing can be improved. This correction can be implemented with a filter bank which applies gain to various frequency sub-bands [10]. This method is also mentioned by Sebastian *et al.* and is referred to as selective amplification [8].

Sebastian *et al.* provide an frequency response matching (FRM) investigation between uniform and non-uniform filter banks [8]. The frequency sub-bands in uniform filter banks are equally spaced whereas non-uniform filter banks are arbitrary spaced. Non-uniform filter banks are the preferred choice as they allow for an audiogram to be corrected at the specific, poor hearing threshold frequency bands [8]. According to Sebastian *et al.*, the frequency bands at the critical hearing frequencies should be narrow for the best compensation [8]. Therefore, this design will consider narrow frequency band, non-uniform filter banks for audiogram correction.

Chang *et al.* give further insight into the types of filter banks by providing four different designs namely; uniform, critical-like, symmetric and 1/3 octave [11]. The 1/3 octave filter bank proved to have the best audiogram matching. This filter bank had a $78ms$ delay at $24kHz$ with the FIR filter design compare to a $27ms$ delay when parallel IIR filters were used. Shearman [12] states a better understanding of how people perceive sound can be achieved by using octave analysis. This form of analysis allows for the human perceived quality of the audio signal to be measured. This design aims to optimise the sound quality to improve the patients quality of life and thus, octave analysis is considered.

Chang *et al.* [11] and Yang *et al.* [13] simulate a filter bank using the ANSI S1.11 specifications. ANSI S1.11 is a standardised set of performance requirements for designing octave and fractional octave filters for both analog and digital domains [14]. These requirements were developed by the Acoustical Society of America specifically for audio applications [14]. These standards are commonly used throughout literature and thus, are considered in this design. Furthermore, Chang *et al.* and Yang *et al.* present the concept of multirate filters. These techniques are used to reduce the computational complexity of the filters by adjusting the frequencies at which different filters operate at [11], [13].

Reference [15] discusses the use of dynamic range compression. Patients with hearing loss battle to hear soft sounds but can hear loud sounds fine. Within the patients auditory range, soft sounds should be amplified to an audible level but loud sounds must not be amplified such that the sound is uncomfortable. The dynamic range compression algorithm aims to adjust the amplified soft, speech and loud sound levels into a comfortable range. This is an essential component for patient quality of life and thus, must be considered.

Dhawan *et al.* simulate filtering in a hearing aid while using a wavelet filter to reduce noise. Additive white Gaussian noise (AWGN) was used to simulate the auditory noise [16]. Noise reduction is a critical component of this design and thus, the effect of these noise reduction techniques will be investigated within this system.

Ricketts [7] provides hearing aid performance metrics. These metrics measure to what extent, a hearing assisted patient can hear compared to a normal hearing person. Normal hearing people, require an SNR[1] of at least $+6dB$. as suggested by [17]. Hearing impaired individuals however, require

[1]Signal to noise ratio.

an SNR greater than $+6dB$ to achieve sufficient hearing [18]. This increase SNR requirement is because impaired hearing caused by poor hearing thresholds, results in noise related problems [7]. For children, a SNR of $+15dB \rightarrow +30dB$ is required for educational purposes. Ricketts also stated that by tuning the hearing aid to listen in the direction of the source can improve the SNR by about $3dB \rightarrow 6dB$.

Chen *et al.* [19] suggest that using multiple microphones results in better extraction, separation of frequencies and localization of the source whilst reducing noise, interference, echo and reverberation. This provides motivation for utilising a multiple microphone array in this design. Resource [20] provides a comparison between two microphone array designs namely; broadside and endfire. Chen *et al.* show that the broadside configuration attenuates the signal less at low frequencies and has a larger bandwidth compared to the endfire configuration. The endfire configuration however, had better directivity when placed in the desired sound direction [20]. In addition, McCowan [21] states that voice is a broadband signal. Therefore, in terms of array configuration, a trade-off must be made between bandwidth and directivity performance.

Munir [22] states that in telephony systems, a frequency range of $300Hz$ to $3.4kHz$ is used because this is the frequency range of voice. The human ear however, can detect a frequency range of $20Hz$ to $20kHz$. Audiograms however, typically test upto a maximum frequency of $8kHz$ [23]. Since this system aims to correct audiograms, a frequency of $20Hz \rightarrow 8kHz$ is considered.

McCowan [21] provides a comprehensive explanation of the wave propagation and various algorithms to determine directivity. This paper evaluates the performance of these algorithms within this context to determine the optimum solution.

3. Design Objectives

This design aims to provide an adaptive hearing aid for patients with conductive hearing loss. This device should allow for patients to operate comfortably in social situations and thus, improving their quality of life. To achieve this, the hearing aid must compensate for the hearing loss of the patient, allow the patient to listen in different directions and hear within a comfortable loudness range. Factors such as response time should be designed such that the hearing aid resembles the human auditory system. These requirements are technically defined in Section 5. and 4..

4. Design Requirements

The maximum matching error, within the conversational frequency range ($500Hz \rightarrow 4kHz$), must lie within the preferable compensation limit of $3dB$ [11]. A system delay of $6ms \rightarrow 8ms$ is noticeable and a $20ms$ delay results in confusion between the audio sound and visual movements [24]. Therefore, this system must have a response time of $20ms$ or less.

5. Design Assumptions and Constraints

This paper presents a theoretical design and hence, assumptions can be made. This design is not limited by cost, power or space. This design is considered for adult patients only. The directionality component considers a $2D$ wave in a plane instead of a $3D$ wave. Finally, it is assumed that the speaker is deep inside the ear and hence separated from the microphone. This separation is considered to be significant enough such that feedback cancellation is not required. The directionality assumes that audiogram will be corrected to a flat response. This design assumes that the limitations of the hardware are negligible and that only the effects of the system itself contributes to delays. This system is constrained to a bandwidth of $20Hz$ to $8kHz$ to correspond to the audiogram frequency range.

6. Design Methodology

The design of this system consists of two main subsystems. The first is applying directionality techniques to allow for the user to listen in specific directions. This aspect of the design was completed by Arunima Pathania. More information regarding this design can be found at resource [25]. The second is the audiogram correction subsystem. These subsystems will be amalgamated into a single system. This paper focuses on the adaptive hearing subsystem. The audiogram correction subsystems consists of a filter bank, dynamic range compression, DAC[2] and a speaker.

In this system, the adaptive filtering component is optimised for the audiogram matching error and response time. Various filter bank designs are presented in literature. Therefore, a comparison will be drawn between each design within the conductive hearing loss context. The best performing design will be further optimised for the best trade-off between matching error and response time. Dynamic range compression will be applied to the output of the filter bank to maintain the sound levels within a comfortable range. Once a design is selected, the suitable hardware will be selected and a circuit realised. Digital filters were considered, as opposed to analog filters, since the specifications of the digital filter can be easily adjusted to satisfy the gain requirements of various prescription formulas.

7. System Overview

An overall system diagram is shown in Figure 1. The directionality component required multiple microphone inputs. Therefore, it is placed before hand. Otherwise, a filter bank would be required for each microphone which is impractical and will require more components and introduce additional delays.

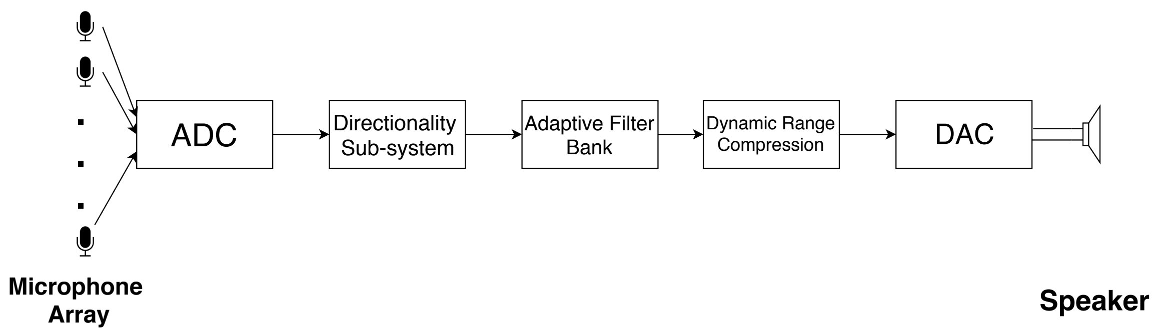

Figure 1: Overall System Block Diagram

8. Adaptive Filter Design

The adaptive filter is a filter bank which consists of an array of bandpass filters. Each of the bandpass filters operate with a particular frequency range and gain. The filter bank is adaptive in the sense that the filter parameters can be adjusted by the audiologist to meet a specific requirement. This design subsection aims to utilise a frequency bank to meet an audiologists prescription. The audiogram considered in this design is presented in Section 8.1.

Approach: Research proved that there are multiple filter bank types used to rectifying a patients audiogram, typically categorised as uniform and non-uniform. To determine the optimum design, each of these systems will be considered. Sections 8.4.1, 8.4.2, 8.4.3 and 8.4.4 present the design of the uniform, symmetric, critical-like and octave filter banks respectively. These designs will also be compared to the ANSI S1.11 specification presented in Section 8.5. The directionality sub-system relies on the phase response of the system and thus, only *FIR* filters (because of their linear phase

[2]Digital-to-Analog-Converter

characteristics) will be considered as oppose to *IIR* filters. Furthermore, because *FIR* filters are stable, an accurate behaviour of the filter bank can be simulated and predicted. Two metrics are used to measure a filter banks performance namely; matching error and group delay. The matching error is defined as the difference between the filter bank response and the perscription. The group delay (T_g) is calculated as shown in equation 1 [11].

$$T_g = \frac{P}{2f_s} \tag{1}$$

P is the order corresponding to the filter which bottlenecks the filter bank (i.e. the filter with the highest order) and f_s is the sampling frequency. Both metrics are considered when choosing a filter bank design.

8.1 Audiogram

The audiogram considered in this design corresponds to a patient with conductive hearing loss. Audiologists typically measure a patients hearing at $125Hz$, $250Hz$, $500Hz$, $1kHz$, $2kHz$, $4kHz$ and $8kHz$ [26]. `MATLAB`'s `pchip` function was used to interpolate these values to provide hearing threshold values for the full $20Hz$ to $8kHz$ range. Reference [26] provides audiograms for normal hearing and conductive hearing loss. Figure 2 illustrates these audiograms.

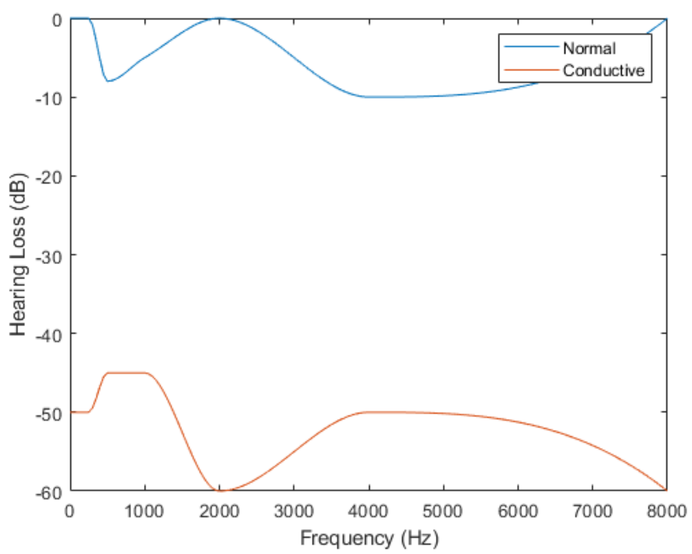

Figure 2: Audiogram for Normal Hearing and Conductive Hearing Loss

The filter bank aims match the conductive hearing loss patients audiogram to the normal hearing audiogram.

8.2 Insertion Gain

The filter bank corrects the audiogram by applying gain to particular frequencies sub-bands. Each sub-band gain is defined as insertion gain. The accuracy of the filter bank is based on it's ability to match the prescription formula. Therefore, the prescription formula is considered to be a black box and as a result, the performance of the filter bank is independent of the prescription formula used.

For simplicity, this design calculates the insertion gain using the NAL-R formulas given in equation 2.

$$\begin{aligned} H_{3FA} &= (H_{500} + H_{1k} + H_{2k})/3 \\ IG_i &= 0.15 \times H_{3FA} + (0.31 \times H_i) + k_i \end{aligned} \tag{2}$$

Where IG_i is the insertion gain, H_i is the audiogram value at the i^{th} sampled frequency and k_i is a constant at the i^{th} sampled frequency given by Table 2 in Appendix B. Similarly to the audiogram, the insertion gain values were interpolated using `MATLAB's pchip` function. The insertion gains for the full frequency range is illustrated in Figure 3.

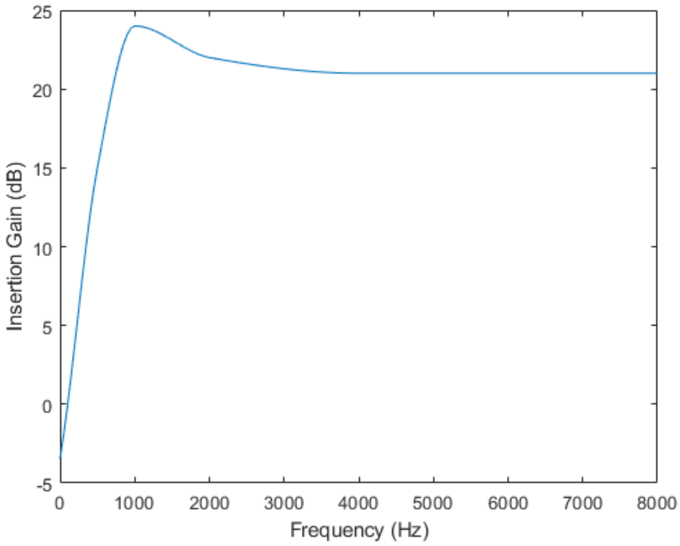

Figure 3: Interpolated Insertion Gain for Full Frequency Range

A sub-band's insertion gain is taken as the average insertion gain within the sub-band range.

8.3 Number of Filters

As stated above, the filter bank consists of multiple bandpass filters, each operating over a specific sub-band. Therefore, the number of filters used must be investigated; particularly for the uniform, critical-like, symmetric and octave filter banks designs. A uniform filter bank was used to investigate the effect that increasing the number of filters has on matching error and computational complexity. Appendix C provides the details of the filter banks used in this investigation. Figure 4 and 5 illustrate the matching error for each band across the frequency range and the mean error for each number of frequency bands respectively.

Figure 5 illustrates that increasing the number of bands, decreases the mean matching error. This is because increasing the number of filters emphasises the frequency sub-bands that need to be adjusted. Therefore, the design should aim to maximise the number of filters.

8.4 Uniform, Critical-like, Symmetric and Octave Filter Bank Design

This section aims to optimise the structure of the filter bank's frequency bands. Therefore, a comparison between a uniform, symmetric, critical-like and octave filter banks will be made. These filters will then be compared to the ANSI S1.11 filter bank. To draw a fair comparison, all of the filters will be design with the parameters given in Table 3 in Appendix C. Each filter bank will consist of 8 bandpass filters. Initially, the transition band per sub-band is set to 10% of the sub-band's bandwidth. How-

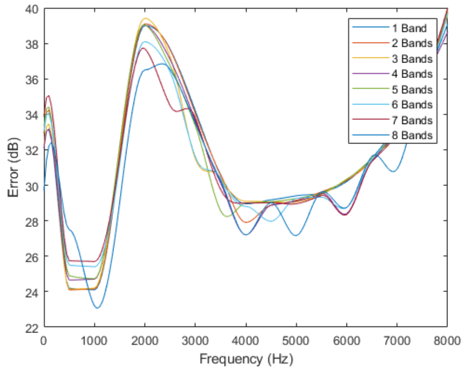

Figure 4: Matching Error Across Frequency Spectrum per Filter Bank

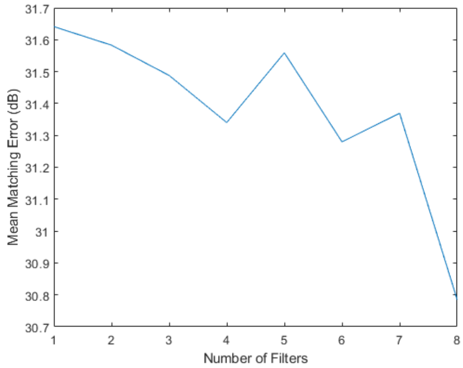

Figure 5: Mean Matching Error per Filter Bank

ever, through simulation, it was found that adjusting the transition bandwidth significantly affected the performance. Therefore, each sub-band's transition bandwidth is iteratively adjusted to achieve the optimal performance. This allows for the best performing filter banks to be compared. Therefore, concrete performance decisions can be made. The same insertion gain method as in Section 8.2 will be used.

8.4.1 Uniform Filter Bank Design A uniform filter bank consist of an array of evenly spaced, filters with equal bandwidths [11]. Uniform filter banks are simple and easy to implement. However, for a sufficient resolution, uniform filter banks requires more bands than non-uniform filter banks for a good fit. The additional filters implies that more computations are required and thus, the filter bank's group delay is potentially increased [24]. This design optimised Section 8.3's 8 sub-band design by adjusting the filter bands to minimise aliasing effects. The filter bank characteristics are given in Table 13 of Appendix E. This design achieved and average and maximum matching error of $1.36dB$ and $11.25dB$ respectively with a group delay of $5.48ms$. The matching error across the full frequency spectrum can be seen in Figure 10 of Appendix E.

8.4.2 Symmetric Filter Bank Design Symmetric filter banks provde an improvement on the uniform filter bank. The non-uniform sub-bands are symmetric about the center frequency of the frequency spectrum, in this case $4kHz$. Symmetric filter banks have the ability of enhancing the matching error of low and high, or mid frequencies [8]. Figure 2 illustrates that the threshold of a conductive hearing loss patient is worse at the low and high frequencies. Therefore, the frequency bands will be chosen such that these frequencies are emphasised. Table 14 in Appendix F illustrates the bands used in this investigation. Simulation illustrated that the symmetric filter bank achieved an average error of $1.34dB$ with a maximum matching error of $9.4dB$. This filter bank achieved a group delay of $14.92ms$. The matching error for the full frequency spectrum is illustrated in Figure 11 in Appendix F.

8.4.3 Critical-like Filter Bank Design This filter is forms part of the non-uniform filter bank category. This filter design attempts to account for the psychoacoustic characteristics using the critical bands of the Bark Scale [27]. Table 12 in Appendix D provides the Bark scale's critical frequency band information. This investigation is limited to using 8 filters and a maximum frequency of $8kHz$. Therefore, the sub-bands are set as shown in Table 15 of Appendix G. This design achieved an average and maximum matching error of $0.87dB$ and $8.15dB$ respectively and a group delay of $16.18ms$. The matching error for the full frequency spectrum is illustrated in Figure 12 of Appendix G.

8.4.4 Octave Filter Bank Design As stated above, separating a frequency spectrum into octaves allows for the quality of sound to be measure and by derivative, improved [12]. It is therefore natural for a filter bank to be designed using this principle. Each filter bank sub-band will have a center frequency relative to the reference sub-band center frequency $f_c[0] = 1000Hz$ [28]. The center frequency of each sub-band is calculated using equation 3. The corresponding upper (f_{cu}) and lower (f_{cl}) passband frequencies are calculated using equation 4 [28].

$$f_c[k-1] = f[k]/2 \tag{3}$$

$$\begin{aligned} f_{cu}[k] &= \frac{f_c[k]}{2^{1/2}} \\ f_{cl}[k] &= 2^{1/2} \times f_c[k] \end{aligned} \tag{4}$$

The center frequencies for this filter bank are chosen to be $63Hz$, $125Hz$, $250Hz$, $500Hz$, $1kHz$, $2kHz$, $4kHz$ and $8kHz$ which results in the filter specifications given in Table 16 of Appendix H. Simulations demonstrated that this filter bank achieved an average and maximum matching error of $0.72dB$ and $10.44dB$ respectively. Figure 13 in Appendix H illustrates the matching error across the full frequency spectrum. The group delay for this filter was $37.45ms$.

8.5 ANSI S1.11 Design

A octave filter bank was designed using the ANSI S1.11 specficiations. The class 2 attenuation specification is chosen for two reasons. The first being that as it allows a $1dB$ ripple which allows for an accurate comparison with the aforementioned filter banks. Secondly, the larger ripple decreases the delay [29]. The stop band attenuation is set to $60dB$. This is considered to be sufficient for hearing loss applications [13]. ANSI S1.11 compliant filter banks are typically implemented using IIR filters. However, for an accurate comparison, this investigation implements the ANSI S1.11 standard using FIR filters. The frequency specifications are calculated as shown in Appendix I with the resulting specifications summarised in Table 17 of Appendix I. Through simulation, the mean matching error was $0.25dB$ and the maximum error was $2.3dB$. The ANSI S1.11 filter bank achieved a group delay of $47.28ms$. Figure 14 in Appendix I illustrates the matching error across the full frequency spectrum.

8.6 Filter Design Selection

From the several investigated filter bank designs, the optimal design must be chosen. This decision is based on two metrics; namely matching error performance and computational complexity. Figure 6 illustrates the matching error and computational complexity of each design.

From Figure 6, the critical-like and octave filter banks provide the best balance between matching error and group delay. However, experimentation found that only slight optimisations can be applied to these filter banks. The ANSI S1.11 has a significantly lower matching error compared to the other filter banks. Furthermore, the group delay of the ANSI S1.11 filter bank can be significantly optimised by using transition bandwidth relaxation (adjusting the transition bandwidth to reduce the filter order) [11]. Therefore, the ANSI S1.11 filter bank design technique is chosen to design the adaptive filter bank. The optimisations applied to this filter bank are presented in Section 8.7.

8.7 Filter Bank Optimisation

In the previous section, the investigations were constrained to using 8 filters. ANSI S1.11 provides specifications for 1/3-octave filter banks, meaning there are 3 sub-bands per octave. As shown in Section 8.3, increasing the number of sub-bands decreases the matching error. A design was simulated using the frequency sub-bands $F_{14} \rightarrow F_{39}$ from the ANSI S1.11 specification. Simulations revealed extensive aliasing affects and a maximum group delay of $107.7ms$. This is unacceptable. Relaxation

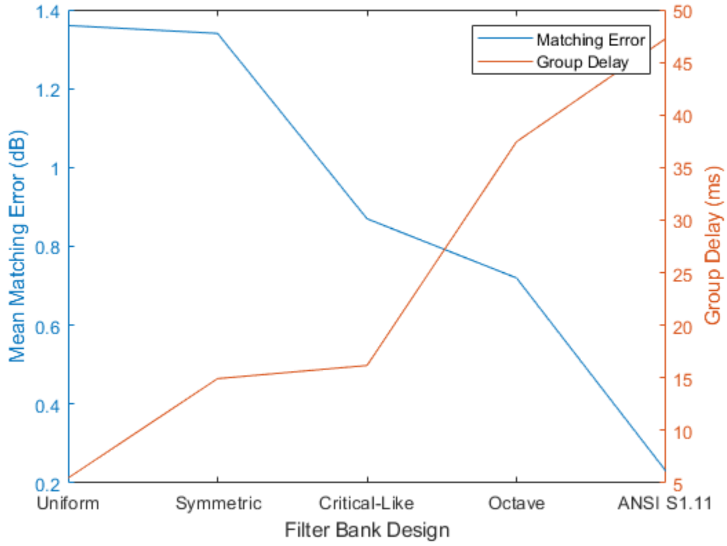

Figure 6: Mean Matching Error and Group Delay for each Filter Bank Design

techniques exist to reduce rigidity of the ANSI S1.11 specification by increasing the relevant transition bandwidths. However, the lowest frequency sub-band (F_{14}) has a lower passband frequency of $23Hz$. However, it is impossible to make the lower transition bandwidth large enough to reduce the order without reducing the attenuation. Therefore, the octave filter bank is chosen over the 1/3-octave filter bank.

As shown in Section 8.6, the ANSI S1.11 octave filter bank is highly accurate, but has an unacceptably large group delay. Therefore, the ANSI S1.11 octave filter bank will be optimised to meet the $10ms$ group delay requirement. As seen in equation 1, the group delay is dependant on the filter order. The maximum filter order to achieve a group delay of $10ms$ is therefore, $P = (10ms) \times (2f_s) = 400$. Using `MATLAB's` Park-McClennan algorithm function `firpmord`, it was found that the first, second and third sub-band filters did not meet this requirement with filter orders of 1350, 1890 and 945 respectively. To reduce the order and hence the group delay, relaxation of the ANSI S1.11 specifications can be applied. Using iterative techniques, it was found that a transition bandwidth of $75Hz$ yields a $10ms$ group delay. The filter bank will be bottlenecked by the filter with the maximum group delay. Therefore, the minimum transition bandwidth is set at $80Hz$ to ensure a group delay $< 10ms$ per filter. The first passband frequency of the ANSI S1.11 filter bank designed in Section 8.5 was $22Hz < 80Hz$. Therefore, the selected sub-bands per octave were adjusted such that the first passband frequency could meet the $10ms$ group delay requirement. Table 18 in Appendix J illustrate the adjusted frequency specifications. The adjusted filter bank yielded a maximum group delay of $8.88ms$ with a mean and maximum matching error of $0.83dB$ and $10.42dB$ respectively. The matching error for the full frequency spectrum is illustrated in Figure 15 of Appendix J.

9. Dynamic Range Compression Design

This section aims to design a compression algorithm to maintain the output of the filter bank within a comfortable auditory range ($40dBSPL \rightarrow 80dBSPL$ [30]). Figure 7 illustrates the loudness range that normal and hearing impaired people can hear, and is comfortable.

As seen from Figure 7, speech is perceived as soft to a hearing impaired individual. Therefore, by merely amplifying the sound to improve speech perception, loud sounds that were once comfortable will be shifted into the uncomfortable region. Dynamic range compression ensures the the output of the filter bank remains in a comfortable auditory region by compressing the loudness range into

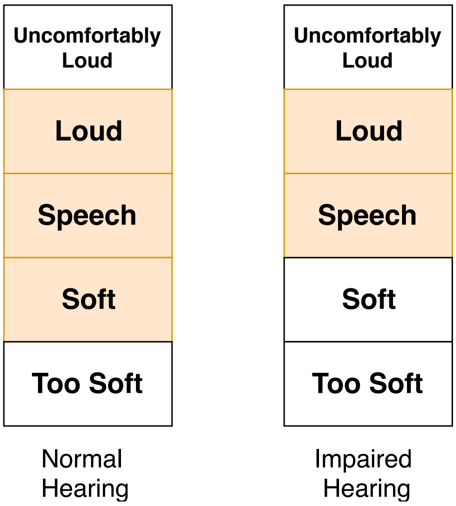

Figure 7: Comfortable Loudness Range for Normal and Hearing Impaired People

the loudness range, audible range to the patient. The design of the dynamic range compression component involves choosing suitable values for the compression threshold, compression ratio, attack and release times and the number of channels. More information about these parameters can be found in Appendix K. The parameters chosen for this design are such that speech intelligibility is maximised. The compression threshold should be set below $50dBSPL$ to amplify the soft speech into an audible range [15]. In this design, the compression threshold is set at $30dBSPL$. To operate over a wide input range, the compression recommended to be below 4 and therefore, is set at 3 in this design. The attack and release times are set at `MATLAB's` recommended $0.05s$ and $0.2s$ respectively. Each of the sub-band filter banks will be followed by a dynamic range compressor. This is chosen to improve speed intelligibility by amplifying the loud vowel sounds separately from the softer consonant sounds. The dynamic range compressor output is calculated as shown in Appendix L. To test this algorithm, gain was applied to a $6kHz$ digital sine wave with an amplitude of 1 before it was passed through the compressor. By adjusting the gain, the affect of the compressor can be seen which is referred to as the static characteristics. Figure 8 illustrates the static characteristics for dynamic range compressor used in this design.

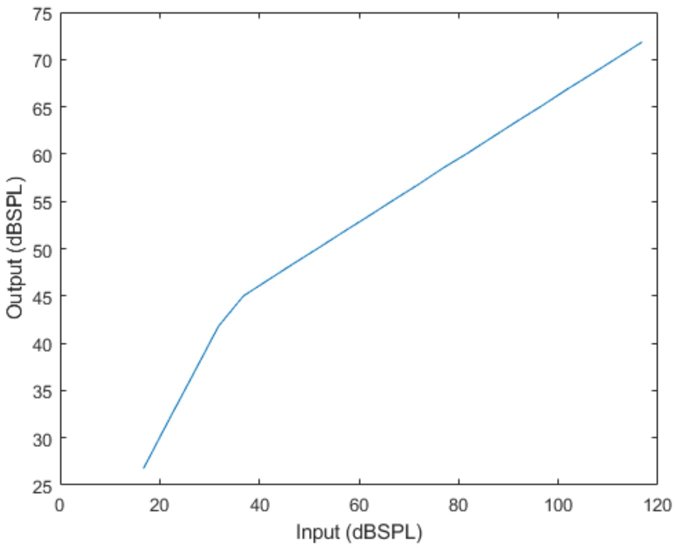

Figure 8: Dynamic Range Compressor Static Characteristics

10. System Realisation

This subsystem consists of an input, an analysis and synthesis stage, and finally an output. For the analysis stage, a DSP[3] is required which will house the filter bank and the dynamic range compression algorithm. The sythesis stage consists of a DAC, reconstruction filter, pre-amplifier which outputs to a speaker. The input to this subsystem is the output of the directionality subsystem. The directionality analysis phases takes placed on an Arduino Mega. An SPI transaction paradigm is used to communicate between the directionality and adaptive filter subsystems. The Analog Devices, *ADAU1463 Sigma Digital Audio Processor* is chosen as the DSP for this system. This DSP can handle 24000 filter taps per sample [31]. The optimised filter bank consists of a total of 1683 filter taps per sample and therefore, this DSP can easily handle the filter bank processing requirements. The *ADAU1463* has 2 communication port types, I^2C and SPI. To synthesise the signal, the output of the DSP is connected to the *TLV320AIC3109-Q1* audio processor. The *ADAU1463* supports both SPI and I^2C as opposed to *TLV320AIC3109-Q1* which only supports I^2C. Therefore, the *ADAU1463* communicates with the *TLV320AIC3109-Q1* via the I^2Cport. The *TLV320AIC3109-Q1* contains the DAC, reconstruction filter and pre-amplifier suitable for a 16Ω headphone output load resistance [32]. The *CE20M-16* micro speaker is specified to output the sound of the system to the patient. The *CE20M-16* micro speaker has a 16Ω load resistance [33] and thus, is suitable for the output of the *TLV320AIC3109-Q1* audio processor. An overall circuit diagram is giving in Figure 16 of Appendix M.

11. Success Criterion

To deem this subsystem successful, the adaptive filter bank and dynamic range compression must be combined and simulated. The results of these simulations should be compared to that of the design requirements and constraints. The following criterion will be used to measure the success of each subsystem.

1. Matching Error: In the previous section, the entire frequency spectrum was analysed. The most common frequency range of conversation however, is $500Hz$ to $4kHz$ with particular importance at $2kHz$ [34]. Therefore, since the focus of this design is maximise speech intelligibility, the matching error between $500Hz$ and $4kHz$ will determine the success of the frequency response matching component.
2. System Response: As stated above, the response time of the system is a critical design consideration. Since the directionality is assumed to have a response time of $10ms$, the overall response time of this subsection cannot exceed $10ms$ to meet the $20ms$ response time requirement.
3. Loudness Compression: The dynamic range compression must ensure that sounds, comfortable to the human ear, remain in the comfortable loudness region ($40dBSPL \rightarrow 80dBSPL$) [35].

12. Critical Evaluation of Results

Simulations were performed on the combined subsystem. The measurements taken from these simulations were compared to the success criteria of the subsystem. The matching error, group delay and dynamic range compression results are provide in Section 12.1, 12.2 and 12.3 respectively. A technical view point regarding this design is presented in Section 12.4 with a non-technical discussion presented in Appendix A.

12.1 Matching Error Results

The optimised filter bank has a mean and maximum matching of $0.73dB$ and $4.5dB$ respectively between $500Hz$ and $4kHz$. Whilst the average matching error falls within the $3dB$ constraint, the maximum error does not. When the transition bandwidths are adjusted, the impact of aliasing is increased. As a result, when the filters are combined, the overall response deviates further from

[3]Digital-Signal-Processor

the desired response and thus, the matching error increases. In terms of matching error, the ideal filter bank would be constructed of ideal filters as no aliasing would occur. However, since this is unrealisable, this is an important design trade-off.

12.2 Response Time Results

Two components contribute to the response time is the delay introduced from the filter bank and the processing time of the dynamic range compression component. The processing time of the dynamic range compression is a simple conversion and hence, is negligible compared to the numerous calculations required for each filter. Therefore, the response time is bottlenecked by the filter bank and by derivative, the maximum group delay. As stated in Section 8.7, the group delay is $8.88ms$. Whilst this falls within the constrained $10ms$ response time, it will still be noticeable to the wearer.

12.3 Dynamic Range Compression Results

The dynamic range compression component adjusted the loudness range of the filter bank. With the parameters chosen in Section 9., the transformation shown in Figure 9 was achieved.

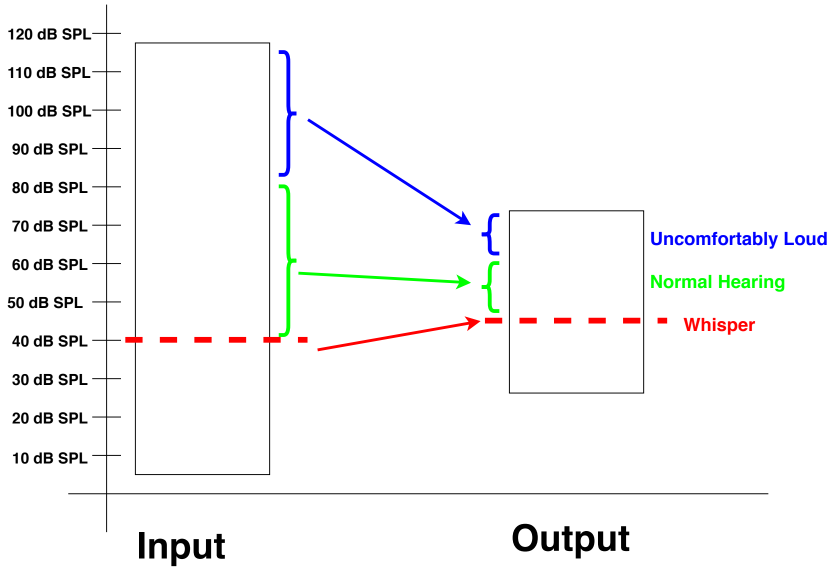

Figure 9: Dynamic Range Compression Transformation

From Figure 9 it can be seen that the output is compressed within the comfortable hearing threshold of $40dBSPL$ and $80dBSPL$. The red line shows that the whisper loudness level is amplified firmly into the audible range and the harmfully loud sounds are suppressed into a comfortable level. This component however can be improved such that the soft sounds and loud sounds occupy only the $40dBSPL$ to $80dBSPL$ range.

12.4 Socio-economic Impacts of Design

The designed adaptive filter bank can adjust the gains of it's various sub-bands such that a patients audiogram can be corrected, with a reasonable matching error. The adaptability of this design implies that hearing aids can be updated when a patients hearing deteriorates, as opposed to constructing a new one, which keeps the recurring cost of the device low. By compensating for the hearing impaired, more professional and capable individuals would be interacting with society and hence, the economy would benefit.

Although the cost of the system is not constrained, it is an important factor to considered in any design. The cost of this design is divided up into the directionality and adaptive filter components. The cost of the directionality is $R4000$. For the adaptive filtering, the total cost is $R300$. A breakdown of the cost for this system is given in Table 1.

Table 1: Bill of Components

Component	Cost (ZAR)
ADAU1463	177,17
TLV320AIC3109-Q1	63,32
CE20M-16	57,43
Directionality	4000

The total cost of the system is therefore, $R4300$ per ear. Typical hearing aids in South Africa cost between $R5000$ and $R50000$ per ear [36]. The design presented here, is therefore a cheaper alternative to typical hearing aids.

13. Future Recommendations

Due to time constraints, the design was not fully optimised. However, this section discusses some suggestions that should be investigated to improve the performance of the design. An investigation into a dedicated noise filter would result in a better performing hearing aid in noisy environments. An iterative approach was taken to handle the trade-off between the matching error and group delay of the filter bank. Extensive investigations into the balance between minimise the matching error and group delay would greatly enhance the performance of the hearing aid and hence, the quality of life of the wearer. As shown above, the number of filters reduces the matching error. The ANSI S1.11 1/3-octave filter bank therefore, should have a lower matching error than the ANSI S1.11 octave filter bank. This ANSI S1.11 design was discarded due to it's inability to reduce the group delay and aliasing effects at low frequency. However, this should be revisited with the intention of minimising aliasing effects and group delay. The filter bank presented is adaptive in the sense that the gain of sub-bands can be adjusted. A further aspect of adaptability would be to allow the bandwidth of the sub-bands to be adjusted. This would allow the audiologist to further correct the patients audiogram. This would involve developing an algorithm to ensure that minimal aliasing occurs with the chosen bandwidths. Multirate filters were not considered in this design as computational complexity was not constrained. However, utilising a quadrature mirror filter with multirate filtering is a possible solution for reducing aliasing affects [37]. The DSP chosen for this design far exceeds the requirements. Although cost is not a constraint, it must be considered when this system is realised. Therefore, a more suitable DSP must be chosen. The dynamic range compression component can be optimised to compress the output into the full comfortable hearing range.

14. Conclusion

This paper presented the design of an adaptive filter bank for a hearing aid, to correct the audiogram of a hearing impaired individual. An octave filter bank, used the ANSI S1.11 standards was shown to have the best matching error, but poor group delay. The specifications of the filter bank were relaxed which resulted in an average matching error of $0.83dB$ with a group delay of $8.88ms$. Dynamic range compression was applied to the output of the filter bank to ensure that the loudness remained within a comfortable range. The cost of this hearing aid is $R4300$. This design can be improve by applying systematic relaxation techniques, introducing a dedicated noise filter and improving the accuracy of the dynamic range compressor.

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Appendices

A Social and Economic Implications of Design

The responsibility of an engineer is to use advances and ideas from science to alleviate problems faced by society. Therefore, an engineer must always consider the implications that their design has on the people that it is intended to benefit. This non-technical report therefore discusses the implications that this design has on society and our economy.

Hearing loss affects many people around the world. Wits University stated that in *2016*, 20% of the population, which is around 11.2 *million* South Africans [1].

The most significant hindrance due to hearing loss, is the inability to communicate affectively with others. This inability to communicate leads to an individual becoming isolated from society and subsequent isolation, dependence, depression and frustration may occur. All of these factos affect the quality of life od the individual. The amount of people affected by hearing loss, combined with the repercussion of hearing loss, shows that systems designed to compensate for a persons hearing loss are required. This compensation, should be focused on improving a persons speech intelligibility so that their ability to communicate effectively is restored.

Hearing aids are a solution for people with hearing loss but some designs have their own drawbacks. Hearing aids simply amplify the sound to a range where the wearer can hear, this also often results in the noise being amplified to a point that is excessively loud and uncomfortable for the wearer. Furthermore, sounds are often amplified in all directions which can confuse and disorientate the wearer. As a result, this design contains a directionality component which allows the wearer to control in which direction they would like to listen. Also, only the auditory regions (known as frequency bands) where the person battles to hear are amplified. This means that noise is less likely to be harmfully amplified and the wearer can participate in social events and activities comfortably, which can alleviate social isolation that they may be experiencing.

Each person has a different hearing characteristic and as a their hearing deteriorates over time, different compensation is required. The hearing aid is designed to be adaptive, meaning that it can be programmed specifically to compensate for an individuals hearing impairment. Since this is a digital system, the hardware will remain the same which makes it an affordable and customizable choice that can simply be updated as hearing deteriorates. The design presented here is expected to cost approximately $R4300$ per ear as opposed to existing hearing aids, which cost between $R5000$ and $R50000$ per ear in South Africa. About 27.2% of South Africans are unemployed and it is unlikely that the unemployed will be able to fund existing hearing aids themselves. The public health sector provides limited hearing aids, which have prolonged waiting periods with no guarantee of receiving one [38]. This cheaper alternative can provide hearing impairment compensation for those less fortunate. Consequently, our society will be uplifted as more people can benefit from verbal education because of the ability to communicate effectively.

Having people with hearing impairments able to function in society is not only beneficial to the recipients of hearing aids, but society and the economy. Opportunities, such as further education and business roles that require effective communicators, that were previously more difficult to attain for the hearing impaired, may be freely available with the use of hearing aids. By wearing a hearing aid, the hearing impaired can contribute significantly to society, without the hindrances of hearing loss, and consequently, improve the economy.

Hearing impairment is particularly devasting for its sufferers during their formative childhood years. These children battle to develop their vocabulary, which affects their speech abilities. The discrepancies between their vocabulary and the vocabulary of other children their age only worsen with age. They also battle to hear the ends of words which produces difficulties understanding tenses, resulting in poor sentence construction. Due to the inability to hear their own voices, they may not speak with correct volume. They also cannot hear certain speech sounds which comprises their pronunciation. The children may consequently lack important social skills, which are essential for their adult life as a member of society [39]. Children with hearing loss consequently suffer academically and in cases of moderate hearing loss, perform on average one to four grade levels below their fully hearing peers. This may deprive of them of opportunities later on in life, as battling academically is usually accentuated with age [39].

Engineering plays a vital role in society. This design has direct social and economic impacts.

B Insertion Gain Parameters

The values of k_i is determined by Table 2.

Table 2: k_i Parameter at Specific Frequency Values

Frequency (Hz)	250	500	1000	2000	3000	4000	6000
$k_i(dB)$	-17	-8	1	-1	-2	-2	-2

C Number of Filters Investigation

In this investigation, a filter bank was constructed. Each filter was designed using `Simulink's` digital bandpass filter. The design settings used are given in Table 3.

Table 3: Filter Design Settings

Impulse Response	FIR
Order Mode	Minimum
Filter Type	Single-rate
Input Sample Rate	$20kHz$
Passband ripple	$1dB$
Transition Bandwidth	$200Hz$
Design Method	Equiripple
Structure	Direct-form FIR

The order of an FIR filter corresponds to the window length. Therefore, it is set to a minimum to keep the results consistent. Tables 4 to 11 provide the filter range and gains used in this investigation. f_{s1}, f_{p1}, f_{p2} and f_{s2} correspond to the lower stopband frequency, the lower pass band frequency, the upper passband frequency and the upper stopband frequency respectively. A_{s1} and A_{s2} correspond to the lower and upper stopband attenuations.

Table 4: Filter Parameters - 1 Band

Parameter	**Filter 1**
$f_{s1}(Hz)$	20
$f_{p1}(Hz)$	250
$f_{p2}(Hz)$	8000
$f_{s2}(Hz)$	8200
$A_{s1}(dB)$	$G1dB + 3$
$A_{s2}(dB)$	$G1dB + 3$

Table 5: Filter Parameters - 2 Bands

Parameter	**Filter 1**	**Filter 2**
$f_{s1}(Hz)$	20	3800
$f_{p1}(Hz)$	250	4000
$f_{p2}(Hz)$	4000	8000
$f_{s2}(Hz)$	4200	8200
$A_{s1}(dB)$	$G1dB + 3$	$G1dB + G2dB + 3$
$A_{s2}(dB)$	$G1dB + G2dB + 3$	$G2dB + 3$

Table 6: Filter Parameters - 3 Bands

Parameter	**Filter 1**	**Filter 2**	**Filter 3**
$f_{s1}(Hz)$	20	2800	5550
$f_{p1}(Hz)$	250	3000	5750
$f_{p2}(Hz)$	3000	5750	8000
$f_{s2}(Hz)$	3200	5950	8200
$A_{s1}(dB)$	$G1dB + 3$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$
$A_{s2}(dB)$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + 3$

Table 7: Filter Parameters - 4 Bands

Parameter	**Filter 1**	**Filter 2**	**Filter 3**	**Filter 4**
$f_{s1}(Hz)$	20	1800	3800	5800
$f_{p1}(Hz)$	250	2000	4000	6000
$f_{p2}(Hz)$	2000	4000	6000	8000
$f_{s2}(Hz)$	2200	4200	6200	8200
$A_{s1}(dB)$	$G1dB + 3$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$
$A_{s2}(dB)$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$	$G4dB + 3$

Table 8: Filter Parameters - 5 Bands

Parameter	**Filter 1**	**Filter 2**	**Filter 3**	**Filter 4**	**Filter 5**
$f_{s1}(Hz)$	20	1700	3350	5000	6650
$f_{p1}(Hz)$	250	1900	3550	5200	6850
$f_{p2}(Hz)$	1900	3550	5200	6850	8000
$f_{s2}(Hz)$	1100	3750	5400	7050	8200
$A_{s1}(dB)$	$G1dB + 3$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$	$G4dB + G5dB + 3$
$A_{s2}(dB)$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$	$G4dB + G5dB + 3$	$G5dB + 3$

Table 9: Filter Parameters - 6 Bands

Parameter	**Filter 1**	**Filter 2**	**Filter 3**	**Filter 4**	**Filter 5**	**Filter 6**
$f_{s1}(Hz)$	20	1300	2800	4300	5800	7300
$f_{p1}(Hz)$	250	1500	3000	4500	6000	7500
$f_{p2}(Hz)$	1500	3000	4500	6000	7500	8000
$f_{s2}(Hz)$	1700	3200	4700	6200	7700	8200
$A_{s1}(dB)$	$G1dB + 3$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$	$G4dB + G5dB + 3$	$G5dB + G6dB + 3$
$A_{s2}(dB)$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$	$G4dB + G5dB + 3$	$G5dB + G6dB + 3$	$G6dB + 3$

Table 10: Filter Parameters - 7 Bands

Parameter	**Filter 1**	**Filter 2**	**Filter 3**	**Filter 4**	**Filter 5**	**Filter 6**	**Filter 7**
$f_{s1}(Hz)$	20	1200	2350	3500	4650	5800	6950
$f_{p1}(Hz)$	250	1400	2550	3700	4850	6000	7150
$f_{p2}(Hz)$	1400	2550	3700	4850	6000	7150	8000
$f_{s2}(Hz)$	1600	2750	3900	5050	6200	7350	8200
$A_{s1}(dB)$	$G1dB + 3$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$	$G4dB + G5dB + 3$	$G5dB + G6dB + 3$	$G6dB + G7dB + 3$
$A_{s2}(dB)$	$G1dB + G2dB + 3$	$G2dB + G3dB + 3$	$G3dB + G4dB + 3$	$G4dB + G5dB + 3$	$G5dB + G6dB + 3$	$G6dB + G7dB + 3$	$G7dB + 3$

Table 11: Filter Parameters - 8 Banks

Parameter	Filter 1	Filter 2	Filter 3	Filter 4	Filter 5	Filter 6	Filter 7	Filter 8
$f_{s1}(Hz)$	20	800	1800	2800	3800	4800	5800	6800
$f_{p1}(Hz)$	250	1000	2000	3000	4000	5000	6000	7000
$f_{p2}(Hz)$	1000	2000	3000	4000	5000	6000	7000	8000
$f_{s2}(Hz)$	1200	2200	3200	4200	5200	6200	7200	8200
$A_{s1}(dB)$	$G1dB+3$	$G1dB+G2dB+3$	$G2dB+G3dB+3$	$G3dB+G4dB+3$	$G4dB+G5dB+3$	$G5dB+G6dB+3$	$G6dB+G7dB+3$	$G7dB+G8dB+3$
$A_{s2}(dB)$	$G1dB+G2dB+3$	$G2dB+G3dB+3$	$G3dB+G4dB+3$	$G4dB+G5dB+3$	$G5dB+G6dB+3$	$G6dB+G7dB+3$	$G7dB+G8dB+3$	$G8dB+3$

D Bark-Scale

Table 12 provides the critical frequency bands of the Bark-scale [40].

Table 12: Bark-scale Critical Frequency Bands

Number	Center Frequency (Hz)	Cut-Off Frequency (Hz)	Bandwidth (Hz)
		20	
1	60	100	80
2	150	200	100
3	250	300	100
4	350	400	100
5	450	510	110
6	570	630	120
7	700	770	140
8	840	920	150
9	1000	1080	160
10	1170	1270	190
11	1370	1480	210
12	1600	1720	240
13	1850	2000	280
14	2150	2320	320
15	2500	2700	380
16	2900	3150	450
17	3400	3700	550
18	4000	4400	700
19	4800	5300	900
20	5800	6400	1100
21	7000	7700	1300
22	8500	9500	1800
23	10500	12000	2500
24	13500	15500	3500

E Uniform Filter Bank Specifications and Results

Table 13 provides the specifications for each sub-band in the uniform filter bank.

Figure 10 illustrates the matching error of the uniform filter bank across the full frequency spectrum.

F Symmetric Filter Bank Specifications and Results

Table 14 summarises the frequency bands used to investigate the performance of the symmetric filter bank design. Because the human hearing frequency spectrum begins at $20Hz$, the first sub-band is not symmetric to the last sub-band. However, there is only a 4% difference which is negligible. The transition bandwidth for the first sub-band however, will be $20Hz$. Since only the transition bandwidth of the first sub-band is affected, this affect is also ignored.

Table 13: Uniform Filter Bank Frequency Bands

	Filter 1	**Filter 2**	**Filter 3**	**Filter 4**	**Filter 5**	**Filter 6**	**Filter 7**	**Filter 8**
$f_{s1}(Hz)$	120	900	1900	2900	3900	4900	5900	6900
$f_{p1}(Hz)$	200	1000	2000	3000	4000	5000	6000	7000
$f_{p2}(Hz)$	1000	2000	3000	4000	5000	6000	7000	8000
$f_{s2}(Hz)$	1080	2100	3100	4100	5100	6100	7100	8100
$Gain(dB)$	17,16	23,07	21,62	21,1	21	21	21	21

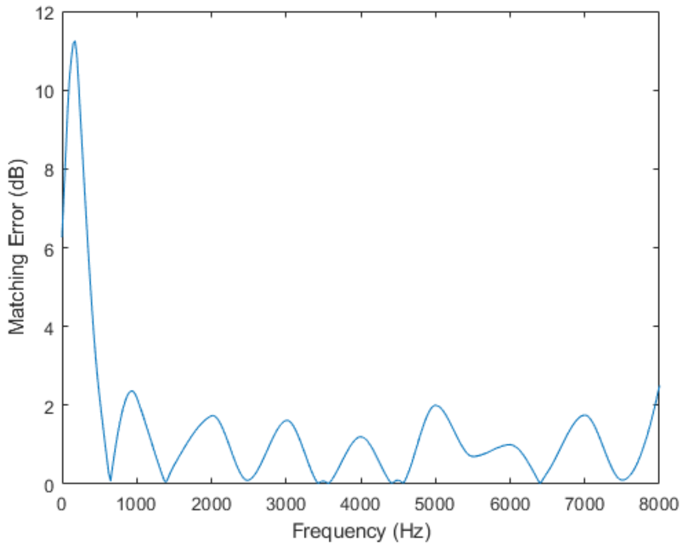

Figure 10: Matching Error for Uniform Filter Bank

Figure 11 illustrates the matching error for the full frequency spectrum of the symmetric filter bank.

G Critical-Like Filter Bank Specifications and Results

Within the Bark range, bands 22,23 and 24 fall outside of the $8kHz$ constraint and are therefore, ignored. The remaining 21 bands are divided into 8 sub-bands by grouping together 2 or 3 sub-bands. According to the audiogram used in this design, the greatest hearing loss occurs within the $1kHz$ to $3kHz$ range. Therefore, a greater resolution is required within this frequency range. The frequency bands used in this investigation are therefore given in Table 15.

Figure 12 provides the matching error of the critical-like filter bank across the full frequency spectrum.

Table 14: Symmetric Filter Bank Frequency Bands

Band Number	**Lower Passband Frequency** (Hz)	**Upper Passband Frequency** (Hz)	**Bandwidth** (Hz)
1	20	500	480
2	500	1000	500
3	1000	2000	1000
4	2000	4000	2000
5	4000	6000	2000
6	6000	7000	1000
7	7000	7500	500
8	7500	8000	500

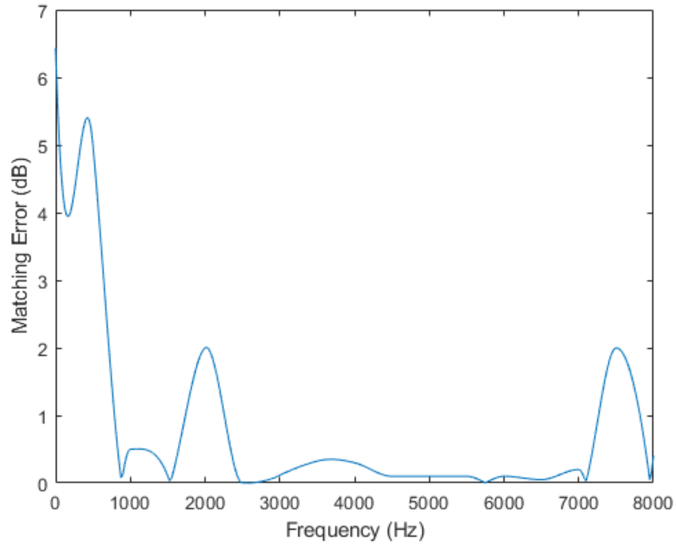

Figure 11: Matching Error for Symmetric Filter Bank

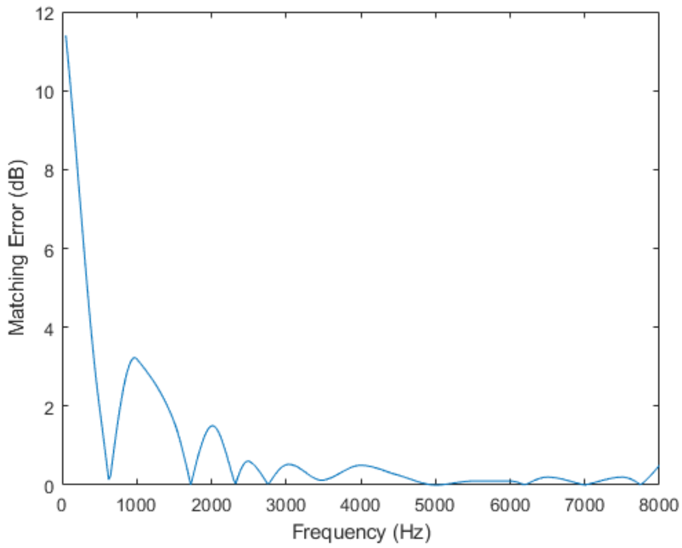

Figure 12: Matching Error for Critical-Like Filter Bank

H Octave Filter Bank Specifications and Results

Table 16 illustrates the frequency bands used in the octave filter bank design. Within the audio range, there exists bands at $16Hz$, $31.25Hz$ and $16kHz$. However, the $16Hz$ and $31.25Hz$ sub-bands are ignored as this design is restricted to using 8 bands and, $16kHz$ violates the Nyquist sampling criteria.

Figure 13 illustrates the matching error of the octave filter bank across the full frequency spectrum.

Table 15: Frequency Range per Sub-band for Critical-Like Filter Bank

Band	Number	Center Frequency (Hz)	Cut-Off Frequency (Hz)	Bandwidth (Hz)
			20	
1	1	60	100	80
	2	150	200	100
	3	250	300	100
2	4	350	400	100
	5	450	510	110
	6	570	630	120
3	7	700	770	140
	8	840	920	150
	9	1000	1080	160
4	10	1170	1270	190
	11	1370	1480	210
5	12	1600	1720	240
	13	1850	2000	280
6	14	2150	2320	320
	15	2500	2700	380
7	16	2900	3150	450
	17	3400	3700	550
	18	4000	4400	700
8	19	4800	5300	900
	20	5800	6400	1100
	21	7000	7700	1300

Table 16: Octave Filter Bank Frequency Bands

Band	Center Frequency (Hz)	Lower Passband Frequency (Hz)	Upper Passband Frequency (Hz)
1	63	45	89
2	125	88	177
3	250	177	354
4	500	354	707
5	1000	707	1414
6	2000	1414	2828
7	4000	2828	5657
8	8000	5657	11314

I ANSI S1.11 Filter Bank Specifications and Results

Table 17 provides the filter specifications for the ANSI S1.11 filter bank. The lower (f_l) and upper (f_u) edge band frequencies are calculated as per equation 5.

$$\begin{aligned} f_l &= (G^{-\frac{1}{2b}}) \times f_m \\ f_u &= (G^{\frac{1}{2b}}) \times f_m \end{aligned} \tag{5}$$

Where $G = 10^{3/10}$ is the octave ratio, b is the bandwidth designator and f_m is the center frequency of the band. Since class 2 is chosen, the attenuation can range from $-0.5dB$ to $+0.5dB$. An iterative approach is taken to calculating the lower and upper stop band frequencies (i.e. transition bandwidth) to achieve the most accurate frequency response.

Figure 14 illustrates the matching error for the full frequency spectrum.

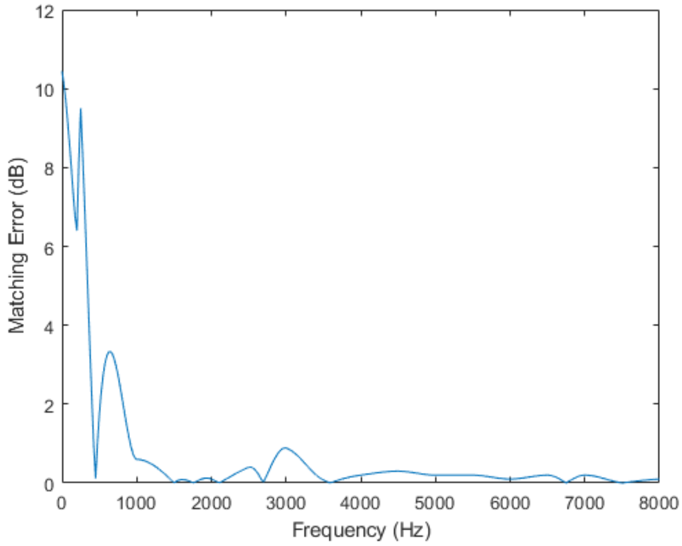

Figure 13: Matching Error for Octave Filter Bank

Table 17: ANSI S1.11 Filter Bank Frequency Bands

Sub-band	**Lower Stopband Frequency**	**Lower Passband Frequency**	**Upper Passband Frequency**	**Upper Stopband Frequency**	**Gain**
1	1	22	45	145	-2,42
2	30	45	90	190	-1,27
3	60	90	180	280	1,26
4	80	180	353	535	6,69
5	153	353	707	907	15,62
6	507	707	1403	1603	23,29
7	1203	1403	2810	3010	22,07
8	2610	2810	8200	8400	21,03

J Optimised ANSI S1.11 Filter Bank Specifications and Results

The frequency specifications were adjusted such that the group delay of the filter bank was less than $10ms$. Table 18 summarises the specifications for the optimised filter bank.

Table 18: Optimised ANSI S1.11 Filter Bank Specifications

Sub-band	**Lower Stopband Frequency** (Hz)	**Lower Passband Frequency** (Hz)	**Upper Passband Frequency** (Hz)	**Upper Stopband Frequency** (Hz)	**Gain** (dB)
1	10	90	140	240	0,47
2	60	145	223	323	3,16
3	123	223	353	535	7,60
4	153	353	561	761	13,68
5	361	561	1100	1300	21,76
6	900	1250	2220	2420	22,76
7	2020	2220	4450	4650	21,23
8	4250	4450	8976	9176	21,00

Figure 15 illustrates the matching error for the full frequency spectrum.

K Dynamic Range Compression Parameters

The Dynamic Range Compression algorithm works by adjusting the output level depending on the input level. These adjustments fit the filter banks output into a suitable loudness range. The specifics of the parameters used in the dynamic range compression algorithm are discussed below.

- Compression Threshold: The compression threshold is the point where the compression algorithm begins to compress the loudness levels. This is also referred to as the threshold knee point and defined as the point where the output level is $2dB$ lower than the uncompressed output level.
- Compression Ratio: This parameter (shown in equation 6) defines how much the input signal will

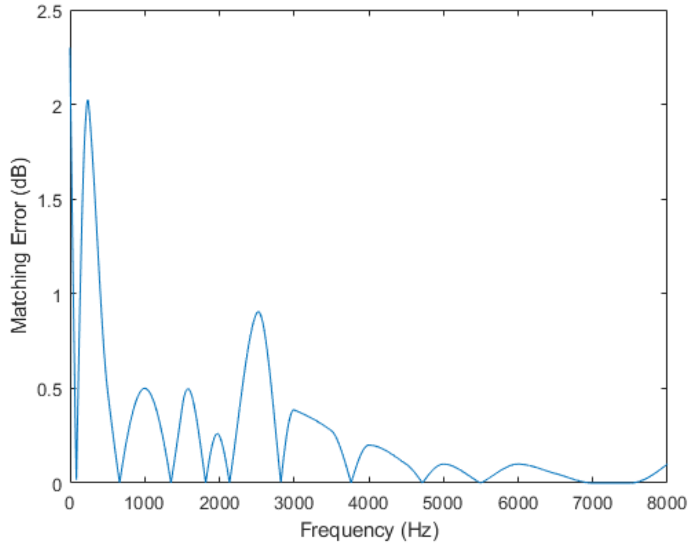

Figure 14: Matching Error for ANSI S1.11 Octave Filter Bank

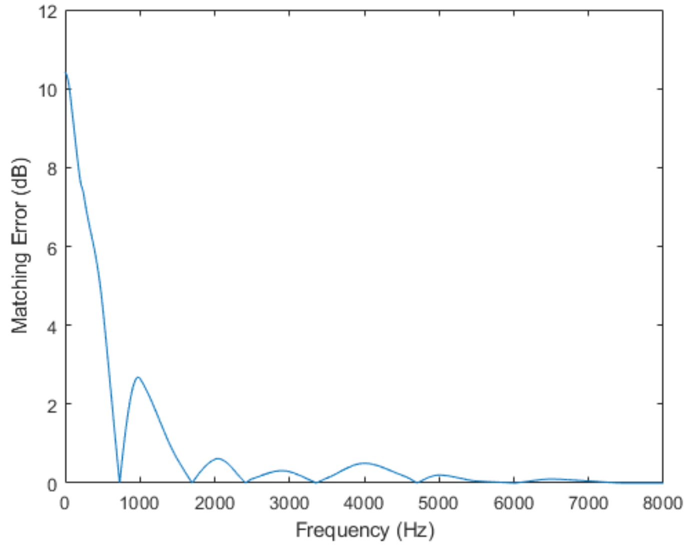

Figure 15: Matching Error for Optimised ANSI S1.11 Filter Bank

be compressed, once above the compression threshold.

$$CR = \frac{\Delta input}{\Delta output} \tag{6}$$

Below the compression threshold, the compression ratio is 1.

- Attack Time: The attack time is the delay from when the input signal surpasses the compression threshold, to when the compression takes affect.
- Release Time: Release time is measured as the time between when the input loudness level drops below the compression threshold, to when the compression deactivates.

L Dynamic Range Compression Calculations

Reference [41] provides the calculations presented in this section. The input signal is sampled ($x[n]$) and converted to decibels ($x_{dB}[n] = 20 \times log_{10}|x[n]|$). This design utilises a hard knee and hence, the static characteristics are calculated as shown in equation 7.

$$x_{sc}(x_{dB}) = \begin{cases} x_{dB} & x_{dB} < T \\ T + \frac{x_{dB}-T}{R} & x_{dB} \geq T \end{cases} \tag{7}$$

Where T is the compression threshold and R is the compression ratio. The gain is then calculated by taking the difference between the static characteristics and the input signal ($g_c[n] = x_{sc}[n] - x_{dB}[n]$). The attack and release parameters are then applied to the gain as shown in equation 8.

$$g_s[n] = \begin{cases} \alpha_A g_s[n-1] + (1-\alpha_A)g_c[n] & g_c[n] \leq g_s[n-1] \\ \alpha_R g_s[n-1] + (1-\alpha_R)g_c[n] & g_c[n] > g_s[n-1] \end{cases} \tag{8}$$

Where g_s is the smoothed gain and α_A and α_R are the attack and release time coefficients calculated as shown in equation 9 and 10 respectively.

$$\alpha_A = e^{\frac{-log(9)}{F_s \times T_A}} \tag{9}$$

$$\alpha_R = e^{\frac{-log(9)}{F_s \times T_R}} \tag{10}$$

Where T_A and T_R are the attack and release times respectively. The make-up gain M, is calculated such that a $0dB$, stead state input will give a $0dB$ steady state output. Equation 11 describes how this value is obtained.

$$M = T - (T/R) \tag{11}$$

The make-up gain is then applied to the smoothed gain ($g_m = g_s[n] + M$). The value g_m has units of (dB) and therefore, the equivalent linear value (g_{lin}), must be calculated. This is achieved using equation 12.

$$g_{lin} = 10^{\frac{g_m[n]}{20}} \tag{12}$$

Finally, the output (y) is calculated as shown in equation 13.

$$y[n] = x[n] \times g_{lin}[n] \tag{13}$$

M System Realisation

Figure 16 is a representation of the circuit diagram, connecting the various processors together in this design.

N Project Planning and Execution

The work breakdown of the project is illustrated in Figure 17.

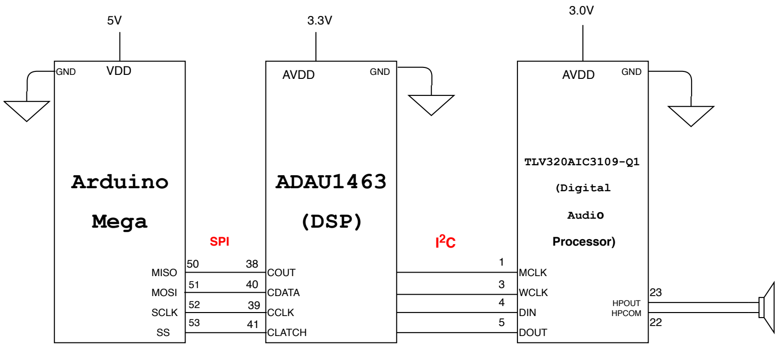

Figure 16: High-Level Realised Circuit Diagram

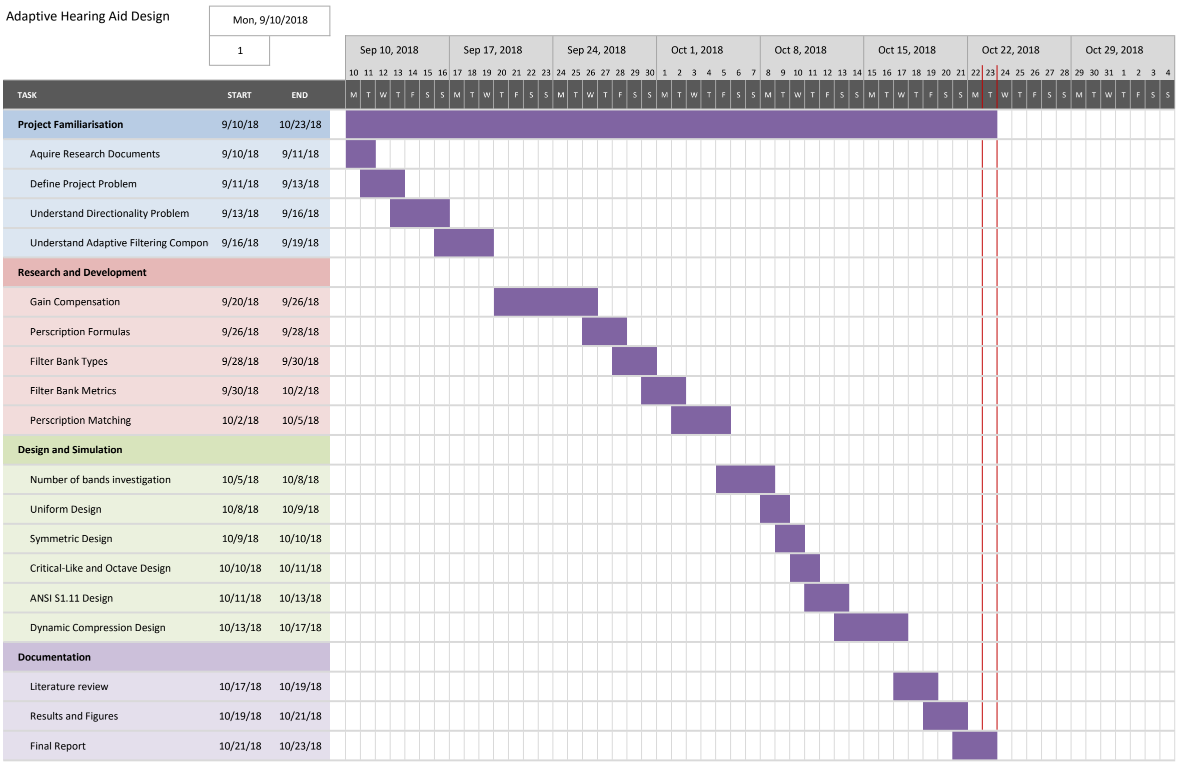

Figure 17: Gantt Chart illustrating Work Progression throughout Project

O. Meeting Minutes

Meeting minutes: Design – Adaptive Hearing Aid 1st meeting

[17 September 2018], [10:15 am]
Wits Chamber of Mines, School of EIE, Seminar Room

Facilitator: Prof. Rubin

Note taker: Arunima Pathania & Fiona Oloo

Attendees: Kavilan Nair, Iordan Tchaparov, Joel Oommen, Arunima Pathania, Kyle Govender, William Becerra, Fiona Oloo, Boitumelo Mantji, Verushen Coopoo, Lindokuhle Mbatha,

Agenda:
Overview

Absent: Daniel Edwards, Jean Jordaan

Agenda item: Overview

A brief introduction to project.

- The design must work in simulation
- The design can be completed in pairs thus a clear work distribution and retrospective Gantt chart will be necessary for Appendices in the report (follow up necessary)
- Read over CBO to ensure ELOs are met. A marking rubric will be made available
- The project is a desktop design i.e. it is not necessary to develop it down to the prototype
- The electronics of a hearing aid is more important viz. a full electronic design is necessary
- “Adaptive”: manually be able to change directionality and the sound level depending on sound environment
- It is not imperative that two hearing aids be designed. Designing as one device is sufficient
- Any simulation package can be used (PSpice, Multisim)
- Note: Hearing aids do not amplify across the entire auditory range
- The success in terms of the efficiency in directionality design is at the students’ discretion with sufficient reasoning
- Report page limit is stated in CBO
- Consider amplification limitations
- It is a sufficient assumption that designing for one ear implies the design would hold for both ears.
- Cost is not a strict design constraint
- For the purpose of this project the audiograms for both ears can be treated identical.
- Official meeting times: 10:15 am Mondays, EIE Seminar room

[11:00 am] Meeting adjourned.

Meeting minutes: Design – Adaptive Hearing Aid 2nd meeting

[27th September 2018], [10:40 am]
Wits University, Flower Hall Entrance

Facilitator: Prof. Rubin

Note taker: Kyle L. Govender

Attendees: Kavilan Nair, Iordan Tchaparov, Joel Oommen, Kyle Govender, William Becerra, Fiona Oloo, Boitumelo Mantji, Verushen Coopoo, Lindokuhle Mbatha, Jean Jordaan, Daniel Edwards

Agenda:
Work division
Project Management
Directionality
Filtering
Miscellaneous

Also Present: None

Apologies: None

Absent: Arunima Pathania

Agenda item: Work Division

- The course co-ordinator, Dr Masisi, emphasises the importance of the ELO pertaining to individual work for this specific project.
- As previously established, students are allowed to work in pairs. An update in this regard is that each student in the pair solely works on either the Filtering or Directionality component of the project.
- Both students in the pairs need to fully comprehend and understand the aspect of the project that their partner is working on i.e. Filtering or Directionality. The design of the Filtering and Directionality needs to be developed such that these two subsystems 'can' be integrated, but actual integration is not required for the final solution.
- The report needs to be centred around the student's allocated aspect of the project, whilst the other should be referenced where appropriate.
- Each member's report should adequately convey that they understand their partner's section of the project.

Agenda item: Project Management

- At this point in the project students should be using the Gantt chart, that should have been already developed in previous weeks, to monitor and track their progress to ensure they are on track.

Agenda item: Directionality

- It can be assumed that the subject has 'normal hearing' i.e. a flat response at all frequencies in the audiogram.
- Polar plots considered must be of a 2D projection, not 3D.
- The achieved design must be at the electronic level of abstraction.
- The resolution of achieved directionality is at the student's justifiable discretion.
- The microphones used in the array need to be omnidirectional.
- Prof. Rubin suggests that a 90° dexterity on either side of the normal to the user's head orientation is a sufficient dexterity of directionality i.e. spanning a 180° spatial field.
- The behaviour of the achieved design should be validated by means of polar plots.
- Directionality behind the user should not be considered.

Agenda item: Filtering

- The achieved design must be at the electronic level of abstraction.
- The designed filters need to be constructed with narrow enough filter bands to compensate for audiogram limitations.
- The filtering needs to be readily adjustable to different audiograms.

- The auditory system and achieved solution can be considered as one composite system i.e. the hearing deficit for each ear is the same.
- Filtering should realistically/practically stop at 8 kHz, students can go higher if they so wish.
- Prof. Rubin suggests that a standard Desktop PC would not be computationally sufficient for the filtering thus, a dedicated chip should be considered.

Agenda item: Miscellaneous

- Cost is not a factor to consider in this design however, there is an upper bound on the practicality of the system that needs to be considered at each student's discretion.
- Prof. Rubin will send a link after the meeting to an image of the overall system block diagram of the group who did the hearing aid for the Lab Project.
- The design of a power system required for the design should not be considered for the purposes of this project i.e. off the shelf solutions or mere bench supplies is sufficient.
- The validation of the system should take the form of simulation and level of functionality according to the well-defined objectives stated at the start of the design, comparison to a 'Gold Standard' is not required.
- A digital system is not required, an analogue or hybrid system is perfectly acceptable.
- The hearing aid is to be designed for the purposes of listening to speech and music.
- Lindo brings forward conflicting ELO requirements in two different versions of the Design Project Course Brief and Outline, Prof. Rubin indicates that he will confirm which is the correct version and convey this information to the students.

[11:30 am] Meeting adjourned.

Meeting minutes: Design – Adaptive Hearing Aid 3rd meeting

[01 October 2018], [10:30 am]
Wits Chamber of Mines, School of EIE, Seminar Room

Facilitator: Prof. Rubin

Note taker: Lindokuhle Mbatha

Attendees: Kavilan Nair, Iordan Tchaparov, Joel Oommen, Arunima Pathania, Kyle Govender, William Becerra, Fiona Oloo, Boitumelo Mantji, Verushen Coopoo, Lindokuhle Mbatha, Jean Jordaan, Daniel Edwards

Agenda:
Recap
Project Management
Directionality
Filtering
Next meeting

Also Present: Dr.Nitch

Apologies: None

Absent: None

Agenda item: Recap

- Students need to work individually as this is one of the requirements for the course.
- However, students are allowed to work in pairs to a certain extent, then diverge in which case one student works on directionality and the other works on filtering.
- For directionality, assume a flat audiogram within the frequency range of 20Hz – 20kHz.
- Regarding polar plots, only consider the plot in the horizontal plain at ear level.

Agenda item: Project Management

- You should have a Gantt chart to help keep track of you progress.
- Apply the project management skills learnt in previous courses.

Agenda item: Directionality

- Jean (Suggestion):
  - ➢ Having an array of 4 microphones arranged in square formation.
  - ➢ Directionality can be achieved using either DSP or analogue (electronics).
- Prof. Rubin (Response)
  - ➢ 4 elements (microphones) seems to be the bare minimum.
  - ➢ Doing directionality using DSP may not be viable.
- Jean (Question): What are the upper limits of the design and to what extent should the directionality be?
- Prof. Rubin (Response):
  - ➢ We have to design for real world application, i.e. we have to specify the degree/resolution of directionality with good justification. This will require some background research.
  - ➢ Consider dB tolerance from lateral direction while someone is listening in one specific direction.
  - ➢ (Side note) Do research on Lombard effect, it may be something to consider in your design ('may' being the operative word).
- Jean (Question): In electromagnetic (EM) systems and array theory, the length of conductors introduce a phase shifts, should we correct for this? Also, in reconstructing the source when considering the signals received from multiple microphones, one has to consider relative phase shifts and the correction thereof, isn't this noise cancellation in itself?
- Prof. Rubin and Dr. Nitch (Response):
  - ➢ In auditory systems, as opposed to EM systems, electronic delays are negligible.
  - ➢ Directionality deals with differential amplification based on direction.
  - ➢ The technique of correcting the signal based on multiple inputs from multiples microphones (receivers) is called maximum-ratio combining (MRC) and it is not a noise cancellation technique.
  - ➢ Noise cancellation is adding the negative or inverted form of a noise signal to the input signal that may contain noise.

Agenda item: Filtering

- Fiona (Question): Is it acceptable to assume that the signal coming from the directionality subsystem has maintained its integrity (i.e. is no different from the source signal), or should we consider any alteration that might have been introduced by the directionality subsystem and will the process of band separation affect directionality?
- Prof. Rubin (Response):
  - Have to consider the fact that both the directionality and filtering subsystems should be able to work as a unit that forms one device.
  - Number of channels may affect directionality.
- Prof Ruben: For directionality, it is not the source of the sound that is being corrected, rather it is the response/detection of the auditory system to that sound that is being corrected. The purpose of the device is to amplify specific bands more than other on an audiogram.
- Note that the sound level is specified in dB with respect to some reference, in this case it is dB SPL (dB of sound pressure level).
- Joel (Question): Why should the students working on filtering worry about the design aspects of the directionality?
- Prof. Rubin (Respnse): The two subsystems should be compatible in terms of electronics. Note that we should not design a hybrid system, but we should consider the interaction of the two subsystems.
- Fiona (Question):
  - How do we go about designing for signal to noise ratio and should we incorporate feedback cancellation of internal noise?
  - Is it necessary to design a compression circuit?
- Prof. Rubin (Response):
  - For filtering the main requirement is to amplify frequency regions where it is required based on the audiogram. Ignore microphone noise, thus we don't have to design feedback compensation (this is an assumption).
  - Can design a feedback cancellation for microphone noise if time permits, but make sure you have met all the required specifications of the project.
  - It is necessary to have a compression circuit.
- Lindo (Question): Should we consider any delays in frequency components of the signal that are introduced by the filtering process?
- Prof. Rubin (Response): Have to do the relevant research to determine if it's necessary.
- Consider looking at the World Health Organization (WHO) Minimal Performance Requirements of hearing aids for more project specs.

Agenda item: Next meeting

- From henceforth all upcoming meetings will be held on Mondays 10:15 am at Chamber of Mines (School of EIE's seminar room) till further notice.

[11:10 am] Meeting adjourned.

Meeting minutes: Design – Adaptive Hearing Aid 4th meeting

[08 October 2018], [10:40 am]
Wits Chamber of Mines, School of EIE, Seminar Room

Facilitator: Prof. Rubin

Note taker: Amprayil Joel Oommen

Attendees: Kavilan Nair, Iordan Tchaparov, Amprayil Joel Oommen, Arunima Pathania, Kyle Govender, William Becerra, Fiona Oloo, Boitumelo Mantji, Verushen Coopoo, Lindokuhle Mbatha, Jean Jordaan, Daniel Edwards

Agenda:
Apology for Delay in Meeting Start
Filtering
Directionality
Report-related matters
Miscellaneous

Apologies: None

Absent: None

Agenda item: Apology for Delay in Meeting Start

- Prof. Rubin had several matters that he had to attend to and was hence, delayed in getting to the meeting on time. Prof had emailed students and notified Jeannine at the reception, to inform students about the delay.

Agenda item: Filtering

- Fiona (Question):
  - ➢ Many dedicated Hearing Aid Digital Signal Processors (DSPs), have predefined filter designs (e.g. centre frequencies, bandwidths); however, this prevents designing our own system with our own considerations.
  - ➢ Hence, once decided on a design for the filter (e.g. orders, gains), are we required to design a DSP which is able to implement the filters?
- Prof. Rubin (Response)
  - ➢ Rather make use of readily available DSPs, capable of handling the filter design you would like to implement.
  - ➢ Can design your own DSP if you want, however, given the current time constraints and even looking at potential commercial South African start-ups, Prof believes it will be easier using readily-available DSPs, components, etc.
- Fiona (Question): Are we able use design considerations from literature, such as attenuation in the cut-off bands, when designing our own filters (e.g. attenuation of 40 dB outside of the passbands, resulting in sounds in those frequency ranges being softer)?
- Prof. Rubin (Response): It is a fair assumption to make.
- Joel (Question): Is the audiogram stored in the DSP? Is it sufficient to provide audiologists with the capability to adjust the gain levels between certain frequency ranges?
- Prof. Rubin (Response):
  - ➢ DSPs are required to be able to store the audiogram of an individual, so that the hearing aid is adjustable for them (e.g. if the individual's hearing deteriorates as they age, then the hearing aid must be able to be adjusted to cater for that).
  - ➢ An acceptable means of achieving this, is by designing the hearing aid with a discrete number of filters which operate on different frequency ranges and where its gains can be adjusted as required – this is a means of "storing and normalizing" one's audiogram.
- Lindo(Question): Last week, you had specified that we require a compression design, for our Dynamic Range Compression. Are we also required an Expansion Section, within the Compression design, which is a suggested solution for a specific psychoacoustic problem?
- Prof. Rubin (Response): You can include expansion in the design, but you do not need to go into detail about the psychoacoustic analysis behind the implementation.

Agenda item: Directionality

- Fiona (Question): With regards to focusing on sounds at a particular angle (i.e. having a main lobe focusing on a specific region), how big should the successive "next" lobes be? What must the dB difference between the main lobe and the successive lobes? Is there some sort of threshold value that we should use for that dB value?
- Prof. Rubin (Response): Use your discretion and design in accordance to what you need / require for your directionality.
- Jean (Question): Depending on the array size, the way we approximate a near-field and far-field sound will be different (i.e. characteristics such as wavelengths of the sound, speed of sound and the distance of the source from the listener). Is it ok for the rest of the design, that we make an assumption that both the near-field and far-filed are treated the same?
- Prof. Rubin (Response): Absolutely, show that you understand the problem at hand, but then move on with the design, by stating the compromises and design decisions. Such a consideration can only be tested and verified using practical experimentation, hence show that you have considered it, but move on with the design.

Agenda item: Simulations and Results

- William (Question): Is it fine to perform simulations of the system on Simulink, using predefined simulation blocks (e.g. Octave bandwidth filters)?
- Prof. Rubin (Response):
  - If elect to go this route, will have high-level system simulations.
  - Will be required to make justifications for design considerations and will need to make certain assumptions of the performance of the device, etc.
- William and Joel (Question): How do we evaluate the errors and performance of the system, seeing that this is a paper design?
- Prof. Rubin (Response):
  - First, you must decide upon design specifications for the Hearing Aid.
  - Then, go about designing and simulating the system.
  - When evaluating and critiquing the design, compare the original design specifications with the simulated design and see if the design meets requirements.
- William (Question): What results are we required to include in the report?
- Prof Rubin (Response):
  - *Filtering*: Outputs from the filters and perhaps also the compression algorithms /techniques.
  - Not practical to test the designed filters using the entire frequency range for speech. Therefore, try get a few input speech samples (of different frequencies) and obtain the outputs of the filters for those input.
  - *Directionality*: Show outputs of the directionality system (i.e. on a polar plot), when different point sources of sound are used.

Agenda item: Report-related matters

- Verushen (Question): Please could more clarification be given, pertaining to how the two partner's systems are to be integrated. Is it for example necessary to simulate a signal being inputted into one of the systems, and its subsequent output being used as the input of the second system?
- Prof Rubin (Response):
  - The report ***MUST NOT*** be a presentation of the hybridization of the two partner's systems – time constraints are too limiting, and this would only be required to be done in a commercial project.
  - Rather, the two systems must be able to be joined in "*principle*" – ensure that communication and thought has taken place in collaboration with your partner, taking reasonable precautions, etc. (e.g. knowing how many outputs will come out of the one system, and using those as inputs of the secondary system).
- Prof Rubin (Comment): This is a paper design, and we do not have the capability of physically testing and making subsequent changes; therefore, be reasonable and make decent design consideration and assumptions.

- Jean (Question): In the rubric, in order to achieve a "Good" or "Excellent" rating, we are required to show evidences of outstanding features in the design, etc. Please could you clarify what that means in this context?
- Prof Rubin (Response):
  - The markers of the report, understand the time constraints and the difficulty of the problem presented to students.
  - Therefore, to evidence outstanding features, show that you have thought clearly through matters (even if they present themselves as problems that are difficult to address), justifying the design, showing the compromises that have had to be made, etc. If you can evidence this clear thinking and can motivate decisions, it is likely that the markers would consider this favourably.
  - What we are more concerned is the design cycle and how you approached solving the problem, as opposed to the actual design.
- Jean and Verushen (Question): Please could you give details about the content and structure of the Appendices?
- Prof Rubin (Question):
  - *Appendix A*: The CBO mentions this must be a non-technical report of the impacts of the engineering solution, and about the improvements that can be applied in reducing the negative impacts.
  - This can detail the social and economic impacts of the hearing aid on the community (both locally in South Africa and even internationally) and vice versa.
  - Format of Appendix A must follow IEEE standards, as well as anything further dictated in the CBO. Can write it like an essay, can insert headings if you desire, etc. (provided the proper standards and dictated guidelines are conformed to).
  - *Appendix B and C*: Previously, had mentioned that a Gantt Chart could be included as an Appendix. Furthermore, it would be valuable including a rough cost breakdown of the entire hearing aid device (you will not be marked down if you do not have one, nonetheless, it can be insightful in giving an idea of how sustainable an idea this is, thus aiding in evaluating the appropriateness of the design etc).
- Joel (Question): Please can more clarification be given on the format of the report.
- Prof Rubin (Response): Use the guidelines given in the CBO to dictate the details of the style requirements of the report (e.g. 15 pages, single column), and use the Blue Book to ensure that these details conform to the IEEE styling standard.

Agenda item: Miscellaneous

- Lindo (Question): Are we allowed to use a FPGA, instead of a DSP, if necessary?
- Prof Rubin (Response): Yes. The FPGA allows you to digitally code the device and dedicate the device for processes / operations / functionality. In addition, it allows for parallelism of operations. Can be a good alternative to the DSP if need be.

[11:35 am] Meeting adjourned.

Meeting minutes: Design – Adaptive Hearing Aid 5th meeting

[15 October 2018], [10:15 am]

Wits Chamber of Mines, School of EIE, Seminar Room

Facilitator: Prof. Rubin

Note taker: Kavilan Nair

Attendees: Kavilan Nair, Iordan Tchaparov, Joel Oommen, Arunima Pathania, Kyle Govender, William Becerra, Fiona Oloo, Boitumelo Mantji, Verushen Coopoo, Lindokuhle Mbatha, Daniel Edwards,Jean Jordaan

Agenda:

Revised Project Brief

Directionality

Filtering

Next meeting

Apologies: None

Absent: None

Agenda item: Revised Project Brief

- Revised brief was sent out in the morning which mentioned that a **low cost** should be a requirement. **PLEASE IGNORE THIS STATEMENT AS THE DESIGN DOES NOT HAVE COST AS A CONSTRAINT.**

Agenda item: Directionality

- Verushen (Question): There are a lot of polar plots that we can show in the report, should we include all?
- Prof. Rubin (Response):
  - ➢ Only some key plots should be necessary
  - ➢ Jean (Suggestion): Can plot a 3D graph to combine and make the graphs concise.
- Kyle (Question): Is it advantageous to increase the range beyond 180 degrees?
- Prof. Rubin (Response):
  - ➢ Not really, you can if you want but make sure it doesn't sacrifice the performance
- Verushen (Question): Can we use digital mics?
- Prof. Rubin (Response):
  - ➢ You can use it but it must be compatible with what your partner needs to work with
- Iordan (Question): Is there a limit to the number of DSPs that the design uses?
- Prof. Rubin (Response):
  - ➢ No hard cap, but it should be within reason and practical.
- Kyle (Question): Is 100 microphones too many?
- Prof. Rubin (Response):
  - ➢ It needs to be reasonable

Agenda item: Filtering

- Fiona (Question): I have picked my DSP device, do I need to explain the concepts of it or have a detailed pinout?
- Prof. Rubin (Response):
  - ➢ Yes, you need a pinout so you can embed it in the circuit diagram
- Joel (Question): What level of detail is required for the electronic circuit diagram and algorithms?
- Prof. Rubin (Response):
  - ➢ Given the allocated time for this project, it doesn't have to be low level
  - ➢ Can provide the pseudocode in the form of flowchart for the algorithms implemented on the DSP

- Fiona (Question): If the DSP already contains a built in DAC, do I need to design my own?
- Prof. Rubin (Response):
    - ➢ Can treat the DSP as a black box
    - ➢ Don't worry about the internals of the device, using an example, you dont need to worry about the transistor level if you are using an operational amplifier
- Boitumelo (Question):
    - ➢ Is the DAC and speaker design components only for the adaptive filtering students?
- Prof. Rubin (Response):
    - ➢ This is where the teamwork comes into play and it depends on the design and how the two systems integrate.
- William (Question): Do we need to simulate the DAC and ADC?
- Prof. Rubin (Response): It isn't necessary.
- Fiona (Question): Is it better to have full simulation or modular simulations
- Prof. Rubin (Response):
    - ➢ Makes sense to modularize and it makes things a bit easier
- Joel (Question): Is Dynamic Range Compression (DRC) essential?
- Prof. Rubin (Response):
    - ➢ Yes, you have to have compression
- Lindo (Question): Can we use libraries?
- Prof. Rubin (Response):
    - ➢ Yes you can, but make sure you show the design thought that you put into setting some of those parameters
- Lindo (Question): In terms of building up the design and justifying, can I use a filter bank with variable bandwidths?
- Prof. Rubin (Response):
    - ➢ Sure, as long as you can justify it.
- Fiona (Question): What is meant by adaptive?
- Prof. Rubin (Response):
    - ➢ The directionality must be adaptive in terms of being able to focus the microphones within a specific range
    - ➢ The filter must be adaptive in terms of the gains adjusting to match the audiogram
- William (Question): Do we need to show the power supply circuitry?
- Prof. Rubin (Response):
    - ➢ Yes you should, show the power to the components with a voltage regulator for example.
- Fiona (Question): Can we map the -1 -> 1 of a signal as the dynamic range?
- Prof. Rubin (Response):
    - ➢ Not too sure, maybe test it yourself or message Kirsten at Optinum?

Agenda item: Report and group aspects

- Joel (Question): How much does the individual report relate to the other group member
- Prof. Rubin (Response): Don't talk about the detail implementation of partners system. Talk in terms of input/output and how the two subsystems will interface with each other.
- William (Question): In terms of the integration of the two systems, will there be a penalty if that parts doesnt make sense.
- Prof. Rubin (Response): There will be a penalty, as group work does make up a component of the project.
- Jean (Question): When we are discussing the other person's design in the report, how do we reference it?
- Prof. Rubin (Response): You should have a section where you discuss the partners design in terms of input/output and you should reference their report like you would reference any other paper.
- Prof. Rubin (Statement): Write the partners name on the front of the report, but it should be small and at first glance the reader should know whose paper this is.
- Verushen (Question): Should the abstract go on the cover page?
- Prof. Rubin (Response): It should not be on the cover page.
- Iordan (Question): Can we have an acknowledgements section?

- Prof. Rubin (Question): It is essential, especially for directionality, with the help from Dr. Nitch.
- Dan (Question): Can I use a design approach where I build up and prove some of the design decisions, such as how many filters to use.
- Prof. Rubin (Response): Sounds good, as long as you link it back to the original spec. One thing to note is that Design is not always objective, it can be subjective.

Agenda item: Next meeting

- There will be a meeting next week on the 22nd of October, this meeting will however, not be compulsory and no minutes will be taken.

[11:15 am] Meeting adjourned.