School of Electrical & Information Engineering
University of the Witwatersrand, Johannesburg
Private Bag 3, 2050, South Africa

ELEN4011 - Design Project

Biomedical Engineering - Design of an Adaptive Hearing Aid

23 October 2018

Supervisor: Prof. D. Rubin

Daniel Edwards
1055301

Contents

Abstract

This is my abstract

1. Introduction

define what adaptive means

2. Background

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 loneliness, isolation, dependence, depression and frustration. These are signs of poor quality of [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 to improve the patients speech intelligibility.

3. Design Research

Many factors contribute to hearing loss. A common type is conductive hearing loss [3]. This is a result of age, excessive noise or various diseases [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 3.1) and a literature review (Section 3.2).

3.1 Existing Solutions

Two main hearing assisting devices exist for sensorineural hearing impairments namely; hearing aids and cochlear implants [5]. Cochlear implants are typically used in cases of extreme hearing loss. These devices work well in quiet environments [6]. However, cochlear implant users battle to perceive speech in noisy environments [6] and thus, the individuals quality of life is not improved. 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[7]. Ricketts [8] 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 [8]. Thus, these hearing aids improve the patient's speech intelligibility which can positively their social experience.

put something in about Ricketts and avoiding noise - isolation, say something about adaptive

3.2 Literature Review

Sebastian *et al.* [9] 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 [10]. An audiogram illustrates the hearing threshold of a person at different pitches and frequencies [9]. 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

[11]. 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 [11]. This method is also mentioned by Sebastian *et al.* and is referred to as selective amplification [9].

Sebastian *et al.* provide an frequency response matching (FRM) investigation between uniform and non-uniform filter banks [9]. 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 [9]. According to Sebastian *et al.*, the frequency bands at the critical hearing frequencies should be narrow for the best compensation [9]. 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 [12]. 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 [13] 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.* [12] and Yang *et al.* [14] 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 [15]. These requirements were developed by the Acoustical Society of America specifically for audio applications [15]. 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 [12], [14].

Reference [16] discusses the use of dynamic range compression. 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 [17]. 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 [8] 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 [18]. Hearing impaired individuals however, require an SNR greater than $+6dB$ to achieve sufficient hearing [19]. This increase SNR requirement is because impaired hearing caused by poor hearing thresholds, results in noise related problems [8]. 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.* [20] suggest that using multiple microphones results in better extraction, separation of frequencies and localization of the source whilst reducing noise, interference, echo and reverbera-

[1]Signal to noise ratio.

tion. This provides motivation for utilising a multiple microphone array in this design. Resource [21] 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 [21]. In addition, McCowan [22] 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 [23] 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 $8kHz$ [24]. Since this system aims to correct audiograms, a frequency of $20Hz \rightarrow 8kHz$ is considered.

McCowan [22] 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.

Put something in about the number of bands that increases the matching error Put the stuff about the audiogram gain factor thingy
mention stuff about why digital is better than analog
talk about group delay
mention stuff about multirate filters

4. 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.

5. Design Requirements

6. Design Assumptions and Constrains

6.1 Assumptions

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.

6.2 Constraints

This system is constrained to a bandwidth of $20Hz$ to $8kHz$ to correspond to the audiogram frequency range. A system delay of $6ms \rightarrow 8ms$ is noticeable and a $20ms$ delay results in confusion between the audio sound and visual movements [25]. Therefore, this system must have an maximum response time of $20ms$ or less.

7. 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. 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 on 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. Once a design is selected, a suitable DSP[3] 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.

Mention the compression aspect

ideal filter bank

8. 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.

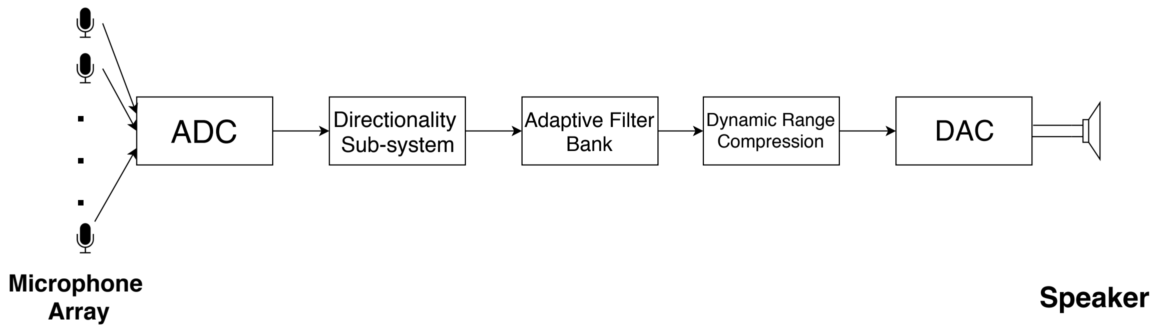

Figure 1: Overall System Block Diagram

9. Adaptive Filter Design

State the factors that contribute to the optimisation
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. This design subsection aims to utilise this frequency bank rectify a patients audiogram. This audiogram is presented in Section 9.1.

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 9.4.1, 9.4.2, 9.4.3 and 9.4.4 present the design of the uniform, critical-like, symmetric and octave filter banks respectively. These designs will also be compared to the ANSI S1.11 specification presented in Section 9.5. The directionality sub-system

[2]Digital-to-Analog-Converter
[3]Digital-Signal-Processor

relies on the phase response of the system and thus, only *FIR* filters (because of their linear phase 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 [12].

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

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.

9.1 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.

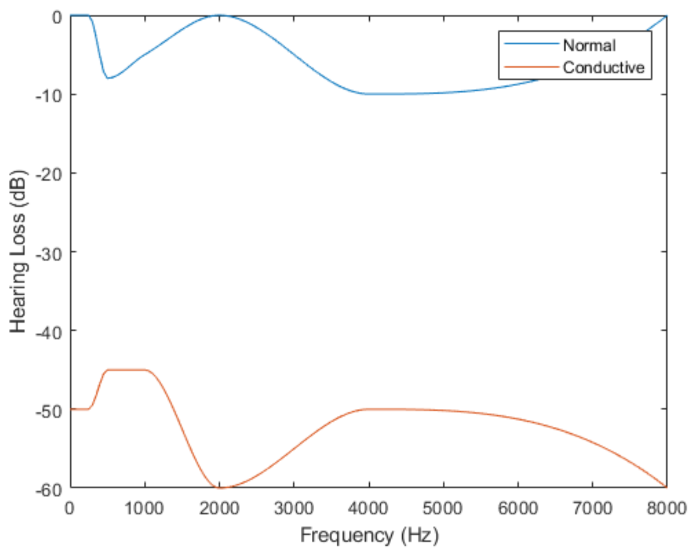

Figure 2: Audiogram for Normal Hearing and Conductive Hearing Loss

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

9.2 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. In this design, the insertion gain is calculated using the

NAL-R formulas given in equation 2.

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

Where X is a constant, 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 1 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.

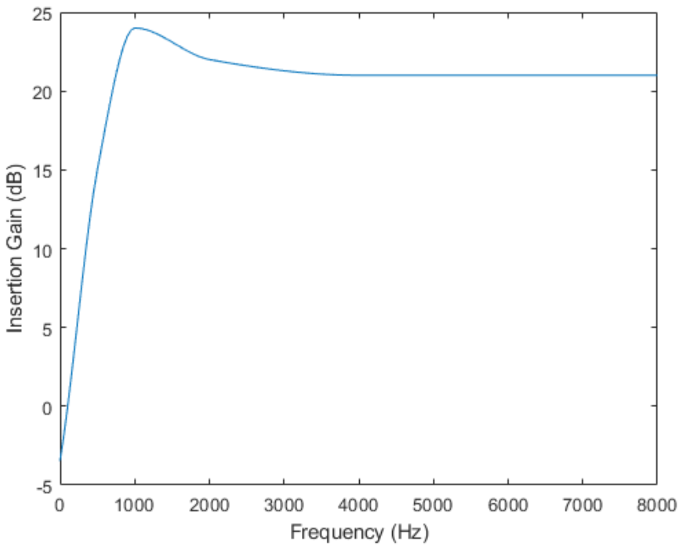

Figure 3: Interpolated Insertion Gain for Full Frequency Range

The insertion gain, per sub-band is taken as the average insertion gain within the sub-band range.

9.3 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.

9.4 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, critical-like, symmetric 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 2 in Appendix C. Each filter bank will consist of 8 bandpass

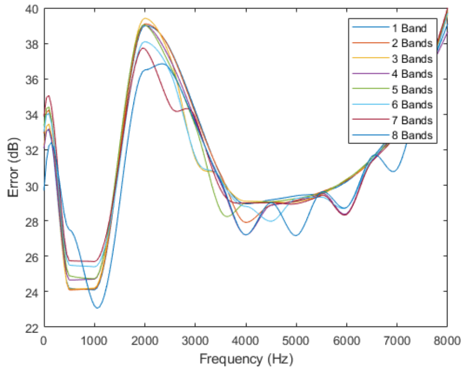

Figure 4: Matching Error Across Frequency Spectrum per Filter Bank

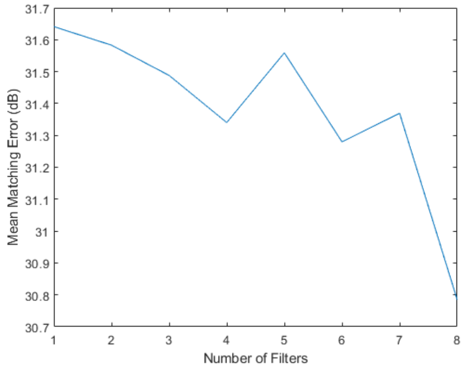

Figure 5: Mean Matching Error per Filter Bank

filters. Initially, the transition band per sub-band is set to 10% of the sub-band's bandwidth. However, 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 9.2 will be used.

9.4.1 Uniform Filter Bank Design A uniform filter bank consist of an array of evenly spaced, filters with equal bandwidths [12]. 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 [25]. This design optimised Section 9.3's 8 sub-band design by utilising the filter characteristics in Table 12 of Appendix E. This design acheived 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 7 of Appendix E.

9.4.2 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 11 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 13 of Appendix F. 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 8 of Appendix F.

9.4.3 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 [9]. 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 G 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 9 in Appendix G.

9.4.4 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 [13]. 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 15 of Appendix H. Simulations demonstrated that this filter bank achieved an average and maximum matching error of $0.72dB$ and $10.44dB$ respectively. Figure 10 in Appendix H illustrates the matching error across the full frequency spectrum. The group delay for this filter was $37.45ms$.

9.5 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 [14]. 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 16 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 11 in Appendix I illustrates the matching error across the full frequency spectrum.

9.6 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 [12]. 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 9.7.

9.7 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 9.3, 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

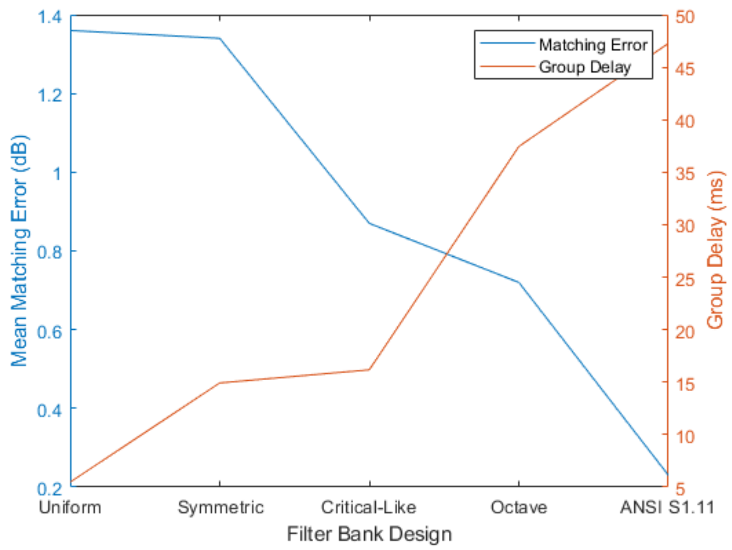

Figure 6: Mean Matching Error and Group Delay for each Filter Bank Design

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 9.6, 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 9.5 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 17 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 illustrates in Figure 12 of Appendix J.

10. Dynamic Range Compression Design

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. To maximise speech intelligibility, the parameters are set as shown in Table XXX [16].

11. System Realisation

This subsystem consists of an input, an analysis and synthesis stage and finally an output. The hardware required to realise this system consists of a DSP, DAC, anti-imaging filter and speaker. TALK ABOUT THE INPUT. The Analog Devices, *ADAU1463 Sigma Digital Audio Processor* is chosen as the DSP for this system.

talk about the DSP, DAC, anti-imaging filter, speaker

12. Success Criterion

what defines success. Look at the requirements. matching error, group delay

13. Critical Evaluation of Results

13.1 Socioeconomic Impacts of Design

14. Future Recommendations

why you dont use multirate filters
divide into what needs to be improved and what the implementation considerations must be

15. Conclusion

References

[1] U. of the Witwatersrand, *International ear care day - act now, here's how!* 2016.

[2] A. Ciorba, C. Bianchini, S. Pelucchi, and A. Pastore, "The impact of hearing loss on the quality of life of elderly adults," *Clinical Interventions in Aging*, vol. 7, pp. 159–163, Jun. 2012. DOI: 10.2147/CIA.S26059. [Online]. Available: https://www.ncbi.nlm.nih.gov/pmc/articles/PMC3393360/.

[3] E. Earinfo.com, *Hearing aid resource for consumers*, 2018. [Online]. Available: http://www.earinfo.com/faq/hearing-loss.

[4] Hear-it.org, *Sensorineural hearing loss (snhl) - causes and treatments*, 2018. [Online]. Available: https://www.hear-it.org/Sensorineural-hearing-loss.

[5] A. H. Centers, *Cochlear implant vs. a hearing aid - what's the difference?* Feb. 2018. [Online]. Available: https://www.atlantichearingcenters.com/blog/2018/feb02-cochlear-implants-vs-hearing-aids.

[6] J. Carroll, S. Tiaden, and F.-G. Zeng, "Fundamental frequency is critical to speech perception in noise in combined acoustic and electric hearing," *The Journal of the Acoustic Society of America*, vol. 130, no. 4, pp. 2054–2062, Oct. 2011. DOI: 10.1121/1.3631563. [Online]. Available: https://www.ncbi.nlm.nih.gov/pmc/articles/PMC3206909/.

[7] C. Woodford, *How do hearing aids work?* Apr. 2018. [Online]. Available: https://www.explainthatstuff.com/hearingaids.html.

[8] T. A. Ricketts, "Directional hearing aids," *Trends In Amplification*, vol. 5, no. 4, pp. 139–176, 2001.

[9] A. Sebastian and J. T. G., *Digital Filter Bank for Hearing Aid Application Using FRM Technique.*

[10] *Audiometry test, hearing test*, Apr. 2018. [Online]. Available: https://www.mayfieldclinic.com/PE-Hearing.HTM.

[11] K. Kakol and B. Kostek, "2016 signal processing: Algorithms, architectures, arrangements, and applications (spa)," in. IEEE, pp. 219–224.

[12] K.-C. Chang, M.-H. Chuang, and C.-H. Lin, "10-ms 18-band quasi-ansi s1.11 1/3-octave filter bank for digital hearing aids," *IEEE TRANSACTIONS ON CIRCUITS AND SYSTEMS*, vol. 60, no. 3, pp. 638–649, Aug. 2012. DOI: 10.1109/TCSI.2012.2209731. [Online]. Available: https://ieeexplore.ieee.org/document/6268303.

[13] S. Shearman, *Analyze signals octave by octave*, Apr. 2001. [Online]. Available: https://www.edn.com/electronics-news/4382916/Analyze-signals-octave-by-octave.

[14] C.-Y. Yang and C.-W. Liu, "A systematic ansi s1.11 filter bank specification relaxation and its efficient multirate architecture for hearing-aid systems," *IEEE/ACM TRANSACTIONS ON AUDIO, SPEECH, AND LANGUAGE PROCESSING*, vol. 24, no. 8, pp. 1380–1392, Aug. 2016. DOI: 10.1109/TASLP.2016.2556422. [Online]. Available: https://ieeexplore.ieee.org/stamp/stamp.jsp?arnumber=7457341.

[15] T. E. D. O. of the Federal Register and A. Accredited Standards Institute Committee S1, *American National Standard Specification for Octave-Band and Fractional-Octave-Band Analog and Digital Filters.* Acoustical Society of America (ASA).

[16] S. H. Technologies, *The Compression Handbook*, 4th ed. The Compression Handbook, 2017.

[17] R. Dhawan and P. Mahalakshmi, "International conference on electrical, electronics, and optimization techniques (iceeot) - 2016," in *Digital Filtering In Hearing Aid System For The Hearing Impaired.* IEEE, pp. 1494–1497.

[18] B. C. J. Moore, *An Introduction to the Psychology of Hearing*, 6th ed. Emerald Group Publishing Limited, 2012.

[19] T. Tillman and R. Carhart, "Interaction of competing speech signals with hearing losses," *Archives of Otolaryngology*, vol. 91, no. 3, pp. 273–279, Mar. 1970.

[20] J. Chen and J. Benesty, *Design and implementation of small microphone arrays*, 2014.

[21] InvenSense, *Microphone array beamforming*, 2013.

[22] I. McCowan, "Robust speech recognition using microphone arrays," PhD thesis, Queensland University of Technology, 2001.

[23] B. Munir, *Voice fundamentals - human speech frequency*, Mar. 2012. [Online]. Available: http://www.uoverip.com/voice-fundamentals-human-speech-frequency/.

[24] *The audiogram.* [Online]. Available: https://www.asha.org/public/hearing/audiogram/.

[25] R. Brennan and T. Schneider, "A flexible filterbank structure for extensive signal manipulations in idigital hearing aids," *ISCAS '98. Proceedings of the 1998 IEEE International Symposium on Circuits and Systems*, vol. 6, pp. 569–572, Aug. 2002. DOI: 0.1109/ISCAS.1998.705338. [Online]. Available: https://ieeexplore.ieee.org/document/705338.

[26] A. Mahmoud, *Reading an audiogram*, Jun. 2015. [Online]. Available: https://www.slideshare.net/amirmah/reading-an-audiogram.

[27] K.-S. Chong and B.-H. Gwee, "A 16-channel low-power nonuniform spaced filter bank core for digital hearing aids," *IEEE TRANSACTIONS ON CIRCUITS AND SYSTEMS—II: EXPRESS BRIEFS*, vol. 53, no. 09, pp. 853–857, Sep. 2009. DOI: 10.1109/TCSII.2006.881821.

[28] R. J. Cassidy and J. O. Smith III, *Third-octave filter banks*, Jun. 2008. [Online]. Available: https://ccrma.stanford.edu/realsimple/aud_fb/Third_Octave_Filter_Banks.html.

[29] C.-Y. Yang, C.-W. Liu, and S.-J. Jou, "A systematic ansi s1.11 filter bank specification relaxation and its efficient multirate architecture for hearing-aid systems," *IEEE/ACM TRANSACTIONS ON AUDIO, SPEECH, AND LANGUAGE PROCESSING*, vol. 24, no. 8, pp. 1380–1392, Aug. 2016. DOI: 10.1109/TASLP.2016.2556422. [Online]. Available: https://0-ieeexplore-ieee-org.innopac.wits.ac.za/document/7457341.

[30] J. O. Smith III, *The bark frequency scale*, Feb. 2018. [Online]. Available: https://ccrma.stanford.edu/~jos/sasp/Bark_Frequency_Scale.html.

Appendix

Appendices

A Impact Appendix - Non-Technical thing

B Insertion Gain Parameters

The values of k_i is determined by Table 1.

Table 1: 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 2.

Table 2: 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 3 to 10 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 3: Filter Parameters - 1 Band

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

D Bark-Scale

Table 11 provides the critical frequency bands of the Bark-scale [30].

E Uniform Filter Bank Specifications and Results

Table 12 provides the specifications for each sub-band in the uniform filter bank.

Table 4: 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 5: 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 6: 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 7: 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 8: 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 9: 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 10: 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
$A_{s1}(dB)$	$G1dB+3$	$G1dB+G2dB+3$	$G2dB+G3dB+3$	$G3dB+G4dB+3$	$G4dB+G5dB+3$	$G5dB+G6dB+3$	$G6dB+G7dB+3$	$G7dB+G8dB+3$
$A_{s2}(dB)$	$G1dB+G2dB+3$	$G2dB+G3dB+3$	$G3dB+G4dB+3$	$G4dB+G5dB+3$	$G5dB+G6dB+3$	$G6dB+G7dB+3$	$G7dB+G8dB+3$	$G8dB+3$

Table 11: 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

Table 12: 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 7 illustrates the matching error of the uniform filter bank across the full frequency spectrum.

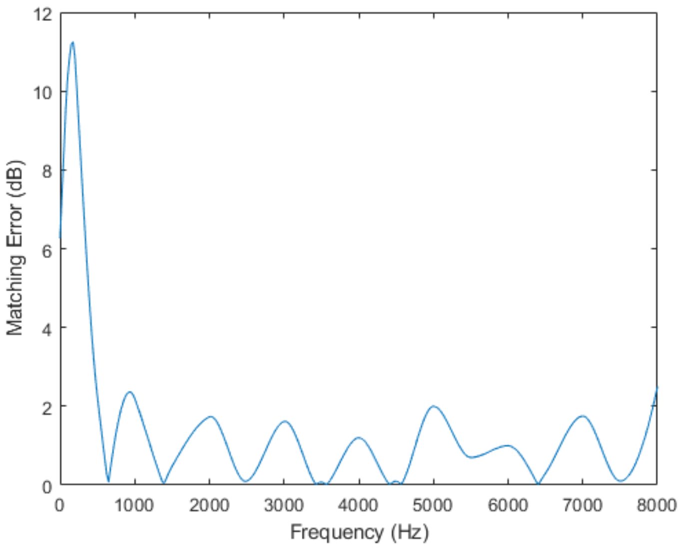

Figure 7: Matching Error for Symmetric Filter Bank

F 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 13.

Figure 8 provides the matching error of the critical-like filter bank across the full frequency spectrum.

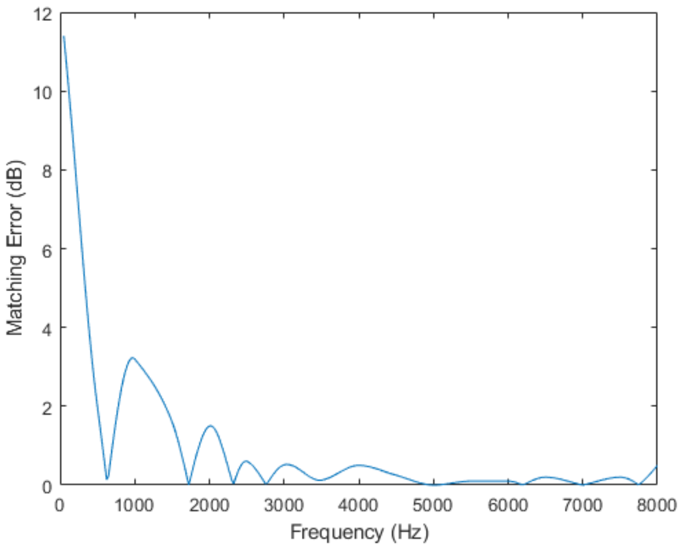

Figure 8: Matching Error for Critical-Like Filter Bank

G 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

Table 13: 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

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

H Octave Filter Bank Specifications and Results

Table 15 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 10 illustrates the matching error of the octave filter bank across the full frequency spectrum.

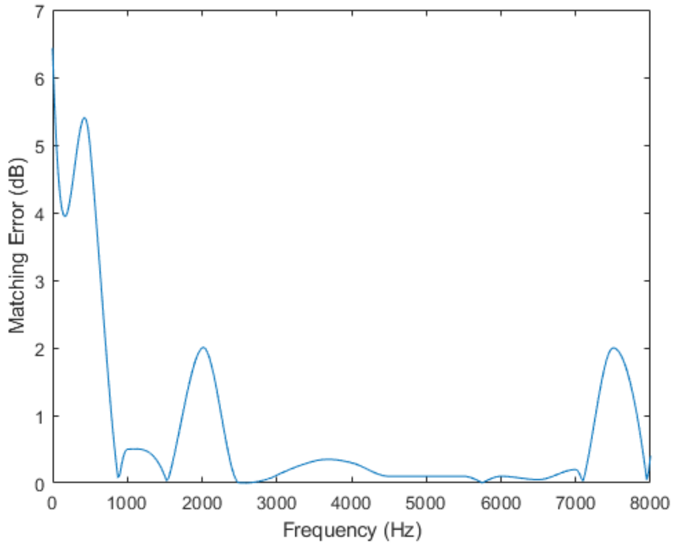

Figure 9: Matching Error for Symmetric Filter Bank

Table 15: 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 16 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.

Table 16: 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

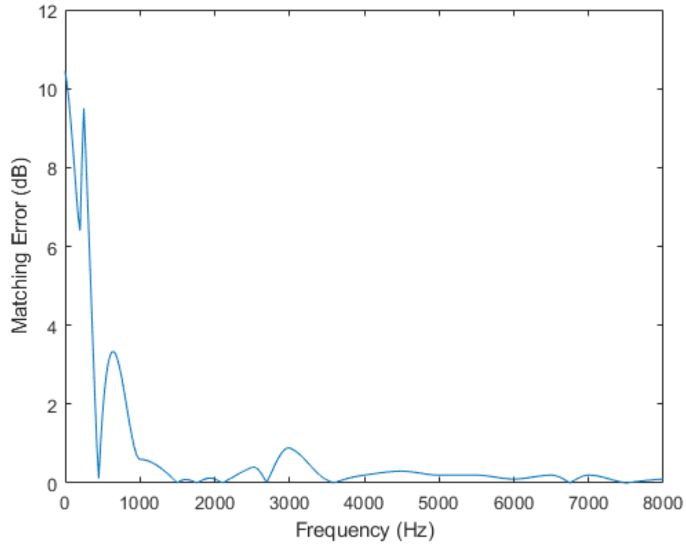

Figure 10: Matching Error for Octave Filter Bank

Figure 11 illustrates the matching error for the full frequency spectrum.

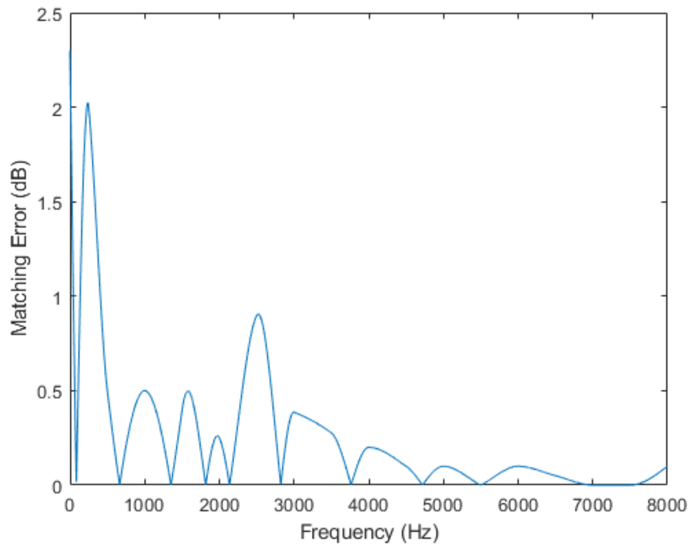

Figure 11: Matching Error for ANSI S1.11 Octave Filter Bank

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 17 summarises the specifications for the optimised filter bank.

Figure 12 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.

Table 17: 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

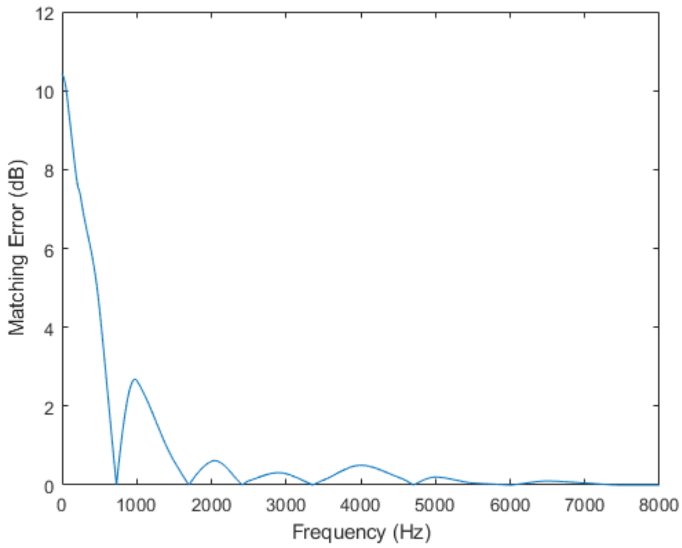

Figure 12: Matching Error for Optimised ANSI S1.11 Filter Bank

- Compression Threshold (CT): 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 (CR): This parameter (shown in equation 6) defines how much the input signal will 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 (AT): The attack time is the delay from when the input signal surpasses the compression threshold, to when the compression takes affect.
- Release Time (RT): Release time is measured as the time between when the input loudness level drops below the compression threshold, to when the compression deactivates.