CHALMERS
UNIVERSITY OF TECHNOLOGY

Electronic Design of an Analog Equalizer for a Bone Conduction Stethoscope

Greeshma Ajayakumar
Raakesh Thiyagarajan

DEPARTMENT OF ELECTRICAL ENGINEERING
CHALMERS UNIVERSITY OF TECHNOLOGY
Gothenburg, Sweden 2024
www.chalmers.se

Master's thesis 2024

Electronic Design of an Analog Equalizer for a Bone Conduction Stethoscope

GREESHMA AJAYAKUMAR & RAAKESH THIYAGARAJAN

Department of Electrical Engineering
Division of Signal Processing and Biomedical Engineering
Unit of Biomedical Signals and Systems
Chalmers University of Technology
Gothenburg, Sweden 2024

Electronic Design of an Analog Equalizer for a Bone Conduction Stethoscope
GREESHMA AJAYAKUMAR & RAAKESH THIYAGARAJAN

Supervisor: Karl-Johan Fredén Jansson, Department of Electrical Engineering
Examiner: Sabine Reinfeldt, Department of Electrical Engineering

Master's Thesis 2024
Department of Electrical Engineering
Division of Signal Processing and Biomedical Engineering
Unit of Biomedical Signals and Systems
Chalmers University of Technology
SE-412 96 Gothenburg
Telephone +46 31 772 1000

Typeset in LaTeX
Printed by Chalmers Reproservice
Gothenburg, Sweden 2024

Electronic Design of an Analog Equalizer for a Bone Conduction Stethoscope
GREESHMA AJAYAKUMAR & RAAKESH THIYAGARAJAN
Department of Electrical Engineering
Chalmers University of Technology

Abstract

The necessity and interest of audiologists to finely tune and customize sound frequencies according to individual patient needs, has inspired us to develop and integrate an equalizer to a bone conduction stethoscope. The project designs an analog amplifier equalizer circuit that is used for volume control and frequency selection. It captures sounds from the skull bone with the help of a skin microphone hoping to provide audiologists a flexible tool that can isolate specific frequency bands for analysis, auditory experience diagnosis and troubleshooting of bone conduction hearing aids. The switching options designed on the equalizer has bandpass filters for the audiometric frequencies with the possibility to focus the listening range to both low and high frequencies and thereby encourage audiologists to select the desired ranges of inspection. The project plan was designed to use the electronic features to categorize audio spectrum into bass (125 Hz to 1 kHz), wide (125 Hz to 8 kHz) and treble (1-8 kHz) for testing. The test is done on two subjects with the use of a skin microphone having a strategy to check one's own voice to understand the perception of booming effect, crispness and clarity along with normal speech auditory range of bone conduction voice. The test subjects also experimented by hearing each other's voice through the equalizer to understand the audio spectrum as mimicked in normal hearing conditions as a reference. The results showed that the auditory experience was in sync with the theoretical expectations. To conclude, the design and testing of the prototype showed that it is a promising tool for audiologists in near future.

Keywords: Skin Microphone, Bone Conduction, Air Conduction, Hearing Aid, Equalizer, Volume Controller.

Acknowledgements

We would like to express our deepest and warmest gratitude to our supervisor, researcher Karl-Johan Fredén Jansson for his guidance and support throughout the exciting journey. We also appreciate for all the ideas, insights and answers regarding the bone conduction technology and electronics surrounding the topic. We also want to heart-fully thank our examiner, Assoc Prof Sabine Reinfeldt, for welcoming us into the topic and encouraging us all along. The research works and the vision of both our supervisor and examiner have sparked us in wanting to explore and dive deep to related topics. We would also like to thank the lab in charge people for supporting us during the design and testing phase and special mention of gratitude to all the researchers, professors, students who have been working all along in bringing advancements in the field of aiding hearing.

Greeshma Ajayakumar & Raakesh Thiyagarajan, Gothenburg, May 2024

List of Acronyms

Below is the list of acronyms that have been used throughout this thesis listed in alphabetical order:

AC	Air conduction
AGC	Automatic Gain Control
BAHA	Bone Anchored Hearing Aid
BAHS	Bone Anchored Hearing Systems
BC	Bone Conduction
BCD	Bone Conduction Device
BCI	Bone Conduction Implant
MPO	Bone-Anchored Hearing Aid

Contents

List of Figures

1

Introduction

Researchers in this field of audiology have long been interested by the complex mechanisms that enable auditory transmission. There are two main ways that sound is transmitted to our senses: through the air and through the bones. Perceiving sound is a complex process. The latter uses sound waves that pass through the ear canal, whilst the former uses direct vibration transmission to the skull bones, avoiding the outer and middle ear. The difficulties associated with bone conduction hearing aids have created a pressing need for creative solutions as technology develops. The goal of this research project is to understand the details of auditory perception through the creation and use of a particular device called bone conduction (BC) stethoscope.

Overcoming to the challenges caused by bone conduction technology is the innovative bone conduction stethoscope. This specialty equipment records change to sound that is emitted by the skull bone via the skin when the microphone is carefully placed on the forehead. With its revolutionary approach to evaluating the effectiveness of bone conduction hearing aids, its unique utility goes beyond traditional diagnostic tools. As the project aims to further the field of audiological research and therapeutic practices, this introduction lays the groundwork for a discussion of the goals, approaches, and anticipated results [1].

The precise tuning of sound frequencies to meet everyone's specific requirements is critical in the field of audiology. An equalization integrated into BC stethoscope is an inventive solution that was made possible by this need and the growing interest in customized auditory experiences. With the help of this work, audiologists will be able to better meet the needs of their patients by providing an adaptable tool that can be used to precisely adjust sound frequencies. This project attempts to transform the examination, diagnosis, and debugging of bone conduction hearing aids by integrating an analog amplifier equalizer circuit with volume control and frequency selection capabilities.

1.1 Purpose of the Project

The primary goal of this project is to design an analog amplifier filter circuit that satisfies the unique needs of the bone conduction stethoscope. Sophisticated tools are needed for volume control and frequency selection on this device, which is essential for capturing sounds from the skull bone through the skin. The initiative hopes to provide audiologists with a flexible tool that can isolate specific frequency bands and adjust volume by putting these characteristics into practice. It is intended that this improved capabilities will make a substantial contribution to the discipline of audiology, namely in the areas of auditory experience diagnosis and troubleshooting related to BC hearing aids.

1.2 Outline

The project proceeds systematically, beginning with an extensive review of the literature to understand the many spectrum ranges that affect speech transmission. This method divides contributing components into discrete frequency ranges and addresses things like voice harmonics, fundamental frequency, discrimination in speech, sharpness and clarity. Multisim software is used during the design to verify that every circuit component functions as intended and to validate the band pass gain amplifier and frequency selector parameters. The upcoming selections will provide a more thorough examination of this creative effort by looking deeper into the approach, design details, and simulation results.

2
Theory

2.1 Auditory Sound Transmission

Air Conduction (AC) and Bone Conduction (BC) are the two basic mechanisms of sound transmission that the auditory system uses to enable the remarkable ability of humans to sense sound. For the cochlea, which converts mechanical vibrations into neural impulses that the brain interprets as sound, these natural channels are essential for carrying sound [2]. As a method of transmitting sound, bone conduction has special qualities for auditory perception. It stands out in especially for its ability to transfer sound waves through the skull bone, offering a substitute for the more common air-to-eardrum approach. BC is known for its ability to improve sound quality in two-way communication systems, particularly in settings where typical AC transmission is difficult, likely very noisy areas that require the use of ear-canal hearing protection equipment [3].

2.1.1 Air and Bone Conduction Hearing

In the field of air conduction hearing, changes in air pressure are used to transfer sound waves through the atmosphere. The start of this process is when outside noises enter the ear canal and vibrate the eardrum. The fluid inside the cochlea moves as a result of these vibrations passing through the sensitive ossicles of the middle ear, the malleus, incus and stapes. The cochlea's hair cells are stimulated by this movement, resulting in electrical signals that are transmitted to the brain and processed to produce the sense of sound. For those whose auditory nerves and cochleae are unharmed, AC hearing is the primary mode [4][5]. Around half of it comes from the bone conduction, which varies somewhat with each unique sound and how it is produced [3].

BC hearing implies the direct passage of sound waves from the skull to the cochlea, which directly stimulates it. However, there are other pathways involved. For instance, sound can radiate into the ear canal, where it then follows the AC pathway to reach the cochlea. Even in cases where the conventional AC channel is blocked, sound can still be perceived through this approach. In BC hearing, vibrations are produced using a BC vibrator, which avoids the middle and outer ear and instead stimulates the cochlea directly. Direct transmission through bone is a helpful alternative for those with certain forms of hearing loss because it allows them to hear sounds even in situations where the typical air channel may be blocked [4][5]. See figure 2.1.

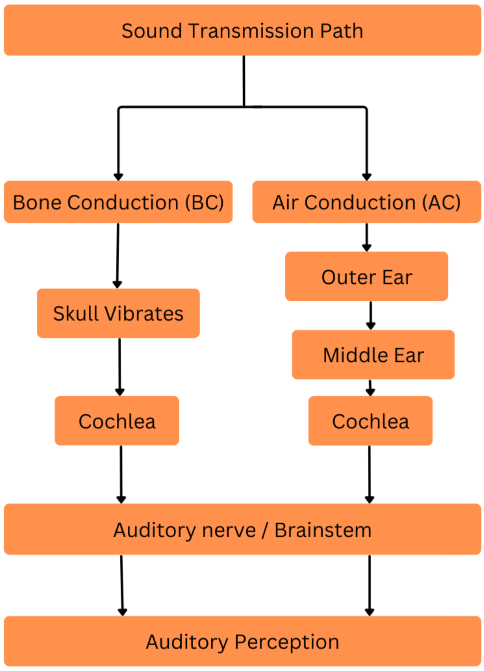

Figure 2.1: Pathway of auditory perception

In audiology, identifying the type and severity of a hearing loss is essential to providing suitable rehabilitation options. An important part of this diagnostic procedure is determining the BC thresholds, which are determined with a bone vibrator. Audiologists can determine the kind of hearing loss and suggest suitable solutions by comparing the hearing threshold in BC and AC. For patients suffering from single-sided deafness, mixed hearing loss, or conductive hearing loss, this diagnostic strategy is essential [4]. The sensitivity differences between AC and BC have been investigated in a number of studies that have focused on different sound sources. Based on the research, it was shown that for most sounds in the 1-2 kHz range, the BC component is predominant. The BC component of the sound was frequently 40 to 60 dB lower than the AC part, according to an analysis of the sensitivity differences between the two types of sound from the surroundings sound field. This knowledge is essential to comprehending the possible advantages of employing BC microphones to record voice in noisy environments, as BC can significantly enhance signal-to-noise ratios [3].

2.2 Hearing Rehabilitation

2.2.1 Hearing Loss

When a person's hearing thresholds in both ears are 20 dB or above, and they are unable to hear as well as someone with normal hearing, they are considered to have a hearing loss [6]. A person may experience one or both ears of hearing loss, which can range in severity from modest to profound [6]. Some of the main causes of hearing loss are age-related hearing loss, noise- induced hearing loss, persistent middle ear infections, ototoxic medicines that harm the inner ear, and congental or early onset children hearing loss [6]. Academic achievement and career prospects may suffer in places where there are insufficient adaptations for hearing impairments [6]. Education is sometimes inaccessible to children who are deaf or have hearing loss in impoverished nations. In terms of health sector expenses, educational support, lost productivity, and societal expenditures, the World Health Organization (WHO) estimates that untreated hearing loss costs the world economy about US $ 980 billion yearly [6].

2.3 Speech Intelligibility

Speech Intelligibility is a measure of how much of a speech signal is able to be perceived or rather comprehensible under conditions of limitations. It usually refers to measure of extend of an acoustic signal produced by a speaker, being correctly recovered by a listener [7]. Sound is created when air passes through the vocal cords. Efforts made in changing pressure levels on the vocal cords can influence the sound levels, pitch and frequency spectrum of the voice sound. The pharyngeal, oral and nasal cavities above the vocal cords are important contributing factors towards the filtering of the voice spectrum. Affecting these filtering cavities and the efforts made on and by the vocal cords are individualistic meaning that the perception differs from one person to another. It has also dependence upon the spoken language. Some languages are tonal while others are non-tonal. The spoken speech signal covers a wide range of the complete audible frequency spectrum. In tonal languages like Chinese or Thai, speakers convey message through the use of their fundamental frequency adapting their lexical tones. In non tonal languages, usually speech signals are made up of vowels and consonants [8].

The vowels are generated by the vocal cords and natural filtering methods employed through affecting the vocal cavities help in the formation of different vowel sounds. Thus the vowels are created by tuning of resonances of the vocal cavities [9]. Generally, at normal intensities, the energy of the vowels start diminishing after 1 kHz [10]. However, when the voice is raised, the speech spectrum shifts to one or two octaves towards higher frequencies and it has to be noted that the sound level of consonants do not increase as much as the increase observed in the level of vowels. The consonants are created due to the air blockages and the noise sounds produced by the passage of air through throat, mouth and tongue. They form one of the important parts of non tonal languages predominantly existing above 500 Hz. specially ranging between 2 to 4 kHz [9]. Studies suggest the importance of consonants in understanding intelligibility. They reveal that a high pass filter of cut off 500 Hz on speech spectrum decreases intelligibility by 5 percentage. However having higher cut off decreases speech intelligibility by higher rates. Also, having a low pass filter at 1 kHz decreases intelligibility by 40 percentage. The frequencies between 1 to 4 kHz are of higher importance to speech intelligibility [9].

Background noise, frequency response with respect to background noise, quality of the signal, reverberation and the acoustic effects of surrounding environment are some of the influencing factors affecting speech intelligibility [11]. An optimum speech level is achieved when the background noise is maintained below 40 dB and for sound signals having noise more than 40 dB. maintaining a SNR of around 12 dB, thus by having the speech signal four times louder than the background noise, a optimal speech signal can be perceived. Emphasis has to be made to maintain strict filtering conditions for frequencies in range of 1 to 4 kHz. Any prevailing music or background sound in this frequency spectrum has to be reduced by a factor of 5 to 10 dB to improve intelligibility [9]. Reverberation, though perceived to enhance steady state vowels smears out consonant words and masks gaps and transitions between sentences. Factors such as direction of speech, position of head and neck also add to be influencing factors predicting intelligibility. In general, the fundamental frequency or pitch for most of men lie around 100Hz and women exhibit one octave higher while children have pitch around 300 Hz [9]. Research suggest that standing at a distance of 1 metre from the speech source help to perceive signals naturally with highest intelligibility [12].

2.3.1 Hearing Aids and Fitting

On the fifth anniversary of the Hear the World initiative, a study was commissioned by the authors involving more than 4000 participants across five countries on topic related to effects of hearing on all areas of life [13]. This study which finally concluded emphasizing on the tagline "Hearing is for living" provided insights that revolved around three important findings.Improved hearing can benefit all areas of life leading to enhanced quality of life leading to better communication with family and friends. Improved hearing can lead to more active and healthy life with perks of having higher concentration,less stress and a relaxed mindset associated with general well being of an individual. Finally, the study suggested that the hearing aids that helps in recovery of hearing loss has open and positive approach among the groups of people surrounding the affected individual having welcoming open hands towards the tech.

There have been attempts to treat hearing loss for hundreds of years, according to historical records [14]. From simple instruments like ear trumpets to revolutionary digital technologies, hearing aid technology has advanced dramatically over time [14]. The BCI is an example of an active transcutaneous bone conduction device (BCD) that was created to stimulate the skull bone directly while maintaining skin integrity [15]. Research on BCI patients' long-term follow-up has shown how safe and successful it is at helping people with mild-to-moderate mixed hearing loss and conductive hearing loss receive hearing rehabilitation [15].

An audiologist is a healthcare professional evaluating and determining the type and degree of hearing loss to suggest and fix hearing aids to people. Results are compounded by certain audiometer tests where there is a repetition of sounds and words at different sound levels to possibly find the quietest level at which the individual can hear. Based upon the results, customised hearing aids with specific features are recommended by the audiologists. They also suggest techniques associated with hearing aids involving communication strategies and assistive technologies to aid during challenging situation. Some of the important features provided by the hearing aids are having directional microphones focusing on speech in noisy environments, Noise cancellation features assisting communication and understanding in noisy situations, Feedback cancellation alleviating the annoyance of buzz and whistle sounds and streaming programs to tap sounds from audio sources.

The design of an analog equalizer that can be embedded to the skin stethoscope promises features that help audiologists during the support and rehabilitation phase. The project design aims to provide options for the bone conduction stethoscope with volume control and frequency selector ranging over particular bandwidth. These bandwidths are categorised with respect to bass, mid and treble frequency ranges. The feature can help the audiologists to focus on particular bandwidth controlling the volume to detect interesting findings regarding individualistic hearing loss perception. The design also includes a tuner for tuning into resonant frequencies to understand the unique perception The inferences made can be useful in customising the hearing aids specifically. The component box providing these processing options,being an embedded part of the stethoscope makes it quite easy to be portable and reliable too.

2.3.2 Bone-Anchored Hearing Aid (BAHA) and Bone Conduction Device (BCD)

Sound waves are directly transmitted to the skull bone using the Bone Conduction Implant (BCI) device. By stimulating the ear canal, eardrum, and middle ear components, AC hearing aids, on the other hand, follow the conventional method of sound waves travelling through air. A minimally invasive yet extremely effective remedy for hearing loss, the BCI's revolutionary design avoids skin penetration, showcasing its benefit over percutaneous bone anchored hearing aid (BAHA) systems. Based on preclinical and clinical data, the BCI system has the potential to transform the field of hearing aids by eliminating the drawbacks of conventional bone-anchored devices [16].

In contrast to standard BAHAs, the transcutaneous BCI appears to provide comparable patient satisfication and audiological rehabilitation. Several audiometric measurements and patient related outcomes were assessed as part of an observational study involving matched pairs of patients in order to perform this comparison [17]. While patient related outcomes modestly favored the BCI, the data showed no statistically significant diffrence in audiological parameters between the BAHA and BCI groups [17]. In addition, the significance of maintaining appropriate audibility for patients is shown by the objective testing of audibility in BCDs. Regardless of the type of BCD (percutaneous, active transcutaneous, or passive transcutaneous), using a skin microphone to objectively tesy audibility in patients offers a promising clinical technique [18]. Better outcomes and increase audibility for patients using bone conduction technology are ultimately possible with this method's accurate assessment and gain adjustment of the device. Through the objective evaluation of audibility using a skin microphone, physician may precisely evaluate and adjust the device settings to assure enhanced speech intelligibility and overall hearing performance for patients with different kinds of BCDs [18].

2.4 Bone Conduction Stethoscope

2.4.1 Skin Microphone

The field of auditory research has made significant progress in recent years, and the introduction of new technologies, such the skin microphone, has been crucial in deepening our understanding of the mechanics underlying sound transmission. The purpose of the skin microphone is to directly record and collect sound waves from the skin's surface [1][19]. In general, a skin microphone consists of an electric condenser microphone applied to the skin's surface, especially the forehead [1]. The positioning of this device enables it to both record and transducer vibrations caused by BC sound, offering a unique point of view for the study of auditory phenomena. Using a skin microphone is essential, particularly when using bone conduction devices such as the BCI. Instead of conventional microphones positioned in the external auditory canal to measure AC, a skin microphone is positioned in a way that directly records vibrations through the skull bone [1]. The skin microphone is an essential tool for evaluating thresholds, maximum power output (MPO), and voice responses in studies pertaining to all BCDs.

2.4.2 Equalizers

Equalizers are instrumental tools that are capable of influencing relative amplitudes of various frequencies in an audio spectrum. They are creative in the way they alter the tonal quality of the audio spectrum. They are made up of multiple filter circuits that have isolated ranges on which certain algorithms can be implemented [20]. The equalizer is provided with the volume control feature in the project which is discussed in detail in the next section. There are filter types depending upon the mechanism and features of filtering such as shelving filters, high pass filters, low pass filters, band pass and band stop filters [20]. High pass filters, which reduce the level of frequencies below the cut off frequency and low pass filters, which reduce the level of frequencies above a certain cut off frequency are used in the equalizer design of the project. The basic characteristics of equalizers are dependent upon their slope, frequency, Q-factor and gain. The slope determines how aggressively the frequencies are attenuated or boosted once the cut off frequency is reached. The Q-factor which stands for quotient of change factor determines the bandwidth of the equalizer. When the Q factor is less than 1, the resulting bandwidth is wide and if it is greater than 1, the resulting bandwidth is more tighter and sharper around the cut off frequency. The gain which can be interpreted as both positive and negative gain corresponds to level boost and level cut respectively [20].

Every sound perceptible to human hearing is composed of a fundamental frequency and its harmonics. The different kinds of audio frequency spectrum ranges and its characteristics [21] are discussed as follows. Fundamental frequency is the lowest audible pitch of the spectrum. Harmonics are integer multiples of the fundamental frequencies adding timbre and brightness to the tonal quality. Deeper exploration into the frequency range groups help us to relate frequencies with the way of feeling connected to the total auditory experience. The sub bass spectrum ranging from 20Hz to 60 Hz covers the spectrum of deep foundations of sound. Sounds of this spectrum adds to the raw power giving edge over feeling than to hear. But generally electronic noises produced by components lies around 50 Hz and steps are taken to filter frequencies below 100Hz to remove these noise signals. Maybe in the near future, research to filter only sub bass sounds corresponding to original audio will be helpful to retrieve the emotional intensity. But for the scope of the project limits the range owing to the challenge created by noise. The bass spectrum covering 100 to 250 Hz forms the basic foundation of sound adding to the power. Human ears are not very sensitive to the range and enhancements made in this range alter the sonic property of the perceived audio. A moderate boost of frequencies around 250 Hz range can increase the warmth, whereas same effect on the low end of spectrum increases the thickness. A high boost on the spectrum leads to a booming effect. The lower mid range frequencies ranging from 250 Hz to 500 Hz covers the spectrum that shapes the character of a sound. The fundamental frequencies of vowels are in this range and low order harmonics add richness and complexity to speech. Boosting frequencies around 300 Hz enhances bass clarity and around 500 Hz adds depth to the voice [22].

The mid range and upper mid range covers frequencies around 500 Hz to 2 kHz and 2 kHz to 4 kHz respectively. In our project, we consider mid range starting from 1 kHz. The mid frequency consonants that convey sounds like m,n,ng and some kind of i and r sounds are in the range of 500 Hz to 1 kHz. This range also includes the harmonics and formant frequencies of vowel perception, thus including them into a total bass unit. The sounds particularly around 1 kHz frequency are more sensitive to vocals and care has to be taken while boosting around this range. The upper mid range frequencies covering 2 to 4 kHz defines the clarity and character of speech. The ear canal resonates at 3.5 kHz showing the natural emphasis of its higher sensitivity as to even a mild boost could easily alter the timbre of the sound produced [23]. The 4 kHz to 6 kHz frequency range defines and creates presence to the sound source. The range more than 6 kHz termed to be the brilliance audio spectrum adds to the sibilant sounds which are the usually the unwanted whistling sounds, for example when stress over s is emphasized. It helps in adding to the sparkle and sharpness of the sound.

The low frequencies travel slightly faster than their high frequency counterparts ,and so, if greater is the distance from the sound source, the greater will be the time lag between the fundamental frequency and its harmonics. The variation in the phase relationship between different frequencies creates a sense of distance reducing the need to give attention. Also when the sound travels through air passage, the higher the frequency the higher is the absorption of it due to frictional loses in air. This accounts to the spectral changes in the audio. Similar kind of effects are generally observed in people with hearing loss. They tend to miss out more on the sharpness and level of higher frequencies considering effects of distance. Sounds similar to one another in the mid range tend to overlap the spectral perception [24].

Audiologists would find themselves aiding patients to overcome these challenges with the help of equalizer. Considering the challenges mentioned above, altering high frequency components and increasing the volume help to compensate for losses due to air friction. The equalizer also compensates for the phase changes induced by reducing the delay of higher harmonics and increasing the illusion of closeness of sound [25]. As how to high frequency sounds can affect the attention to a sound, low frequency sounds affect the subliminal effects. These effects are usually felt strongly when subjected to rhythmic, repetitive sounds and equalization on this band aids in feeling the depth of sound [25].The mid range control can be played upon to restore the original subjective balance and reduce the spectral overlapping.

Thus audiologists can tune equalizers according to patient needs to match for the losses over specific frequency range. Gain or attenuation over their affected frequencies can help them hear normally. The BC stethoscope can give us the perception of every individual and the study of each audio spectrum can help audiologists to tune the directional microphones on the hearing aids to focus on the person based on their head movements [26], thus by increasing attention. Background noise filtering, wind interference cancellation and speech enhancements can isolate words and result in increased clarity [26]. The tool is quite useful capable of mitigating unwanted feedback such as whistling and squealing sounds that generally happen in hearing aids. Equalizers are helpful in identifying the problematic frequencies and customisation on its characteristics can make hearing more pleasant. Tinnitus, a perception of buzzing or ringing sounds often occurring in people with hearing loss, can be healed by using the tool as a part of sound therapy where customisation on various sound source and adjustments on certain frequencies can create sound stimuli masking the effects and perception of tinnitus [27].

2.4.3 Volume control

The use of variable gain amplifiers is a backbone of efficient signal control in the field of communication and imaging systems. Volume control is necessary because of the amplitude of the signals that are encountered throughout the transmission and receiving processes are dynamic. Efficiency and dependability of the system as a whole are improved by their smooth integration into current analog signal processing systems.

2.4.4 State of Art

The state-of-art in BC sound assessing microphone technology is a major step forward in tackling the global issue of hearing loss. Researchers have undertaken extensive investigations to find the ideal position for accurately collecting bone conducted sound created in the skull bone, using a skin microphone based on an electric condenser microphone from Sonion [5]. Modern microphone technology makes it possible to quantify sound quality precisely when wearing bone conduction hearing aids, which helps people with conductive or mixed hearing loss subjectively evaluate their auditory experiences. Researches have discovered important elements including hair interference and ideal placement positions through careful evaluation with a variety of transducers and testing procedures. These discoveries have led to insights that guide the development and application of bone conduction devices. Modern microphone technology not only improves measurement accuracy and reliability, but it also sets the stage for future studies that investigate other factors like sound leakage and pressure effects on measurement results, expanding the field of BC sound evaluating and hearing aid optimization [5].
Research groups focusing on developing methods to optimise strategies for BC hearing aids fitting and usage, proposed the usage of skin microphone to measure vibrations on the skull bone and to use the skin microphone as a stethoscope [28] to listen and reflect is the planning approach of the corresponding research team.

3
Method

3.1 Initial considerations and strategy

In order to build and construct a prototype component box that equalizes sound on a BC audio system, consideration is given to the input signal, the desired output, as well as the logic and distinctive features of the electrical components that are used throughout the process. A thorough review of the literature provides a chance to investigate related subjects and fosters a deeper awareness of the parameters and limitations of our activity. The acoustic impulses that are retrieved from the skull bone are sent by the skin microphone, which serves as the input component. Following the input microphone calibration, attention is directed toward the equalizer's design and development. Various options for electronic design are tried out until the most straightforward and efficient one is chosen in the end to become the prototype. The design section that follows covers different approaches. The operation of electronic counterparts that assist us in realizing the results of frequency range selection and loudness control must be emphasized.

The audiologists adjust the sound level through the BC stethoscope by using the volume control feature. An amplifier's gain can be electronically changed to change the output voltage in relation to the input voltage. The chosen control mechanism determines whether the gain factor is increased or decreased. Resistors and potentiometers are important components in determining the gain factor and gain control, respectively, and the voltages are directly proportional to the sound levels. The stability and likelihood of negative gain are taken into consideration when working on an inverting amplifier.The next stage is filtering. Since we are isolating specific frequency ranges, band pass filters are used. The high pass and the low pass filters are implemented at the desired cut off frequency ranges.

3.2 Functional Blocks

The volume control is a mix up of gain and attenuation from the input sound having control over both increasing volume to listen more intensely on specific frequency and decreasing volume to fade away some range of less importance or to understand whether the corresponding attenuation strategy would anyhow ease patient's comfort.
The figure 3.1 describes a simple inverting operational amplifier. It is a closed loop circuit, hence making the resistor R2, a feedback resistor and the resistor R1

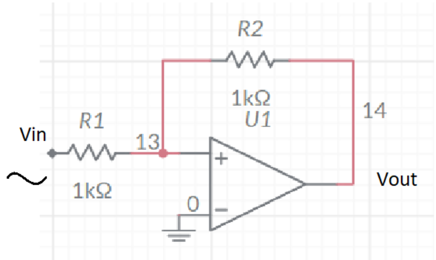

Figure 3.1: A simple inverting amplifier circuit

connected to the input terminal acting as the input impedance. With the help of Kirchhoff's current law, the gain of the inverting operational amplifier can be calculated as follows
Gain(Av) = -R2/R1 = -R(feedback)/R(input). The negative sign indicates the phase inversion.
The purpose of the high pass and low pass filters is to designate particular cut off frequency bands. The high pass and low pass filters are shown in figures 3.2 and 3.3, respectively.

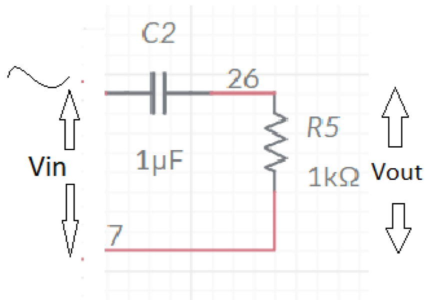

Figure 3.2: A simple high pass circuit

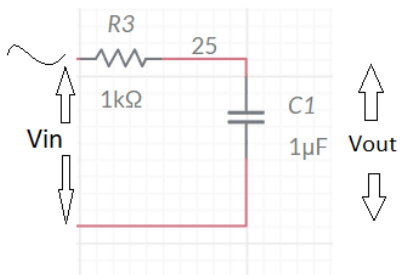

Figure 3.3: A simple low pass circuit

3.2.1 Design Requirement

The design of the volume controller requires the selection of a maximum for both gain and attenuation. After some testing, we were able to identify the boundaries, with an upper limit of +10dB and a lower limit of -40dB being ideal with regard to the output of the skin microphone. To electronically impose control over values between these bounds a potentiometer was utilized. The provided variable resistance modifies the voltage over the specified transduced range. Regarding the choice of frequency range, there are three frequency ranges: 125 Hz to 1 kHz for bass, 125 Hz to 8 kHz for wide, and 1 kHz to 8 kHz for treble.

3.3 Calibration of the device

We use a speaker that can maintain a consistent sound pressure level to calibrate the device. First, the microphone is calibrated to guarantee precise readings. Next, the skin microphone is positioned close by to measure its output. Any relative differences between the two devices are shown in this comparison. The calibration procedure is carried out at a frequency of 1 kHz and a sound pressure level of 114 dB. The microphone is then placed inside a test chamber and wired to an amplifier. It is also connected to an additional amplifier, which helps to filter out unwanted noise interference, especially at frequencies that are outside of the range that produces a flat response. The signal is amplified by 20 dB using this secondary amplifier.

Two 9 V batteries provide the amplifier's power. The source, which functions similarly to a function generator, connects to the signal analyzer's rear input, and the speaker is attached as the amplifier's input. As such, the speaker's input and the microphone's output comprise the signal path. In order to finish the calibration setup for accurate and dependable performance, the technique also includes measuring the input signal to the speaker.

3.4 Headphone compatibility

Headphone compatibility is a crucial factor for people with hearing loss since it affects how they interact with different audio devices, such as computers, tablets, cellphones, and entertainment systems. The development of technology in hearing aids has greatly enhanced their compatibility with headphones, providing users with more mobility and accessibility in their everyday routines [29][30]. Conventional hearing aids presented difficulties with headphones because of problems like feedback, discomfort, and compatibility restrictions. However, compatibility issues have mostly been resolved with the development of digital hearing aids (BAHAs). The seamless integration of modern hearing aids with headphones allows users to enjoy crystal-clear audio without any pain or disturbance [29][30]. Users can wear headphones more comfortably and experience increased hearing according to its percutaneous design, which reduces interference with headphone positioning. Users can completely employ headphones for diverse audio needs due to the system's audiological outcomes, which show major gains against unaided thresholds [31]. Furthermore, improvements in active transcutaneous bone conduction devices, such the BCI have increased the range of headphones that can be used by people with mixed and conductive hearing loss. The devices enable users to wear headphones painlessly and without trouble since they promote bone conduction without damaging the skin's integrity [15].

Modern hearing aid technology, such as active transcutaneous bone conduction devices and bone anchored hearing systems, making them more accessible and offering users better audio quality across a wider range of audio devices. These developments highlight the continued dedication to innovation in the field of hearing healthcare, guaranteeing that people with hearing loss can take full advantage of the modern, interconnected world. In the project the equalizer is made compatible with Sennheiser HDA 300 (Sennheiser, Hannover, DE).

3.5 Simulation and Testing

The intended circuit is simulated using the NI Multisim software and the ELVIS platform prototyping board (National Instrument, US) is used to examine the built circuit, and the results need to be verified for the operation of the functional blocks before being examined for compatibility with headphones.

3.6 Prototype building

On the prototype board, volume control and frequency range selecting functions are combined. In order to keep the prototype safe and straightforward, switching strategies were chosen. To make the tool easy to use, the electronics are soldered, placed inside a component box, and buttons and connections are made on the box.

3.7 Subjective testing

Two test volunteers took part in the prototype's testing. In order to determine the efficiency of the hearing aid and to listen to various perceptions utilizing the tool's options, the equalizer is tested by inserting a skin microphone on the forehead.

4

Design

The analog equalizer combines the functional units of volume control and frequency range selection. There can be different possible approaches for the design plan implementation. The options of using variable resistors to tune between different cut off frequencies, band pass filters to set well defined frequency ranges and frequency resonant circuit to focus on particular frequency along with variable impedances to tune the window are are simulated tested and conclusive decision is made on the final design. Here are the several equalizer design approaches that are shown using the block diagram. It is discussed how to use the functional units and what their limitations are.

4.1 Design1 - Tunable frequency selectors

4.1.1 Stethoscope

The main device used to record sounds conveyed through BC is the stethoscope. A skin microphone that is positioned to detect vibrations coming from the skull bone is integrated into the design. This microphone accurately transmits auditory signals to the circuit's subsequent components because it is sensitive to even the smallest variations in sound waves.

4.1.2 Component Box

The component box contains all of the necessary components for controlling and processing signals. Different blocks inside this enclosure carry out certain tasks to improve the caliber and fidelity of the audio signals that are delivered. The following parts make up the component box:

Source

Incoming audio signals from the stethoscope are first received by the source module. It serves as a channel for carrying the recorded sounds to further phases of the circuit's processing. In addition to guaranteeing effective signal transport, this component lays the framework for additional amplification and filtering.

High Pass Filter with Gain Control

High frequencies are allowed to go through while low frequency components of the audio stream are selectively attenuated or eliminated. The high-pass filter in the circuit attenuates frequencies below a predetermined cutoff point, so removing undesired bass frequencies, noise, and rumble. Furthermore, following high-pass filtering, the gain control features allows signal level adjustment, which assures the best signal-to-noise ratio and compliance with subsequent processing steps.

Low Pass Filter

The "Low Pass Filter"attenuates higher frequencies in the audio stream while enabling only low frequency components to get through. It works in the opposite manner as the "High Pass Filter". It is comprised of a circuit for a low-pass filter that has a cutoff frequency that suppresses higher frequencies earlier. The audio stream can be made cleaner and smoother by using the low pass filter to eliminate high frequency noise or distortion.

Low Frequency Selection

Changing frequencies in the lower range of the audio range is the main goal of low frequency selection. Users can increase or decrease the apparent bass response of the audio signal by adjusting the loudness of the bass frequencies with this component. Low frequency selection is frequently used to adjust the bass output to particular listener's choice or to balance out the playback environment's auditory qualities.

Mid Frequency Selection

An audio spectrum's mid range contains frequencies that can be targeted and adjusted by users through a signal processing technique termed "mid frequency selection". The audio signal's tonal balance and clarity can be adjusted with this capability, which allows for the selective boosting or attenuation of frequencies that are commonly connected to voice, instruments, and other mid range content.

High Frequency Selection

The ability to target and modify frequencies in the upper range of the audio spectrum is provided by high frequency selection. With the exact control this component offers over the audio signal's treble response, users can adjust high frequency material to suit their tastes and make it louder or softer.In audio mixing and mastering applications in particular, high frequency selection is crucial to obtaining a detailed and well balanced sound reproduction.

4.1.3 Headphones

The headphones functions as the last output step, converting processed audio signals into waves of sound that the user can perceive. Comfort and fidelity are given first priority in the design, assuring precise reproduction of filtered and boosted signals.

The headphones' crisp, clear sound output helps audiologists evaluate and interpret auditory data confidently and precisely, which improves patient care and diagnostic accuracy in the end.

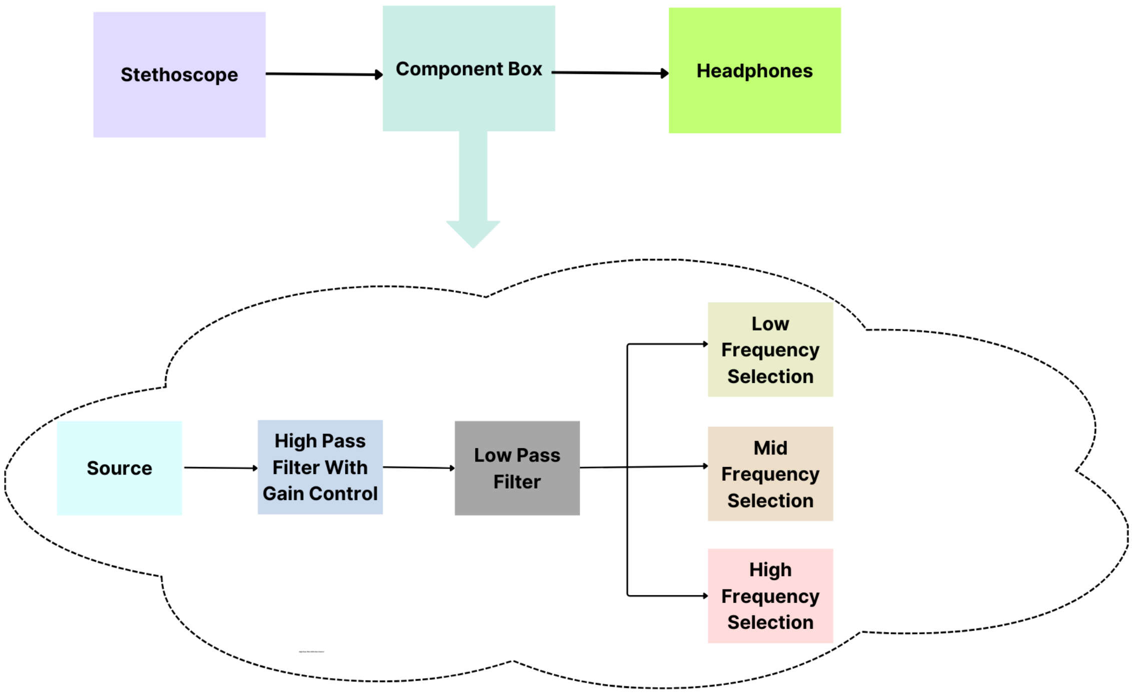

Figure 4.1: Block Diagram of the Design 1

The limitation of the design is the fact that the tunable ranges, though have impact in listening to different window lengths, it was not accurate enough to select particular value range of choice. The feature of adjusting window lengths, though can be more effective and aid audiologists with complex inferences, seems to be lacking accuracy comparing to its usefulness. Complex calibration and adjustments are needed to define switching ranges which goes out of the project scope to keep it simple.
The inside of the component box is the only part displayed in the upcoming block diagrams, as the other sections remain the same:

4.2 Design2 - Resonant circuit to capture frequency of choice

In this design, source, high pass filter with gain control, and low pass filter are the same as design 1.

A circuit consisting of an electronic inductor, a capacitor, and electrically actuated switches is known as resonant circuit with switches. By selectively tuning to a particular resonance frequency, the resonant circuit can be set up to enhance desirable signals while filtering out unwanted ones. Increasing the circuit's adaptability and functionality in signal processing applications are the switches that allow it to switch between several resonant modes or adjust the resonance frequency.

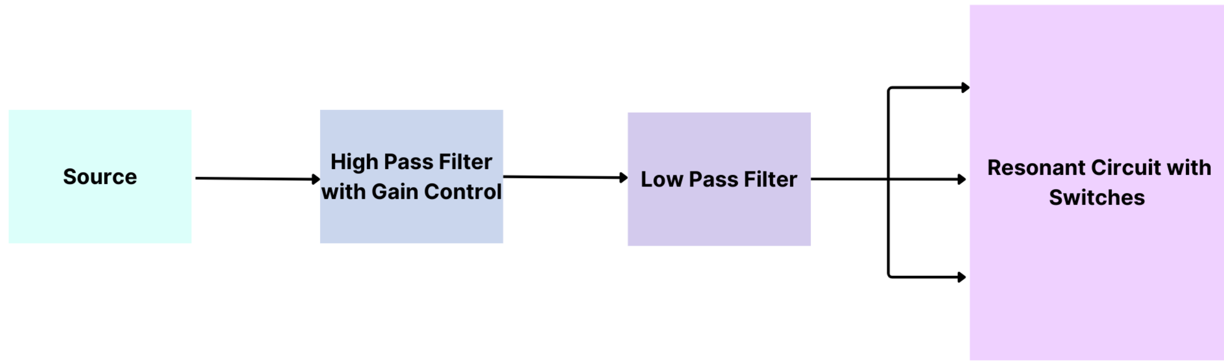

Figure 4.2: Block Diagram of the Design 2

The resonant circuits were able to filter out particular frequency value and the the window width depended upon the resonant components. During the testing phase, the inability to have control over the covered window range was observed. The capacitance and the inductance values are very high to maintain optimal Q-factor, which gives the narrowness of the window width. The switching mechanism to toggle between different selection also is complex and out of scope of he project.

4.3 Design3 - Band pass filters to set defined ranges

In design3 source and low pass filter are same as design1 and design2. It consist of the following parts:

Band Pass for Bass

While attenuating frequencies outside of its designated range, a band pass filter permits a specific range, or "band", of frequencies to flow through. This specific case is designed to pass only the bass frequencies; it usually cuts off both the high and low frequencies that are outside of the intended bass range.

Band Pass for Wide

This band pass filter is set up to pass a specified range of frequencies, similar to the last one, but it covers a wider range of frequencies. Depending on the desired outcome, it is made to pass through frequencies ranging from low to high range.

Band Pass for Treble

This band pass filter is different from the others in that it is designed to pass only treble frequencies through while blocking both nid range and bass frequencies.

High Pass Filter

With this filter, frequencies below the cutoff are attenuated while frequencies above the cutoff are allowed to pass through. It is frequently used to cut out desired bass or low frequency noise from a signal so that higher frequencies can still flow through.

Gain Controller

By precisely adjusting the signal amplification levels, the gain control amplifier module makes it possible to fine-tune the audio output to suit specific needs and preferences. This part offers adaptability and control over signal intensity, guaranteeing the ideal ratio of clarity to volume. Gain control features allows the circuit to adapt to different listening situations and user demands, which improves the user experience in general.

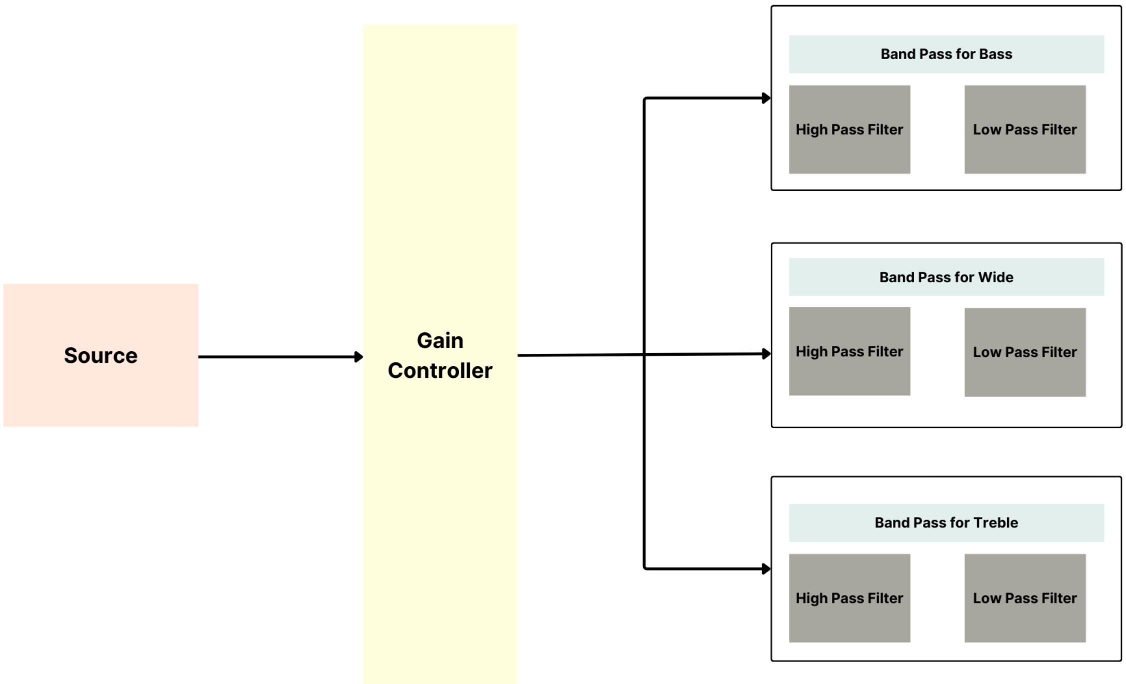

Figure 4.3: Block Diagram of the Design 3

The design made up of band pass filters is the most optimal and suiting design for the project plan. Well defined frequency ranges can be isolated and volume control function can be exercised on it with good accuracy, equalizing sound for further inferences. The design is simple with easy switching options and even though not applicable for detailed distribution and understanding of effects of various audio spectrum, the predefined ranges are good enough to infer a lot of basic information related to hearing loss and hearing aids. Thus this simple design is implemented to create the prototype.

A switching strategy is implemented having separate buttons to activate the band equalization and the block diagram of the switching strategy is given in Figure 4.4.

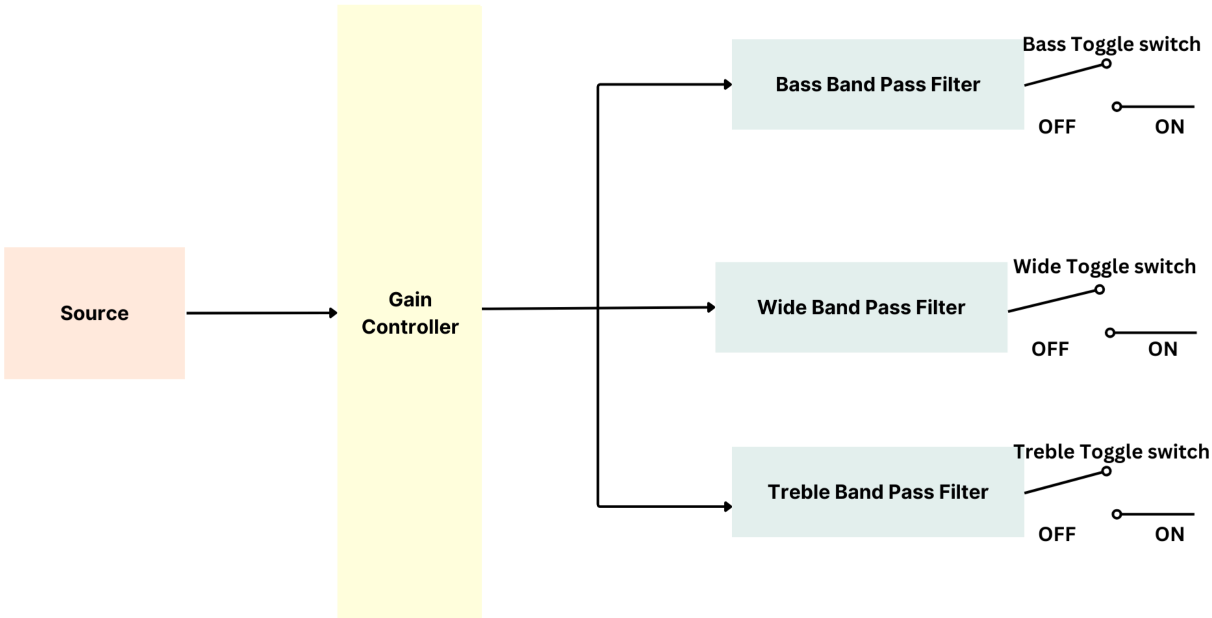

Figure 4.4: Block Diagram of the Switches

The gain controller also allows users to change the amounts of signal amplification in the specified frequency ranges. Users can precisely tune and customize the audio output by toggling these switches between their ON and OFF positions, giving them effective control over the frequency ranges and signal amplification to suit their preferences.

4.4 Circuit Diagram of the Equalizer

The values of the electronic components are calculated and are implemented as in Figure 4.5.

4.5 The Equalizer Prototype

Figure 4.5 showing the circuit diagram of the equalizer, featuring wide, bass and treble bands.

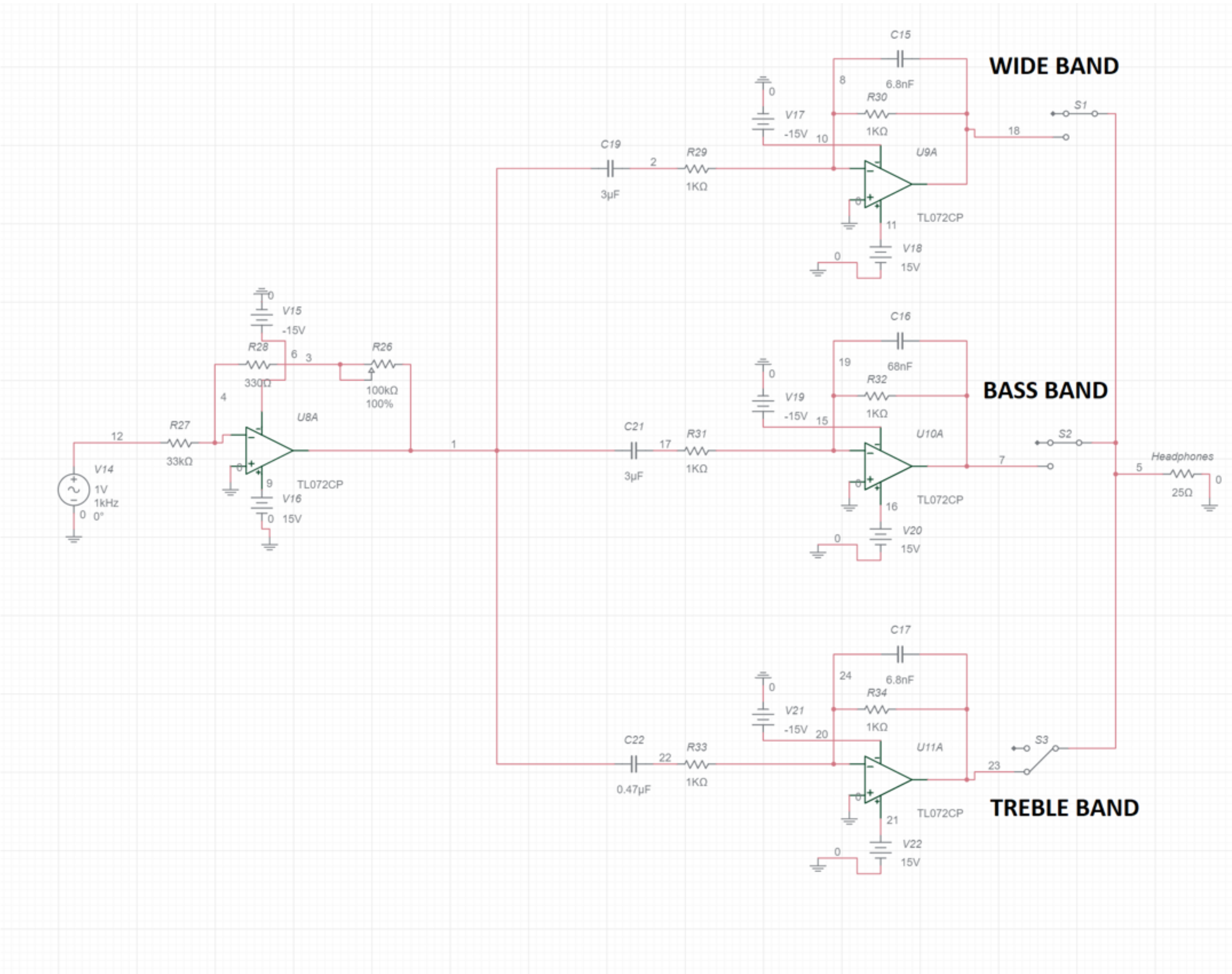

Figure 4.5: Circuit diagram of the Equalizer

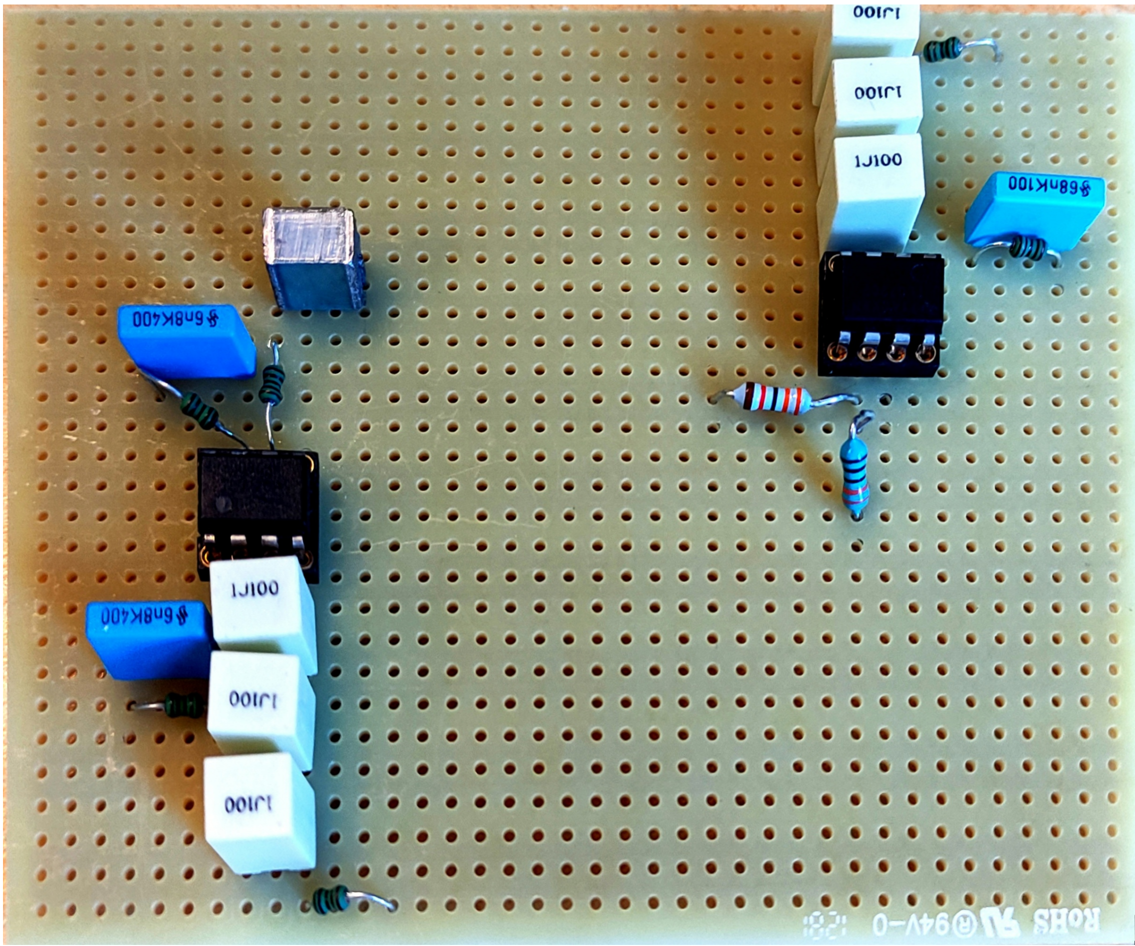

Figure 4.6: PCB design of the equalizer

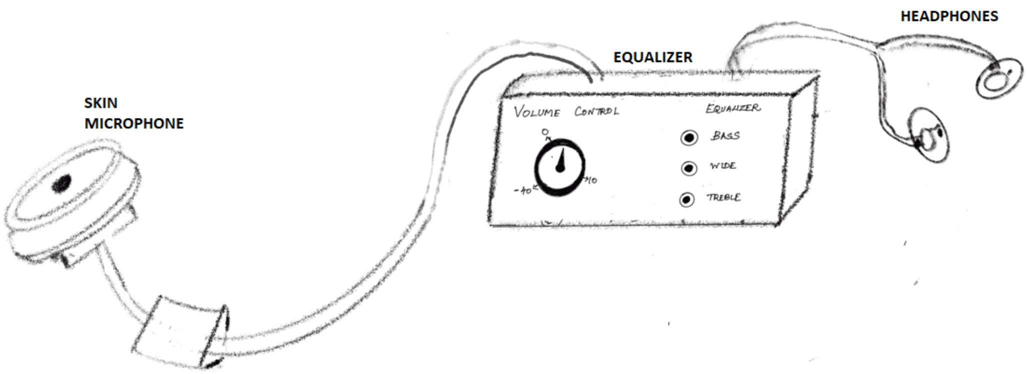

Figure 4.7: A pictorial realization of the equalizer prototype

5

Results

5.1 Simulation results

The simulation findings were based on a 1V input voltage, with the equalization volume gain set to +10dB. The input signal was an AC Sweep signal to determine the system's response to all frequency ranges (1 Hz to 100 kHz).

5.1.1 Band pass filters to set defined ranges

The simulations were run by pushing three switches to pick the desired frequency band. Figure 5.1 , figure 5.2 and figure 5.3 represent the wide, bass and treble band respectively. The gain (dB) is plotted against frequencies (Hz).

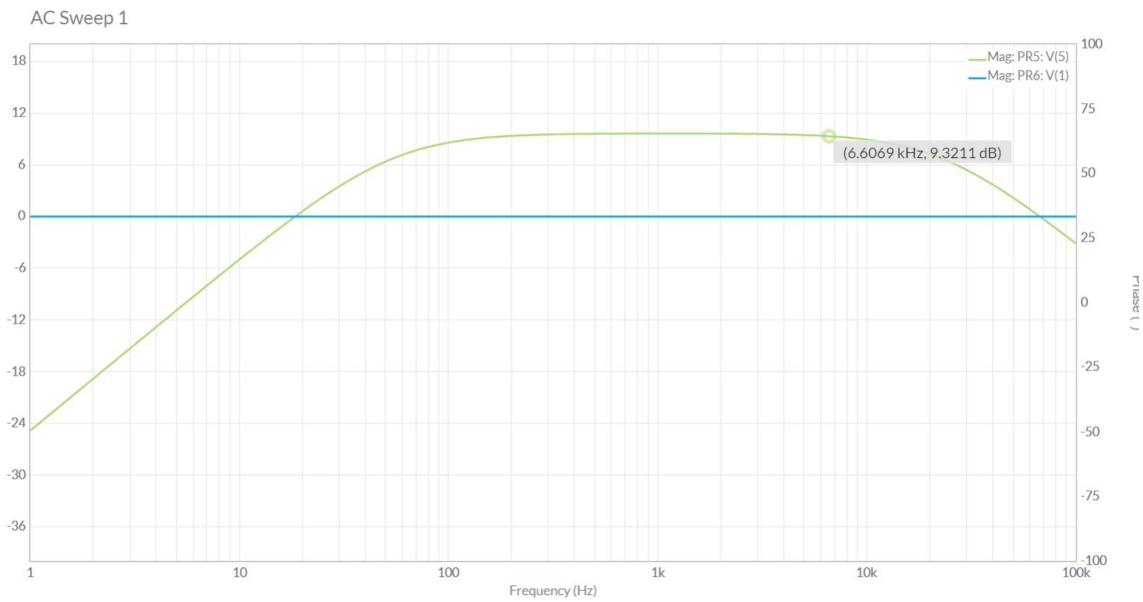

Figure 5.1: AC Sweep simulation of Wide Band(125-8000)Hz

Figures 5.4, 5.5, and 5.6 depict simulation results at various volume tuning levels. There are three cases, one examined for gain attenuation (i.e., less than 0 dB), another for gain amplification (i.e., more than 0 dB), and the last example checking for the most minimal volume. The gain (dB) is plotted against frequencies (Hz) in all the figures.

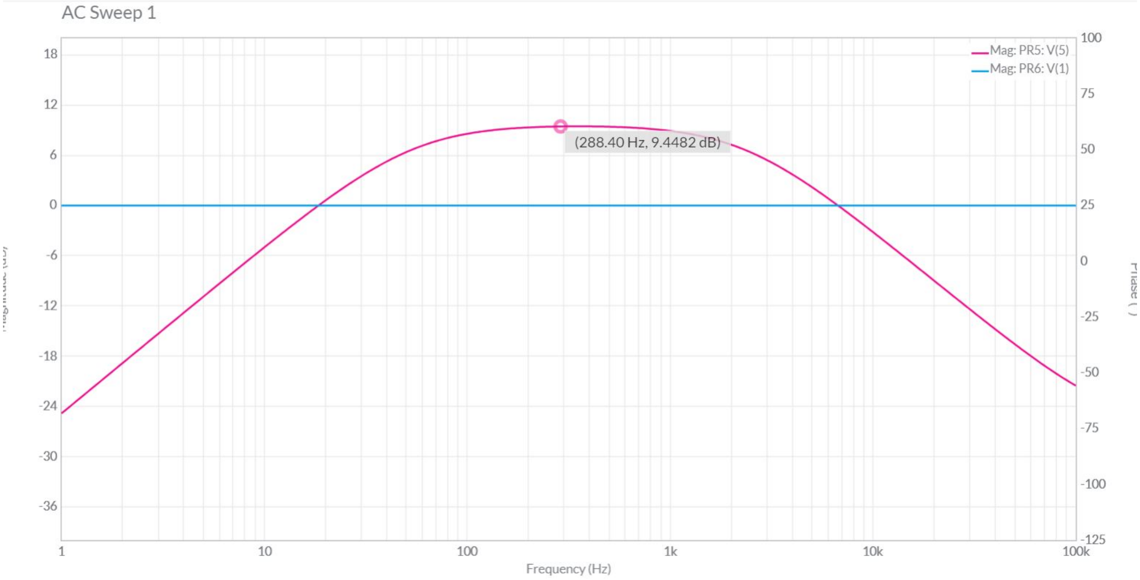

Figure 5.2: AC Sweep simulation of Bass Band(125-1000)Hz

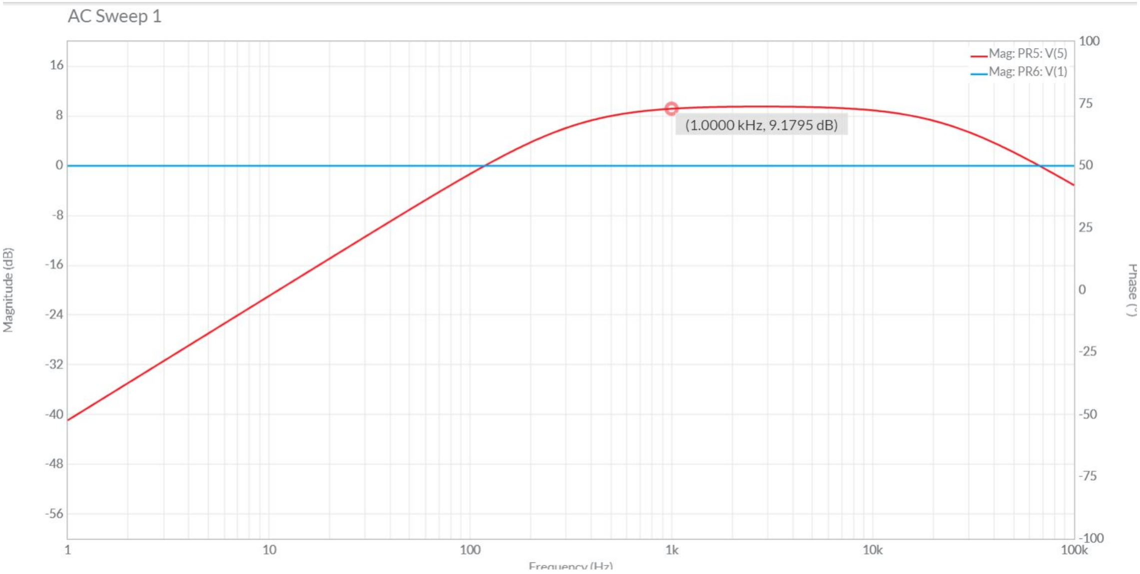

Figure 5.3: AC Sweep simulation of Treble Band(1000-8000)Hz

5.1.2 Resonant Tuner circuit

Resonant frequency selector and window length tuner was designed and implemented through the circuit diagram as follows. The changes in the capacitance values with having a constant inductance value of 100 mH helps in realising tuning of resonant frequencies. The capacitance values range from 16uF to 0.0025 uF for covering resonant frequencies of range 125 Hz to 10 kHz. With having a 10 k potentiometer to tune the window length, we observed that window bandwidth tuning is ranging from 50 Hz to 550 Hz approximately. The results of the multisim simulation are also plotted.

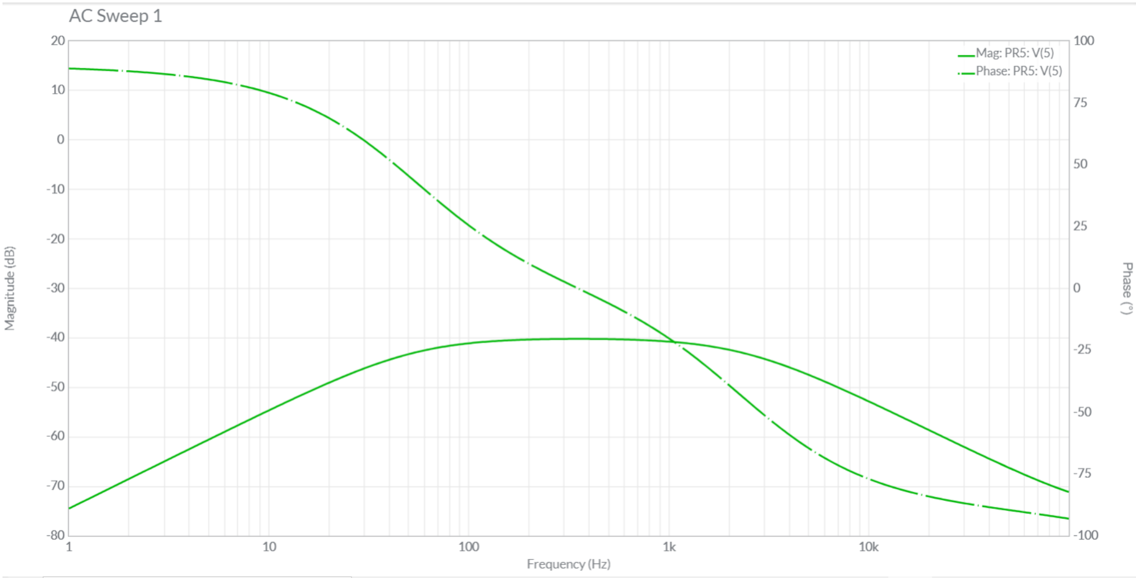

Figure 5.4: AC Sweep simulation of full volume down

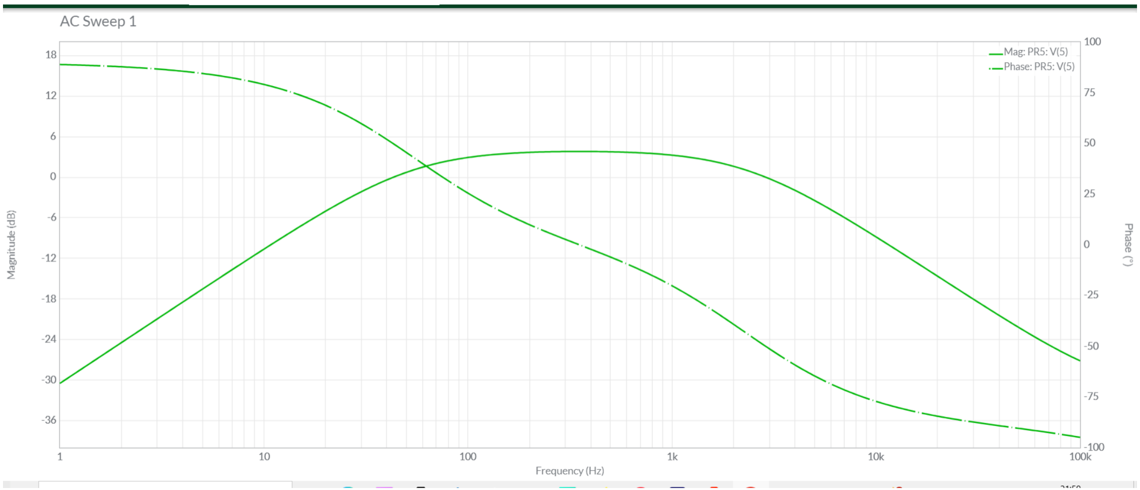

Figure 5.5: AC Sweep simulation of gain amplification

5.2 Tunable frequency selectors

Bass equalizer

The circuit design of bass equalizer is as follows.The bass frequency spectrum covers a frequency range of 100 Hz to 1000 Hz.The equaliser is designed to tune to adjust the frequency from a range of (100 Hz-200 Hz) bandwidth to (100 Hz-1000 Hz) bandwidth . The results of the boundary values using a bode analyser is also plotted having a maximum volume gain of 10 dB.

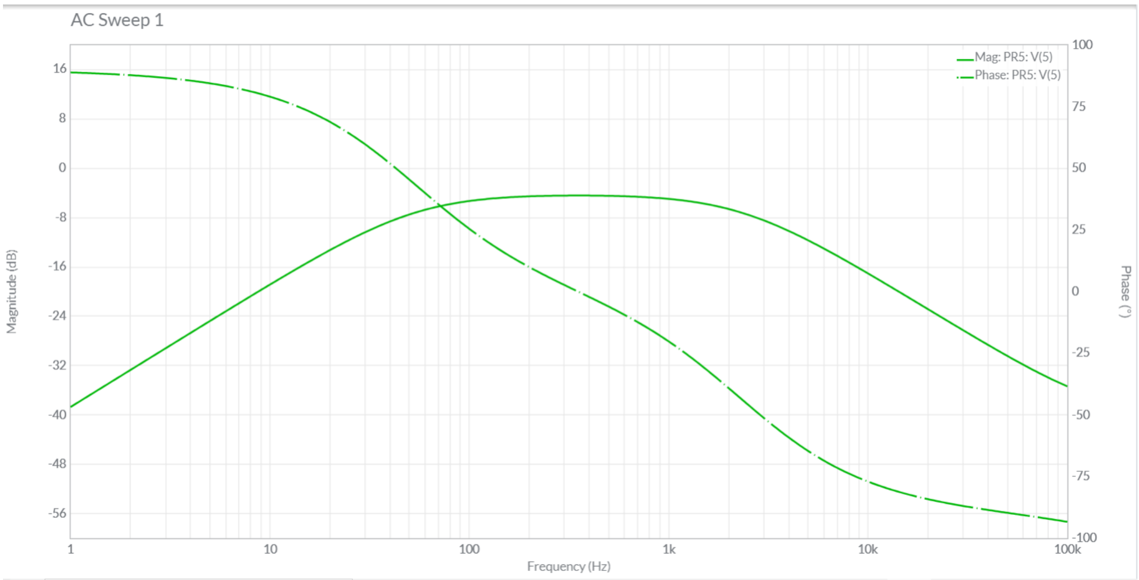

Figure 5.6: AC Sweep simulation of gain attenuation

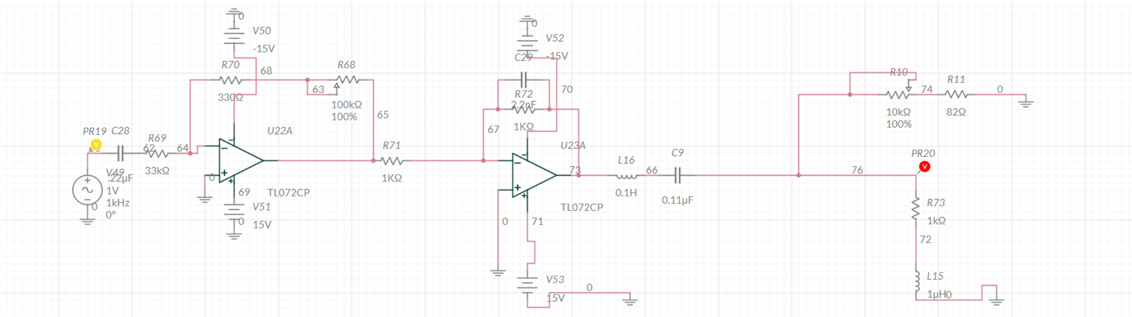

Figure 5.7: Resonant Tuner Circuit

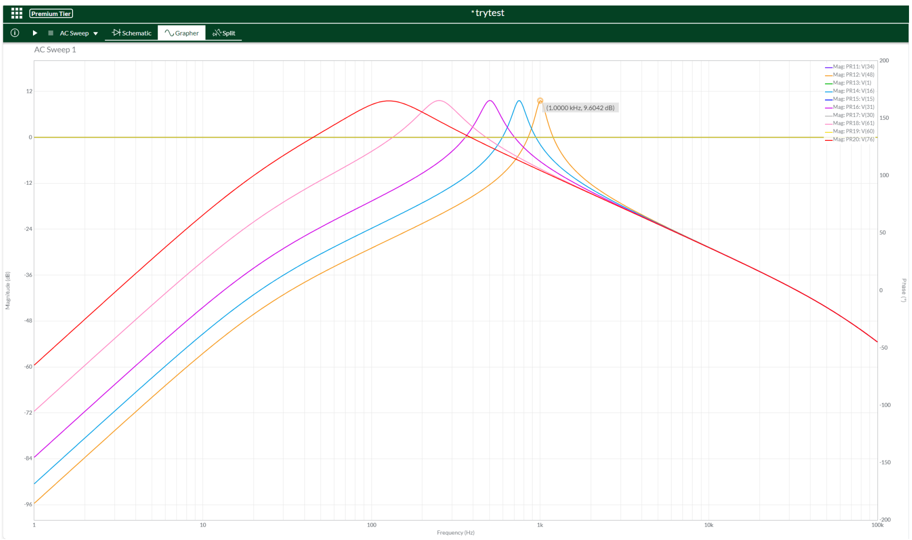

Figure 5.8: 125 Hz to 1 kHz - 50 Hz window length

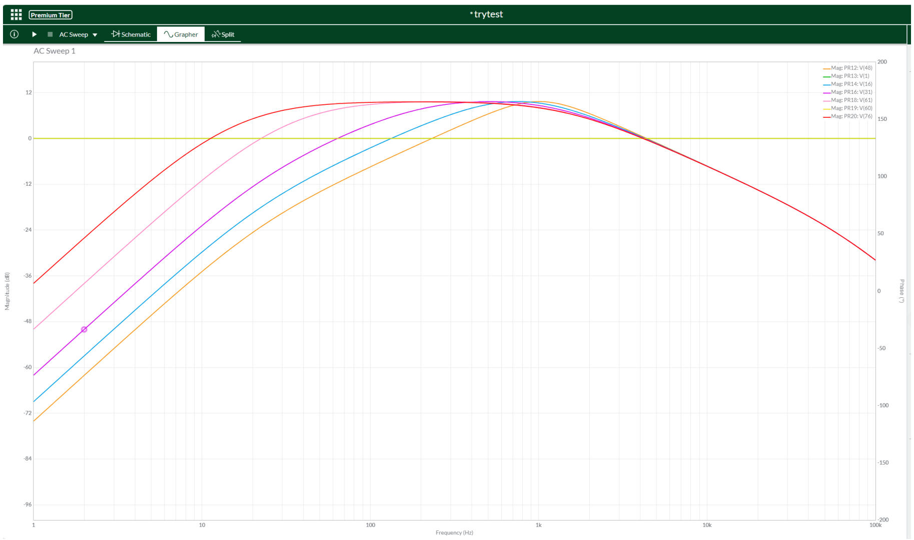

Figure 5.9: 125 Hz to 1 kHz - 550 Hz window length

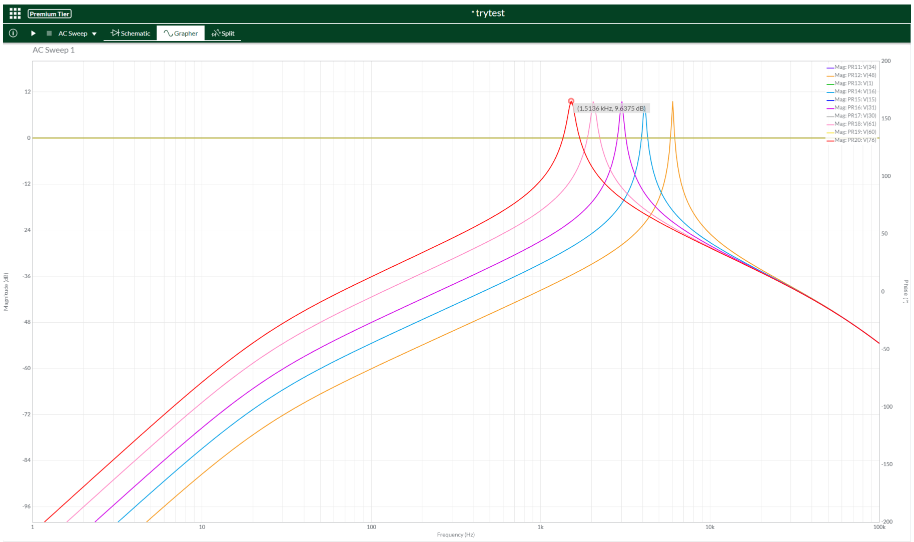

Figure 5.10: 1 kHz to 6 kHz - 50 Hz window length

Mid equalizer

The circuit design of mid equalizer is as follows.The mid frequency spectrum covers a frequency range of 1000 Hz to 4000 Hz.The equaliser is designed to tune to adjust the frequency from a range of (1000 Hz-2000 Hz) bandwidth to (1000 Hz-4000 Hz) bandwidth . The results of the boundary values using a bode analyser is also plotted having a maximum volume gain of 10 dB.

Treble equalizer

The circuit design of treble equalizer is as follows.The treble frequency spectrum covers a frequency range of 4000 Hz to 10000 Hz.The equaliser is designed to tune to adjust the frequency from a range of (4000 Hz-5000 Hz) bandwidth to (4000 Hz-

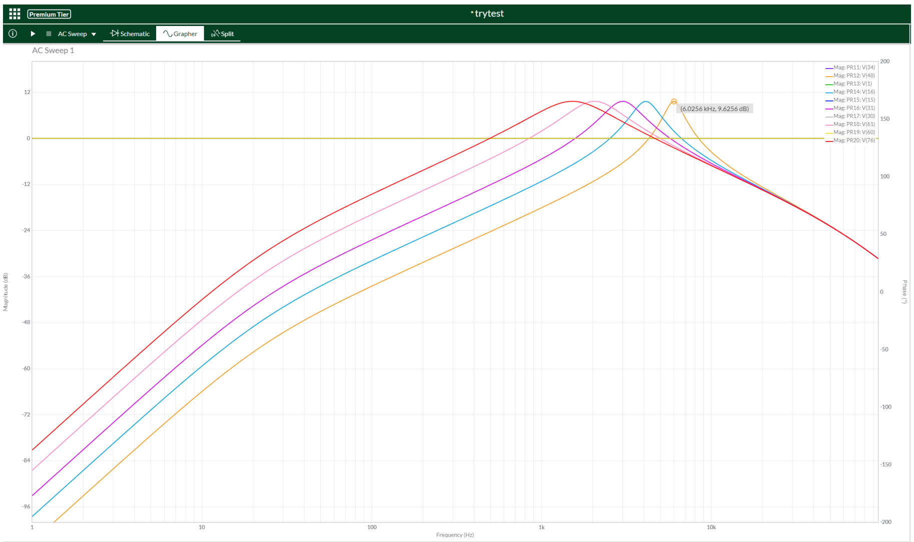

Figure 5.11: 1 kHz to 6 kHz - 550 Hz window length

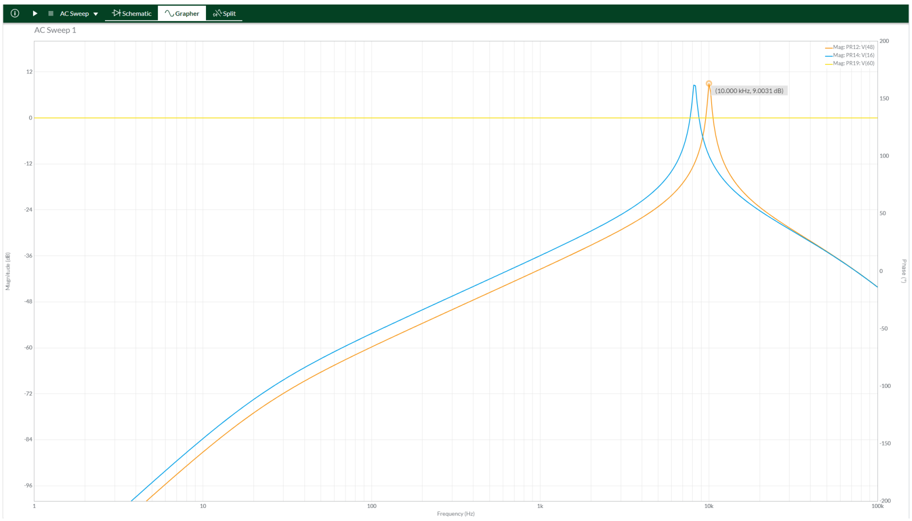

Figure 5.12: 8 kHz to 10 kHz - 50 Hz window length

10000 Hz) bandwidth . The results of the boundary values using a bode analyser is also plotted having a maximum volume gain of 10 dB.

We were able to observe that the difference between perception of bass, mid and treble equalizers showed similar traits that there was only difference in the sound level perception between different bandwidth ranges. For example in bass equalizer, the bandwidth of (100 Hz-200 Hz) had less sound levels comparing (100 Hz-1000 Hz). This may be due to A-weighted filtering that occurs in our hearing perception.

5.3 Measurement results

The prototype was developed and tested to ensure its functionality. The frequency bands were switched between bass, broad, and treble modes, with the results shown

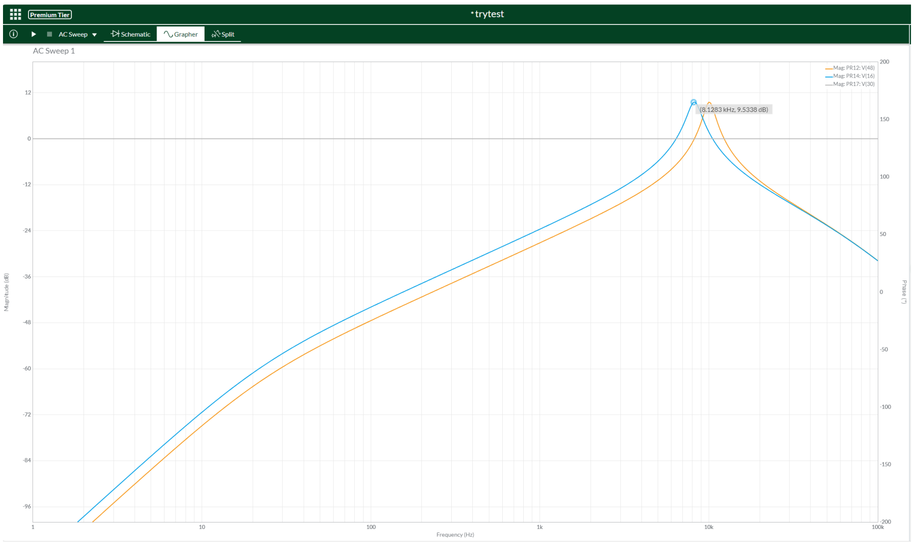

Figure 5.13: 8 kHz to 10 kHz - 550 Hz window length

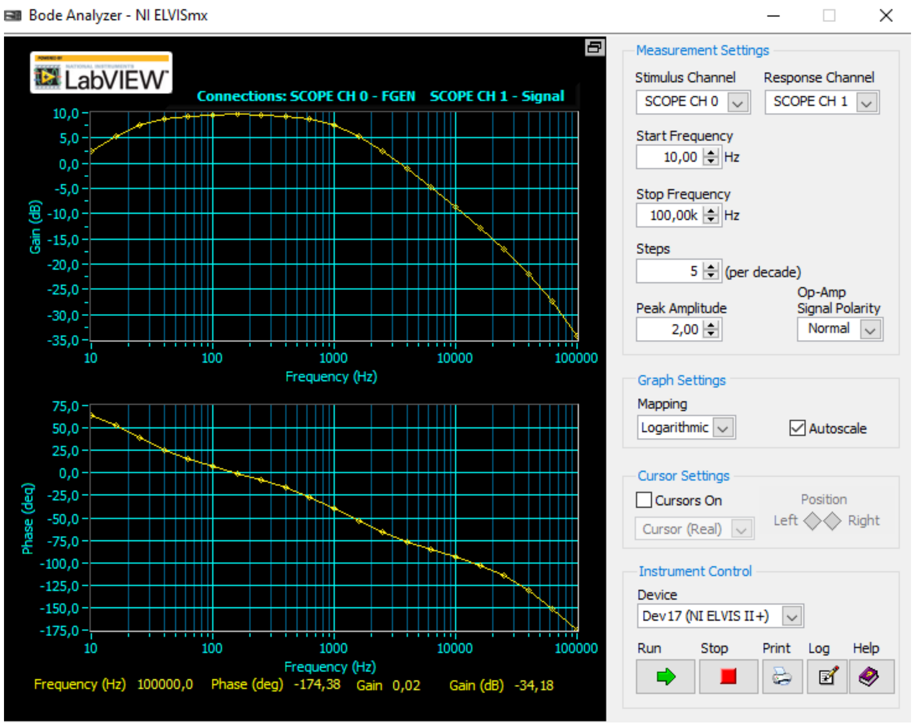

Figure 5.14: Bass equalizer min range

in figures 5.7, 5.8, and 5.9, respectively. The input signal was an AC sweep signal created by the NI Elvis board with frequencies ranging from 10 Hz to 100 kHz having peak voltage of 0.02 V. The circuit's response is demonstrated by plotting the gain (dB) versus the frequencies (Hz). The volume tuner is kept at its maximum setting in all three circumstances when the system's reaction must be checked at the greatest feasible level.

The volume controller is tested for the same simulation testing plan.Figure 5.11 represents the full volume down test case and a clear -40 dB is visible as the attenuated

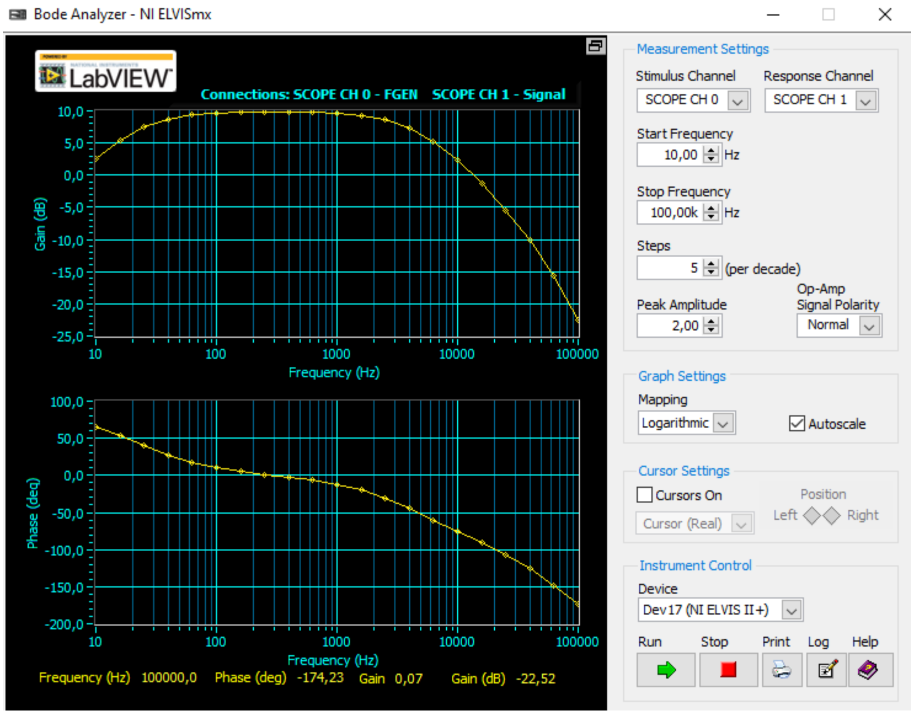

Figure 5.15: Bass equalizer max range

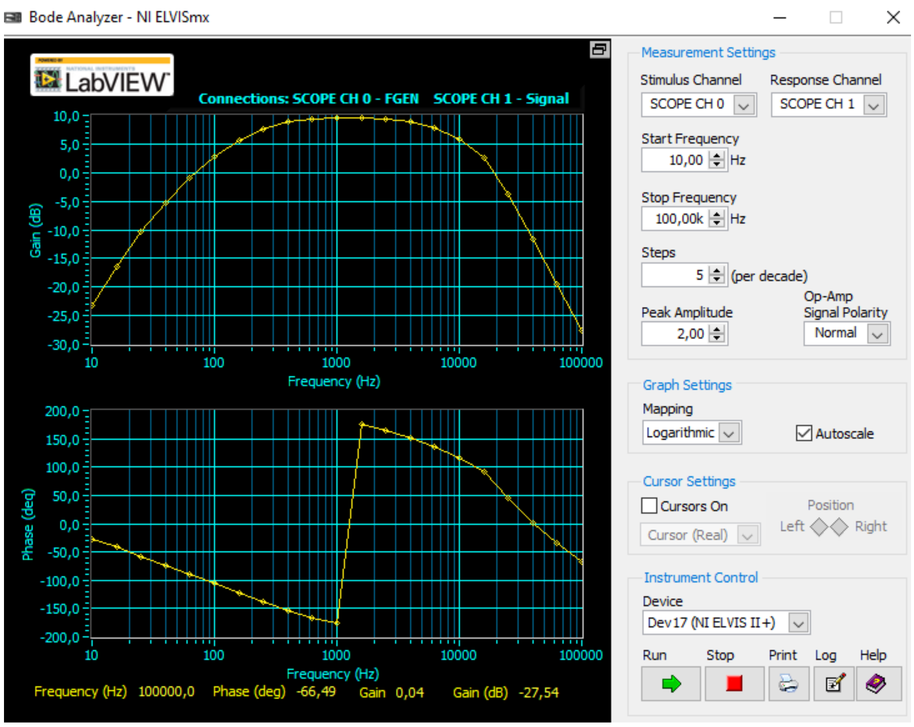

Figure 5.16: Mid equalizer min range

gain level. Similarly, figures 5.10 and 5.12 represent the gain attenuation and gain amplification test cases.

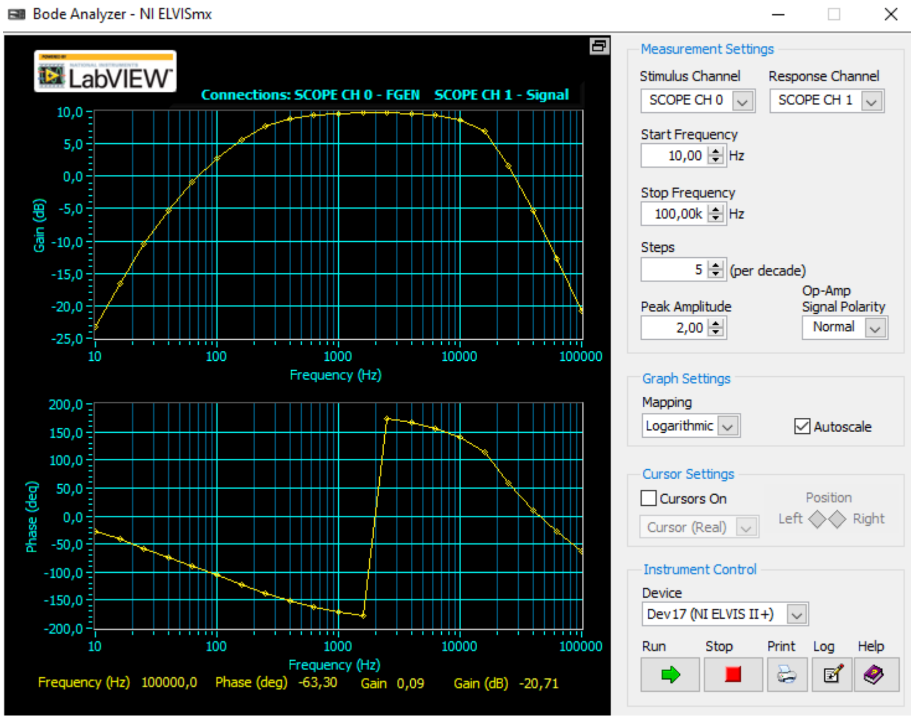

Figure 5.17: Mid equalizer max range

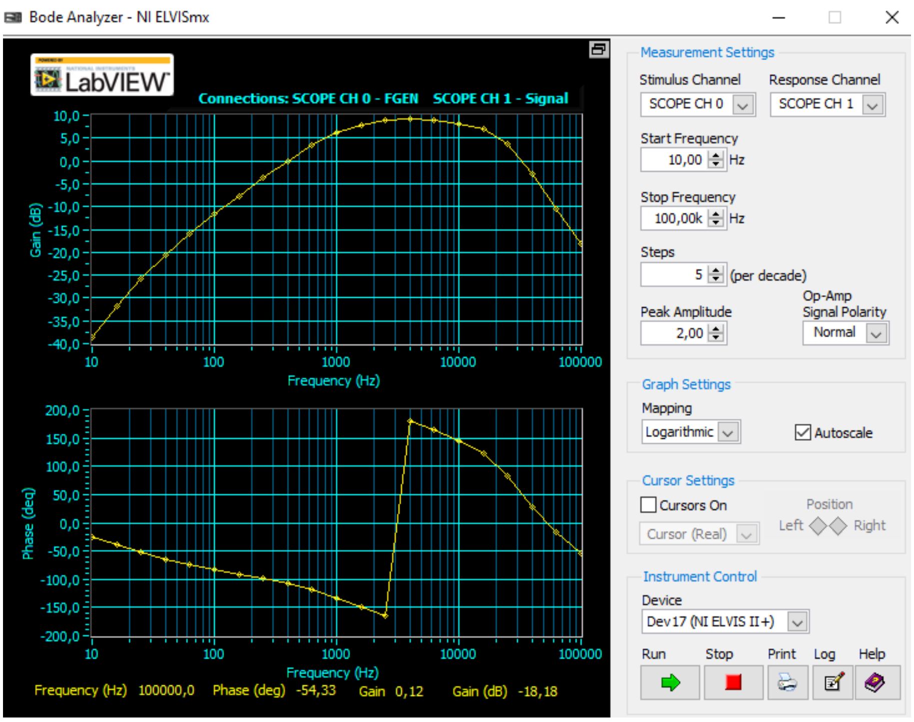

Figure 5.18: Treble equalizer min range

5.4 The built prototype

The prototype is designed with a switching strategy that includes three push buttons for selecting the frequency range.It also features a volume controller with input and

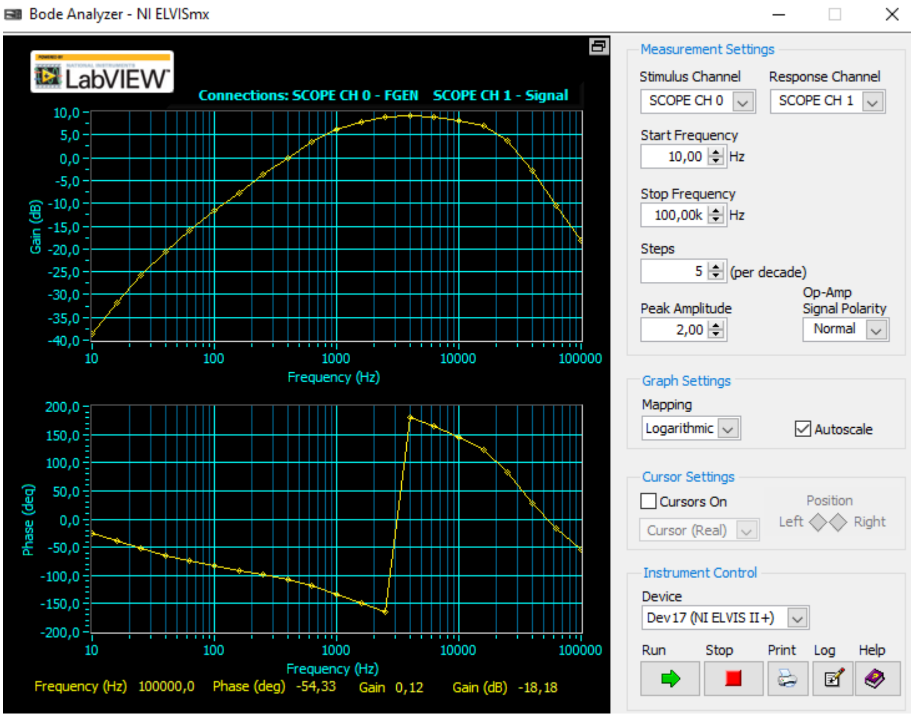

Figure 5.19: Treble equalizer max range

output ports located on top. Figure 5.13 shows a picture of the built prototype

5.5 Subjective Testing

Two test case scenarios were used for the subjective testing.The prototype was used to listen to teammate's bone conduction voice in the first instance.To reduce the effects of air transmitted sound, a skin microphone was appropriately insulated and placed on the head of a colleague.In the second test instance, listening to one's own voice was the task.

Insights from Bone Conduction Sound Perception

In the first case, using the wide push button option produced a sound that was quite similar to the natural sound of the original voice as heard in a regular listening environment. Gain amplification and volume control modifications made in response to another person's voice, relayed through bone conduction, allowed for the fine tuning of sound levels and the appearance of close proximity to the speaker. Notably, the choice of bass amplification gave the reverb effect more depth, however it was said to be less noticeable than one's own voice, which can be related to natural gender variances in voice characteristics. In addition, the second person's vocal tone balanced the high frequencies quite well, producing a distinct audio impression.

In the second case, hearing one's own voice via bone conduction led to an unique experience. When closed the eyes, the bass frequencies created a really booming

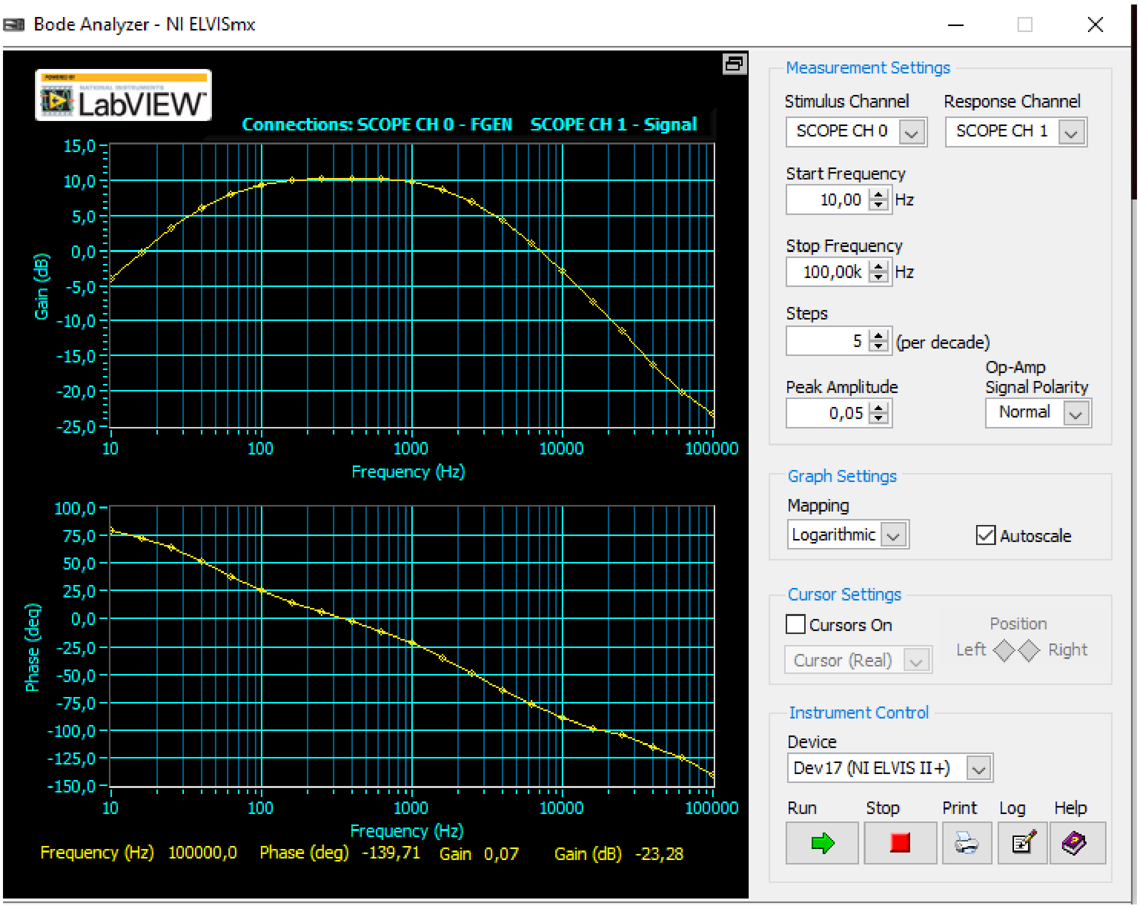

Figure 5.20: Bode analyser plot of bass band(125 Hz -1000 Hz)

impact that changed the way that perceived the acoustics of the surrounding area. It was like the echo chamber or cave reverberations. A unique character was added to the audio by the direct conduction of sound through one's bones, which covered a wide range of speech frequencies with remarkable clarity. Furthermore,it was believed that the high frequencies added to the feedback loop, improving the speaker's own articulation. The ability to control word stress levels was proved by purposeful changes in voice clarity, which also highlighted the complex interaction between bone conducted sound and individual vocalization approaches.

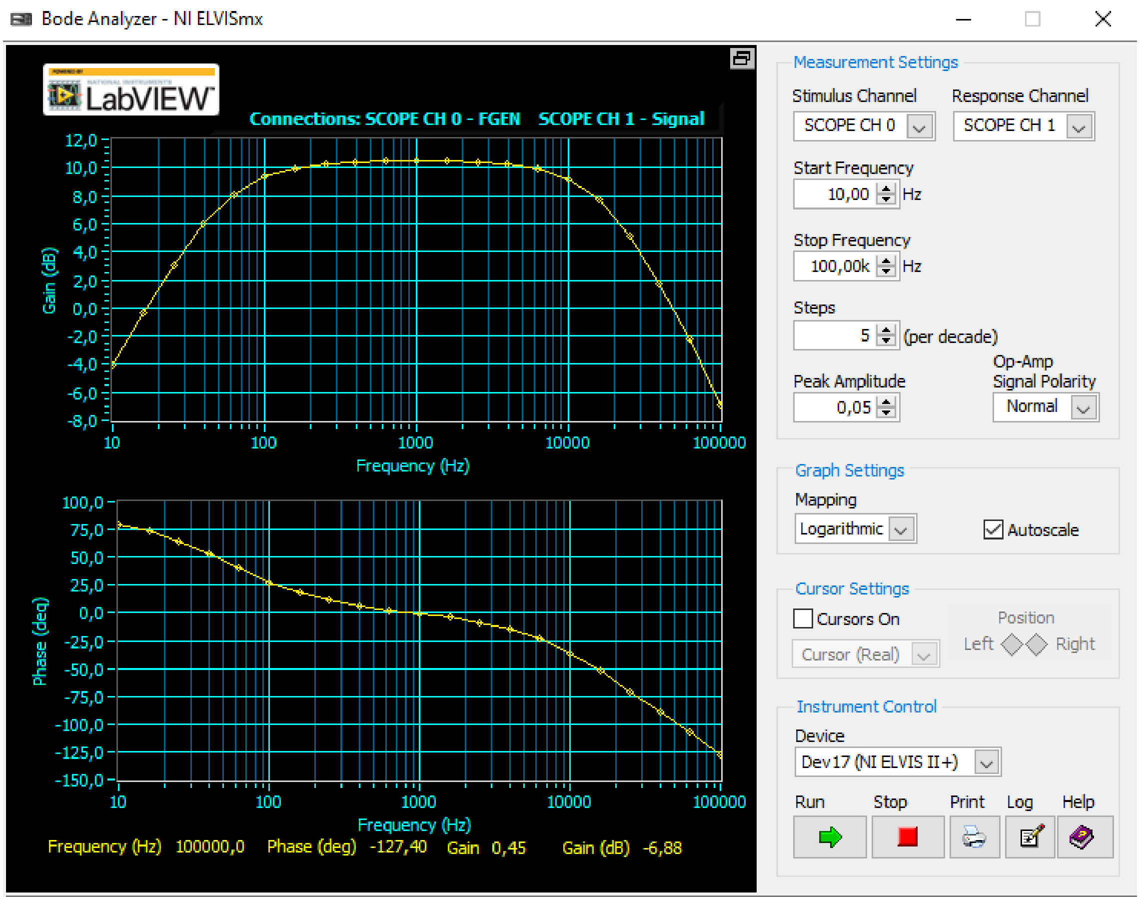

Figure 5.21: Bode analyser plot of wide band(125 Hz -8000 Hz)

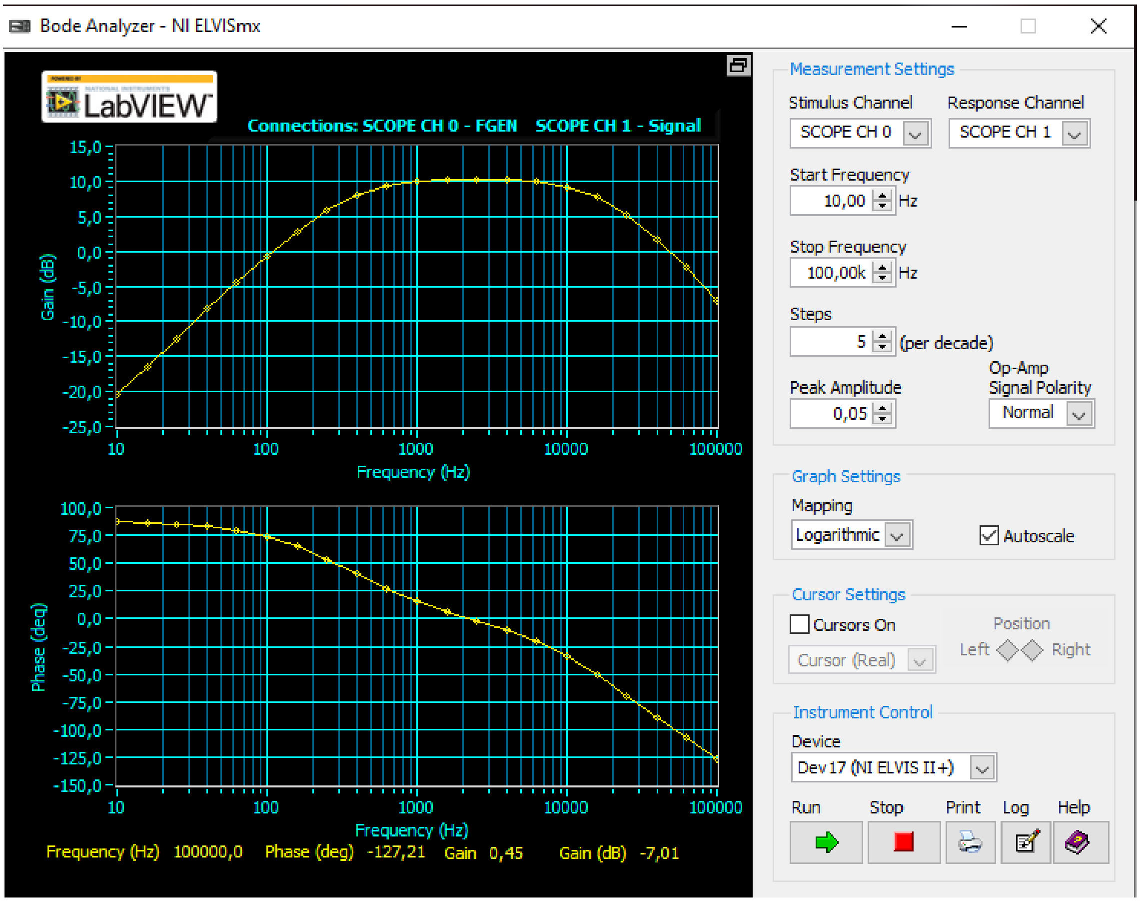

Figure 5.22: Bode analyser plot of treble band(1 kHz -8 kHz)

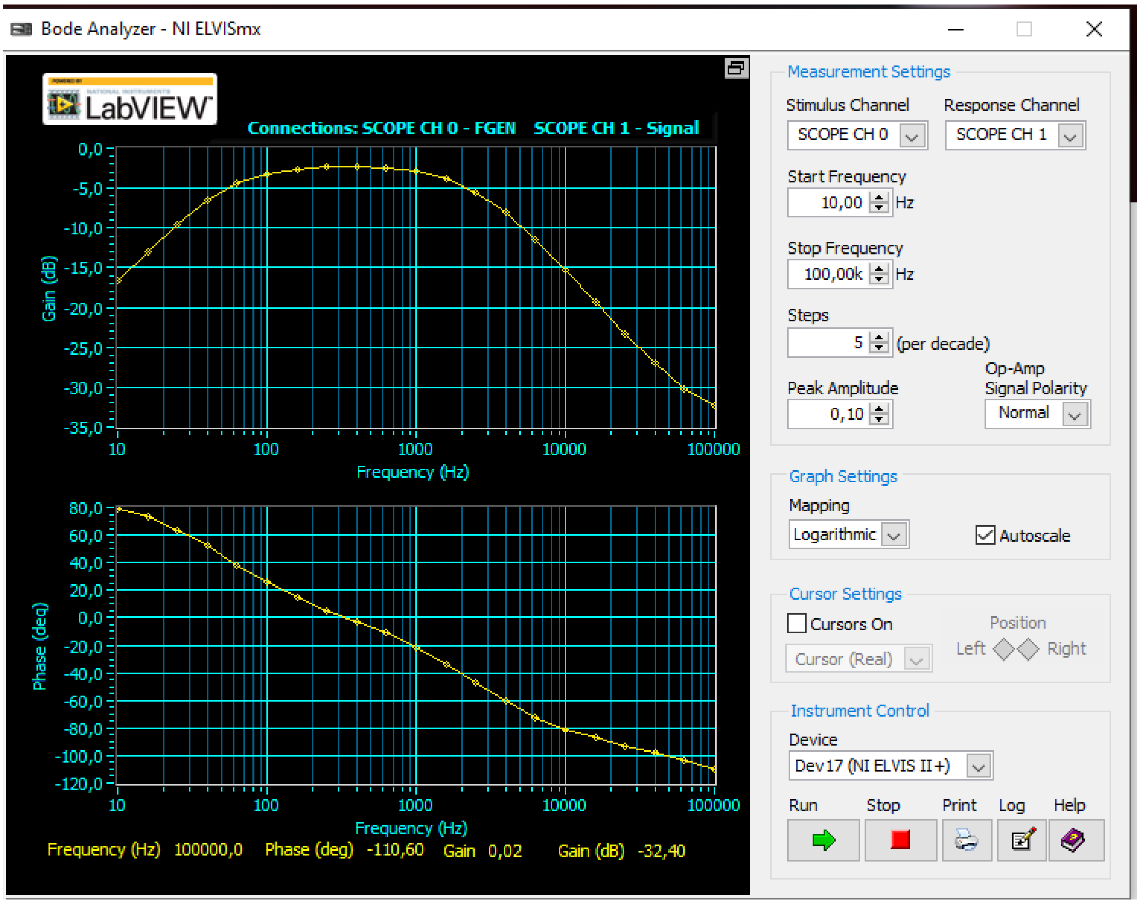

Figure 5.23: Bode analyser plot of volume tuning - gain attenuation

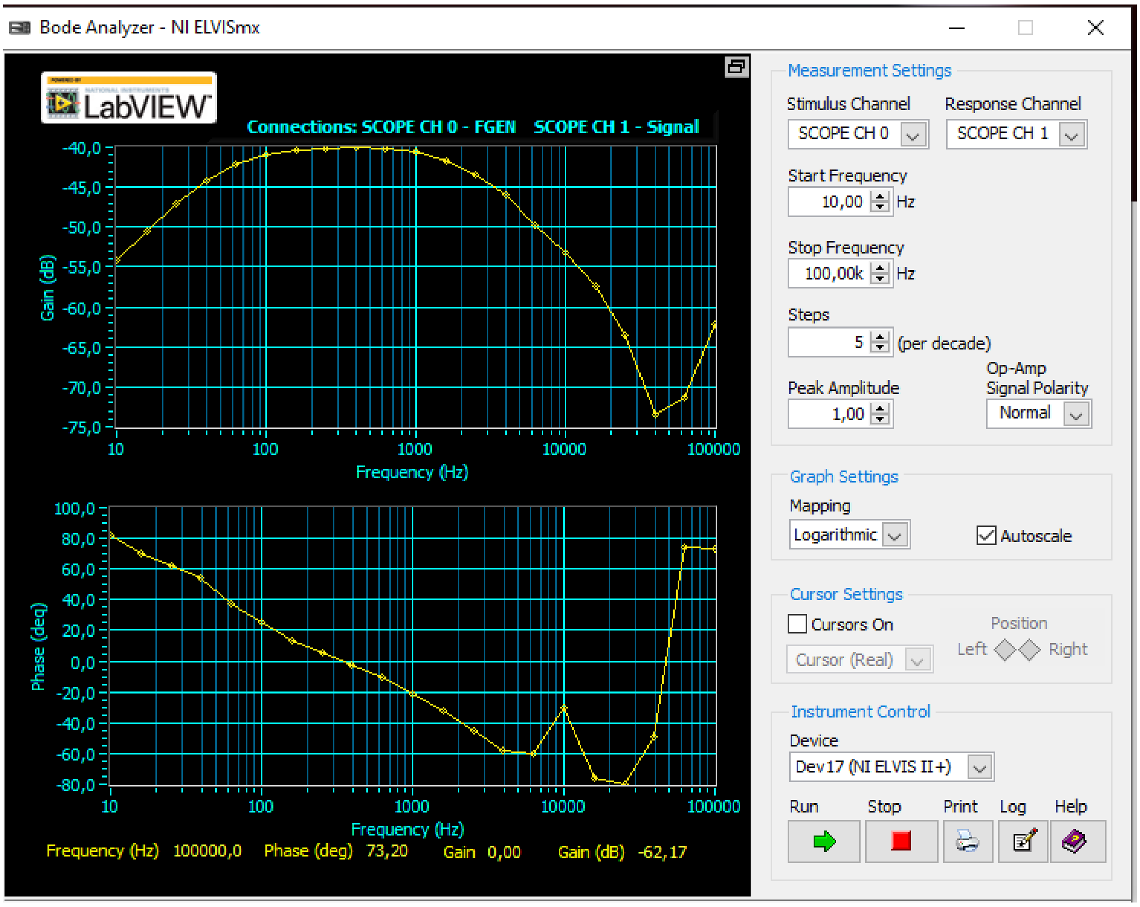

Figure 5.24: Bode analyser plot of volume tuning - full volume down

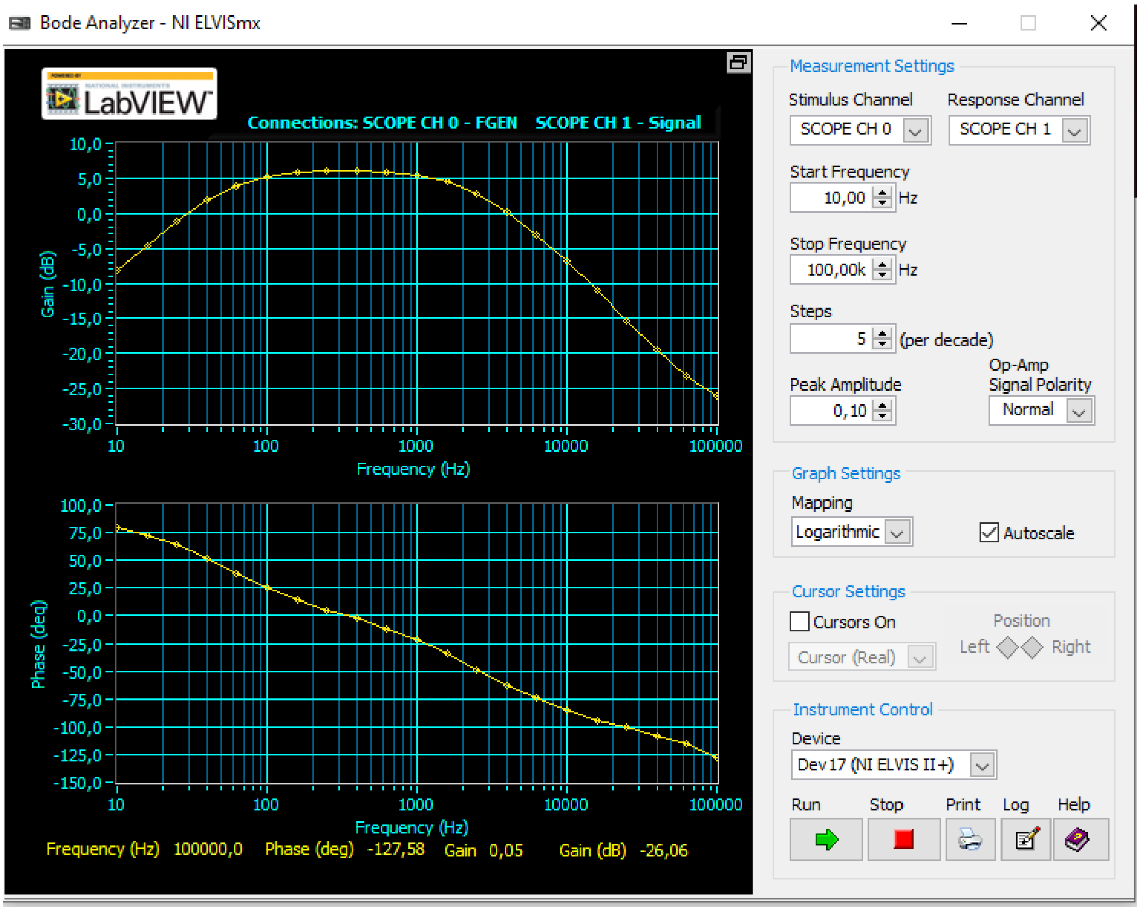

Figure 5.25: Bode analyser plot of volume tuning - gain amplification

Figure 5.26: The equaliser prototype connected with skin microphone and headphones

6
Discussion

The design of the analog equalizer investigated three different approaches to include frequency range selection and volume control features. Design3 which used the band pass filters was developed as a result of a final conclusion.Every design choice was carefully weighed in terms of its functionality components, constrains, and suitability for achieving the project's goals.

Tunable frequency selectors were utilized in Design 1 to modify the bandwidth. This design included a skin microphone incorporated to precisely detect vibrations from the skull bone, and a stethoscope as the main device for capturing sounds delivered through bone conduction. A source module, low-pass filter, high-pass filter with gain control, and components for low, mid, and high frequency, and frequency selection were among the crucial part of the signal processing component box. Though useful in altering window lengths, the adjustable ranges were not precise enough to choose certain value ranges, requiring laborious calibration and tweak beyond the scope of the project. Despite this drawback, Design 1 offered useful information about adjusting the frequency range, especially for the treble, mid and bass bands.

The focus of Design 2 was to record desired frequencies by use of resonant circuitry. It included a component box including necessary signal processing components and a stethoscope as the primary recording device, much like Design 1. Instead of using tunable frequency pickers, Design 2 used switches to filter out particular frequency values from resonant circuits. Despite having the ability to filter certain frequencies, Design 2 had difficulty managing the covered window range because of the narrow window widths caused by high capacitance and inductance values. Furthermore, more work than needed for the project was involved in the switching process. Resonant circuits have potential in frequency selection, as demonstrated by Design 2, despite these drawbacks.

The most effective approach was found in Design 3, which uses band pass filters to select certain frequency ranges. Design 3 made use of a stethoscope and a component box containing signal processing parts, just as the earlier versions. But rather than having resonant circuits or tunable frequency selectors, Design 3 had high pass, low pass and band pass filters for the treble, bass and wide frequency ranges in addition to a gain controller. Because of the clearly defined frequency ranges provided by this architecture, precise isolation and volume control were made possible along with straightforward switching options. Despite lacking the level of details found in other designs, Design 3's material was adequate to draw fundamental conclusions about hearing loss and aids, which was in line with the project's goals.

The outcomes validate the equalizer's operation. The results closely resemble the predicted values. The lowest volume level is approximately -40dB, while the maximum is approximately +9.6dB. By definition, cut off frequencies are where the amplification is 3 dB lower than the maximum intended amplification level. Design has been made to elevate the cut off frequencies to be 1 dB lower than the intended amplification or attenuation, taking into account the sensitivity of speech and audio signals. Equalization is established and the bandwidth is isolated within the appropriate range using the frequency selector switches.

Different observations were initially observed when high input peak voltage(more than 0.3V) and operational amplifiers with a weaker class AB amplifier were used. The instability was observed only when headphones were connected as load to listen to the equalizer circuit. Headphones have inductors which add reactive components leading to phase shifts and instability. An inclusion of a proper class AB amplifier and taking care of the simulated peak voltage when using AC sweep helps us to overcome the low power. Usually the peak voltage of skin microphone output is very low(less than 0.30 V), therefore the peak voltage issue is automatically sorted out when the tool is used for its corresponding input parameter The TL072 operational amplifiers are dual amplifiers embedded with class AM amplifiers already in them. Hence using TL072 amplifiers automatically cancel out the instability issue. Headphones compatibility is also crucial factor as the loading effects change between different headphones. The headphones used in the testing has a resistive load of 25 ohms along with its varying inductive load and becomes half when connected in parallel.

The prototype's objective was clear: it should be built as the simplest and most user-friendly form of an assistance tool for audiologists. The necessity to keep things simple helped us define the extent and boundaries of technology and possibilities. The scope for making the tool very simple included preferring analog design over digital, implementing logic and algorithms through electronic components rather than supporting embedded microcontrollers, using replaceable batteries instead of rechargeable batteries, and omitting indicators to represent or highlight functionality. Even the functional blocks function simply yet are quite useful. Even if the necessity to meet complicated criteria and high performance with many extra features is compromised, the efficacy of the tool to aid audiologists in listening to basic equalization frequency ranges remains constant. Thus, a simple and efficient tool is created.

The functional blocks that comprise the equalizers are those for frequency selection and volume control. The volume control provides a framework for implementing or improvising various frequency selection techniques. Additional features that could be readily added to the current base enables the prototype to support upgrade options. A configurable bandwidth window and aresonant circuit that can pick audiometric frequency values for analysis are examples of potential improvements.

6.1 Future work

The functional blocks that comprise the equalizers are those for frequency selection and volume control. The volume control provides a framework for implementing or improvising various frequency selection techniques. Additional features that could be readily added to the current base enables the prototype to support upgrade options. A configurable bandwidth window and aresonant circuit that can pick audiometric frequency values for analysis are examples of potential improvements.

Future development will focus on other applications that may use the equalizer tool. Control choices can be improved from the standpoint of the prototype. Additional bandwidth possibilities can be defined by a thorough calibration and design. Adding adjustable resistors and capacitors can aid in more precisely choosing desired frequency values. In order to delve further into pitch and tonality, resonant circuits can be implemented as an extra feature to separate tonal frequencies, their octave, and harmonics. Using an embedded controller board, digital signal processing methods can be integrated to add more capabilities. With the current developments in machine learning and signal processing, digital technology can assist patients in controlling and personalizing their sense of hearing.
The instrument can be used as a biofeedback and control assisting device or as a diagnostic and BC testing verification tool. Other fields of study interested in deciphering the secrets of bone conductivity can also use the instrument.

7

Conclusion

The effective appearance of the equalizer prototype's capabilities is emphasized by the design implementation's completion. The prototype efficiently uses signal processing techniques to accomplish simple inference tasks through careful planning and implementation. The preset frequency ranges and volume control functions, which enable accurate adjusting of the audio output to fit specific requirements, are crucial to its functionality. The equalizer prototype improves listening quality overall by allowing users to easily isolate and modify different audio spectrum sections through the provision of predefined frequency bands.

The design's portability and ease of use are two of its most noticeable features. The portability option and ease of deployment in a variety of environments are ensured by using 9V replaceable batteries. This power source selection improves the system's adaptability and removes the need for complicated wiring or additional power supplies, making it user friendly and available to a variety of users. Additionally, the design achieves an acceptable equilibrium between usability, simplicity, and accuracy, all factors that are crucial to achieving the project's goals. The equalizer prototype has sophisticated signal processing capabilities, but despite this, its user interface is simple and easy to use, making it easy for users to navigate and change settings. The resulting balance assures that the equalization system stays useful and effective in practical implementations, satisfying the various demands of users in various scenarios.

The conclusion essentially emphasizes the effective conversion of design thoughts into a practical and useful equalization prototype. The prototype successfully satisfies the project's criteria while providing a smooth user experience by carefully taking into account a number of variables, including power source selection, signal processing methodologies, and user interface design. In order to maintain the equalization system's relevance and influence in the audio processing industry and beyond, more improvements and alternations can be done as the project progresses to maximize its functionality and performance.

The prototype's capacity to provide predetermined frequency ranges and volume control. which let users customize the audio output to their unique needs and preferences, is one of its main strengths. The prototype makes precise modifications that improve the overall listening experience easier by offering preset frequency bands, which make it easier to isolate and manipulate different audio range parts. A gain controller is also included to further increase the prototype's adaptability by allowing users to adjust the signal amplification levels for the best possible clarity and integrity.

Conclusively, the equalizer prototype's effective implementation marks a remarkable breakthrough in the project's development path. The project's goal of offering a useful and effective solution for audio processing duties is realized in the prototype through careful design and execution. In order to maintain the prototype's relevance and influence in the field of audio processing and beyond, more improvements and optimizations can be performed in the future to improve its functionality and performance.

Bibliography

[1] Persson, A. C., Håkansson, B., Caveramma Mechanda, M., Bill Hodgetts, W., Fredén Jansson, K. J., Eeg-Olofsson, M., Reinfeldt, S. (2023). A novel method for objective in-situ measurement of audibility in bone conduction hearing devices – a pilot study using a skin drive BCD. International Journal of Audiology, 62(4), 357–361.

[2] Pickles, J. (1998). An introduction to the physiology of hearing. In An Introduction to the Physiology of Hearing. Brill.

[3] Reinfeldt, S. (2009). Bone Conduction Hearing in Human Communication - Sensitivity, Transmission, and Applications. Ph.D Thesis. Chalmers University of Technology: Available at: https://research.chalmers.se/publication/93679 (Accessed: 7 May 2024).

[4] Fredén Jansson, K.J. (2017). The Balanced Electromagnetic Separation transducer for bone conduction audiometry and hearing rehabilitation. Ph.D thesis. Chalmers University of Technology. Available at: https://research.chalmers.se/publication/249149 (Accessed: 7 May 2024).

Fredén Jansson, K.J., Håkansson, B., Rigato, C., Eeg-Olofsson, M., Reinfeldt, S. (2019). Robustness and lifetime of the bone conduction implant-a pilot study. Medical Devices (Auckland, NZ), 12, 89-100.

[5] Lukic, E., and Blomster, E. (2023). Finding the Optimal Head Position for Bone Conduction Sound Measurements when Using a Skin Microphone. Master Thesis. Chalmers University of Technology. Available at: https://odr.chalmers.se/server/api/core/bitstreams/90a0ef29-b0be-4fbf-b7b2-238913f58b62/content (Accessed: 7 May 2024).

[6] WHO (2024). Deafness and hearing loss. World Health Organization. Available at: https://www.who.int/news-room/fact-sheets/detail/deafness-and-hearing-loss (Accessed: 7 May 2024). https://www.who.int/health-topics/hearing-loss

[7] Hustad, K. C., Mahr, T., Natzke, P. E. M., Rathouz, P. J. (2020). Development of Speech Intelligibility Between 30 and 47 Months in Typically Developing Children: A Cross-Sectional Study of Growth. Journal of speech, language, and hearing research : JSLHR, 63(6), 1675–1687.

[8] Braun, B., Johnson, E. (2011). Question or tone 2? How language experience and linguistic function guide pitch processing. Journal of Phonetics, 39(4), 585–594.

[9] DPA Microphones (2021). Facts about speech intelligibility: human voice frequency range'. Available at: https://www.dpamicrophones.com/mic-university/facts-about-speech-intelligibility (Accessed: 7 May 2024).

[10] Brimijoin, W. O., Akeroyd, M. A., Tilbury, E., Porr, B. (2013). The internal representation of vowel spectra investigated using behavioral response-

triggered averaging. The Journal of the Acoustical Society of America, 133(2), EL118–EL122.

[11] Baese-Berk, M. M., Levi, S. V., Van Engen, K. J. (2023). Intelligibility as a measure of speech perception: Current approaches, challenges, and recommendations. the Journal of the Acoustical Society of America/the Journal of the Acoustical Society of America, 153(1), 68–76.

[12] Neely, K. K. (1956). Effect of visual factors on the intelligibility of speech. The Journal of the Acoustical Society of America, 28(6), 1275–1277.

[13] Desai, D. R. (2016). HEARING AID. Available at: https://drrajivdesaimd.com/2016/09/29/hearing-aid/ (Accessed: 7 May 2024).

[14] Valentinuzzi M. E. (2020). Hearing Aid History: From Ear Trumpets to Digital Technology. IEEE pulse, 11(5), 33–36.

[15] Reinfeldt, S., Eeg-Olofsson, M., Fredén Jansson, K. J., Persson, A. C., Håkansson, B. (2022). Long-term follow-up and review of the Bone Conduction Implant. Hearing research, 421, 108503.

[16] Taghavi. (2014). The Bone Conduction Implant (BCI) - preclinical studies, technical design and a clinical evaluation

[17] Rigato, C., Reinfeldt, S., Håkansson, B., Jansson, K. J., Hol, M. K., Eeg-Olofsson, M. (2016). Audiometric Comparison Between the First Patients With the Transcutaneous Bone Conduction Implant and Matched Percutaneous Bone Anchored Hearing Device Users. Otology neurotology : official publication of the American Otological Society, American Neurotology Society [and] European Academy of Otology and Neurotology, 37(9), 1381–1387.

[18] Persson, A. C., Håkansson, B., Fredén Jansson, K. J., Reinfeldt, S., Eeg-Olofsson, M. (2024). Objective verification of audibility in bone conduction devices. International journal of audiology, 1–7. Advance online publication.

[19] Hodgetts, W., Scott, D., Maas, P., Westover, L. (2018). Development of a Novel Bone Conduction Verification Tool Using a Surface Microphone: Validation With Percutaneous Bone Conduction Users. Ear and hearing, 39(6), 1157–1164.

[20] Martin, D. (2023). How do equalizers work?

[21] Young, C. (2023). Audio frequency range explained, Gear4music Blog, 11 August. Available at: https://www.gear4music.com/blog/audio-frequency-range/ (Accessed: 7 May 2024).

[22] Lenc, T., Keller, P. E., Varlet, M., Nozaradan, S. (2018). Neural tracking of the musical beat is enhanced by low-frequency sounds. Proceedings of the National Academy of Sciences of the United States of America, 115(32), 8221–8226.

[23] Menezes, P.deL., Cabral, A., Morais, L.daS., Rocha, L. P., Passos, V. (2004). Ressonância: um estudo da orelha externa [Resonance: a study of the outer ear]. Pro-fono : revista de atualizacao cientifica, 16(3), 333–340.

[24] Best, Virginia van Schaik, André, Jin, Craig Carlile, Simon. (2005). Auditory Spatial Perception with Sources Overlapping in Frequency and Time. Acta Acustica united with Acustica. 91. 421-428.

[25] Sound on Sound (1996). EQ: a New Perspective. Available at: https://www.soundonsound.com/techniques/eq-new-perspective (Accessed: 7 May 2024).

[26] Francis Audiology Associates. (2023). 6 High-Tech hearing aid features you may not have known about, Francis Audiology Associates Blog, 30 March. Available at: https://www.francisaudiology.com/blog/6-high-tech-hearing-aid-features-you-may-not-have-known-about (Accessed: 7 May 2024).

[27] Wang, H., Tang, D., Wu, Y., Zhou, L., Sun, S. (2020). The state of the art of sound therapy for subjective tinnitus in adults. Therapeutic advances in chronic disease, 11, 2040622320956426.

[28] Chalmers News (2023). Groundbreaking research in bone conduction technology. Available at: https://www.chalmers.se/en/current/news/e2-groundbreaking-research-in-bone-conduction-technology/ (Accessed: 7 May 2024).

[29] Valentinuzzi M. E. (2020). Hearing Aid History: From Ear Trumpets to Digital Technology. IEEE pulse, 11(5), 33–36.

[30] Nadol, J. B. (1993). Hearing loss. New England Journal of Medicine/the New England Journal of Medicine, 329(15), 1092–1102.

[31] Lagerkvist, H., Carvalho, K., Holmberg, M., Petersson, U., Cremers, C., Hultcrantz, M. (2020). Ten years of experience with the Ponto bone-anchored hearing system-A systematic literature review. Clinical otolaryngology : official journal of ENT-UK ; official journal of Netherlands Society for Oto-Rhino-Laryngology Cervico-Facial Surgery, 45(5), 667–680.