Design and Analysis Study for a Cost Effective and/or Better Quality Hearing Aid Device

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بسم الله الرحمن الرحيم

In the name of Allah, the most

Beneficent and the most Merciful

Declaration

It is hereby declared that this research study has been done for partial fulfillment of requirements for the degree of Master of Science in Biomedical Engineering. This work has not been taken from any publication. I hereby also declared that no portion of the work referred to in this thesis has been submitted in support of an application for another degree or qualification of this or any other university or other institute of learning.

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Dedicated ….

To a Person who is "The Rehmat" for all the universe,

And

To our Parents and to whom we love and respect

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LIST OF ABBREVIATIONS

AC	Alternating current
ADC	Analog to digital converter
B.T.E	Behind the ear
BAHA	Bone anchored hearing aid
BER	Brainstem evoked response audiometry
C.I.C	Completely in the canal
DAC	Digital to analog converter
DAI	Direct audio input
dB	Decibel
DC	Direct current
DFR	Digital feedback reduction
DNR	Digital noise reduction
DSE	Digital speech enhancement
DSP	Digital signal processing
ELI	Ear level instrument
ENT	Ear nose and throat
FDA	Food and drug administration
HIMSA	Hearing instrument manufacturers software association
Hz	Hertz
I.T.C	In the canal
I.T.E	In the ear
I/O	Input/output
LSB	Least significant bit
MSB	Most significant bit
NIDCD	National institute on deafness and other communication disorders
NIH	National institute of health (U.S. Department)
NIOSH	National institute for occupational safety and health
PCB	Printed circuit board
PTA	Pure tone audiometry
Q	Quality factor
REM	Real ear measures
SFR	Special function register
SNR	Signal to noise ration
SPI	Serial peripheral interface
T-coil	Telecoil
USART	Universal asynchronous receiver/transmitter
WDRC	Wide dynamic range compression
WHO	World health organization
w_o	Cutoff frequency

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ABSTRACT

World Health Organization report (2012) shows that 360 million people are suffering from disabling hearing loss worldwide. Hearing loss affects the developing and learning processes in children, while in adults, it affects and creates problems in the education, employment and general wellbeing. Hearing aid is the solution to it but a large number of hearing impaired people in Pakistan, either cannot afford the high prices of digital hearing aids, or they do not know how to choose a better quality hearing aid which suits their requirements. Moreover, since the analog hearing aids amplify the speech and noise signals equally, they are not much flexible and also have functional limitations. This research work is focused to primarily assist the hearing impaired people to choose better quality hearing aid devices keeping in view their budget. The research work also includes analysis, design and testing of a cost effective digital hearing aid. More than hundred digital hearing aids from different brands have been studied and comparatively analyzed. For user/patient, a selection procedure has been proposed in order to choose a better quality device keeping in view his/her budget/ hearing profile. Following the above objective, a microcontroller based cost effective, flexible, portable and small sized digital hearing aid has been designed and tested. The main components of the proposed system are sound detection circuit, a microcontroller and a digital to analog (DAC) converter. Audio signal from the microphone is amplified and converted to digital signal by 10 bit built-in analog to digital converter (ADC) of AVR atmega32 microcontroller and after the desired processing, the 8 bit DAC converts the signal back to analog signal and feed it to earphone. The internal clock of the microcontroller and built-in ADC avoids the extra circuitry and the external DAC helps in suppressing the low intensity background noise. The device is aimed to process the human voice frequencies only. It is tested in lab, implemented on PCB and it shows satisfactory results for the low frequency signals up to 3.5 kHz, which caters for the frequency range of most human voices. The minimum power (gain) of the device is 40 dB and has an adjustable gain of up to 80 dB. Based on the power range, this device is comparable to most of the lower cost digital hearing aids (mentioned in chapter 3) and the goal is to match it with Lotus Pro M device. Thus in future, the system will be developed for more effective noise suppression and broader frequency range, by adding frequency channels.

CHAPTER 01

INTROCUCTION

1.1. BACKGROUND

To understand the working of hearing aids, it is equally important to know the working of human ear [1]. The ear is very complex organ and its structure is divided into three parts, the external, middle and the inner ear [2]. The external ear function is the localization of sound. It picks up the sound signal and direct it toward the middle ear same like the directional microphone in the hearing aid [3]. The sound from the external ear vibrates the tympanic membrane, which in turn vibrates the ossicular bones of the middle ear and then these bones transfer the vibration to the oval window of the inner ear. So functionally, the middle ear performs three tasks. First, it has an impedance matching network, secondly, it amplifies the sound and thirdly it splits the signals into different frequency bands. Analogues to the middle ear, a hearing aid have some kind of impedance matching network, a pre and post amplifier, and has a signal processing unit which works with different frequency bands [3]. The oval window vibrates the inner ear cochlea, filled with fluid and contains hair cells. The vibration in the fluid stretches these hair cells, which in turn generate the neural signals and transfer it to the brain through auditory nerve[1]. The hair cells at the base of cochlea respond to higher frequencies while that of at apex responds to low frequencies [2]. In analogues to the inner ear, a hearing aid analyzes different frequencies and processes them accordingly to the user requirements [3].

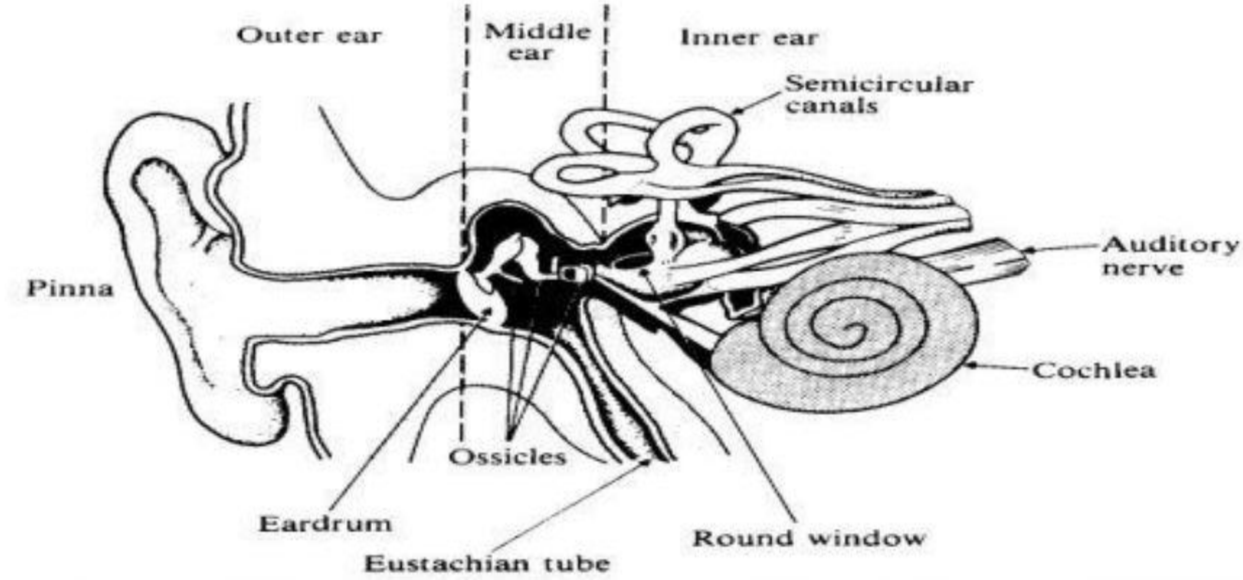

Figure 1.1: Structure of human ear. It is divided into three parts i.e the outer, middle and the inner ear.

1.2. HEARING RANGE

The normal human can hear the frequencies in the range of 20-20kHz., but the ear is most sensitive to the frequencies in the range of 500-4kHz which is the most important range of speech processing [4]. Hearing is affected by two parameters, the loudness or intensity of sound and the pitch or the fundamental frequency component in the sound. The ear is designed such that it perceives sound on logarithmic amplitude and frequency scales, i.e. if we double the amplitude, it will be perceived as same. Same is the case with frequency. If we increase the frequency to double the initial value, it will be perceived the same as before [3]. Sound level is normally expressed in decibel (dB) and it ranges between 0 dB (silence) and 140 dB (shotgun) [2]. National Institute of Occupational Safety and Health (NIOSH) classify the sound level above 85 dB for 8 hours as hazardous noise [5].

1.3. HEARING LOSS

Hearing loss is also known as deafness and it is the decrease in ability or inability to hear sound. According to the World Health Organization (WHO) reports, 360 million people are suffering from disabling hearing loss worldwide. Hearing loss causes negative effects on the life of both children and adults. In children, it affects the development and education, etc. while in adults, it affects many fields of life like employment and daily activities, etc. in short, affects both the personal and professional lives of a patient. Hearing loss varies from a degree that a patient, even doesn't know about it and up to complete deafness. Normally the ear can detect sound up to 25 Decibels (dB) [6].

Deafness varies from patient to patient; depending upon the severity of loss, it may be from partial to complete deafness. According to WHO, classification of deafness is given in the table 1.1;

Table 1.1: WHO classification of deafness/hearing loss

Degree of hearing loss	Hearing loss range (dB)
Normal hearing	0-25
Mild deafness	26-40
Moderate deafness	41-55
Moderately severe	56-70
Severe deafness	71-91
Profound/stone deafness	>91

1.4. TYPES OF DEAFNESS/HEARING LOSS

Hearing loss comes with the most common medical conditions that are presented to doctors (ENT Specialists). Etiologically, hearing loss can be divided into the following types

- Sensorineural hearing loss
- Conductive hearing loss
- Mixed hearing loss

1.4.1. Sensorineural Hearing Loss

It is the most common of all types and about 90% of losses are sensorineural and most of these losses involve the damage to the cochlea or the eighth cranial nerve. Sensorineural loss occurs due to the dysfunctioning of the inner ear receptors. This loss is irreversible and mostly the loss occurs at higher frequencies. The causes of sensorineural loss can be congenital, parental, aging, birth related disorders, ototoxic drugs, trauma, bacterial infections, intense noise environment, meniere's disease, fractures of the temporal bone, congenital malformation and cochlear otosclerosis.

1.4.2. Conductive Hearing Loss

This type of hearing loss occurs whenever there is a problem with the conduction of sound wave along the sound conduction pathway, i.e. problem in the conduction of wave either along external ear, ear drum or the middle ear.

In conductive hearing loss, the sound is not distorted at all and the quality of voice is maintained. The sound appears quieter and the patient can clearly hear or listen to her own voice. As this loss may be due to external or middle ear, so the common causes of this deafness is divided as follows

- External ear causes
  - Impacted wax
  - Ear infection (otitis externa)
  - Fungal infection (otomycosis)
  - Boil, if large enough to occlude the ear canal
  - Tumors of external ear
  - Ear drum perforation

- Middle ear causes
  - Cholesteatoma
  - Otoslerosis

1.4.3. Mixed Hearing Loss

Mixed hearing loss is the combination of both sensorineural and conductive hearing loss and it occurs when the cause is due to both of them.

1.5. HEARING AID

A hearing aid is an electro-acoustic device which converts the acoustic power into electrical power, amplify it electronically and convert it back to sound. The basic components of a hearing aid are a microphone, an amplifier, a speaker and a battery. The microphone picks up the sound signal from the air and converts it into an electrical signal and this signal is then amplified by the amplifier and is fed into the speaker which converts the electrical signal back into the sound signal. The battery provides the power for the whole process to occur [1, 4]. It helps the hearing impaired person in amplifying sound for him/her so that he/she can listen and participate well in daily activities [7]. It helps the user more in both quiet and noisy environments [8] but cannot correct the hearing loss completely [9].

In the last couple of years, much innovation has occurred in the hearing aids [1]. Some of the recent innovations are the open ear hearing aids, smaller light weight hearing aids, wireless hearing aids which provide connectivity to the household electrical devices, powerful behind the ear hearing aids and noise reduction algorithms [10]. Functionally a hearing aid can be analog or digital hearing aid.

1.5.1. Analog Hearing Aid

Most of analog hearing aids cannot differentiate between the speech and noise signals and amplify both of them equally. Some of them are programmable, having a microchip through which the device is set to different situations like in noisy environments or in silent places [7,8]. They can store multiple programs for various environments and a push button is used to set the device to a particular environment. They are less expensive than the digital hearing aids but are becoming less common due to some functional limitations.

1.5.2. Digital Hearing Aid

Functionally, these hearing aids contain all the features that analog hearing aids have. They convert the sound signals into binary codes of computer and exactly duplicate the sound signal at the output [7,8]. The digital signal processing (DSP) makes them more preferable than the analog hearing aids. They amplify sound in a much more complex way and hence perform better in noisy environments. They provide greater flexibility during the programming and fulfill the sound requirement of a user or specific pattern. They have multiple programming memories and due to the flexibility, its circuitry can be used for all types of hearing aids.

1.5.3. Features Of Hearing Aid

Hearing aids come with different features which allow hearing impaired person to listen sound in different environments and automatically adjust the aids accordingly to the situations. However, as the number of features in a device increases, it increases the cost of a hearing aid and vice versa. Some of the important features are discussed as below.

1.5.3.1. Directional microphones

This feature allows the user to listen to speech, even in noisy situations. It amplifies the sound coming from a specific direction or angles in greater amount compared to other directions. The main purpose of directional microphone is to increase the signal to noise ratio (SNR),i.e. during the face to face talk or when the sound comes from the front direction, then it amplifies it greater than the sound coming from the other sides [9] and hence allows to user to enjoy conversation without any disturbance.

1.5.3.2. Telecoil

This feature allows the direct connectivity to the magnetic signals like the landline phones and inductive transmission systems and eliminates the environmental noise [10]. In t-coil mode, the microphone is automatically turned off. This option can be used in auditoriums, theaters and all other places which have FM installation or an induction loop.

1.5.3.3. Direct Audio Input(DAI)

This feature allows the direct connectivity of a hearing aid to a TV, computer or a CD player etc. and it can also be plugged in FM assistive systems or a remote microphone [7]. This mode is less susceptible to EM interference and produces a high quality signal.

1.5.3.4. Feedback suppression/management

The important source of noise in hearing aids is the feedback signal i.e. when the hearing aid has a loose fitting or when it gets close to the telephone, then the feedback signal appears at the input and it acts as a noise. So to remove or minimize the noise, this feature is used and it suppresses the squeals [7].

A lot of many other features are available in the hearing aids like digital noise reduction, no of channels and bands etc., but as the no of features increases, it also increases the price of a hearing aid.

1.6. OBJECTIVES OF RESEARCH

The objectives of the study are to;

- Briefly study and analyze human conductive hearing loss problem
- Carry out a comparative analysis of various hearing aids available in the market and suggest the best ones with respect to quality and cost for a prospective patient keeping in view his budget.
- Carry out a feasibility study and attempt if possible, to test a sample hearing aid device, in order to reduce cost and/or improve quality.

1.7. REASON/JUSTIFICATION FOR THE SELECTION OF THE TOPIC

While NUST student volunteer team was visiting Sir Syed School for special Education in Rawalpindi, many deaf students pointed out the issue of poor quality and high costs of hearing aids available in the market. This project has subsequently been taken up keeping in view the community needs.

1.8. RELEVANCE TO NATIONAL NEEDS

According to the Pakistan Association of the deaf, there are approximately 7.43 % of deaf people in Pakistan suffering from different types of hearing loss. This study will assist them in choosing a hearing aid device, keeping in view their budget and will provide queues, whether a better quality/ low cost hearing aid is possible or feasible.

1.9. ADVANTAGES

The study will serve following purposes at Community / national level:

- Will assist a patient suffering from conductive hearing aid loss to choose an appropriate and best quality hearing aid device, keeping in view his budget.
- Will provide queues using testing/reverse engineering whether the cost can be brought down keeping the quality same or quality could be increased keeping the price same, for the existing device in the market.

1.10. Areas of Application

- Study and analysis of human conductive hearing aid loss problem
- Study and analysis of medical devices: Hearing aids
- Comparative analysis of various hearing aid devices
- Design, development and reverse engineering of hearing aid devices, if possible

Chapter 02

RELEVENT RESEARCH

2.1. INTRODUCTION

According to a WHO report, 360 million people are suffering from disabling hearing loss in the world [11]. Hearing loss causes negative effects on the development and learning processes in children, while in adults, it creates problems in their education, employment and general well being. Overall, it affects personal and professional relationships [12].

Hearing aids and cochlear implants have been developed to assist people with hearing disabilities [12] but a small portion of adults seek help for hearing aids and use it [13]. Some studies show that a large number of people do not afford and have hearing aids who could actually take advantage of it [14, 15, 16].

Hearing aids started their journey almost few centuries ago when the ear trumpets were used to transmit sound to the ear. It probably acquired its modern shape from the telephone invention of Alexander Graham Bell and Thomas Edison invention and was considered as one of a principal site in the component miniaturization of electronics. Significant improvements in hearing aids came in parallel with the developing technology. Withthe development of digital technology, complete digital hearing aids came into the market at the end of the 20th century. DSP chips not only have improved the existing features of hearing aids, but also have given many new features to a modern hearing aid. A directional microphone, noise reduction algorithms and wireless technology have increased the signal to noise ratio (SNR) and have improved speech recognition in noise in the modern digital hearing aids. The coming sections review the history of hearing aids and presents an overview of the recent developments and future trends.

2.2. HISTORY OF HEARING AIDS

Till 17th century ear trumpets were used as hearing aids [17,18], they were a little modified in the 18th century and now they keep focused on the source of the sound [18]. Advancement in the ear trumpet was initially made by introducing a hearing tube used to collect the sound through a cone shaped opening at one end and directed that to the ear at the other end. Heinrich August Van Danker filed the first patent on hearing tubes in 1819 [19].

The consecutive inventions by Alexander Graham Bell and Thomas Edison played an important role in establishing a new foundation of the modern hearing aids [20].Alexander made a telephone and patented it in 1876, it was a first electrical device that could transmit the signal [21] while Thomas Edison made a carbon transmitter in 1878 that was used to amplify the electrical signals [21,22]. Alexander used the transmitter, battery and a receiver, and made a hearing aid device for his mother [23]. In the beginning, Electrical hearing aids were as same as the small telephone system [22].

The race for developing better and efficient hearing aid devices has started and different patents were filed in 1890's but the 1st patent by Mr. Alonzo E. Miltimore qualified the production level in 1892 [24]. Several companies, including Kitchener and Wilhelm of Stuttgart, George P. Pilling and son of Philadelphia made the hearing aids of their own versions [20]. Dictograph company introduced the carbon-type hearing aids in 1898 [20]. A year later Miller Reese Hutchison patented the 1st practical hearing aid device that contained a carbon microphone that captured the sound and passed on it to the carbon receiver that finally converted it into an output and sent it to the ear through headphones [20, 25, 26]. It was very heavy in weight, placed on a table and was sold in 400$. Later on it was named as "Akouphone"[20,25].

In 1913 Siemens introduced the electronically amplified hearing aids for the first time, these devices were about the size of a cigar box, they were not much portable but their speakers fit in the ear [8]. In 1920s, vacuum tubes were introduced in the hearing aid device. This modification made the sound amplification more efficient [25]. Earl Hanson patented the vacuum tube hearing aid device in 1920 [27], but these devices were still heavy by weight due to large batteries limited to life time of 1 day because of the high power consumption [25]. Advancement in the military technology in the World War 1 helped in the improvement of hearing aids too, and around 1925, the first portable hearing aid was made by Acousticon (manufacturer) [18]. It was a model “56” but still heavy by weight to carry [18]. In 1932, Sonotone Corporation introduced a new type of hearing aid device called a bone anchored hearing aid which could be used in different kinds of hearing impairments, but is more effective in assisting people with middle ear disease [25].

Step-next advancement was a wearable hearing aid device developed by Aurex Corp (an electronic manufacturer) in 1938. In this device, an amplifier-receiver, clipped to wear clothes, was connected to earpiece via a wire and receiver was wired to a battery which strapped to the legs [21]. A year back in 1937, Norman Krim developed the Subminiature vacuum tubes that made the size of amplifier smaller and required less power [21]. Later these tubes were combined with the two innovations from the World War 2—button batteries and printed circuit boards--and a compact and reliable model was produced in which amplifier, microphone and battery were combined in one unit that could fit in a pocket and the earpiece was connected to it through a wire [21].

Till the 1947, hearing aids were heavy in weight and body worn only, but the transistor invention brought a great change in the hearing aids because they were cost effective, smaller in size and their power consumption was also low [22,25] and due to these on/off switches, it became possible to reduce the size of hearing aids. Efforts were made to design hearing aids that could fit within eyeglasses or can be placed behind the ear called B.T.E [18]. Finally Sonotone introduced the first transistor amplified hearing aid device in 1952, it hit the market and was sold for 229.50$ [22]. By 1953, more than 200,000 transistorized hearing aids were sold by different companies [21], and by 1960 hearing aids like “hearing glasses”, I.T.E and I.T.C were available in the market [21,25]. Hearing aids were the main site for component miniaturization and the integrated circuits found their first commercial application in hearing aids [22].

The struggle has been started to improve listening comforts by reducing noise. In 1969, the built-in directional microphone was manufactured [29,30p87] and in 1970s, integrated circuits for non-linear compression were made for hearing aids, which could reduce the environmental noise from speech [30p87]. Up to early 1980s, analog speech processing was in the market [30] and digital hearing aids were not much developed and it was considered a science fiction to reduce the size of computer to fit it in the ear [31].

Late 1980s and early 1990s was the age of hybrid analog-digital models [32]. In 1988, programmable devices were introduced that could be electronically programmed from a computer software or a user operated remote control [33]. In the early 1990s, most hearing aids were analog but with the availability of audio signal processing technology and miniature computer components, the digital hearing aids were born [33]. In 1991 for the first time a fully automatic device without volume control was introduced by Oticon [34]. In 1992, GnDenavox commercialized the DSP hearing aid device which was not completely digital and still having some analog circuitry [35]. In 1993-94, "completely in the canal" (CIC) hearing aid device was introduced by Argosy Electronics [36].

The era of digital hearing aids was begun in 1996 [21] and a first 100% digital B.T.E hearing aid device was made on the bases of Adaptive speech alignment, which uses two different processes for consonant and vowel sounds and divide sound into seven bands [33]. The progress continued and in 1997 a new chip set was made that allows the advance feedback management, acclimatization and also allows the user to adjust his/her own choice audio range in which he/she could listen comfortably [25]. The same year, Philips introduced a model D72, which was a remote controlled [32]. By 2000, hearing aids were available with greater flexibility and the wearer could program it according to his/her need [21].

From the start of the 21st century, manufacturers introduced different features and technologies in the hearing aids which made them more comfortable and reliable. In 2001, Oticon introduced the "Voice Finder" feature in its Adapto hearing aid. This feature could detect voice in the noise and process it, while when there is no noise it automatically switches to comfort processing mode [37]. In 2003, SeboTec introduced a PAC system, which is a receiver in the canal hearing aid; this system features directional microphone, multi memory, WDRC and digital signal processor [38].

By 2005, more than 80% of the users had digital hearing aids [21]. Further, an ELI was introduced to hearing aids that connected the hearing aid with digital wireless technology, Bluetooth [39]. In 2008, "Lyric" was introduced by Insound which could be worn for 24 hrs and up to 4 months [40]. In 2009, SoundAMP was introduced by Ginger Labs, which is software and through itan iPhone can be used as a hearing aid device [41]. In 2011, Aquaris was introduced by Siemens which is a dustproof, shockproof and a waterproof hearing aid device [42].

The computer technology has brought a remarkable change in performance of hearing aids, sizes are made smaller and smarter and have the ability to adjust to different listening conditions, but there is still a space for improvement especially with the background noise. Researchers are working on different algorithms like the DNR and directional microphones to filter out this noise.

2.3. ADVANCEMENT IN HEARING AIDS

Since the introduction of digital signal processing to hearing aids in 1996 [43], much advancement occurred in the hearing aids. In modern hearing aids the DSP chips have improved the features very much, but still limitationsexist and some users have complained regarding the speech recognition in noise. The MarkeTrak V data about the hearing aid users show that about 25% usershave a hard time listening in the presence of background noise and 1% has difficulty while using telephone [44].

The SNR (Signal-to-Noise Ratio) and speech recognition has much been improved with the development of directional microphones and wireless technology in the recent years. Development in the noise reduction algorithms helped the users to listen in noise with ease and comfort. Some of the other development in the recent years like frequency compression, which helps users having higher frequency loss problem, self learning and REM (real ear measures) makes the fitting of aids much easier for both the patients and audiologists [45].

2.3.1. Wireless and Bluetooth Technology

Wireless technology is not new in hearing aids; it dates back to 1930 when t-coils or induction coils were used in hearing aids [10]. Wireless communication in hearing aids has improved with time as soon as some progress occurred in wireless communication technology. Hearing devices make use of the wireless transmission through FM transmission AM transmission, induction coils or any of these combinations and helps the users to listen in different listening situations

[45]. The digital wireless technology like Bluetooth was introduced by Hearing Instrument Manufacturers Software Association (HIMSA) in 2003 through a NOAH Link device [46]. It is used in programming a hearing aid wirelessly, but the NOAHLink system still uses some cables to connect to aid while programming.

Bluetooth has FM wireless technology and provides wireless connectivity between different devices like MP3 players, mobiles phones, computers and various other devices [45, 47, 48]. It is a short range device of almost 30-300ft transmission range which depends on the class of blue tooth being used [45,47].

The Bluetooth technology development in the hearing aids increases the connectivity between the patients and technology [45] i.e. the Oticons streamer is a bluetooth compatible central unit and it enables the hearing aid to connect to cell phone or other bluetooth devices wirelessly or can be directly connected to a microphone jack through a wire. It provides the connectivity through a digital magnetic wireless transmission [49].

Today a wireless technology has many features in hearing aids i.e. wireless technology can be used for the command code (add-to-aid), audio signal streaming (add-to-aid), remote control, streaming and companion microphones [50].

2.3.2. Frequency Lowering

Frequency lowering option is for those who have severe to profound hearing loss. The idea behind the frequency lowering is that the high frequencies are shifted toward the lower frequencies where it is easy for patient to listen [45].

Frequency transposition and Frequency compression are two major approaches for frequency lowering technique implementation. In the frequency compression approach the whole band of frequency is compressed to a narrow bandwidth, i.e. if the patient inaudible range is above 4000 Hz and compression ratio of 2 is used, then after the frequency compression, the signal at 8 kHz will now appear at 4 kHz and a signal previously at 4 kHz will now appear at 2 kHz. Now the overall bandwidth is 0-4 kHz, which now also contains the frequencies previously inaudible [51].

While in contrast, in frequency transposition higher frequencies are shifted towards the lower frequencies such that they superimposed. i.e. a patient with an inaudible range above 4 kHz, and a one octave transposition target is used, then a signal at 8 kHz will be shifted to 4 kHz and a signal at 4 kHz will be shifted to 2 kHz, but the original 4 kHz, 2 kHz and 1 kHz (and so forth) will remain in their original positions and will mixed with the frequencies above 4 kHz. This approach has a main advantage of keeping the temporal structure of original signal preserved, although it causes initial confusion and masking of the original signal [51]. Widex used the frequency transposition feature in their hearing aid model and referred to it as audibility extender. The starting point for transposition frequency is set by software after examining the patient's sensogram. It uses a linear transposition algorithm and transposes sound into an audible frequency range of a patient by one octave [45]. As it shifts the higher frequencies to lower and middle frequency regions, so allows the user to listen to consonants and other high frequency sounds that were previously inaudible to him/her [52]. Audibility extender as a program option allows the user to set it as a primary or secondary listening mode in different listening situations. The PhonakNaida model uses the frequency compression and it uses the cutoff frequency as a limit for stopping amplification and start higher frequency shifting towards lower frequencies [45]. Frequency lowering shifts higher frequency sound into low frequency sounds. Although it is beneficial, but initially users feel uncomfortable as they percept sound as unnatural and it may be objectionable [53]. But it may be a frequent reaction of a user based on his past experience and not that what he will experience in the future [53]. i.e., many patients with severe-to-profound hearing loss, that were previously using linear aids, complaint about insufficient loudness at the start but after some experience with non-linear hearing aids they accepted it and showed improved performance with it [64].

2.3.3. Directional Microphones

Directional microphones in the hearing aids have been improved very much since in the last fifteen years [55]. Different algorithms are made for directional microphones using digital signal processing [56]. Before the DSP introduction in hearing aids, analog hearing aids either had dedicated directional microphones or a push button was used to switch between the directional or omnidirectional settings [45]. The trends in directional microphones algorithms can be explained as a step-by-step process [55].

Directional microphones were developed in hearing aids in order to increase the SNR [45]. There are two sound ports in fixed directional microphones, signals entering at the back port are acoustically delayed and are subtracted from the signals entering in the front microphone [57]. Research shows that speech recognition in noise is improved with directional microphone, but it has some limitations [45]. Directional mode is preferred when there is background noise [58]. However, many users do not switch appropriately to directional mode or even don’t know it and hence don’t take benefit of it [45]. A comparative study about the directional microphone’s lack of success and benefits was conducted by Walden and Dyrlund [58], the overall result of this study shows that although directional microphones are very successful in the laboratory, but in real life its performance may not be the same for all users as many of user could not control it correctly [45]. To overcome the directional microphones limitation, the automatic directional microphone was the next development. Its algorithm is such that the user automatically switches between the directional and omnidirectional mode. Auto switching in it is dependent on SNR, input level and signal location [56]. This feature works well for some user who could not do it manually. However, it can create problems for those who do not want to switch, but it switches automatically, or it switches so fast and amplify the unwanted or noisy sound like dogs barking or cough sound [56]. Keeping in view these and fixed null limitations, the next advancement was automatic adaptive directional microphones [45]. The main advantage of an automatic adaptive directional microphone is that it automatically adjusts itself to the source of noise coming from any direction. i.e. if the noise angle is between the back and side, then it will switch automatically to a hypercardioid polar design [56].Studies showed that when noise is from a moving source, then a user with automatic adaptive directional microphone shows similar performance like a normal listener as compare to that one with directional microphone [59]. Many researchers have shown advantages of it. However, in the presence of moderate level uniform noise, it shows a reduce performance [60]. The next advancement in directional microphone technology is Multichannel automatic adaptive directional microphone. This kind of directional microphone contains different polar designs in frequency channels of hearing aids and switches automatically between the directional and omnidirectional modes [45]. This feature tackles with different scenarios and different kinds of noise sources present in the environment.

i.e. in a windy environment, it may be in directional mode in the higher frequency region and omnidirectional mode in lower frequency regions [45]. The next development in this regard is an A;2symmetric directional microphone. The standard way for using it is binaural directional fitting. One way to use it in noise is that both the binaural aids are in directional modes (standard way) while another option is to use one aid (right sided) in directional mode while the other aid (left sided) in omnidirectional mode [55]. This feature may be the much beneficial for a student with one aid in omnidirection and other in directional mode, with directional mode he will concentrate upon lecture while with omnidirectional he will listen from rest of the class [45]. When the sound source is from the front, near to the listener and the noise source is separate from signals then asymmetric directionality shows improved performance [61] while in noisy situations the binaural directionality shows good performance [62]. Taking advantage of wireless technology, another approach narrowing the beam-width is used for directionality. In this approach the two microphones on one aid communicates with the two microphones on the other aid. This moves the null point further to the front and creates a narrow beam (approximately 45), this narrow beam then improve SNR [55]. Another approach is using an array of microphones outside the hearing aids, but it has cosmetic issues [55].

The step-by-step progress in directionality is directional microphones, automatic directional microphones, adaptive directionality, Band-split directionality, manual and auto beam width adjustment, manual and auto steering of directional pattern, asymmetric directionality, narrowing the beam-width and multiple microphonearrays [55].

2.3.4. Noise Reduction

Like other algorithms, noise reduction has also been much developed with digital signal processing. In early analog devices, algorithms were intended to reduce the gain of low frequencies and hence filter out noise [45]. It prevented the analog compression circuitry to be activated by a strongly low frequency signal. The limitation of analog algorithm lies in that the algorithm was implemented in one channel only and gain reduction was based on the level of input signal [63].

At the start, the wiener filter and the spectrum subtraction were used for DNR (digital noise subtraction) [63, 64]. In spectral subtraction a noise sample is selected during the speech and it is subtracted from whole the signal (speech + noise) leaving only the speech signal [63, 64] while the wiener filter generates a filter based on characteristics of noise and sound signal and gives optimal SNR at the output [64]. The limitation in both these techniques is stationary noise signal requirement [63]. DNR does not rely on the spatial separation of speech and noise, it determines whether the channel is dominated by noise or not and then reduce the gain in a frequency channel [45]. Gain reduction also depends on the weighting of specific frequencies, SNR in each channel and the on off time of DNR algorithms [57].

In Multichannel noise reduction algorithms, the presence of speech is determined in a channel based upon modulation. If the difference between the characteristic of speech and competing signal is narrower,then the DNR may not be as effective as it is with steady noise [57]. This approach is advantageous but if the filter is much narrower, then there is a possibility of greater group delay [65]. Algorithm Synchrony detection is based on the co-modulation of voice as the vocal folds open and close. Vibration of vocal cord creates human voice and its fundamental frequency is approximately 250 Hz. Opening and closing of vocal folds generate frequencies which are burst of periods and synchronous energy [57, 66]. The gain reduction in this algorithm depends on the presence of synchronous energy, when speech (synchronous energy) is present in channels, then this system decreases compression (provide amplification) otherwise increase compression or reduce gain [66]. This system has a drawback of speech recognition when both the noise and the speech signals have similar spectral patterns [57].

The DNR feature has become a standard in hearing aids. Most studies regarding the hearing aids features and speech recognition in noise have not isolated DNR, but they evaluated the effectiveness of DNR combined with directional microphones [45]. DNR algorithms reduce gain in frequency regions where primary signal is the noise; however, ina realistic environment, often the noise and speech spectrum overlaps [67]. Therefore, in the presence of both speech and noise, many DNR algorithms don’t change the signals significantly and preserve or maintain the speech audible [68]. The goal of DNR is to reduce noise effect and improve speech, as the DSP technology improves the more complicated and newer algorithms may be applied to improve voice quality in noise.

2.4. Evaluation Of Conductive Deafness/Hearing Loss

ENT Specialists refer to the following methods for the patients complaining about their decrease in ability to hear sounds or speech.

- Clinical Examination
- Voice Test
- Tuning Fork Test
    - Rinni’s Test
    - Weber’s Test
- Audiometry

2.4.1. Clinical Examination

On clinical examination, the ENT Specialist inspects and examines the pinna along with its surfaces, mastoid region and external auditory meatus. The purpose of the examination is to check scar, edema, swelling or any discharge. As the shape of external auditory canal is sigmoid so it is first straighten before the examination by pulling it backward, upward and laterally. By doing so, it is then checked for any foreign body, fungal growth, impacted wax, and bulging or eardrum perforation [69]. The tympanic membrane mobility is examined by valsalva maneuver and then it is looked for bulging or any change in color or perforation, that may be a cause of conductive deafness [6].

2.4.2. Voice Test

Voice test is performed in order to know or assess the severity of hearing loss. During this test, the patient is asked to turn on one side such that the affected ear of the patient faces the examiner. The other ear of the patient is blocked by keeping the index finger of the patient on his/her external auditory canal. Examiner puts his/her hand on the patient's eyes in order to reduce the chances of guessing by lip reading. Then the examiner whispers some words from the distance of one arm length and the patient is asked to repeat the words, if he/she does not perceive the words, then the volume of speaking is increased and the procedure is repeated until the patient repeat the words correctly.

This test gives the rough approximation of a patient's severity of deafness.

2.4.3. Tuning Fork Test

Tuning fork test is normally performed in clinical examination and the normally the tuning fork of frequency 512 Hz is utilized.

Following are the types of tuning fork tests that are applied in clinical practices

- Rinni's Test
- Weber's Test

2.4.3.1. Rinni's test

In Rinni's test, the comparison of air conduction of a patient's ear is done with that of a bone conduction (keeping the same ear for comparison). The RINNI's test has two outcomes, namely

- o Rinni's positive
- o Rinni's negative

Normally the human has batter air conduction then that of bone conduction (in case of tuning fork only), so if the patient stops hearing through air conduction then he will also not be able to hear through bone conduction. This is called Rinni's positive.

Now if the patient has conductive type of deafness, then his/her air conduction of the ear will be less than bone conduction and he/she will only hear through bone conduction after his/her lack of hearing through air conduction. This condition is called Rinni's negative.

2.4.3.2. Weber's test

In Weber's test, the patient's bone conduction of both ears was compared and for test performance, the tuning fork is placed over the midline of the forehead. Then the patient is asked whether he is hearing or not? Normally the patient will hear equally in both the ears.

In case of a conductive type of deafness, the patient will hear a lateralized sound in that ear which is affected. Therefore, this test is also called a test of localization.

2.4.4. Audiometry

The science of hearing is termed as audiology and it covers all the aspects and topics in physiology of hearing, acoustics, disorders of hearing, education and rehabilitation of deaf, functional examination of hearing, cochlear implants and hearing aids.

Audiometry is one of the fields of audiology and it relates to measuring the hearing acuity. i.e the frequency, loudness and timbres of sound. This procedure is noninvasive and painless.

Audiometry is broadly classified into two types.

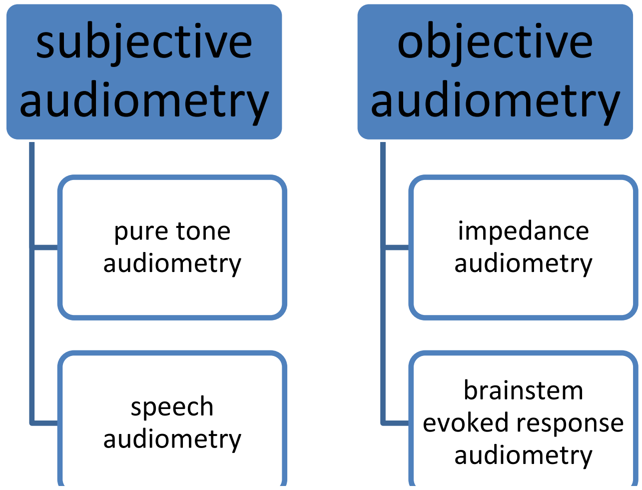

Figure 2.1: Types of audiometric tests. The subjective audiometry requires and active cooperation and a behavioral response from the patient while the objective audiometry does not necessarily require it.

2.4.4.1. Pure Tone Audiometry

Pure tone audiometry is the most common method of measuring the hearing equity. In this method, pure tones are generated by audiometer and are delivered to the ear through an earphone. The frequency of tones varies from 125 Hz to 8000 Hz and the loudness of tones varies from 0 dB to 120 dB. Both the air conduction and bone conduction of a normal human ear lies between the 0 dB and 25 dB and has no air-bone gap [6].

In case of a conductive type of deafness, the graph of air conduction goes down the bone conduction graph while the bone conduction graph remains within its normal limits. This gap between the air conduction and bone conduction graph is called the air-bone gap and it is used to measure the degree of deafness [6].

Figure 2.2 is a sample PTA test of a patient and it shows the level of deafness of a particular patient at different frequencies.

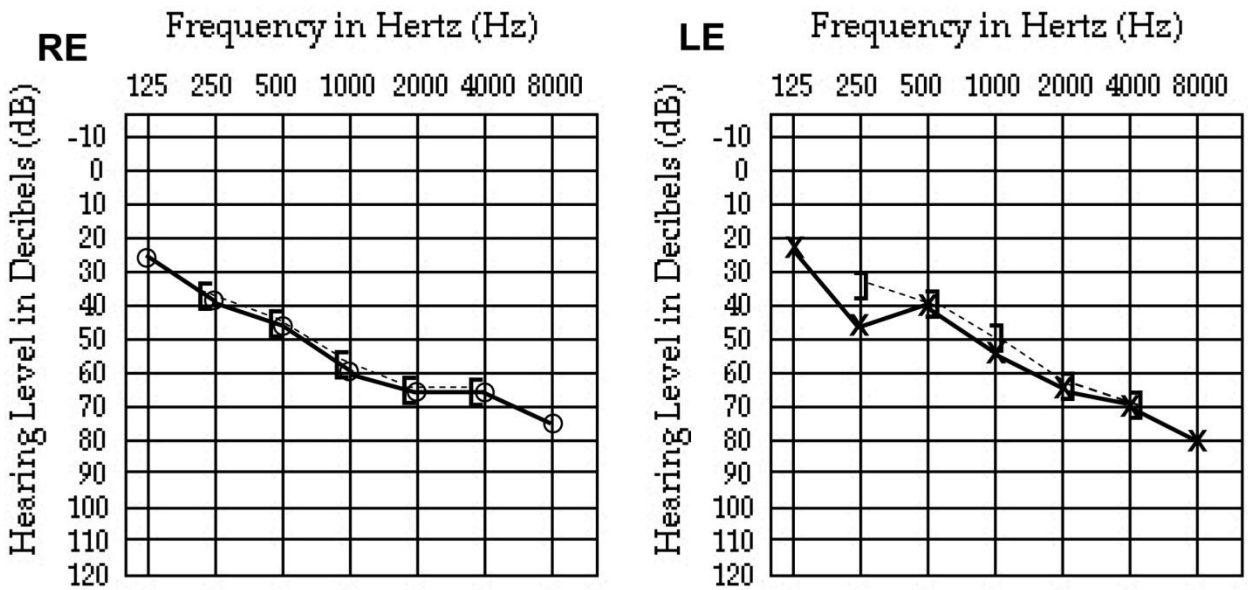

Figure 2.2: Sample PTA test result of both ears. This type of test shows the hearing loss of patient at individual frequencies over a wide frequency range.

CHAPTER 03

COMPARASION OF DIGITAL HEARING AIDS

3.1. INTRODUCTION

Audiology is the field concerned with the selection of hearing aid and the satisfaction of the hearing impaired person with it. An audiologist is the person who specializes in the evaluation and treatment of hearing loss. As a first step, he conducts the different hearing test of a patient and based on their results, suggests the different treatment options for the patient. A hearing aid is one of the options for hearing impaired person. Digital hearing aids come with different features and technologies that help the users in listening with greater comfort and better cosmetic appearance. They have a different number of channels that divides the audio frequency into multiple bands and allows the audiologist to programme each channel individually, keeping in view the user requirements. But as the features and channels increase in hearing aids, it also increases the price. The audiologists suggest the device based on the severity of hearing loss and user comfort, but the user has to tradeoff between the price and available features in hearing aids. This chapter looks over the trends in digital hearing aid, cost estimation, patient requirements and overall procedures from audiologist visit up to fitting the hearing aid in the ear.

3.2. THE DIGITAL ADVANTAGE

For both the patients and dispensing audiologists, advanced digital signal processing algorithms and features available in Digital hearing aids provide a large flexibility and great advantages over the conventional analog hearing aid. Some of the key advantages are;

3.2.1. Flexible Gain Processing

The main advantage of flexible gain processing algorithm is to increase the audibility of sound in the interested frequency bands with comfort and avoiding high intensity discomforted sounds. It is the great benefit of compression scheme rather than the DSP. Generally the sound is divided into a number of channels and compression is applied independently to each channel accordingly to the results of human hearing loss tests. Another strategy, opposite of compression, an Expansion is introduced into the digital hearing aids which help in greater user satisfaction by reducing the role of intensities of low level sounds in the environment and microphone noise that may be otherwise annoying.

3.2.2. Feedback Reduction

Advanced Digital Feedback Reduction (DFR) algorithms continuously monitor for different feedbacks while the user is wearing the digital hearing aid. It eliminates or reduces the moderate feedback using the notch filter or the feedback cancellation system. Normally the jaw movement, loose fitting of hearing aids or close proximity to objects creates the feedback in hearing aids. The DFR algorithms benefit those users who experience such occasional feedbacks.

3.2.3. Noise management

The Digital Noise Reduction (DNR) algorithm reduces the gain when it detects the steady state noise signal either in low frequencies or in specific frequency bands. The outcomes of the DNR's efficacy finding research are mixed. They show that DNR improves the speech recognition, and reduces the annoyance in the presence of steady state (non-fluctuating) noise. Sometimes the

DNR is considered as the complementary process to directional microphones. The directional microphones just reduces the background noise without knowing its temporal contents, hence are limited to reduce the noise from the side or behind the user, but not from the front which is the work of the DNR.

3.2.4. Speech Enhancement

The Directional Speech Enhancement (DSE) is currently introduced in the hearing aids. The DSE increase or improve the intensity of some particular speech segments. It identifies the speech segments based on their temporal or spectral contents and increase their relative intensity. This system is new to digital hearing aids and its effectiveness is still unknown.

3.2.5. Directionality

The directional microphones in the hearing aids improve the signal to noise ratio and its effectiveness is well established. Basically the directional microphones focus towards the direction of the sound coming from, and hence decrease the role of noise coming from background or other sides. Directional microphones combined with digital signal processing (DSP) further increase its benefits. Utilizing DSP, it calibrates the microphone, controls the shape of the signal, automatically adjust between the omnidirectional and directional mode and sometimes using the expansion technique, it reduces the noise generated by directionality.

Digital hearing aids are able to generate as well as process the sound. This capability is used to perform threshold tests and loudness growth, such that to obtain the fitting information of the individual's ear in combined with particular hearing aid. Today's digital hearing aids are much more flexible and certainly exciting and their future success possibilities are endless.

3.3. COMPARATIVE ANALYSIS OF DIGITAL HEARING AIDS

Today there are more than twenty-five digital hearing aids brands and over four hundred models available in the market. Some of the best brands in the hearing aid industry include the Phonek, Unitron, Siemens, Rexton, Sonic, Widex and Oticon etc. These manufactures introduces the hearing aid with different series names and each series contains a number of models differed by their features and power ranges.

Digital Hearing aids are differed by their size, style, power range and unique features. Generally the BTE hearing aids are for the patients with high level of hearing loss. As their batteries are large, so they can amplify the sound up to higher levels as compared to that of ITE or CIC devices which have smaller batteries due to their reduced area and hence can amplify the sound up to certain limits.

It is a very difficult process to choose a hearing aid from such a variety that suits best for a specific user. I.e. patients with the same degree of deafness may require a different kind of hearing aid device and it totally depends on different factors, like their priorities towards different styles, cosmetic appearance and requirement as per their professional life. Moreover, different hearing aids are focused to meet with different kinds of hearing losses, so a hearing aid may be helpful for one patient but it may not suit another patient. Generally smaller hearing aids like ITE and CIC are made for the patients with mild to moderate type of hearing loss.

One of the important considerations in the selection of a device is its price and it can vary up to thousands of dollars. The cost of a hearing aid totally depends on features in that particular device. There are also many cheap digital hearing aids available in the market, but they have limited features as compared with that of advanced devices and it may not provide the patient with benefits, he/she may need. So a patient has to choose the device based upon his requirements.

An Audiologist, after examining the hearing test of a patient, suggests the device based on the severity of hearing loss and other requirements of a specific user. But it may not be the final option for a user, because the hearing aids have a large of variety in their cost and features and patient has to trade off between the cost and his/her required features and needs.

3.3.1. Higher Price Range Advanced Hearing Aids

The digital hearing aid has been much improved with the advances in DSP technology. Today a lot many features are available in common hearing aids and these features are been improving day by day. The progress in digital hearing aid is so rapid that as some new feature develops then it becomes a standard next day. As the number of features along with number of channels increases in a particular device, then it also increases the cost of that device. Table 3.1 highlights over some of the most advanced models of hearing aids from different manufactures and their cost estimation.

Table 3.1: Higher price range digital hearing aids

Hearing Aids Brand Name	Model Name (Only BTE series)	No of channels (Max no in a series)	Listening Programs (max no)	Wireless Capability	Speech Enhancement and Noise Reduction	Feedback Cancellation	Water Resistant	TV-integration / Bluetooth	Price-Range Dollars ($)
PHONAK	Audeo Q	20	5	Yes-Adv	Yes	Yes	-	Yes	1,599-2,799
	Bolero Q	20	5	Yes-Adv	Yes	Yes	Yes	Yes	1,599-2,799
	Naida Q	20	5	Yes-Adv	Yes	Yes	Yes	-	1,599-2,799
	Nios-Micro	16	4	Yes	Yes	Yes	-	Yes	1,849-2,299
	Milo	04	4	No	Yes	Yes	-	-	1,599
UNITRON	Quantum 2 S	20	4	Yes	Yes	Yes	-	-	1,499-2,799
	Moxi2	20	4	Yes	Yes	Yes	-	-	1,499-2,799
	Moxi2 Kiss	20	3	Yes-Adv	Yes	Yes	-	-	1,499-2,799
	Quantum HP	20	4	Yes-Adv	Yes	Yes	-	Yes	1,599-2,799
	Max	20	4	Yes-Adv	Yes	Yes	-	Yes	1,499-2,699
REXTON	Finesse 2c	36	6	Yes	Yes	Yes	-	-	2,499
	Strata 2c	36	6	Yes	Yes	Yes	-	-	1,999-2,499
	Accord 2c	36	6	Yes	Yes	Yes	-	-	1,999-2,299

	Charismo 2c	36	6	Yes-Adv	Yes	Yes	-	Yes	1,999-2,499
	Onyx	12	8	Yes-Adv	Yes	Yes	-	Yes	1,599-1,999
	Day	06	6	Yes-Adv	Yes	Yes	-	-	1,499-1,599
SIEMENS	Sirion	12	4	No	Yes	Yes	-	-	1,599
	Ace	20	6	Yes-Adv	Yes	Yes	-	-	2,299-2,999
	Aquaris	16	5	Yes-Adv	Yes	Yes	Yes	Yes	1,899-2,999
	Pure	16	5	Yes	Yes	Yes	-	Yes	1,899-2,999
	Motion	16	5	Yes	Yes	Yes	-	Yes	1,899-2,999
	Pure Carat	16	5	Yes	Yes	Yes	-	Yes	1,949-2,899
	Nitro	16	5	No	Yes	Yes	Yes	Yes	1,899-2,999
	Life	16	5	Yes	Yes	Yes	-	Yes	1,899-2,999
SONIC	Bliss	10	5	Yes	Yes	Yes	-	Yes	2,199-2,699
	Charm	6	3	Yes	Yes	Yes	-	Yes	1,599-1,999
	Pep	6	4	No	Yes	Yes	-	Yes	1,499
	Flip	10	4	No	Yes	Yes	-	Yes	1,799-2,699
	Endura	12	4	Yes	Yes	Yes	-	-	1,999-2,499
WIDEX	Dream	15	4	Yes-Adv	Yes	Yes	-	-	1,499-2,999
	Clear	15	4	Yes	Yes	Yes	-	Yes	1,899-2,799
	Mind	15	4	Yes	Yes	Yes	-	-	1,899-2,799
	Passion	15	4	Yes	Yes	Yes	-	-	1,999-2,999
	Flash	5	4	No	Yes	Yes	-	-	1,699

OTICON	Alta Pro	16	4	Yes	Yes	Yes	-	-	2,699-2,999
	Nera , Nera Pro	16	3	Yes	Yes	Yes	Yes	Yes	1,899-2,299
	Agil , Agil Pro	10	4	Yes	Yes	Yes	-	Yes	2,999-2,999
	Intiga	10	5	Yes-Adv	Yes	Yes	-	Yes	1,799-2,999
	Acto , Acto Pro	09	4	No	Yes	Yes	-	Yes	1,799-2,399
	Chili	09	4	Yes	Yes	Yes	-	Yes	1,999-2,999
	Ino , Ino Pro	06	1	No	Yes	Yes	-	Yes	1,849-1,949

Hearing aid manufacturers introduce their models in a series and each series contain a number of models differed by their features, power ratings and style, etc. In this comparative study, top models from different brands are chosen and analyzed based upon their features and approximate cost.

As shown in the table 3.1, the Bolera Q and NaidaQ models are the most advance featured of all the Phonak series of hearing aids and they are featured with the latest water resistant technology. Averagely, both these models have greater number of channels and listening programs from the rest of models and have advanced wireless technology and both have almost same price range. But the main difference in both of them is that of Bluetooth, so the Bolera Q is considered as a good model series of Phonak. In the same way, Quantum HP of Unitron and Onyx of Rexton are also good options.

Siemens is considered as one of the best brand in the hearing aids industry and all its model series are good. Its models are easily available in Pakistan as compared to some other brands. Among these models, “Ace” is best of the all others, with almost twenty channels and about six listening programs, but its prices are much higher as compared to others. The second best option in the Siemens models is Aquarius and Nitro. Both of them have sixteen frequency channels and up to six listening programs and another good option is their water resistant technology. Price ranges of both models are same, but the Nitro series lacks the wireless feature. So the best option in the Siemens devices is Aquarius.

The Bliss model of sonic is well featured and is comparable to Unitron and Rexton models. But its prices are too high to be considered. In the same way, Dream and Clear models of Widex are also considerable, but the models of Dream series lack the Bluetooth feature and although Clear models have Bluetooth but their cost is high enough to be considered.

The prices of Oticon series are much higher than those of other brands. The only considerable series of Oticon models is Nera and Nera pro, which have almost sixteen frequency channels but have only three listening programs. The good thing about these models is that they have the water resistant technology.

If we narrow the selection criteria of a device to having minimum ten frequency channels, five listening programs, having wireless and Bluetooth technology, optionally water resistant and having minimum price, then following models are considerable

- Bolero Q Phonak
- Quantum HP Unitron
- Onyx Rexton
- Aquaris Siemens
- Pure Siemens

Keeping in view the average Pakistan patients seeking help for hearing aids, then the cost of these devices are high enough, starting from fifteen hundred to eighteen hundred dollars, but they may be quite satisfactory for some patients, if they can afford it. The most suitable devices for the local users are Aquaris and Pure, because Siemens has a large network of hearing clinics in Pakistan where these devices are easily available. They provide the free programming facility for the second time and maintenance of the device.

3.3.2. Lower To Mid Price Range Digital Hearing Aids

The hearing technology is so much improved that most of digital hearing aids have some basic features as standard; like feedback suppression, noise reduction algorithms and direct audio input and T-coil, etc. In contrast to the previous section, this section is focused to look over some of the low to mid range digital hearing aids from various manufacturers.

In this comparative study, models of those manufacturers are chosen which are easily available in Pakistan. These manufacturers include the Siemens, Widex, Unitron, Phonak and Resound etc. all of these low to mid range selected devices have some standard features. Table 3.2 overlooks these devices, their price range, power range and programmable memories.

Table 3.2: Low to mid price range digital hearing aids

Brand Name	Series & Model Name		Power range (dB)	No of channels	No of programs	Price ($)
Widex	Bravo	Bravo B1	20 - 80	2	3	199
		Bravo B11	40 - 105	2	3	275
		Bravo B2	20 - 80	2	3	255
		Bravo B12	40 - 105	2	3	299
	Bravissimo	Bravissimo BV-8	20 - 80	3	3	395
		Bravissimo BV-18	40 - 100	3	3	450
		Bravissimo BV-38	50 - 120	3	3	550
	Real	Real RE-9	10 - 90	3	3	589
		Real RE-19	40 - 105	3	3	625
		Real RE-Micro	10 - 85	3	3	645
	Flash	Flash FL-9	0 - 90	5	4	785
		Flash FL-19	40 - 100	5	4	785
		Flash FL-Micro	0 - 80	5	4	785
Phonak	Argosy	Argus	10 - 105	4	2	230
		Argus D	10 – 105	4	2	260
		Argus P	50 – 110	4	2	230
		Argus DP	50 – 120	4	2	260
	Milo	Milo Micro	0 - 90	2	3	300
		Milo SP	40 - 120	2	3	300
		Milo UP	60 - 120	2	3	300
	Una	Una M	10 - 105	4	4	360
		Una M AZ	10 - 105	4	4	400
		Una SP	50 - 120	4	4	360
		Una SP AZ	50 - 120	4	4	400

	Milo Plus	Milo Plus Micro	0 - 90	4	4	500
		Milo Plus SP	40 - 120	4	4	500
		Milo Plus UP	60 - 120	4	4	500
	Certena Art	Certena Art M	10 - 105	6	6	700
		Certena Art P	40 - 120	6	6	700
		Certena Art SP	60 - 120	6	6	700
		Certena Art Micro	0 - 100	6	6	700
Bernafon	Win	Win 102	10 - 85	3	2	265
		Win 112	40 - 110	3	2	350
	Neo	Neo 102	20 - 85	5	2	490
		Neo 112	40 - 110	5	3	585
	Xtreme (High Power)	Xtreme XT 121	50 - 120	5	2	699
		Xtreme XT 120	50 - 120	5	3	1080
	Avanti	Avanti AV-106	0 - 85	7	3	700
	Inizia 1 / 3	Inizia IN 1N	10 - 80	-	3	499
		Inizia IN 3N	10 - 80	-	3	699
Siemens	Infinity Pro	Infinity Pro S	20 - 80	1	3	199
		Infinity Pro P	40 - 110	1	3	210
		Infinity Pro SP	60 - 120	1	3	235
	Lotus Pro	Lotus Pro M (twin Mic)	20 - 80	2	3	250
		Lotus Pro P	40 - 110	2	3	260
		Lotus Pro SP	60 - 120	2	3	280

Resound	Match	Match 2T 70V	20 - 90	2	2	130
		Match 2T 80V	20 - 110	2	2	165
	Essence Lite	Essence Lite 70 VI	20 - 95	4	3	200
		Essence Lite 80 VI	25 - 115	4	3	240
	Essence	Essence 70 DVI	20 - 95	6	3	360
		Essence 80 DVI	25 - 115	6	3	425
	Dot	Dot 10 RIE	20 - 100	9	3	400
Unitron	Shine	Shine	0 - 100	2	3	270
		Shine HP	0 - 100	2	3	290
	Shine+	Shine+	0 - 100	4	4	390
		Shine+ HP	0 - 110	4	4	420
	Quantum 6	Quantum 6 P	0 - 90	6	6	650
		Quantum 6 HP	15 - 110	6	6	690

Power range of a device shows the hearing aid's usage for a specific type of hearing impairment. Low power devices are for mild to moderate hearing loss while that of higher power are for severe and profound type of impairment. The increasing number of channels shows the device efficiency in signal processing and the number of programs represents its ability to handle different listening environments.

Different criteria can be applied to choose the device that better suits a specific user. We will discuss those devices which have at least two frequency channels, three program memories and their price is below three hundred dollars. If we apply these criteria, then the following models can be selected for further discussions

- Bravo B1, B2 series (Widex)
- Milo series (Phonak)
- Lotus pro series (Siemens)
- Essence Lite series (Resound)
- Shine series (Unitron)

Again, if we comparatively look over them, then the Siemens Lotus pro is the best option for the local patients because it is packed with the latest features among all of these devices. Some of the important features of Lotus pro are two channels with three programmable memories, directional microphone, antiphase feedback suppression, push button, T-coil and DAI etc.

This comparison is done based on the some standard features and cost of the device. If we increase the cost limit, then certainly some other good devices are also available. The final device for a particular patient is selected based on his/her audiogram tests. One he/she knows about the severity of hearing loss then he/she can take help from this table and can choose the best device keeping in view his/her budget.

CHAPTER 04

MICROCONTROLLER BASED CIRCUIT DESIGN

4.1. INTRODUCTION

Hearing aids can be classified into two types, the analog and digital hearing aids. The analog hearing aid processes the signal in the analog domain while the digital hearing aid first converts the signal into digital domain and then processes it. The conventional analog aids amplify all the frequencies equally, hence along with the desired audio signal, it also amplifies the noise. It cannot differentiate between the signal and background noise. Moreover, it cannot be programmed during the fittingprocess. Some analog aids have different listening profiles, which the user can select using a button on the hearing aid [70]. The advancement in the digital technology and introduction of Digital Signal Processing (DSP) to the hearing aids has much improved the hearing aid [71]. DSP offers many advantages over the analog hearing aids, including the programmability, self monitoring, acoustic feedback control, signal level control and adaptive adjustment etc. [72]. The Digital hearing aids first convert the signal into digital domain and then process it. Hence utilizing the software and proper configuration of the system, speech signal can be much improved even in the presence of noise.

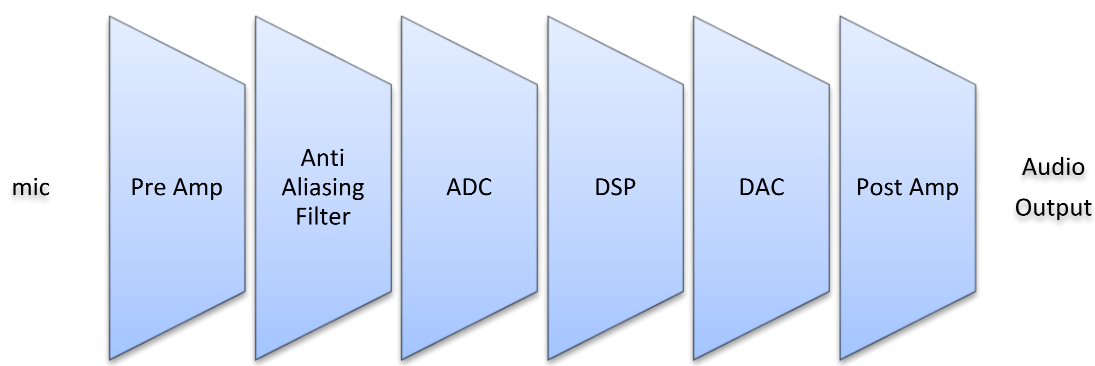

Figure 4.1: Block diagram of a digital hearing aid

As shown in the figure 4.1, a digital hearing aid contains a mic which captures the audio signal and converts it into an electrical signal. The Preamplifier circuit amplifies the audio signal such that it matches the input signal level of ADC. An anti-aliasing filter is necessary in order to avoid the aliasing noise in the output signal. The DSP section processes the digital signal accordingly to the user requirement and DAC convert the signal back into analog domain which is received by the user through earphones.

In the coming sections, we will explore the design of microcontroller based cost effective and flexible digital hearing aid design which has a low-power consumption and reduced circuitry. Utilizing the internal clock of the microcontroller and built-in ADC avoids the extra circuitry and hence reduce area. The 8 bit external DAC helps in suppressing the background noise which is very low in intensity. Hence, using the readily available components and circuitry, the system is simple, portable and cost effective.

4.2. DESIGN CONSIDERATION WITH HEARING AID

The aims of audiologists are to make the hearing aid most beneficial and satisfactory and in order to achieve these goals, following points are necessary [73];

.

- Improved audibility
- Make the average and soft sound more comfortable
- Make the speech intelligible in noise
- Keep loudest sound in comfortable limits

Most of the hearing aid users have sensorineural hearing loss in which the patient is unable to listen to low intensity sounds. Moreover, different patients have hearing impairment problems at different frequencies. In order to tackle these problems, recently the Wide Dynamic Range Compression (WDRC) feature has been introduced into the hearing aids. In WDRC feature, the gain of the different frequencies is set by the audiologist during the programming and it is based on the tests of the patient taken by audiologists. The test clearly shows the frequency regions where the patient needs more amplification and keeping in view the results of a particular patient, the audiologist programmed the device accordingly to the patient requirements.

4.3. CIRCUIT DESIGN

The main components of the designed system are sound detection circuitry, a microcontroller and DAC. The overall circuit of the hearing aid is designed such that the analog signal from the microphone output is converted to digital signal before the processing of a signal. To achieve this conversion, the built-in ADC of a microcontroller is used and analog signal is amplified in order to match the input signal level of the microcontroller. The anti-aliasing filter is used to avoid the aliasing effects of analog to digital conversion. Microcontroller processes the signal and after desired processing, the digital signal at the output of a microcontroller is converted back into an analog signal in order to be picked up by earphone. For this purpose, Digital to Analog converter (DAC) is used. The analog signal from the DAC output is then fed to the earphone. Figure 4.2 and 4.3 shows the software and hardware design of the system respectively.

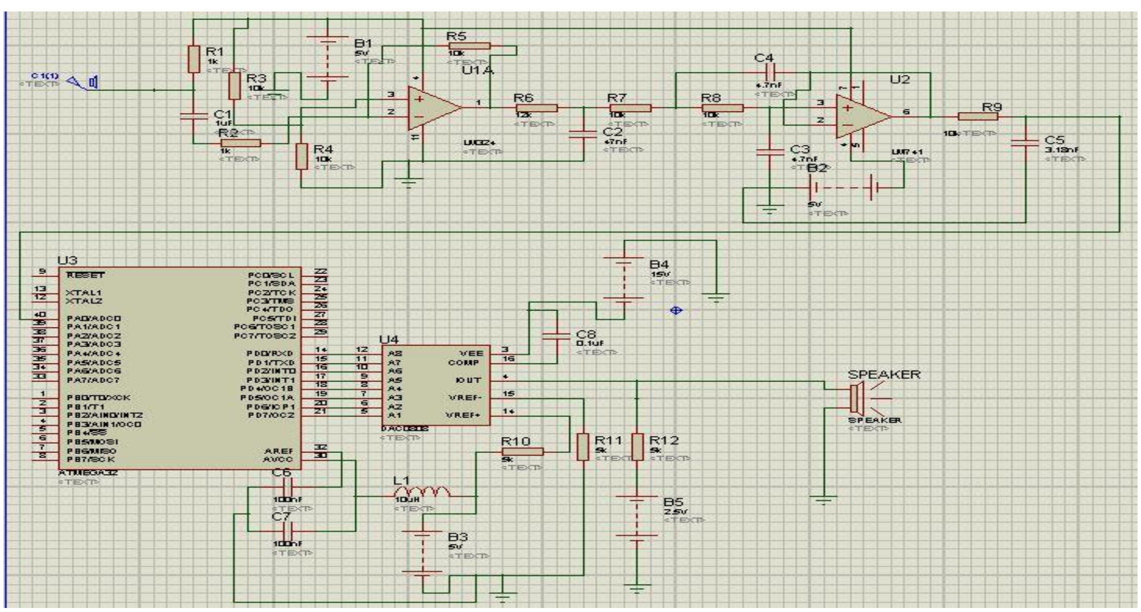

Figure 4.2: Software design of the system. Different components are combined together and their software simulations are performed.

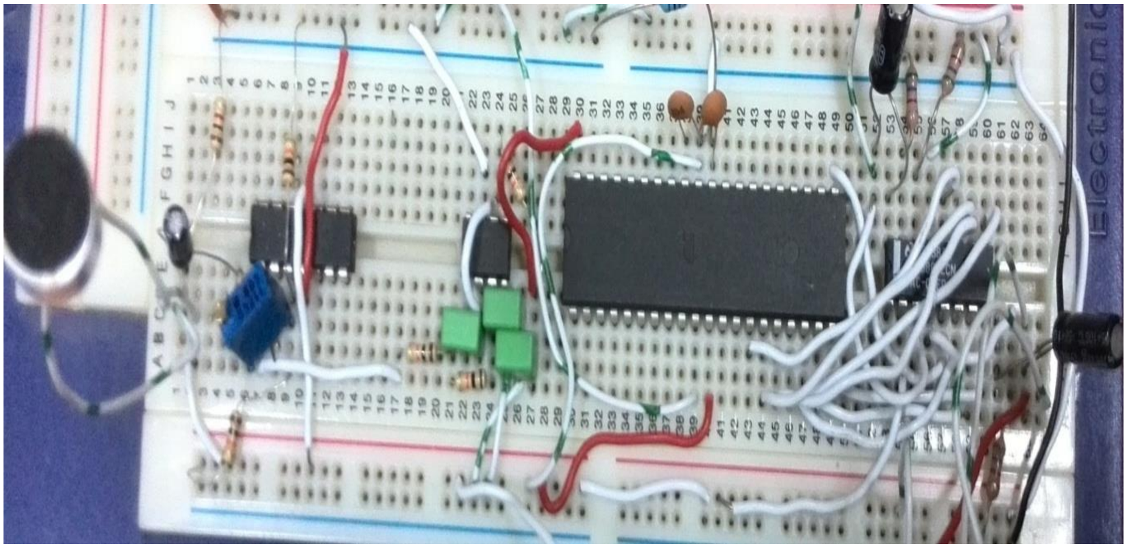

Figure 4.3: Hardware design of the system. Individual circuits are tested on breadboard and then combined together as one unit.

Some of the components used in the design are as fallows

- Electret mic
- Opamp (lm324, lm741)
- Built-in ADC of AVR ATmega32 microcontroller
- Microcontroller ATmega32
- Digital to Analog converter (DAC0808)
- Resisters
- Capacitors
- Inductors
- Earphone

Along with the major components, the individual blocks and circuits are specially designed and focused to work accordingly to the requirements. The sub-circuits and components used in the system are discussed as below;

4.3.1. Sound Detection Circuit

The sound detection circuitry contains the two main components, Microphone and Preamplifier circuit.

4.3.1.1. Electrets microphone

The microphone can be broadly called a transducer as it converts the sound energy into electrical energy. The electret mic is like a parallel plate capacitor. One of its plates is stable and is called a back plate while the other plate is movable and is a diaphragm. The external voltage is supplied to it, which keeps the stable charge over the parallel plates. Normally it has a stable charge, but when the sound signal strikes the diaphragm then it vibrates and an alternating current is produced [74] which is basically the sound signal in electrical form. As shown in the figure 3.4, a capacitor at the output acts like a filter for Alternating Current (AC) to pass and blocks Direct Current (DC).

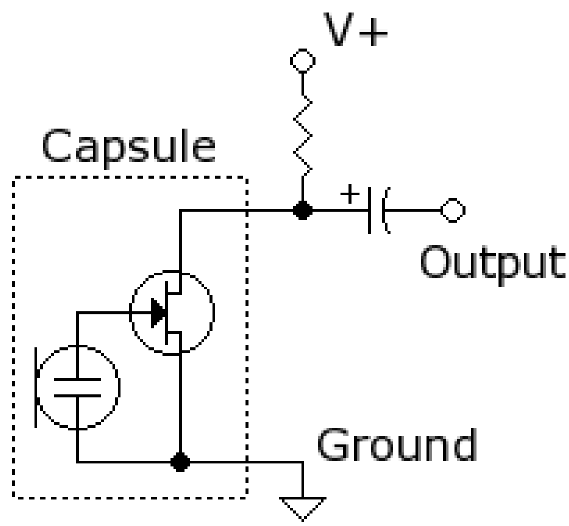

Figure 4.4: Basic circuit of electret microphone

The output of electret microphone is in millivolts and Frequency response of electret microphone, which is its sensitivity performance, is measured over the frequency range of 0-20k Hz and its sensitivity is the measure of output voltage when the sound signal strikes the microphone. It is measured in units of volts/pascal (V/Pa) and normal electrets mics have sensitivities range from 5mV/pa to 17.8 mV/pa.

The mic circuit in figure 4.3 draws a total current of0.21 mA, hence this circuit has a total power dissipation of 1.05 mW.

4.3.1.2. Pre-amplifier circuit

The purpose of preamplifier circuit is to amplify the signal such that to level it to the input signal requirement of analog to digital converter, which is from 0-5 volt. To achieve this level, an operational amplifier LM 324 is used.The Electrical signal from the mic is fed to the inverting input of the opamp, while the non-inverting input is kept at stable 2.5 V from the battery. This combination of inputs provides the DC offset of 2.5 V at the output and now the 180 degree phase shifted audio signal oscillate across offset value.

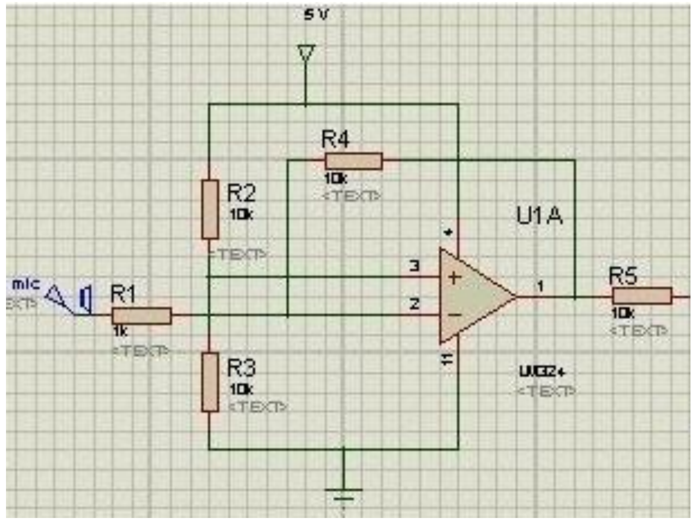

Figure 4.5: Preamplifier circuit diagram

The gain of the preamplifier circuit is as according to the formula

$$V_{out} = - R_2 / R_1 (V_{in}) \quad (1)$$

Where R_2 is the feedback potentiometer and its value can be set maximum up to 100k Ohm. The purpose of this potentiometer is to adjust the gain accordingly to the user requirement.

The preamplifier circuit in figure 4.3 draws a total current of 0.51 mA, hence this circuit has a total power dissipation of 2.55 mW.

4.3.2. Anti-Aliasing Filter

Thepurpose of anti aliasing filter is to avoid the aliasing noise at the output of analog to digital converter (ADC). According to Nyquist-Shanon sampling theorum, the sampling frequency of ADC should be atleast two times the highest frequency componant in a signal. Mathematically;

$$Fs(min) = 2*f_{max} \quad (2)$$

If the sampling frequency is less then the double of max frequency component in the signal, then the higher frequencies will fold back over the lower frequencies and it will create aliasing effect. Normally the human voice frequency reange is between 300 to 3000 Hz. As in the proposed system, we are only intrusted in capturing the human voice only, so here the filter has two main purposes

- avoid aliasing effect
- Pass human voice frequencies

So to achive these goals, a second order R.P Sallen and E.L Key low pass filter is used [75].

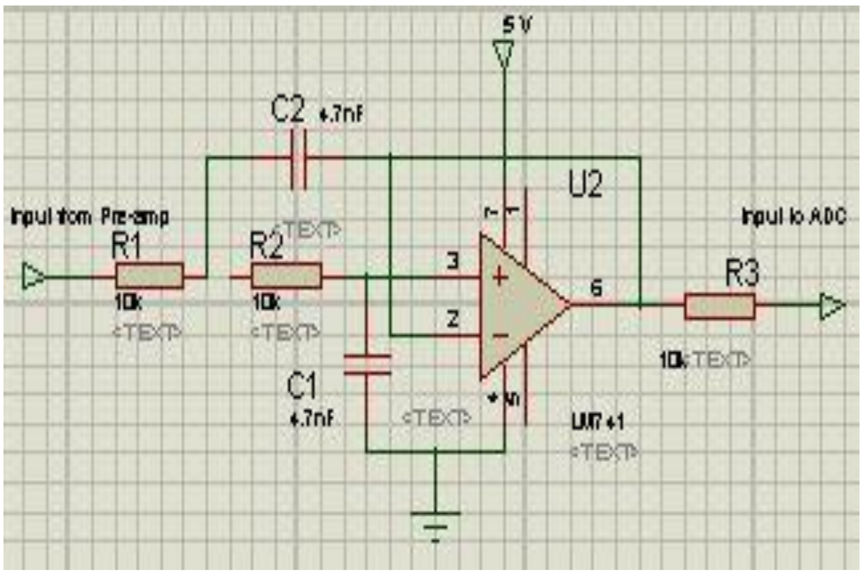

Figure 4.6: Second-order anti-aliasing filter diagram

Some of the formulas for parameters like cutoff frequency (w_0) and quality factor (Q) are as below;

$$w_0 = 1 / R_1R_2C_1C_2 \qquad (3)$$

And

$$f_{cutoff} = w_0 / 2.pi \qquad (4)$$

For quality factor

$$Q = (R_1R_2C_1C_2)^{1/2} / C_1(R_1+R_2) - (K-1) R_1C_2 \qquad (5)$$

Where

$w_0 = 2*pi*f_{cuttoff}$

R_1 = resistance of resister 1

R_2 = resistance of resistor 2

C_1= capacitance of capacitor 1

C_2=capacitance of capacitor 2

Q = quality factor

K = gain of filter

In this design, the gain of the filter is kept one and quality factor is kept ½ because this value of Q gives good results in rise time and settling time.

Hence putting $R_1 = R_2 = 10k$, $K = 1$ and $Q = 0.5$ in equation (3) and (5) and solving for C_1 and C_2 , we get

$$C_1 = C_2 = 1/ R.w_0 \qquad (6)$$

If putting cutoff frequency f_{cutoff}= 3.5k Hz, we get

$$C_1 = C_2 = 4.7 \text{ nF}$$

This value of filter's cutoff frequency gives a good approximation of capturing human voice frequency along with setting the maximum frequency limit for ADC.

The anti-aliasing circuit in figure 4.3 draws a total current of 0.29 mA, hence this circuit has a total power dissipation of 1.45 mW.

4.3.3. Microcontroller (Atmega 32)

The atmega 32 is specifically chosen for this work because of its several advanced levels features like analog to digital conversion which makes it ideal in applications like automotives, medical and consumer applications. It is readily available cost effective microcontroller which has a built in 10 bit ADC and has 8 bit architecture. The large programming memory, extra ports and analog channels make it flexible here in this application because it can be used for adding extra features in hearing aids like direct audio input and directionality etc. some of the features of Atmega 32 are as follows [76]

Table 4.1: Features of microcontroller Atmega 32

CPU	○ Max 16 MIPS throughput ○ RISC architecture
PIN count	Total 40 pins ○ Two power pins: pin.10 = +5v, pin.11= ground ○ Two oscillator pins, 1 reset pins ○ 3 for ADC reference and power ○ 32 (4*8) I/O pins
I/O pins	32 (4 ports *8 pins) can be used as i/o pins. ○ Port A can be used as analog input. i.e ADC ○ Port B,C,D all can be set as i/o pins or ports
Timers	Total three built-in timers/counters ○ Two 8 bit timers/counter timer0, timer2 ○ One 16 bit timer/counter timer1
ADC	Total 8 single channels ○ Reference voltage cab be selected either as internal 2.56 v or external through AREF pin. Detail is in next section
Communication Options	Total three data transfer modules ○ Two wire interface ○ USART

	○ Serial peripheral interface (SPI)
Analog comparator	On-chip analog comparator to which an interrupt is assigned
External Interrupt	Total three external interrupt acceptable and configurable
Memory	Memory is discussed in separate table
Clock	Clock frequency range 1-16 MHz ○ External quartz crystal ○ Internal RC oscillator
Debug	On-chip debug through JTAG boundary scan
Programming	○ Possible through in-system via SPI or parallel programming ○ Via JTAG

Table 4.2: Memory of the chip

Code ROM	32 k bytes
Data RAM	02 k bytes
Data EEPROM	01k bytes

4.3.3.1. Analog to digital converter (ADC)

In this work, the built-in analog to digital converter of omega 32 is used which has the following specifications

- 10 bit ADC
- 8 analog and 7 differential input channels
- Two differential channels have gains with option
- Two SFR for 10 bit binary data. ADCL (for lower byte) and ADCH (for upper byte)

- As total bits are 10 while SFR has 16 bits.so it is optional to ignore either upper or lower 6 bits
- Three options for reference voltage or internal 2.56 V, AVCC (same as Vcc) or external source
- Conversion time depends on clock frequency which can be from 1-16 MHz.

4.3.3.2. Digital to analog converter (DAC)

The external DAC 0808 is used in this work which is 8 bit and uses R/2R method for analog output generation. Basically it gives negative current at the output whose range is between 0-2 mA and using a resister at the output, it is converted in to a voltage output.

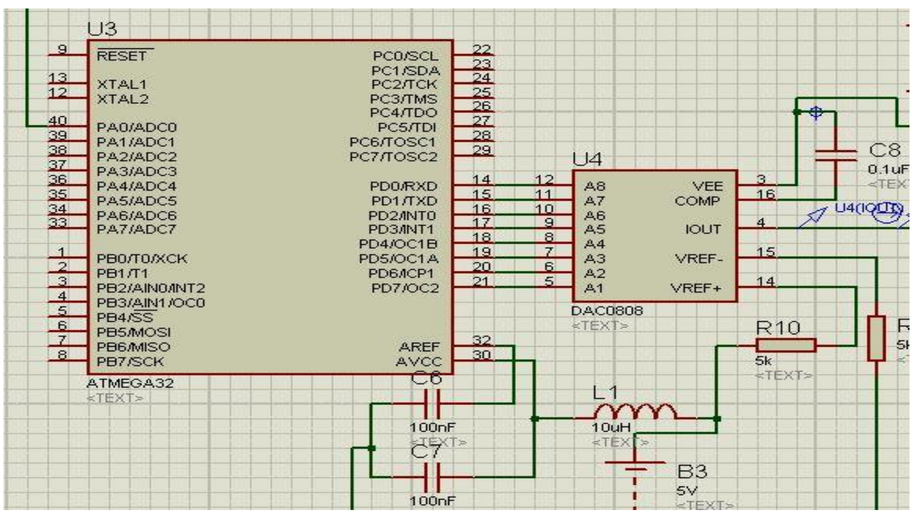

Figure 4.7: DAC interface with Atmega 32

The above figure shows the interfacing of DAC with microcontroller. The internal ADC is of 10 bit but here an 8 bit DAC is used. The lower two bits are ignored as to suppress the low intensity background noise.The dac circuit in figure 4.3 draws a total current of 9.5 mA, hence this circuit has a total power dissipation of 47 mW.

4.4. WORKING OF THE SYSTEM (SUMMARY)

The electret microphone pickup the audio signal and convert it into the analog electrical signal whose magnitude is in millivolts. The signal level at this stage needs to be amplified and for this purpose, it is fed to the pre-amplifier circuit which levels the intensity or amplitude of a signal such that it is matched to the input level of ADC. The potentiometer used in the feedback path of a preamplifier circuit gives control over the amplitude of a signal at the output of preamplifier circuit. Before the ADC conversion, the signal is passed through the Anti-aliasing filter which is specially designed accordingly to the human voice frequency range and user requirements. This filter minimizes the aliasing effects in the reconstructed signal and passes the voice frequencies without any attenuation and fed it into the 10-bit ADC of a microcontroller, which convert it into a digital signal. Microcontroller process the digital signal and after required processing, the signal is converted back to analog signal using an external 8-bit DAC. The two least significant bits of ADC are ignored at this stage, which helps to suppress or reduce the amplitude of low intensity background noise. The audio signal at the output of the DAC is fed to the earphone whose one end is plugged into the jack on the circuit and the other end is in the ear of the user.each sub-circuit of the system like preamplifier, ant aliasing filter. ADC and DAC woks individually and contribute to the overall system.

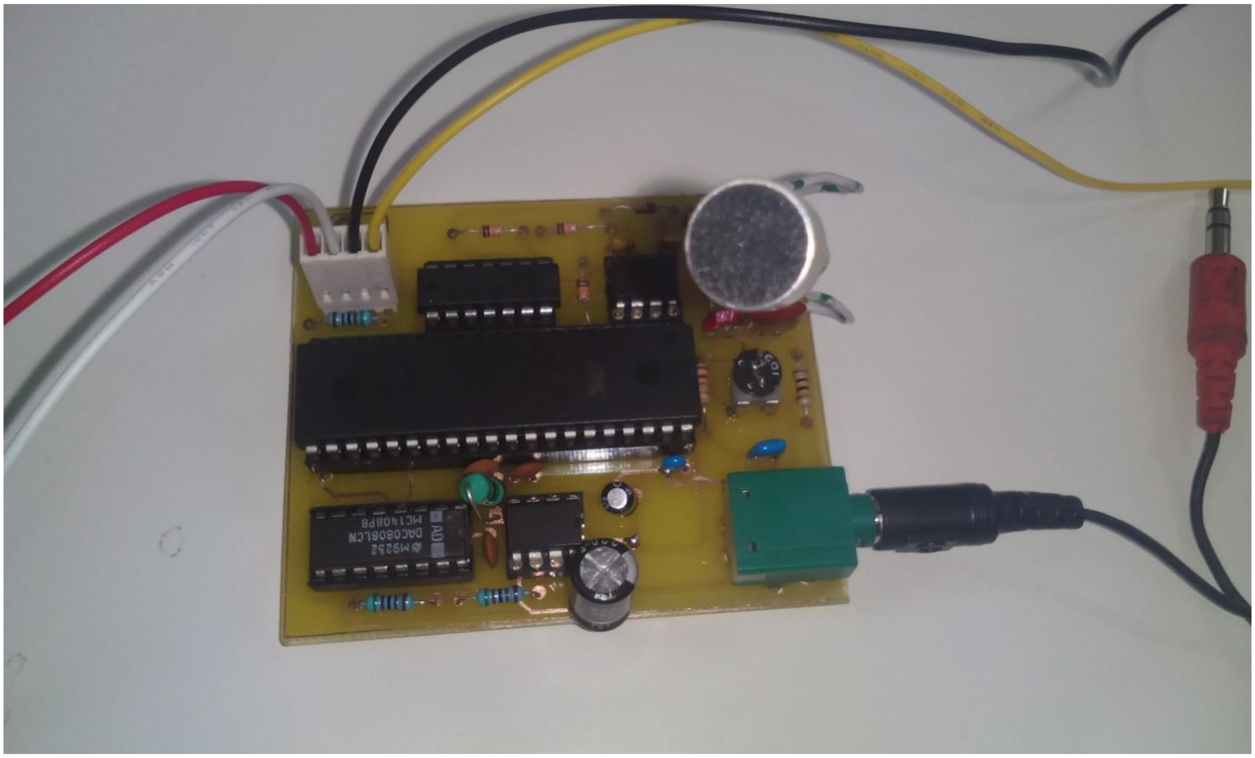

Figure 4.8:PCB implementation of the system. After final tastings,
the device is implemented on PCB

Some of the specifications of this device are

- Frequency bandwidth is up to 4000 Hz
- Total current of individual circuits sums up to 40-50 mA
- Power consumption is approximately 200 mW
- Maximum sound gain is 80 dB

CHAPTER 05

RESULTS

In this research work, a cost effective and portable digital hearing aid is designed and successfully implemented on PCB. The minimum gain of the device is between 30-40 dB and is adjustable to a maximum of 80 dB. The system showed acceptable results especially for low frequency signals including the human voice frequency range.

The sub-circuits of this microcontroller based design, including the sound detection circuitry, anti-aliasing filter, microcontroller and digital to analog converter (DAC) are individually tested and then are combined to contribute to overall system.

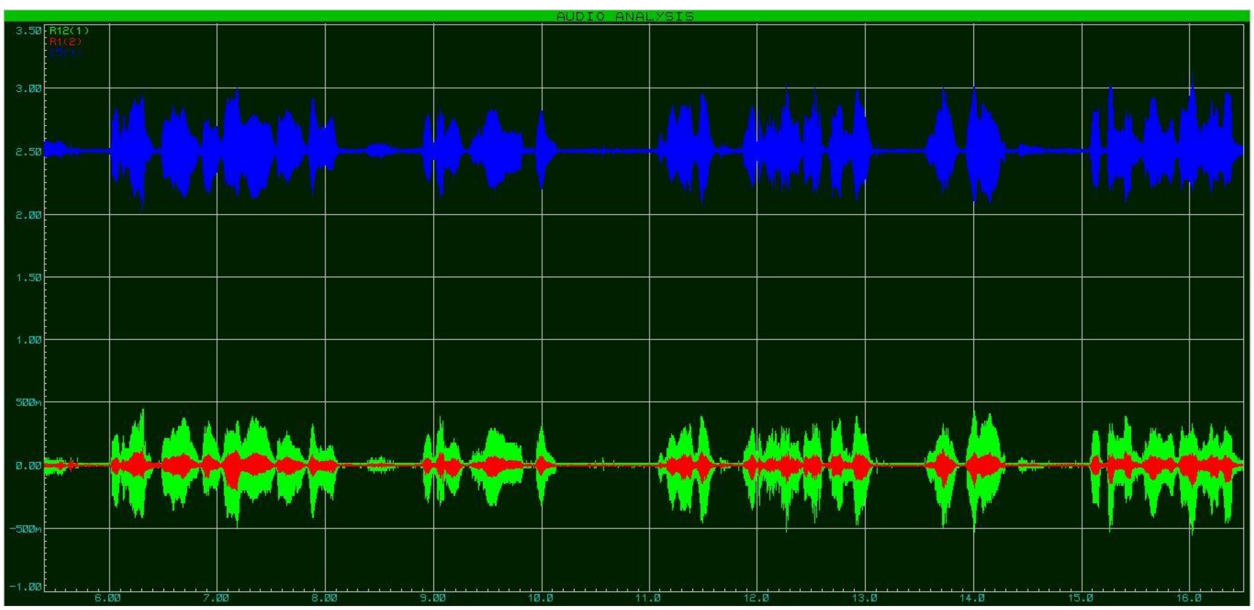

Figure 5.1: Audio processing of the system. The input audio signal is amplified by preamplifier and has given an offset of 2.5 V in order to match it with the input signal level of microcontroller and after the desired processing, the offset is removed and audio signal is bring backed to ground level.

Figure 5.1 shows the signal flow of the system. Red color is the input signal in millivolts which is inverted and amplified by the pre-amplifier with a dc offset of 2.5V. This signal is filtered by anti-aliasing filter and is passed through ADC (shown with blue color). After DAC, dc offset is removed and the signal is inverted back. Hence the processed signal with desired gain is fed to the ear via headphone.

For the test purpose, an audio signal of 50-8000 Hz is used. The achieved results are discussed as under.

5.1. FILTER RESPONSE

For the frequency filtering purpose, 2^{nd} order R.P sallen and E.L key low pass filter is used. Normally the human voice frequency range is up to 3k Hz, so the filter's cutoff frequency is kept at 3.5 k Hz. The gain of the filter is 1 and the quality factor is kept at 0.5 which gives a good compromise between the rise and settling time of a signal.

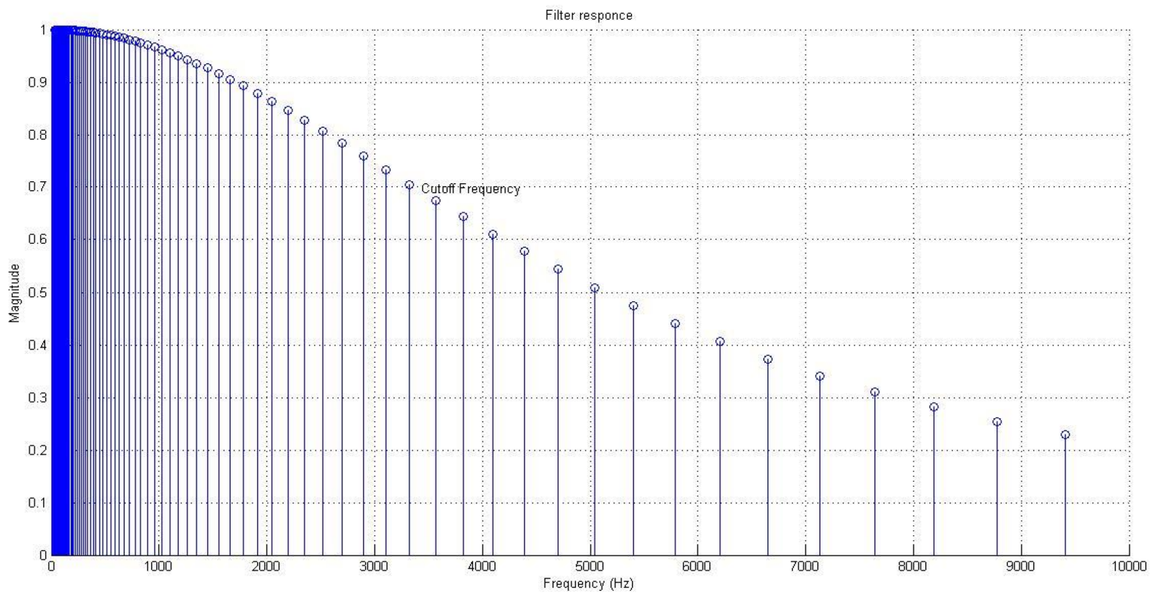

Figure 5.2(a): Frequency response 1. It is the matlab generated graph of the filter frequency response over the desired frequency range and the cutoff frequency is labeled.

Figure 5.2(a) shows a matlab generated graph of a filter's frequency response and it is clear from the above figure that the desired frequencies are processed without any attenuation while the higher frequencies are very much attenuated.

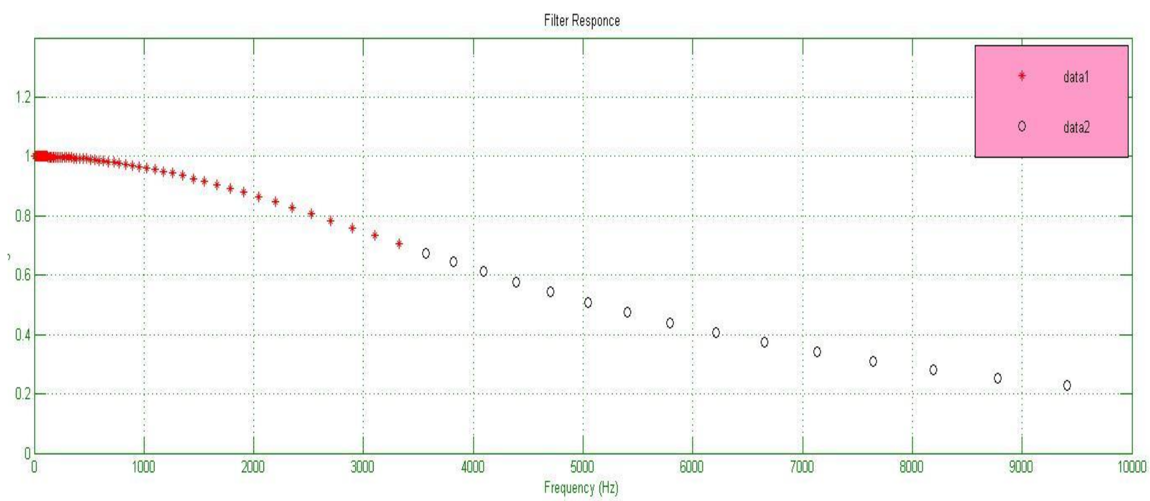

Figure 5.2(b): Frequency response 2. Data 1 (red stars) shows the required frequencies that are passed with desired gain while data 2 (black circles) shows the undesired frequencies which are comparatively attenuated.

Figure 5.2(b) shows another angle of processed frequencies in which the passing frequencies are labeled with red stars while the unwanted frequencies are labeled with circles and their magnitudes are attenuated by filter.

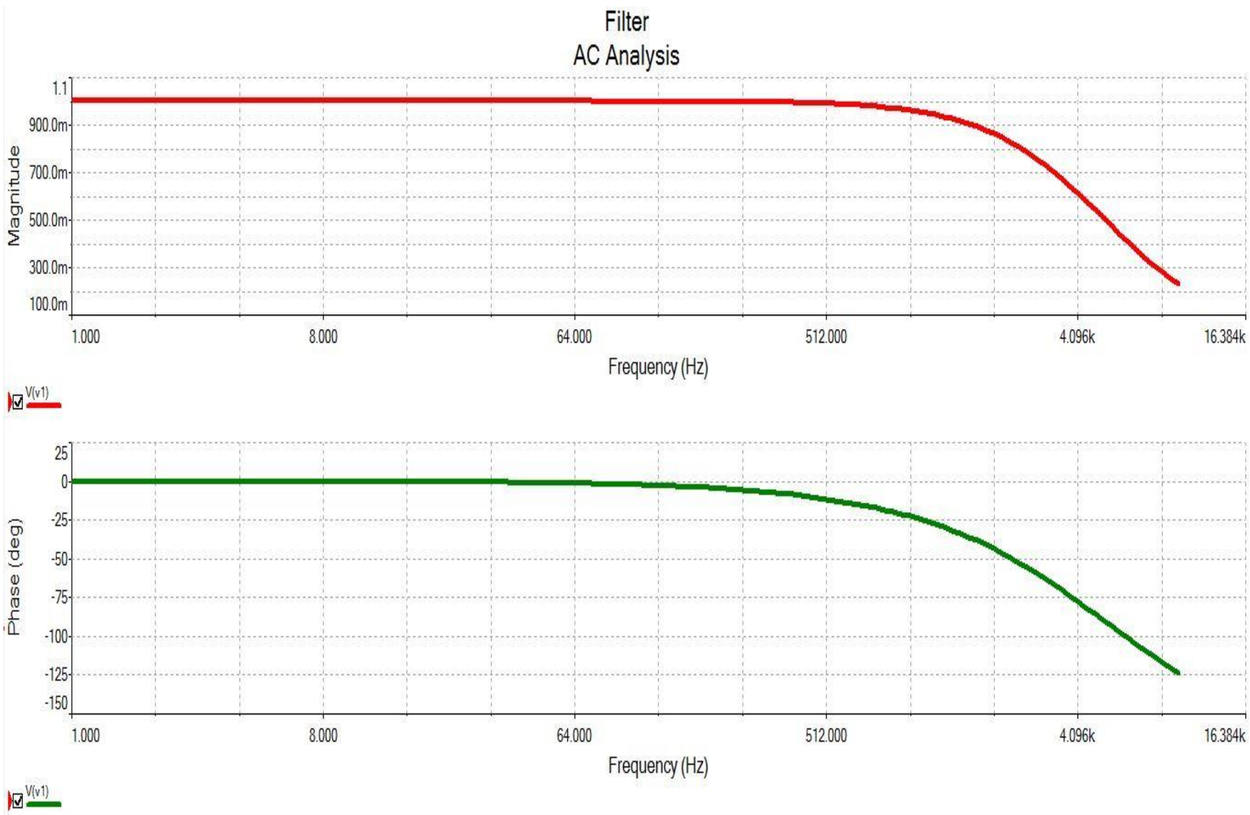

Figure 5.3: Phase response of a filter. It shows both the phase and frequency response of the filter generated in labview,

Figure 5.3 is a graph of frequency vs. phase and magnitude and it is clear from the figure that the phase of passing frequencies is approximately remain the same as of the original signal and at cutoff frequency, it experience a phase difference of almost 25^0 which is acceptable.

5.2. SOUND AMPLIFICATION ANALYSIS

The minimum gain of the system is set by preamplifier and is adjustable and smooth over the desired frequency range. The preamplifier sets the minimum gain of 30-40 dB and after signal processing; the post-amplifier can adjust it to a maximum of 80 dB. The fix gain of the pre-amplifier is shown below;

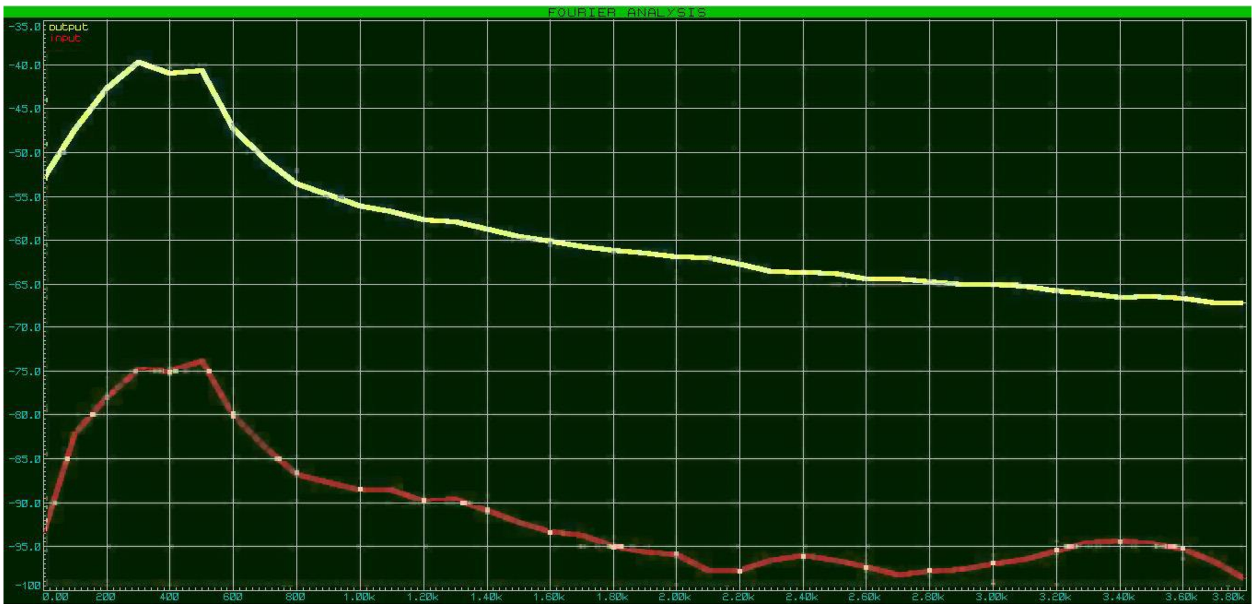

Figure 5.4: Gain of the system over desired frequency range. It shows the fix gain of the preamplifier which is between the 30-40 dB over the required frequency range while the overall gain is adjustable up to 40 dB.

Figure 5.4 shows the magnitude (in dB) of the signal over the desired frequency range. A test audio signal was given as input to the system (shown red here) and the output (shown yellow) has almost a smooth gain over the desired frequency range but as the frequencies get higher, they are attenuated and the gain reduces. After this fix gain, the post amplifier (lm 386) can increase it further and can adjust it up to a maximum of 80 dB.

Hence this kind of hearing aid is effective for the patients having mild up to moderately severe hearing loss.

5.3. TEST SIGNAL ANALYSIS

A test signal of frequency range 50-8000 Hz was given as input to the system. The signal is such that the frequency first varies from 50 to 8k Hz and then decrease back to 50 Hz and it is repeated two times. The response of the system is shown as below;

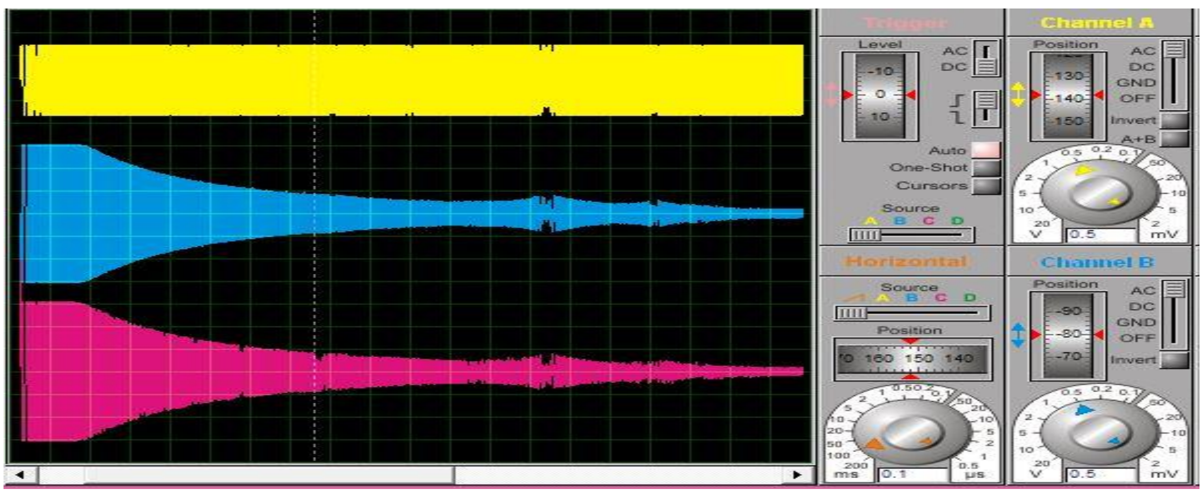

Figure 5.5(a): Audio test signal caption 1 (when the input frequency increases from 50-8000 Hz, the gain decreases and attenuation increases as the frequency increase).

Figure 5.5(b): Audio test signal caption 2(when the frequency first decrease and then increases, the output gain increases as frequency gets lower and attenuation increases when the frequencies gets higher).

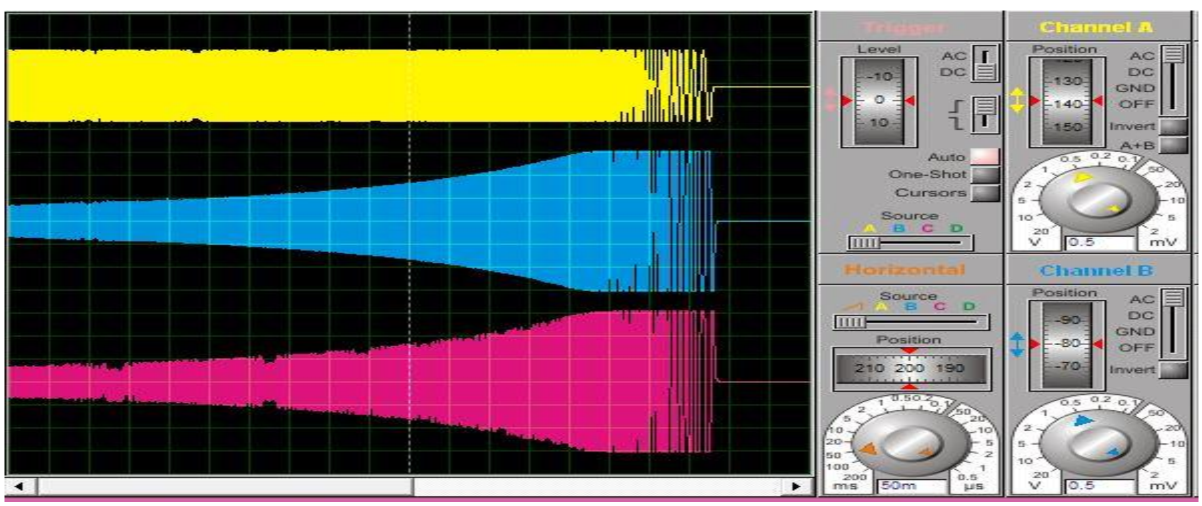

Figure 5.5(c): Audio test signal caption 3(in the end, when frequency moves from higher towards lower end, the output gain increases and the attenuation increases).

In above figures, the yellow one is the input test signal, the blue one is the filter output while the pink signal is the system output.

Figure 5.5(a) shows the response when the frequency increases from 50 up to 8000 Hz. In the start; when the frequency is below cutoff, all the frequencies are passed with desired gain. But as the frequencies get higher, their magnitude becomes approximately neglected. Figure 5.5(b) shows the response when the frequency moves toward the lowest point and then increase gradually. At this time, the magnitude of the output signal becomes maximum when the frequencies gets lower and decreases gradually as the frequency increases. In the last figure 5.5(c), as the input frequency again decreases towards the lower points, the magnitude of the output signal increases simultaneously.

CHAPTER 06

DISCUSSION

The main objectives of the thesis were to assist the patients in choosing better hearing aids available in the market and to design a cost effective hearing aid device.

After the study of more than hundred digital hearing aids available in the market, the devices were categorize in to two categories, the higher price range and lower to mid price range. As the price of device gets higher, the quality along with the luxury increases but the minimum requirements of a low budget patient can be fulfilled with some low price devices as listed in table 3.2. Once if the user decides about his maximum affordable price and the minimum listening environment, he/she can choose the suitable hearing device for his/herself by assisting the table 3.1 and 3.2.

Following the second objective, the system contains the sub-circuits includes pre-amplifier, anti-aliasing filter, microcontroller and DAC. The pre-amplifier circuit serves for two purposes here. First it amplifies, invert and give offset to the audio signal so that it could be match with the input signal level of microcontroller and secondly it controls the minimum gain of the system at the output. Following the amplification purpose, the low level background noise is also amplified along with the desired audio signal and another source of noise is the internally generated noise of different components which disturbs the magnitude of audio signal. Therefore, to tackle these factors the lower two bits of the built-in ADC of microcontroller are neglected and the 8 bit DAC is used which processes the 8 most significant bits (MSB) of the microcontroller, hence it helps in suppressing the low level background noise or internally generated noise of the system and after this stage, the gain is adjustable using the post-amplifier.

Normally the slow rise time of the filter creates problems as the frequency gets higher and the very fast rising time creates high overshoots and large settling time. Keeping in view these points, the 2nd order R.P Sallen and E.L Key low pass filter is used with unity gain and a quality factor (Q) of 0.5 as this much of quality factor gives a good compromise between the rising and settling time specifically for the audio signals.The microcontroller and external DAC are interfaced parallel and it consumes much power, but one possible solution is to use the external switch for controlling the power dissipation of the device.

Impaired hearing affects the individual in different fields of life. He/she cannot be able to communicate properly with others and his/her surrounding. Although technological advancement in the recent years has significantly falls the price of electronic equipments but the prices of medical devices are still high. But the microelectronics field development reduces the price of components and instruments. The microcontroller based hearing aids will take a long time to become common among the users but it provides an easy and cost effective way to enjoy the clear hearing with ease and comfort.

CHAPTER 07

CONCLUSION

This thesis is associated with assisting the hearing impaired people in choosing a better quality hearing aid device for themselves keeping in view their budgets and to designing a better quality and cost effective hearing aid device. Based upon the market price ranges of devices, two different tables of higher price range and lower to average price range are tabulated which will assist a particular user to select a better device after comparing available options accordingly to his/her listening profile's requirements and allocated budget. Fallowing the second objective, the proposed microcontroller based design is first implemented in software and after successful simulation results; it isnow implemented on hardware. The system showed the satisfactory results for the low frequency signals. The system is flexible in the sense that the large memory of the microcontroller, its extra analog channels and ports can be used for other special features like direct audio input and directionality etc. The system showed good results in the human voice frequency range and it overcomes the limitation of analog hearing aid. Based upon the gain limitations, the device is affective for the patients having mild up to moderately severed type of deafness and the system is now further being developed for more effective noise suppression and broader frequency range by introducing frequency channels.

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APPENDIX A

1. HUMAN EAR

The ear is a part of the human auditory system and it helps in balancing and positioning of the human body. Working of ear is like an astounding transducer in the sense that it first converts the sound energy into mechanical energy and then converts it to electrical nerve impulses which are transmitted to the brain. This process helps us in perceiving pitch of the sound by detection of wave frequency and the loudness of sound by detection of the wave intensity or amplitude. It also helps us in perceiving timbre of the sound by the detection of band of frequencies that combines to make up the complex sound wave [6].

Hearing impairment is the total or partial disability to hear sound. There may be many reasons for hearing loss, some are benign that can be corrected or modified by treatment, while some of the causes are malignant and the hearing loss cannot be corrected or recovered completely [77].

2. ANATOMY OF EAR

The ear is considered as one of the complex organ in the body and anatomically, developmentally, clinically and functionally, it is divided into the three main parts, i.e the external, middle and the inner ear [77]. Each of these parts has their own importance and performs specific functions.

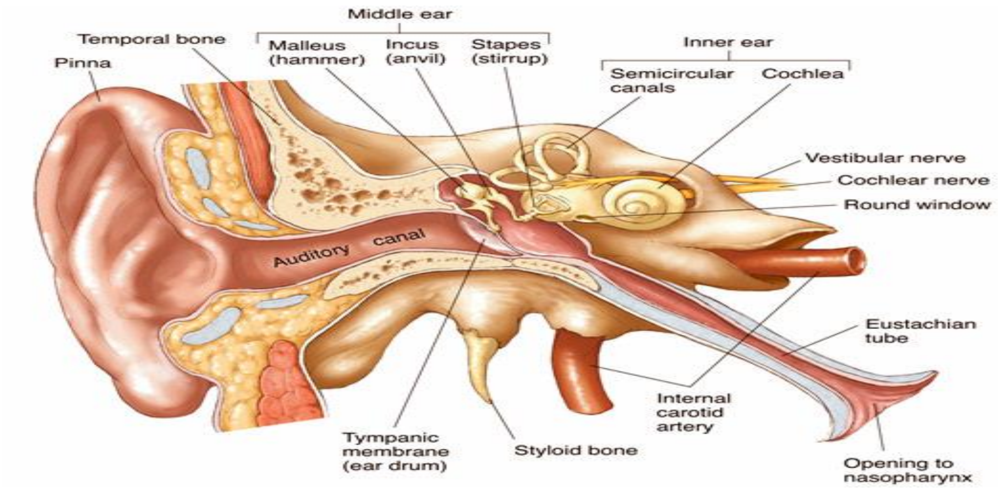

Figure A1: Anatomic division of human ear

2.1.External Ear

The external ear consists of a three parts, i.e pinna or auricle, the external portion of an auditory canal and the outside surface of the tympanic membrane. The pinna has further two surfaces, the medial surface and lateral surface [78]. The first surface is almost smooth and featureless while the different portions of the surface of lateral surface area as shown in the figure below. Its main function is to collect or pick the acoustic energy (sound wave) and to channelize it toward the auditory pathway.

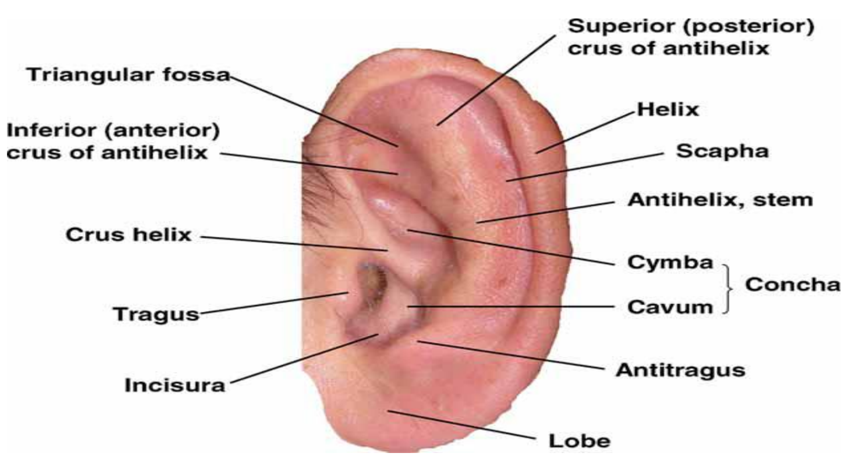

Figure A2: Parts of lateral surface of bone

2.2. The Middle Ear

The middle ear is the air filled closed cavity and is located between the outer and the inner ear. It is connected to the upper throat trough an Eustachian tube. Middle ear consists of a four walls i.e. interior, medial, lateral and posterior. The main function of the middle ear is to transmit the sound energy from the outer ear to the middle ear via oval window and this work is done by the three ossicular bones namely the malleus, incus and the stapes. The three ossicular bones and the middle ear cavity are as shown in the figure below.

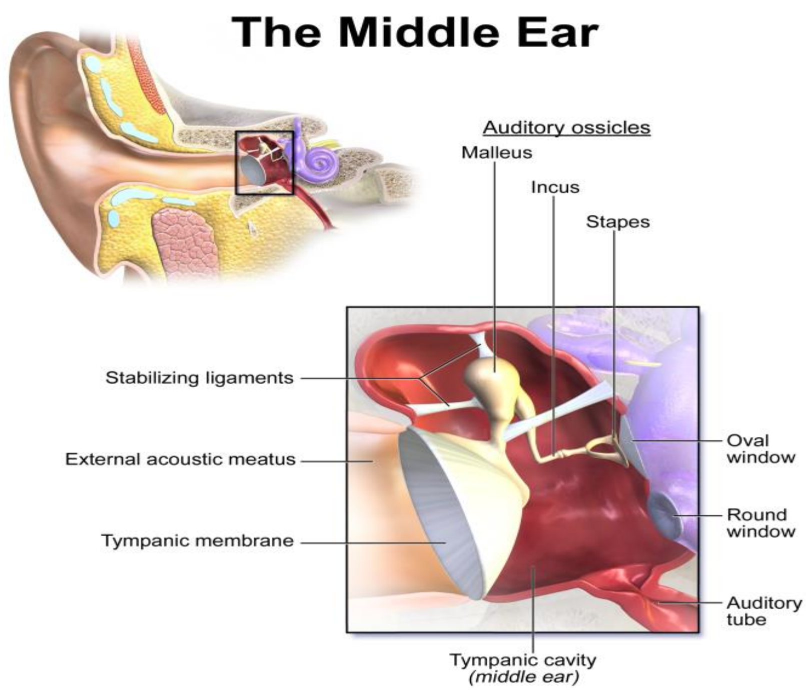

Figure A3: middle ear cavity and ossicular bones

2.3. The Inner Ear

The inner ear is also known as the labyrinth [6] and it is located in the temporal bone of the skull. It consists of a bony and membranous portion and contains the sensory organ for the sound detection named cochlea. The main component of the cochlea is the organ of Corti and its working is the responsibility for sound perception [77]. The different parts of the inner ear are as shown in the below figure. Overall the inner ear function is to convert the sound wave to nerve impulses and transmit them to the brain via auditory nerves.

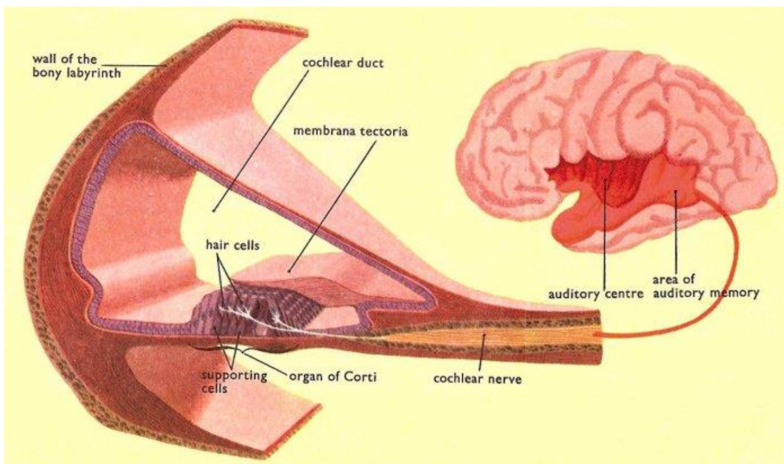

Figure A4: Parts of the inner ear

3. AUDITORY PATHWAY

The auditory pathway can best be understood from the figure as under. The conduction of cochlear nerve starts or originated from the organ of Corti in the cochlea. Then theses nerve fibers go in the pon through the interior and posterior cochlear nuclei. The neurons originated here from the pon, runs in the trapezoid body and relay in the superior olivary nucleus of the opposite side. Here the axons ascend as a lateral leminiscus. Some of them relay here in the nucleus of lateral leminiscus while the other ends in the nucleus of the inferior colliculus and medial geneculate body. From here the nerve fibers are now called the auditory radiations and they ends in the primary auditory cortex [6].

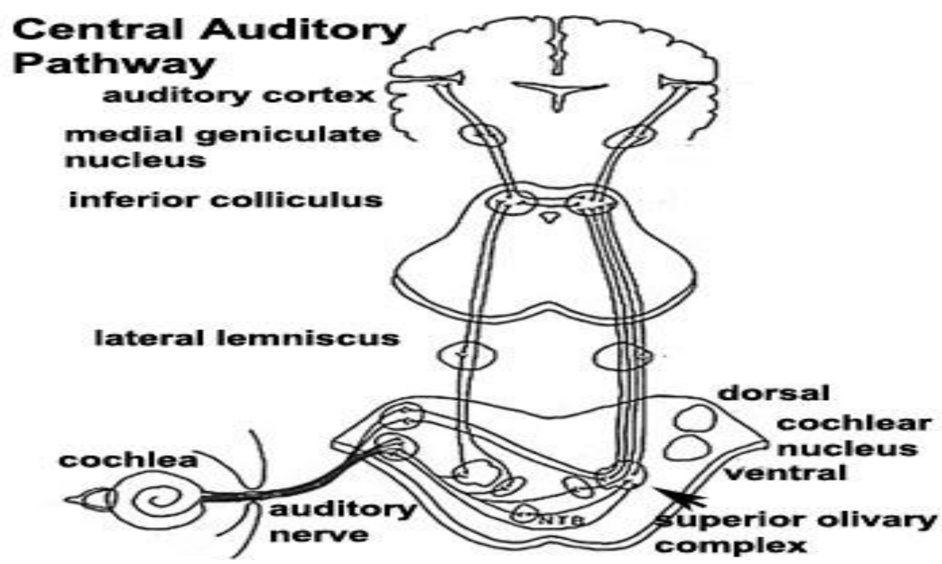

Figure A5: Auditory pathway

Appendix B

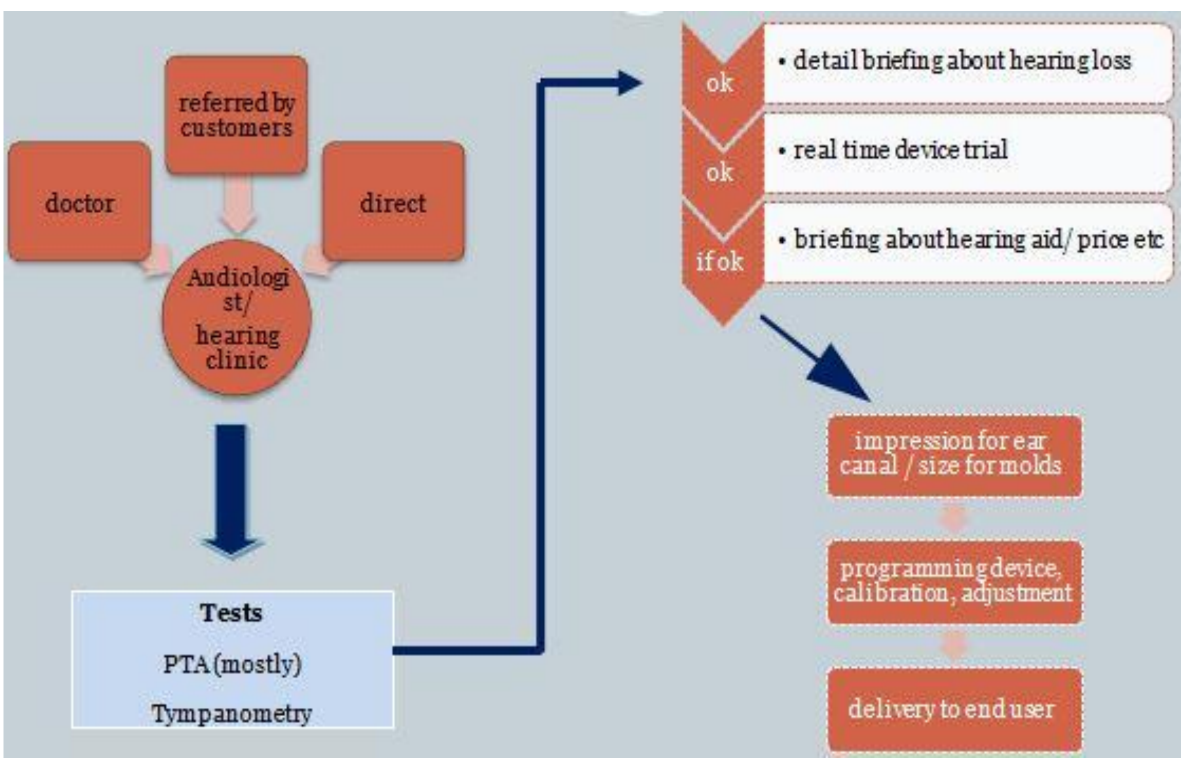

Figure B1: Clinical process for a hearing impaired person. Once a patient visits the audiologist then his/her audiometric tests are performed and based upon the result of these tests, a hearing aid is suggested for him/her.

APPENDIX C

1. TYPES OF HEARING AIDS

Hearing aids are classified into two main groups the external and implantable hearing aids [79].

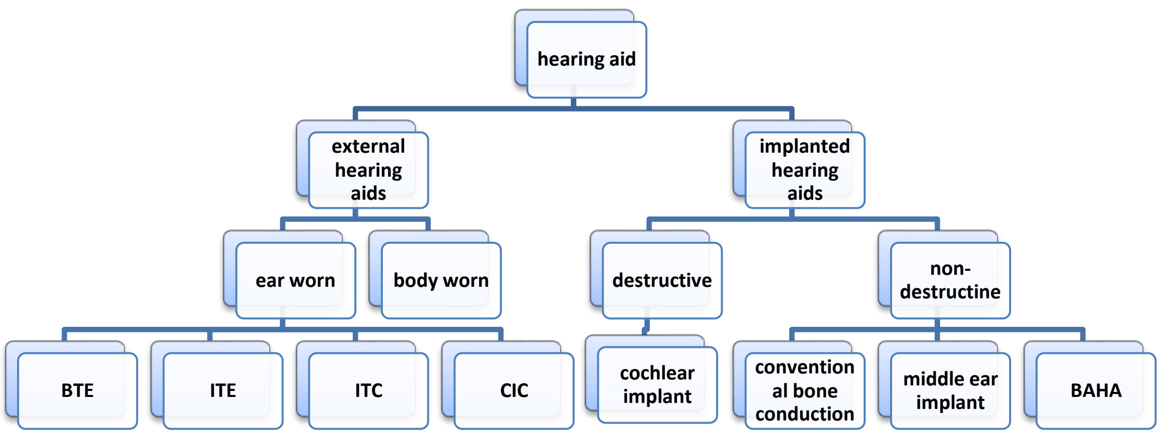

Figure C1: Classification of hearing aids

1.1.Implantable Hearing Aids

Implantable hearing aids can be further divided into destructive and non-destructive hearing aids.

1.1.1. Destructive hearing aid

Cochlear implants are for those patients who are profoundly deaf [79]. The main components of cochlear implant are microphone, a signal processer, transmitter receiver and an electrodes array which is placed inside the cochlea. Signals are transmitted to these electrodes through bones, skin and cartilage using FM radio signals. They help the patient in understand speech and cannot restore hearing and they are irreversible.

1.1.2. Non-destructive hearing aid

These are those implantable aids which rely on direct bone conduction and conventional bone conduction. They are further classified in to the conventional bone conduction, BAHA and middle ear implants.

In **Conventional bone conduction**, an exciter attached to the mastoid area of temporal bone is pressed with constant pressure and it transmit signal, but this signal is attenuated by soft bone and hence its fidelity is poor. The exciter can cause problem like headache, eczema, pain and skin irritation etc. a further extention to conventional bone conduction is BAHA [80].

In **Bone Anchored Hearing Aid** (BAHA), signal conduction is occurred directly through bone and it bypasses the external ear. BAHA contains a titanium screw, an abutment and a sound processer [81]. A screw is implanted behind the ear in the skull and the sound processer after osseointegration, transmit the vibration directly through the bone to the fluid filled cochlea [82]. In this way, the fidelity of signal is increased as compared to conventional bone conduction method and user feels comfortable and this method is surgically reversible.

Another non-destructive aid is the **middle ear implant**which converts sound directly into mechanical vibrations just as in bone conduction. A tiny exciter directly excites the ossicular chain.

Overall the implantable hearing aids have no occlusion effects as they don’t need ear molds, further they don’t have feedback problems or very negligible. They are especially for the patients, having external or middle ear malformation and suffering chronic otitis [79].

1.2.External Hearing Aid

This group is further divided in to body worn and ear worn hearing aids.

1.2.1. Body worn hearing aid

A body worn hearing aid consists of an earphone and a separate case in which the amplifier, control circuitry and battery are kept. The size of the case is such that it can be easily kept in a pocket. The earphone is connected to the case through a wire. Such types of aids provide a large amplification and long battery life. The only problem could be their comparative large size but they are cost effective.

1.2.2. Ear worn hearing aids

They are the basic types of hearing aids and they differ from one another based on their size, amplification level and placement whether inside or on the ear. They are also called air-conducted hearing aids and are of four types. i.e behind-the-ear, in-the-ear, in-the-canal and completely-in-the-canal.

Figure C2: Styles of ear worn hearing aid

1.2.2.1.Behind-the-ear (BTE) hearing aid

The BTE hearing aids basically consist of two main parts, an earmold and a case. The case contains all the electronic circuitry like amplifier, microphone and battery etc and it is placed behind the ear and is connected to an earmold through a tube which carries the sound signal to the ear [7,8]. These aids are used for mild to profound hearing loss and are used by the users of all ages. They are easily cleaned and handled and are sturdy.

A **Mini BTE**is a smaller version of BTE hearing aid in which a very small almost invisible tube is placed in the canal which carries sound and is connected to the aid. It is comfortable and has a reduced feedback and occlusion effect, and has a good cosmetic appearance.

1.2.2.2.In-the ear (ITE) hearing aid

In ITC, all the circuitry is kept in a case which is placed outside of the ear. Its size is larger than ITC and CIC and is smaller than BTE. As the ear grows, its case needs to be replaced. So it is not suitable for children and younger.

1.2.2.3.In-the-canal (ITC) hearing aid

The case of ITC hearing aid is fitted such that it is partially placed in the ear canal and is made accordingly to the shape and size of user’s ear canal. Due to its smaller size, it is a bit more discrete then the ITE and is used for mild to moderately severe hearing loss.

1.2.2.4.Completely-in-the-canal(CIC) hearing aid

CIC hearing aids are the most conspicuous of all the hearing aid types as they are deeply placed in the canal. Their case is almost hidden in the ear canal and is used for mild to moderately severe hearing loss.

ITC and CIC, both types are very small in size so their adjustment is very difficult and have limited features. Their battery sizes are smaller which limits the power and volume of the device. So they cannot be used for severe to profound hearing loss. They are not suitable for children and younger. They have some listening and cosmetic advantages.