CHAPTER 1

OVERVIEW OF AUDIOLOGY AND HEARING MEASUREMENT ANALYSIS

The basic concepts about anatomy and physiology of the ear, hearing loss, audiometers, hearing aids and prescriptive procedures for the recommendation of gain values to various frequency bands of digital hearing aid are discussed in this chapter. The significance and objectives of the present work are also explained in this chapter.

1.1 EAR AND HEARING LOSS

The ear is an important organ in human beings in respect of perceiving worldly knowledge to enrich and makes our life meaningful. Hearing problems cause the affected persons to be dull and aloof from others. If the hearing problem is identified sufficiently early, it can lead to a better and efficient solution. This section briefs the anatomy and physiology of the ear, perception of sound, the problems occurring in the ear and various causes of hearing loss.

1.1.1 Anatomy and Physiology of the Ear

The anatomy of the human ear could be understood from Figure 1.1. The human auditory system comprises of two major sections. A peripheral section, which is our ear, and a central section located in the brain, which carries the sensation perceived by the ears to the auditory area of the
cerebral cortex. The following sequences of operations are performed before the sound is interpreted in the brain:

- The ear receives the sound energy in the form of vibration. This vibrating energy enters in external part of the ear, which is the external auditory meatus and vibrates the eardrum called as the tympanic membrane.

- The vibrations of the tympanic membrane are picked up by a chain of small bones called the malleus, incus and stapes in the middle ear, which conduct vibrations to a specialized organ called the cochlea.

- The function of the cochlea is to convert the vibrating energy into electrical energy.

- This electrical energy enters the auditory nerve and carries the sensation through different parts of the brain to the auditory cortex, where the sensation of sound is analyzed and interpreted (John Hernandez 2004).

The phenomenon by which sound reaches the inner ear through the ear drum is called air conduction. The phenomenon by which the low frequency range sounds reach the inner ear through the bones in the head, bypassing the eardrum is called bone conduction (Biswas 1995). Normally, only a small fraction of sound is received by bone conduction. However, deaf people whose inner ear still functions normally may be able to hear the sound conducted to the ear in this way. For proper hearing, each and every part of this system from the external auditory meatus to the auditory cortex has to be normal (Martin 2008).
1.1.2 Sound Perception

Sound is generated whenever an object vibrates in an elastic medium specifically air. By nature, sounds are combination of multiple frequencies and not a single frequency pure tone or sine wave. However, all complex sounds can be considered as a combination of different pure tone sounds of various frequencies. The ear is not equally sensitive to the different frequencies of the perceived sound, specifically in the lower and higher frequency ranges. The frequency response over the entire audio range was charted originally by Fletcher and Munson in 1933, with later revisions by other authors, as a set of curves showing the sound pressure levels of pure tones that are perceived as being equally loud. The curves plotted for each 10 dB rise in the level with the reference tone being at 1 kHz is called as...
loudness level contours. The lowest curve represents the threshold of hearing, and the top curve represents the threshold of pain. Even though the human hearing ranges from 20 Hz to 20 kHz, very little speech information is available above 8 kHz. The perception of frequencies below 100 Hz is increasingly tactile in nature, making it difficult to assess. The perceived sound with different frequencies heard by humans and various sound sources can be understood from Figure 1.2.

![Figure 1.2 Speech banana and different sounds perceived by humans](image)

**Figure 1.2 Speech banana and different sounds perceived by humans**

**Source:** A hearing loss & late deafened blog SWC site

The speech banana shown in green colour covers the basic and essential speech components. It includes the vowels and consonants of language used in verbal communication. The majority of the speech information lies in the mid frequency range. The sound arising from various other sources falls outside the speech banana. The dB value of the sound alone does not indicate the real implication of the sound perceived by the human. The duration of the exposure of the sound heard is also of
significance. As an example, a tone heard for duration of 10 ms will not be as loud as a tone heard for duration of 10 seconds having the same loudness level. In respect of frequency, lots of discrepancies occur in reality. Low frequencies are not perceived as loudly as signal frequencies in the middle frequency of the audible range.

The next important consideration is the direction of the sound signal which influences the perceived sound level and spectrum. It can influence the production of surround sound. Apart from these, a part of the assessment of a signal lies in the fact that, the perceived loudness will fall in the extended exposure due to the effects of fatigue. The last important fact is that, the perceived level is influenced by a liking for the program, and it is totally a subjective issue because it is very difficult to quantify. If the loudness level is measured, many factors will have to be considered before arriving at a final value. The measuring instrument has to perform like the average perception of a sound signal with a group of young adult with normal hearing. Hence, the determination of the rms value of a voltage alone will not give the required information.

1.1.3 Hearing Loss

Hearing loss in an individual refers to depreciation in the amplitude level of the perceived signal, compared to people with normal hearing functionality. In a majority of the hearing impaired subjects, reduction in the clarity of speech also occurs. Under normal circumstances, hearing loss occurs either due to the middle ear problem or due to the damage in the part of the inner ear called the cochlea. Hair cells are small sensory cells in the inner ear that convert sound energy into electrical signals that reaches the brain. Once the hair cells are damaged, they cannot grow again. The destruction of hair cells created holes in hearing at specific frequencies and
the individual gets a higher value of the minimum threshold of hearing at the particular frequency.

Hearing loss can be broadly classified as:

- conductive hearing loss
- sensorineural hearing loss
- mixed hearing loss.

Conductive hearing loss occurs with a blockage in the external or middle ear. The subject having air conduction loss alone is concluded as having conductive hearing loss and the subject having a decrease in bone conduction is termed as having sensorineural loss. Sensorineural hearing impairment occurs because of the inner ear problem. It reduces the ability of the person to perceive and recognize the signal of a particular frequency in the presence of other frequencies. A damaged cochlea gives rise to several perceptual difficulties, and it is very difficult to hear the sounds when background noise is present. This is an important problem that affects the perception of different types of sound. Mixed hearing loss occurs when the subject is suffering from conductive hearing and as well as sensorineural hearing loss (Katz 2004). A dip in the hearing threshold occurred almost at all the frequencies in both Air Conduction (AC) and Bone Conduction (BC). The audiograms of two different subjects, which are the results of audiological investigations indicating the hearing loss level as conductive and sensorineural hearing loss are shown in Figure 1.3 a-b.

Hearing impaired subjects have difficulty in hearing the essential component of some phonemes. This decreased frequency resolution, and temporal resolution cause noise to be louder than speech when compared to the normal hearing person. In a majority of the hearing impaired cases, the
high frequency components are less audible than the low frequency components. Since the loudness of the speech is due to the low frequency components, the impaired can hear normally till the impairment grows further.

a. Subject having conductive hearing loss

b. Subject having sensorineural hearing loss

Figure 1.3 Audiograms of subjects having hearing loss
The hearing loss of an individual is classified into different types as shown in Table 1.1 based on the value of PTA, which is an average of thresholds of hearing measured in the frequencies of 500 Hz, 1000 Hz, and 2000 Hz (International organization for standardization 1994). Figure 1.4 shows the results of the National health interview survey conducted during 2000-2006 among the civilians in USA.

### Table 1.1 Classification of hearing loss based on the PTA value

<table>
<thead>
<tr>
<th>Sl.No</th>
<th>PTA in dB HL</th>
<th>Types of hearing loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>–10 to 25</td>
<td>Normal hearing</td>
</tr>
<tr>
<td>2</td>
<td>26 to 40</td>
<td>Mild hearing loss</td>
</tr>
<tr>
<td>3</td>
<td>41 to 55</td>
<td>Moderate hearing loss</td>
</tr>
<tr>
<td>4</td>
<td>56 to 70</td>
<td>Moderately severe loss</td>
</tr>
<tr>
<td>5</td>
<td>71 to 90</td>
<td>Severe hearing loss</td>
</tr>
<tr>
<td>6</td>
<td>91 to 120</td>
<td>Profound hearing loss</td>
</tr>
<tr>
<td>7</td>
<td>Above 120</td>
<td>Deaf</td>
</tr>
</tbody>
</table>

Figure 1.4 Percentage classifications of adults aged more than 18 years based on hearing loss level, age and sex in the United States observed during 2000-2006

Source: CDC/NCHS, National Health Interview Survey, 2000-2006
The following inferences can be made from the survey results. The percentage of persons having hearing loss increases with age. The percentage of male subjects suffering from hearing loss is more when compared to the females of the same age group. The hearing loss caused by excessive noise exposure is called as Noise Induced Hearing Loss (NIHL).

The survey conducted in India reveals the following facts about the prevalence and statistical analysis of the hearing impaired cases.

- In India, 63 million people (6.3%) suffer from significant hearing loss. The National Sample Survey (NSS) 58th round (2002) surveyed disability in Indian households and found that hearing loss was 2nd most common cause of disability and the top most cause of sensory deficit (Vishwambhar Singh 2015, Garg et al 2009).
- In urban areas, hearing loss was 9% of all disabilities and in rural areas, it was 10%. It was estimated that the number of persons with hearing disability per 100000 persons was 291; it was higher in rural (310) compared to urban regions (236).
- In the same survey, nearly 32% of the people had profound (person could not hear at all or could hear only loud sounds) and 39% had severe hearing disability (person could hear only shouted words).
- The survey results revealed that about 7% of people were born with hearing disability.
- About 56% and 62% reported the onset of hearing disability at $\geq$ 60 years of age in the rural and urban areas, respectively (Garg et al 2009).
The WHO survey (World Health Organization 2009) indicates the major reasons for the loss of hearing among the people in India as follows:

- Ear wax (15.9%) was the most common cause of reversible hearing loss.
- Non-infectious causes like aging and presbyacusis leads to auditory impairment in India (10.3%).
- Middle ear infections such as Chronic suppurative otitis media (CSOM) (5.2%).
- Serous otitis media (3%) is another leading cause of hearing loss.
- Dry perforation of the tympanic membrane (0.5%).
- Bilateral genetic and congenital deafness (0.2%).

1.1.3.1 Noise induced hearing loss

Human hearing may deteriorate due to different reasons. It has been firmly confirmed that noise is the first and foremost reason for the deterioration in hearing levels (Malcovati 2001). Every day sounds from television, radio, household appliances, and traffic are experienced. If these sounds are heard at safe levels they do not affect the hearing. However, when people are exposed to harmful noise - either sound that are too loud or loud sounds that last for a long time -, the sensitive structures in their inner ear could be damaged, causing NIHL. The loss of hearing sensitivity is normally observed first at high frequency (8 kHz) and later on as the loss progresses its effect is observed in the mid-frequency region (1-2 kHz) as clearly shown in Figure 1.5. As the age or years of deafness increases, the progression of hearing loss is also fast. NIHL can be caused either by an intense “impulse” sound, such as an explosion, or by continuous exposure to louder sounds over an extended period of time, such as noise generated in the working environment. Sources of noise that can cause NIHL include motorcycles,
firecrackers, and small firearms, all emitting sounds from 120 to 150 decibels. Long or repeated exposure to sounds at or above 85 decibels can cause NIHL. If the sound is louder, the NIHL will occur earlier. The proportion of subjects with NIHL, their profession and age can be analyzed from Figure 1.6. The graph is based on a survey conducted in Norway during 2005-2009.

The following inferences are made from the survey results:

- The percentage of people working in the manufacturing sectors suffered from hearing loss.
- Next to the manufacturing sector, people working in the transportation sector suffer.
- People working in the construction field are also affected by a decrease in hearing.

![Figure 1.5 Origin and progression of noise induced hearing loss](image-url)
In general, the deterioration in hearing capability starts as early as in the age of twenty (Norio 2004). Hearing loss is normally due to problems in the middle ear or inner ear. An inner ear problem is usually corrected by fixing a hearing aid after it is identified by the audiometer (Katz 2004).

1.2 AUDIOMETERS AND AUDIOLOGICAL INVESTIGATIONS

Audiometry is the technique to identify the nature of hearing loss and to determine the minimum threshold of hearing by recording the responses of the subject after presenting him with auditory stimuli with varying frequency and different intensity levels. There are different audiological investigations and procedures used for identifying the level of
hearing. Important among them are the pure tone and speech audiometric tests. Tests like tone decay test, small increment sensitivity index test, etc. are mainly performed on subjects having more specific hearing difficulties.

An audiometer is a specialized instrument, which is used for carrying out audiometric tests and procedures. It is capable of generating pure tones at different octave frequencies with different intensity levels (ranging from -20 to +120 dB), and duration. It also produces speech information in the form of words for assessing the hearing capability of an individual. Audiometers must be calibrated at regular intervals to ascertain the standards. This ensures that a gradual loss of hearing among the impaired subjects is because of the deterioration in the hearing level of an individual and not because of the instrumental error. Audiometer can be of different types depending upon the frequency range, range of acoustic output, mode of acoustic presentation, masking facility, procedures used, and types of acoustic stimuli. Due to the advent of modern technology, audiometers have also improved a lot, and their usability has also developed. Even though, the audiometers have improved to a large extent, they are not user friendly, and it requires an expert to understand the controls and operations for better and reliable test results. If not handled by experts, it will result in wrong diagnosis.

1.2.1 Audiometric Tests Conducted in the Ear

Pure tone and speech audiometric tests are the important audiological investigations performed with the hearing impaired subjects to identify and locate the place of disorder.
1.2.1.1 Pure tone audiometric tests without masking

In pure tone audiometry, the thresholds of hearing of the subject in specific octave frequencies 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz, and 6000 Hz are measured. It measures the hearing acuity and the following interpretations could be made with the test results:

- Conductive hearing loss occurs because of the disorder in the external auditory meatus or middle ear
- Sensorineural hearing loss occurs because of the disorder in the inner ear or in the nerve of hearing in the brain
- Mixed hearing loss occurs because of the disorder in the conductive apparatus as well as in the inner ear.

The pure tone audiometric test is done by presenting the stimuli through

i) headphones

ii) bone vibrator.

The pure tone audiometric test encompasses the air conduction test and bone conduction test. They are performed to assess the conduction level of the sound by the middle ear and inner ear. These tests assess the conduction level of sound signals through air and bone respectively (Epstein et al 2009).

- The air conduction test is used for the assessment of middle ear conduction. In that, the sound is fed through headphones and this sound reaches the inner ear by air conduction. The minimum threshold level in dB for different frequencies of pure tone is measured.
The bone conduction test is used for the assessment of inner ear conduction. In that, the sound is applied through a bone vibrator placed into the mastoid process of either the right or left temporal bone. The minimum threshold measuring procedure is the same as that of the air conduction test except the way in which the testing signal is fed.

An audiologist selects the initial level of the stimuli. In this technique, the subject is instructed to signal the audiologist each time a tone is perceived. For air conduction thresholds, earphones are comfortably positioned and the better ear is tested first, if it is known to the subject. If not known, audiologists will quickly screen each ear using the same initial frequency, and the better ear is tentatively determined. Tones are often presented in an ascending series (Price 1971, Greer & Magistro 1997). The following steps are performed to arrive at the value of the minimum hearing threshold:

- Initially a particular frequency stimulus at a predetermined dB level is presented to the subject. A pure tone signal of 30 dB HL is presented to the subject.
- If the subject hears the signal, the tone will be decreased by 10 dB.
- If the subject is still able to hear the tone, the tone will be decreased by 5 dB.
- If the subject does not hear it, the tone will be raised by 5 dB. In this way, by several presentations, the minimum threshold of hearing is obtained.

The set of procedures is repeated for all the testing frequencies. The measured test results of the pure tone audiometry are plotted in a special
graph called the audiogram. The audiogram is plotted between the measured hearing thresholds in dB with the corresponding octave frequency. The shapes of audiograms are associated with different types of hearing loss. To prescribe hearing aid, the audiogram is used as basic information to decide the degree of amplification required at various frequencies. For the bone conduction test, the signal is directly fed to the skull bone instead of the ear canal. The bone vibrator is placed over the mastoid process. In the bone conductor, the metal mass is moved back and forth by the action of the audio current passing through the coil (Carhart & Jerger 1959, Martin 2008).

1.2.1.2 Pure tone test with masking

The need for a pure tone test with masking is decided by calculating the difference between the air conduction thresholds of bad ear with the bone conduction threshold of a good one. If it is greater than or equal to 10 dB, then a pure tone test for the bad ear will be done with masking. In air and bone conduction audiometry, if the signal or speech is applied to one ear, the cochlea of the other ear will also be stimulated by the transmission of the signal through the bone of the skull. If the sound presented to one ear stimulates the other ear, it is called cross hearing (Katz 2004, Brookler & Hamid 2006, Lightfoot 2000). The stimulus applied to the test ear in the air conduction test will pass from the test ear to the cochlea of the non-test ear with reduction in sound energy.

The reduction in the sound level is called Inter-aural Attenuation (IA), and the value of IA varies from 45 to 80 dB. The cochlea’s of both the ears are equally stimulated during the bone conduction test. In this, the inter-aural attenuation is 0 dB. Hence, cross hearing is a serious issue in the bone conduction test. Applying a suitable noise signal to the non-test ear in order to prevent it from hearing the test signal, will solve the cross hearing problem. In the pure tone test with masking, the non-test ear is presented with a narrow
band noise signal with the test frequency as the center frequency. The amplitude of the noise signal is varied such that it prevents the non-test ear from hearing the test words; and also it should not affect the testing procedure conducted with the test ear (Roeser & Clark 2000, Turner 2003).

1.2.1.3 Speech audiometric tests

Pure tone threshold testing is used to assess the sensitivity level of the ear, whereas speech audiometric testing attempts to identify the level of integrity of the entire auditory system by assessing the ability to hear clearly and to understand speech communication. The main use of speech audiometry is the identification of neural types of hearing loss, in which both the reception as well as the dissemination of speech is impaired more markedly than in cochlear or conductive hearing loss. There are two different types of speech audiometric tests, namely, the Speech Discrimination Threshold (SDT) test and Speech Reception Threshold (SRT) test (Olsen & Matkin 1991, Brandy 2002).

1.2.1.3.1 Speech discrimination threshold test

In the speech discrimination test, a set of monosyllabic speech discrimination words (normally 50 words) are presented over headphones to each ear, in which the subject is asked to repeat the words that they heard. The percentage of the number of words the subject is able to identify correctly with the total words presented to them gives the Speech Discrimination Score (SDS). The value of the SDS varies from 0 to 100 %. The classification of hearing loss based on the value of SDS is shown in Table 1.2. Generally, a high score is associated with normal hearing, and low score is associated with hearing loss. The score indicates the subject’s capability of distinguishing the words perceived by them and the score obtained and the value of the pure tone average is normally observed for consensus.
Table 1.2 Interpretations of the speech discrimination score

<table>
<thead>
<tr>
<th>S.No</th>
<th>Speech discrimination score</th>
<th>Interpretation</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>90-100</td>
<td>Normal</td>
</tr>
<tr>
<td>2</td>
<td>76-90</td>
<td>Slight problem (i.e. Tough to understand telephone Speech)</td>
</tr>
<tr>
<td>3</td>
<td>60-76</td>
<td>A moderate problem</td>
</tr>
<tr>
<td>4</td>
<td>50-60</td>
<td>Poor recognition, i.e., Difficulty in interpreting words</td>
</tr>
<tr>
<td>5</td>
<td>&lt; 50</td>
<td>Very poor recognition, i.e., Not able to recognize words</td>
</tr>
</tbody>
</table>

1.2.1.3.2 Speech reception threshold test

The procedure for the SRT test is similar to that of the SDT test except for the fact that this test uses bisyllabic words with equal stress called as spondee words. The SRT is the lowest hearing level in dB HL at which 50% of a list spondee words are correctly identified by the subject. For estimating SRT, a group of 6 spondee words is presented at 25 dB above the average pure tone audiometry threshold for 500 Hz and 1000 Hz, and then at successively lower intensities. The level at which the subject is able to identify 3 out of 6 words correctly, is taken as the SRT. The SRT of a normal subject is very closely related to the pure tone hearing threshold, and is generally ±12 dB HL as that of the pure tone average of the subject. If the SRT value is does not lie in this range, the pure tone audiometric and SRT tests are performed once again to ascertain the quality of the audiological investigations. A list of bisyllabic words is prepared and used to perform the SRT test. In case of disagreement between the pure tone audiometric test and
SRT test inferences, a different set of six spondee words are presented and verified for consensus.

1.3 HEARING AIDS

Hearing impaired subjects are able to hear comparatively better after fixed with a hearing aid, because it is a safe and non-invasive procedure. The hearing impaired patients are usually unaware of their deterioration in hearing. So they delay wearing of hearing aids, and they used to live with minor difficulties. To understand and know well about the functioning of hearing aids, the physical characteristics of the signals have to be understood first. The characteristics are the rate at which the signal oscillates, the time taken for a repetitive fluctuation, the wavelength, the way in which the signal bends around obstacles, the Sound Pressure Level (SPL), the frequency spectrum of the complex sound and the pure tone components ranging over several octave frequency bands.

The objective of the hearing aid is to intensify the perceived speech signals by varying the loudness level of speech and also to provide an optimal sound so that the understanding of the speech signals amidst noise must be high in different environmental conditions (Williams 2004, Williams et al 2004, Williams et al 2004, Kochkin 2007, Williams 2008, Keidser et al 2009, Glyde et al 2011). The amplifier used in the hearing aids may be linear or nonlinear. In the case of linear amplifier, the signal is amplified flat, irrespective of the presence of other complicated frequency sounds. The highest level produced by the hearing aid is called as the Saturation Sound Pressure Level (SSPL). The sound output of the hearing aid is measured in the ear canal or using a coupler or with the help of an ear simulator that imitates a real ear. The hearing aids are classified based on different aspects. Based on fixation they are classified as body aid, behind-the-ear, in-the-ear,
The digital hearing aid of modern days uses different frequency bands in an audible range of sound. REIG is the additional gain required with hearing aid usage, to get a clear perception of sound by the hearing impaired patients. The integrated real ear measurement improves the accuracy of fitting initially, and subsequently, when the hearing aid is fine tuned to achieve better speech intelligibility to the patient (Yanz et al 2007, Yanz et al 2008). Most of the hearing aid users are comfortable with the lesser gain than what is actually prescribed for them. Prescribing the same gain for all individuals, because they have the same hearing thresholds, will result in inaccuracies of too little as well as too much gain (Gerling 1992).

The performance and parameters of the hearing aid can be easily measured with the hearing aid analyzer by connecting it through the coupler. A coupler is a small cavity used to take up the signal from the hearing aid to a microphone. The optimum size of the coupler selected is 2 cc so as to reflect the size of the real ear and hence, it is called as the 2 cc coupler. But still it is slightly larger than the real ear; hence, it produces lesser SPL than the normal ear. The difference is called as a Real Ear to Coupler Difference (RECD). Broadband sounds consist of many frequency signals which are used to measure the performance of the hearing aid, especially in the case of nonlinear hearing aids. The measurements commonly performed with the sound signal, and the output is plotted with the gain versus frequency for different sound input levels. The gain values measured with the hearing aid are classified based on the type of measurements done.

The gain measurements made normally are Real Ear Aided Gain (REAG) and REIG. REAG is the calculated difference between the sound level in the ear canal and the sound level nearer to the subject. REIG is the
calculated difference between the sound level in the ear canal with the hearing aid and the sound level at the same point without using the hearing aid. REIG indicates the required gain provided to the sound signal when the subject uses hearing aids.

1.3.1 Digital Hearing Aids

Subjects who used analogue hearing aids for years together have often become accustomed to the analogue processing of sound as the brain becomes adapted to the sound perceived from it. When the subjects changed their hearing aids to digital ones, they are exposed to different sound processing and features. They often perceived quiet sounds. The digital hearing aids often perceive clear sounds and there is not much harmonic distortion in the sound signal. Digital hearing aids provide amplification that can be more accurately designed to take into account the hearing loss. In digital hearing aids, the sound picked up by the microphone is split into different bands based on the frequency, and then it is amplified depending upon its pitch and loudness to match an individual prescription. The gain values for the different frequency bands of a digital hearing aid can be programmed to meet the needs of the hearing impaired subjects. It allows to be reprogrammed if the hearing condition of the impaired subject changes. The digital hearing aid is inbuilt with an important signal process called Wide Dynamic Range Compression (WDRC), which helps to present quiet sounds so as to make it audible. It also enables the speech signal to be amplified to a comfortable level so as to prevent loud sounds perceived as uncomfortable. The noise reduction and feedback management in the digital hearing aid provides better sound in comparison to an analogue hearing aid.
To summarize the advantages of a digital hearing aid are:

- automatic control of the gain-frequency response of the hearing aid based on the calculation of signal to noise ratio in different frequency regions.
- effective adaptability aspect in terms of the response shaping and modifying the compression characteristics of the hearing aid.
- adjusting the gain instantaneously by incorporating suitable digital signal processing based algorithms.
- intelligent control to adjust the gain-frequency response of the hearing aid depending on the direction of the sound in order to minimize the background noise level.
- inbuilt adaptive noise cancellation algorithms to cancel or prevent the noise from reaching the ear.
- low power consumption over the equivalent analogue version hearing aid of the same specification.

1.3.2 Objective Measures for Hearing Aid Speech Quality and Intelligibility

Two important aspects of speech quality are the perceived overall speech quality, and the speech intelligibility. Perceived overall quality is the overall assessment of speech. Speech intelligibility is the accuracy with which the subject can identify the words pronounced. The intelligibility is measured as the percentage of the correctly identified to the number presented. Speech quality assessment classified into two categories; subjective and objective quality measures. Subjective quality measures are based on a comparison of
the original and processed speech data by listeners. They rank the quality of the speech according to a predetermined scale subjectively.

Objective speech quality measures are based on physical measurement, like acoustic pressure, and mathematically calculated measurements. Objective speech quality measures are calculated from the original undistorted speech and the distorted speech using specific mathematical formula. There are a number of objective measures depending on the application to be tested. The important objective measures used to assess the speech quality and speech intelligibility are:

- SNR Measures
- LP-Based Measures
- Weighted Spectral Slope Measures
- Articulation Index
- Speech Transmission Index
- Speech Intelligibility

Signal-to-Noise Ratio (SNR) is one of the oldest objective measures. It requires both distorted and undistorted (clean) speech samples. SNR can be calculated using the equation 1.1.

\[
SNR\ in\ dB = 10\log_{10} \frac{\sum_{n=1}^{N} x^2(n)}{\sum_{n=1}^{N} \{x(n) - \hat{x}(n)\}^2}
\]

where \(x(n)\) is the clean speech, \(\hat{x}(n)\) the distorted speech, and \(N\) the number of samples. There are a number of objective measures that use the distance between two sets of Linear Prediction Coefficients (LPC) calculated on the original and distorted speech. The Log-Likelihood Ratio (LLR) measure is a
distance measure directly calculated from the LPC vector of the clean and distorted speech. LLR is shown in the equation (1.2).

\[ d_{LLR}(a_d, a_c) = \log \left( \frac{a_d R_c a_d^T}{a_c R_c a_c^T} \right) \]  

(1.2)

where \(a_c\) is the LPC vector of the clean speech, \(a_d\) is the LPC vector of the distorted speech, \(a^T\) is the transpose of \(a\), and \(R_c\) is the auto-correlation matrix of the clean speech.

Weighted Spectral Slope (WSS) distance measure is a direct spectral distance measure. It is based on a comparison of the smoothed spectra from the clean and distorted speech samples. The smoothed spectra can be obtained from either the LP analysis, Cepstrum lifting (a term coined for filtering in the Cepstrum domain), or filter bank analysis. Articulation Index (AI) is the widely accepted quality measure, which can estimate speech intelligibility. AI assumes that distortions can be calculated on a per-critical frequency band basis, and distortion in one frequency band does not affect other bands. AI can be obtained by calculating the SNR for each band, and averaging them. Speech Transmission Index (STI) is an objective measure that can estimate the speech intelligibility for a wide range of environments. STI uses an artificial speech signal as input, which is a spectral-shaped noise that has a long-term spectrum envelope identical to speech. This test noise in each band is modulated so that the modulated envelope is sinusoidal. STI assumes that the loss of intelligibility is related to the loss in the modulation depth. The loss in this modulation in each frequency band is calculated, weighed and averaged by the receiver.

Speech intelligibility is a measure of accuracy with which the speech under test carries its spoken content. In our present study, though there
are different quality measures as discussed above, with regard to the possibility of utilizing them in the analysis, we focused on improving the speech intelligibility of the perceived speech as this significantly contributes to the evaluation and prediction of speech intelligibility, as all NAL prescriptive procedures are based on maximizing the speech intelligibility. Hence, in our study we used the quality objective measure, the speech intelligibility and its related parameters only. The betterment in the speech intelligibility depends on the speaker characteristics, the listener, and numerous types of degradation encountered during transmission. It has been used widely to evaluate building or room acoustics, hearing aid performance, speech codec degradation, speech synthesis performance and many others. It can be divided into two aspects as Word and Sentence intelligibility. Word intelligibility is measured by the correct number of words identified by the listener. Sentence intelligibility tests use question or command sentences, and are measured by the number of correct responses made by the listener. Sentence intelligibility may also be measured by the correct identification of key words embedded in the test sentences, although there are arguments that this is merely a word intelligibility test. Speech Intelligibility Index (SII) is a measure highly correlated with the intelligibility of speech, and the value of SII ranges from 0.0 to 1.0. SII is calculated using the equation (1.3)

\[
SII = \sum_{i=1}^{n} I_i A_i
\]

(1.3)

where, \( I_i \) is the frequency importance function of each frequency bands of speech. If the speech frequencies are divided into ‘n’ bands then the sum of \( I_i \) of all the bands is equal to 1. \( A_i \) is the band audibility of each frequency band, and it ranges between 0 and 1. The methods of calculating SII can be obtained from ANSI S3.5 (1997) standards.
Hearing aid benefit was measured using more objective assessments of speech perception by means of speech recognition materials. In recent years, several speech perception tests, Speech in Noise, Hearing in Noise Test, Lexical Neighbourhoods Test, Quick Speech in Noise Test, and Words in Noise have been developed with the goal of maximizing their validity and reliability, in order to provide a more accurate reflection of a listener's speech understanding. Comparative hearing aid evaluations employed speech audiometry as an important measure of performance. Audiologists need to perform unaided and aided speech recognition testing as part of the hearing aid evaluation. Unaided speech recognition testing can help determine hearing aid candidacy and provide patients with realistic expectations about speech understanding. Conducting aided speech recognition testing can also help to demonstrate when an aided performance is better than an unaided one, and the advantages of special features of the hearing aids, and help obtain information for counseling.

1.4 PRESCRIPTIVE PROCEDURES

1.4.1 Evolution of Prescriptive Procedures

The digital hearing aid amplifies the sound components by segmenting them in terms of frequencies and applies specific gain to the frequency bands. The user gets the facility to adjust the gain values so as to achieve optimum speech audibility and clarity. Initially, it was performed randomly and it takes more trial and error time. To speed up the process, a set of protocols called as prescriptive procedures was developed by a group of researchers. Prescriptive procedures are a specific algorithm developed for proposing the hearing aid technical parameters especially the gain values to be fixed for different frequency bands of a digital hearing aid. Various prescriptive procedures have been developed in the last three decades to calculate the REIG of a digital hearing aid, based either on loudness
equalization or speech intelligibility. Earlier, the prescriptive hearing aid
selection had been done only with linear hearing instruments. The National
Acoustics Laboratory (NAL) of Australia developed the NAL-R formula, and
subsequently, the NAL-RP formula for hearing aid models with linear
circuits. With the development of nonlinear hearing aids, a suitable
prescriptive procedure was needed to provide multiple gain curves for
different input levels (Gatehouse 1993).

Then the researchers developed the successful prescriptive
procedures DSL I/O, IHAFF, FIG6, NAL-NL1 and NAL-NL2 for the non-
linear hearing aids. NAL-NL1 and NAL-NL2 are procedures from NAL,
which are based on maximizing the important parameter speech intelligibility
and used very much for nonlinear hearing aids. Depending on the feedback
and preference of the subjects undergoing hearing aid trials, the final gain
settings will vary from the initial settings. With an increase in the number of
prescriptive formulae, the quality of service can be defined as the extent to
which a particular fitting procedure will give satisfaction to the hearing aid
users.

To enhance the utility of the hearing aid, the electro acoustic
characteristics of the hearing aid must be suitably varied based on the
recommendations of specific prescriptive procedures by considering the level
of hearing impairment of the subject (Studebaker 2002, Keidser et al 2003,
Johnson et al 2011). These prescription formulae were based on the threshold
and supra-threshold of audiometric data. More specifically, they calculate
how much gain is to be provided to different frequencies and also they are
different for various sound input levels (Dillon et al 2003, Ching et al 2010,
Keidser et al 2011). The majority of the so far proposed prescriptive
procedures have been based on the results of the pure tone audiometry only.
In addition, the systematization of the adjustment rules is to ensure the
optimization of the hearing aid functioning, especially in terms of speech intelligibility. The compression characteristics and gain-frequency response of the hearing aid may also be gradually adjusted to attain the manufacturer’s target (Ching 2002, Kwiatkowski et al 2008, Convery et al 2011, Ching et al 2001, Keidser et al 2012).

The adaptation level of the hearing aid by the impaired subjects is based on the adjustment of gain values continuously with the help of the software built into the hearing aid. This is because a new subject with no prior exposure to the hearing aid normally prefers a lower gain for the output sound level, and subsequently, the subject adapts to the higher gain levels. The audiologist may also choose to use a middle adaptation level as a tradeoff between a conservative and aggressive approach for prescribing the gain and output (Glyde et al 2013, Keidser 2009, Keidser et al 2008, Thorne et al 2008). The development of Digital Signal Processing (DSP) technology in hearing aids enhances the flexibility of the programming capabilities of the hearing aid, which allows the audiologists to be more precise to fit a hearing aid to a prescribed target. However, recent survey data indicate that a few audiologists are verifying their hearing aid fittings in the initial visit itself (Kirkwood 2004, Kirkwood 2006).

1.4.2 The Existing Strategy followed in using the Prescriptive Procedures

Presently, different strategies are practiced in prescribing the required parameter values of the hearing aid. The general practice and the experimental setup followed in the selection of prescriptive procedure and programming the hearing aid with the gain values suggested by the procedure can be well understood from Figure 1.7.
The following steps are followed presently to suggest gain values for the hearing aids:

- The subject is initially identified for hearing loss by performing the basic audiological investigations such as pure tone and speech audiometric tests using the specialized hardware, i.e. conventional audiometers.

- The results of the pure tone audiometric test are plotted as an audiogram.

- The SRT and SDS values arrived at, with the speech audiometric test, are noted.

- As soon as the subject completed the audiological investigations, and if the subject found is suitable to get benefited with the use of a hearing aid, he/she will undergo hearing aid trials. During the trials the subject will be first
identified with a appropriate hearing aid for the type of hearing impairment.

- The hearing aid is connected to a specialized hardware called hearing aid programmer which is incorporated with different existing prescriptive procedures.

- The audiologist has to select the appropriate prescriptive procedure based on the audiogram.

- The suggested gain values for different frequency bands of a hearing aid recommended by the procedure are fixed in the hearing aid.

- After programming, the subject is tested with the speech discrimination threshold test, and the SDS value is calculated manually. These steps are repeated at least for three procedures. The procedure which gives the best score will be considered as an appropriate procedure for that type of hearing loss.

- If the subject does not get a reasonable SDS value, the audiologist will have to change the gain values to make the subject to attain the maximum SDS value possible.

Over the past several years, many prescriptive procedures have been developed and tested to fit hearing aids. The appropriate selection of procedure determines the success of the prescriptive hearing aid fitting. But the problem for the audiologists lies in selecting an appropriate procedure for particular types of hearing impairment. Even after identifying the best procedure that suits the particular type of audiogram, the adjustment of gain is a tough task. The trial and error method in selecting the prescriptive procedure and modifying the gain values consumes more time for the subject
and audiologist. Arriving at the value of adjustment in the gain settings is highly challenging for the audiologists. Some of them don’t have time to spare to perform this task and hence, the satisfaction level among the hearing impaired subjects is very low. Hence, there is a need for developing an appropriate procedure and suitable solutions to save time. Analyzing all the parameters and standards used in the prescriptive procedures, the present work finds an appropriate solution to these issues.

1.5 OBJECTIVES OF THE RESEARCH WORK

In the present scenario, the various difficulties faced by the hearing impaired subjects made this research work a necessary and meaningful effort. The important problem faced by the hearing impaired subjects was in getting utmost satisfaction with the performance from the hearing aid. The reasons for this issue were probed and analyzed.

The identified reasons were:

- the impaired subjects delayed the use of a hearing aid, because they were not aware of the deterioration in their hearing level
- the gain values of the hearing aid were not appropriately fixed to get maximum satisfaction

The low frequency sounds contain useful speech information almost for all languages and it is carried to the brain through bone conduction. Initially, for all the hearing impaired subjects, the hearing loss started in the air conduction only. The bone conduction normally does not get affected for an individual till the hearing loss progresses to a certain level. Hence, in the initial stage of hearing loss, it is more pertinent to identify it through audiological investigations.
To get utmost benefit from the performance of the hearing aid,

- identifying the hearing loss at the earliest is essential
- fixing appropriate gain values for different frequency bands of the digital hearing aid is also important.

The core objective of the research work is to develop an adaptive expert system to perform these tasks on a single platform. A computerized audiometer is inbuilt in the proposed system and is used to perform audiological investigations in a mass screening procedure because of the fact that a computer with inbuilt suitable modules is sufficient for proper solutions. To propose the gain values, many prescriptive procedures were developed so far, but each procedure has its unique solutions and it is an uphill task for the audiologists to select an appropriate procedure for a particular subject having a specific problem. Hence, the aim of the present work is to analyze various linear and nonlinear prescriptive procedures and validate the recommendations of these procedures by performing proper audiological investigations with the hearing impaired subjects, so as to develop an expert system to propose the appropriate gain values for the digital hearing aid in an automated system.

The percentage of success of the recommended gain values of each procedure is analyzed based on the calculated values of SDS. The expert system developed in the present work, proposes successful gain suggestion for those subjects who were not satisfied with the recommendations of the existing prescriptive procedure. The important requirement of the present work is to develop a database having the gain values satisfied subjects. Hence, the recommended successful gain suggestions of the satisfied subjects are stored in the database and the expert system, subsequently used this database as a guide and for training, so as to recommend gain values for the new
subjects adaptively. The significance of the expert system in decision making can be obtained from Cornelius (2002). Hence in order to enhance the satisfaction level among the hearing impaired subjects, a system has to be developed to perform mass screening of audiological investigations, to identify the hearing loss at the earliest and to propose appropriate gain values to raise the speech intelligibility of the hearing impaired subjects.