1.1 Sound

The waves in the surrounding are similar to ripples on the watery surface.

![Sound waves](image)

The "wave" into atmosphere contains very less variation in the pressure about normal atmospheric variations. Here also some exceptional cases were in the acoustic wave the air pressure is around the atmospheric level. Acoustic or sound wave as a source generates these compressions and rarefactions as shown in Figure 1.1. Usually a vibrating entity such as a string of violin, a loudspeaker diaphragm, motor in a machine and vocal cord in a human body generates sounds. It is also known that in a vacuum, the sound cannot travel. The eardrum vibrates in direct reply, when the pressure variations in the wave reach the ear and the pressure fluctuations are heard as sound.
1.2 Ear Structure and Working

The human ears are in pair situated around left and right side of human head can be considered as a sensory organs comprising the auditory system, concerned in the detection of sound and the vestibular system is responsible for maintaining body balance equilibrium [1]. The ear can be divided into three parts anatomically which can be known as external, middle and the inner ear which can be seen in Figure 1.2. Sound collection and amplifying mechanism

- It can work as a transducer which convert sound vibration into action potentials
- Action potential can be delivered by nerves

As per the above functions three major regions can be described as follows in terms of the inner ear, the middle ear and external ear. Area and functional supporting parts are shown in Figure 1.2.

![Figure 1.2  Anatomy of ear](image-url)
1.2.1 The External Ear

The external ear used to look after the tympanic membrane recognized as eardrum. It also collects and directs sound waves through the ear canal to the eardrum.

1.2.2 Middle Ear

The middle ear bones recognized as malleus, incus and stapes amplify the sound waves. The middle ear is differentiated from the external ear by the eardrum is a cavity with air also identified as tympanic cavity carved out of the temporal bone. It joints the throat/nasopharynx via the eustachian tube. Eustachian tube equalizes the air pressure on both sides of the eardrum. In general the walls of the tube are distorted. The tube is opened and allowed air in or out as needed for equalization by swallowing and chewing actions [1]. Equalization of air pressure guarantees that the eardrum vibrates maximally when it is stroked by acoustic waves. Functional process of hearing follows below flow.

- The malleus hammer is attached to eardrum.
- The malleus attaches to the incus and that then joints to the stapes.
- The stapes and oval windows are linked together which is the part of cochlea.

1.2.3 Inner Ear

The inner ear is formed with a network of fluid-filled tubes running through the bone of the skull. The bony tubes are filled with a fluid called perilymph. The actual hearing cells contains this membranous and the hair cells of the organ are very important. There are three major sections of the bony labyrinth:

- First part is cochlea which is like snail in the shape.
- Semicircular canals are in the back rest part which is responsible for balance maintenance.
1.2.4 The Cochlea

- In the auditory nerve the cochlea changes acoustic vibrations into action potentials.
- Cochlear fluid (perilymph & endolymph) is vibrated because of vibration of the oval window.
- Vibrations in the fluid in turn cause the basilar membrane to vibrate.
- Vibrations of the basilar membrane cause hair cells bending generator potential.
- To produce action potentials fibers will be stimulated if generator potentials are huge enough.
- Pitches are generated in the different part of cochlea such a way that base is responsible for producing high pitches and apex is generating low pitches [2].

1.3 Hearing Impaired

Most importantly in the human ear cochlea is responsible for hearing. In the cochlea thousands of hair cells are there at the time of born of human. These cell are responsible for sensing different frequency generation of electronics waves which is given to mind. Because of some reasons like aging, very loud voice, drug administration and because of some serious infections these cell are reduced in their numbers. Dynamic reduction in the number of cells produces problem in hearing and which results in the deafness [3]. Deafness can be divided in basically two types which are described as conduction deafness and sensorineural deafness.

- Conduction deafness: The term conductive relates to external ear and middle ear disorders where the transmission of sound to the cochlea is impaired. To diagnose conductive disorders is fairly comprehensive and robust as a consequence of their mechanical and relatively peripheral nature and the possibility of visual observation or surgical confirmation. This type of deafness is often directed to medical or surgical treatment.
- Sensorineural deafness: Sensorineural implies an organic disorder of the cochlea and/or subsequent parts of the auditory system. Sensory is intended to relate to the
cochlea and neural to the subsequent sections of the auditory pathways but is must be remembered that these terms refer to the audiological type rather individually confirmed sites of lesion. Hair cell loss or damage is generated by auditory nerve.

- Central deafness: The term central neural is used to describe hearing disorders where there is a defect in auditory pattern processing. Often without appreciable hearing loss. The disorder is often most evident when speech stimuli are used.

1.4 Methodology of Measurement of Sensitivity: The Audiogram

The hearing sensitivity of patients in the audio logic clinic can be described in terms of the number of dB by which the threshold sound pressure of a person is higher than the normal threshold is referred to as dB of hearing loss. Mapping in a graph of hearing loss against frequency is recognized as an audiogram [3]. The extent of hearing impairment is usually measured primarily in terms of audiogram [3]. The extent of hearing impairment is usually measured primarily in terms of loss of sensitivity. Because of the fundamental difference in the causes, characteristics and management of conductive and sensorineural types of hearing loss, It is desirable to enumerate separately the loss of sensitivity caused by any conductive or sensorineural component. Essentially, this reduces to measurement of both the sensitivity at the cochlea and the overall sensitivity of the ear, the conductive loss being the difference between the two. Pure tone audiometry involves estimating the threshold of hearing for certain standardized stimuli usually via the air conduction and bone conduction routes.

![Audiogram format](image)

Figure 1.3 Audiogram format
Threshold of hearing is variously defined but is often taken to be the lowest sound pressure or alternating force level at which, under specified conditions, a person gives a predetermined percentage of correct detection responses on repeated trials. Threshold definitions are usually based on 50% correct detection. The stimuli used are calibrated on the hearing level scale which has been obtained from normalization studies involving large numbers of subjects and has at the 0db hearing level points the modal value of hearing threshold levels measured in ontologically normal subjects aged [3]. Because it is impractical to obtain a biological calibration for every audiometer. The national and international standards objectively defines the biological baseline for certain combinations of earphone or bone vibrator and acoustic or mechanical coupler. The standards are also specific to particular audiometric test frequencies. For air conduction frequencies of 0.125, 0.25, 0.5, 1, 1.5, 2, 3, 4, 6 and 8kHz are included. For bone conduction at least followings are included: 0.25, 0.5, 1, 2, 3 and 4kHz although there is some variations between national standards. Figure 1.3 illustrates the standard audiogram format used for plotting results. The use of particular symbols for air conduction and bone conduction for the (R) and (L) ears.

1.5 Digital Hearing Aids

Deafness is an often underestimated and misjudged handicap that seriously limits patients’ capabilities in life. Although the handicap of the deaf is not as conspicuous as for instance blindness or a physical handicap, deaf people find themselves often excluded from the normal society because of the large problems they have in communicating with other people. Unfortunately, medical science is not capable to cure deafness, at least for most of the affected patients [4]. On the other hand a number of techniques exist that enable the patients to communicate, such as lip reading, sign language and of course hearing aids. Most of these techniques are of limited help because for instance lip reading only provides a small part of the information the spoken word gives and sign language is only known by a relatively small amount of people.

Previously analog hearing aids are more popular which pick up and amplify the sound before returning it to the ear through a kind of miniature loudspeaker are an effective
rehabilitative means for mild to moderate hearing losses. Moreover, the conventional analog hearing aids have the following disadvantages.

1. Analog hearing aids give restricted performance.

2. Their characteristics are function of mechanical variations time and temperature.

3. Their usefulness is highly limited in case of people suffering from sensorineural loss, where in which, the frequency response be optimally designed depending on the condition of the persons residual auditory area.

In digital hearing aid, a microprocessor or ASIC replaces the hardware used to process the signal like filtering and compression. The analog output of the microphone will be low-pass-filtered to prevent aliasing errors sampled at discrete intervals and will be converted to binary form using an A/D converter. Programmed processor will treat digital signal accordingly. The processed digital signal will then be transformed back to an analog signal to achieve compatibility with human ear via the D/A converter and sent to the hearing aid receiver. The first generation of digital hearing aids is likely to be similar to currently available analog hearing aids with regard to the type of processing that would be done. The major difference between digital hearing aids and the present generation of analog hearing aids will be the degree of control over parameters of the hearing aids. Because the characteristics of the hearing aid like frequency response, maximum power output and compression parameters etc., they will be specified in the software, the constraints imposed by the hardware will be eliminated. The characteristics most suitable for an individual hearing aid user can be specified precisely in the software. Digital hearing aids assure many compensations over conventional hearing aids [5]. These include

1. As because of internal processor digital hearing aids can be programmed.

2. Electro acoustic parameters can be adjusted with much higher precision.

3. Some feedback mechanism can be implemented for self monitoring capabilities.

4. Logical operations can be developed for testing and calibration internally.
5. The noise reduction is possible through advanced processor.

6. Automatic control of signal levels loudness can be implemented accordingly.

Hearing loss is widely recognized as one of the most common human disorders. On average one out of thousand newborns is affected by a severe hearing loss either congenital or acquired. Moreover, the prevalence of hearing loss increases monotonically for older populations as the patient’s hearing is irreversibly affected by, for instance, noise induced trauma and age-related hair-cell degeneration. As a result, about half of the people of 65 and older suffer from a mild to severe hearing loss.

It is fact that the concept of a digital hearing aid was expected at an early date but two major technical troubles had to be resolved before developing a wearable digital hearing aid. The first was the development of a DSP processor extremely fast to operate in real time. Another difficult problem is that the developing digital circuitry which should small in size and with very lower in power consumption for real time use in a small wearable unit like hearing aids [5].

Main focus is in why hearing impaired people are so seriously handicapped in everyday listening situation seem to be very scanty. This lack of knowledge particularly manifests itself in the uncritical way in which hearing aids are assumed to be of benefit. If proper transmission will not conduced then it results in conductive deafness which breaks transmission chain. Most of them cured by help of proper surgery [6]. It is generally recognized that electronic amplification only cannot compensate satisfactorily for these losses. On the other hand many hearing impaired persons appear to be rather disappointed about their hearing aids and take little interest in using them.

Hearing aids designing professional taking challenge that cleanest speech is forwarded to hearing impaired. The common grievance of those with hearing impairment is the reduced ability to recognize speech in everyday communication in a noisy environment. These difficulties are often experienced especially at the working place during activities as a burdensome handicap and it is also major constraint to aged persons. Current statistic shows among total hearing impaired half are suffered from sensorineural losses and only about twenty percentage would be suffering from conductive losses [7]. By several
investigation it can be demonstrated that around 5-15dB more signal to noise ratio is need to treat that type of hearing impaired through hearing aids. Every 4-5 dB improvement of the SNR may raise speech intelligibility by about fifty percentage. Although many techniques provide substantial attenuation of narrow band noise none of them appear to be capable of improving the intelligibility of the enhanced speech.

1.6 Problem Definition

The key parts of a digital hearing aid are the microphone, A/D and D/A converters along with suitable DSP processor, the receiver and a two-port memory.

Microphone with good gain converts analog signal into digital signal. Filtering operation is needed to removed almost known high frequency noises. Sigma-delta process is used along with 16kHz sampling frequency which seems ideal treatment for analog to digital conversion process. To improve quality of signal sampling rate might be extended up to 48KHz to get more resolution.

Memory is used to store the processing parameters that can be down loaded from the audiometer/programmer system to the user. Digital signal processor contains an array of adders, multipliers and registers which provide the fundamental operations necessary for implementing various digital algorithms. Whenever data is converted into digital domain everything would be controlled by DSP processors. Different types of filtering operations are executed. Storage of different audio processed data different set of memory is taken. In the process by the help of various filter parameters peak output is optimized. The set parameters are checked out before fitting by audiologist. Same algorithm logic is used again to convert digital signal into analog which can drive speaker of the hearing aids. Loud speaker is properly designed that can be driven by generated analog signal by considering its impedance and attenuation.

The configuration is in contrast with the use of a general-purpose DSP, where considerable power is consumed in executing program instructions. Now with the development of ASIC, the circuits are implemented in CMOS technology to reduce the size and power consumption [8]. Functional diagram of digital hearing aids is shown in Figure 1.4.
Motivation behind innovation is to detect and reduce noise signals. Any time to reduce noise from the signal it should be operated in frequency domain. For filtering operation properly designed filters bank is required with trade off more coefficient filter more multiplication and more power is required for the same. Depends on the complexity of computation power consumption is vary. Therefore it is important to design the filter bank for consuming as little processing power as possible. The filter banks should be designed with a minimum number of multiplications as the multiplication consumes power.

Now in the presence of noise hearing aids not functioning well so some filtering operation should be required. For that adaptive signal processing has found widespread practical applications. The key reasons for the widespread use of adaptive filter is their ability to optimize their own performance through recursive modifications of internal parameters. There are numerous applications of adaptive filtering, such as adaptive beam forming, noise canceling, speech formant estimation, array processing, etc, in the fields such as telecommunications, radar, sonar, navigation system and biomedical electronic.

Major discrepancy with above concept is that designing of filter bank needed advance prediction of noise characteristics which is almost very difficult in real time operations whereas adaptive filter is a kind of automatic filter which adjusts its parameters as per the requirement by following some of the equations and rules so as to achieve some specified objective.

Whenever there is a requirement to process the signal whose characteristics may not be known exactly or even statistically, in such circumstances, the adaptive filter offers an attractive solution and provides significant improvement in performance compared to a fixed filter. Speech signal processing, especially the improvement of speech signal for the
hearing impaired is a potential area in which the adaptive filtering theory can be effectively applied to improve the intelligibility of the speech signal.

It is required to establish algorithm on the DSP processor using code composer studio, for implementation of the real time operation of the same algorithm. Speech signal should be processed such a way that separate frequency analysis is possible for that. Analysis should specify each and every band of the recorded or incoming speech. As per the audiogram of the patient it should be modified by multiplying with different coefficient to get loudness level up to desired level. Still above mentioned process will not give better quality of the speech in presence of the noise. Noise processing is necessary for good reproduction of the speech signals.

Generally linear or non linear methods can be adopted to carry out process with speech signal. Compared to linear methods non linear methods are more useful in the speech enhancement because most of the path trough which signal travels is sort of non linear like all converter and functionality of loud speaker. Non linear model given better noise suppression. Comparatively to the linear modes of enhancement non liner way of improvement more complex. Generally because processor status only linear scheme attracts designer but non linear environment of signal motivates designer to work on that and takes challenges to improve signal quality than to deal with complexity. Different methods are developed for improving the intelligibility and SNR for speech using LMS, NLMS as well as RLS.

1.7 Objective of Present Work

Deafness is most often caused by deterioration inner cells which are very small like hair, in fact it is not a major problem with the associated neurons. This implied that if the neurons can be stimulated by a means other than hair cells, some hearing can be re deposited. One of the most suitable solutions to fight with deafness is to wear digital hearing aids because of following advantages like greater control, affordability and flexibility, easier maintenance. In many cases most widely preferred option is digital hearing aids with given prime requirements.
1. As per the conclusion of audiogram individual frequency can be amplified

2. Loud noises are bothersome with speech processing.

Mentioned limitations of hearing aids can be tolerated with some software modification in the system. Audiogram of any patient can be measured and as per that given modification could be deployed. For that speech can be analyzed and individual frequency component can be separated from the given speech signal.

Better solution for quality improvement: Hearing aids with Speech frequency separation and loudness increment of individual frequencies using wavelet transform.

Speech enhancement using wavelet transform algorithm is possible with individual steps utilized for the implementation. Same algorithm with similar concept is implemented and results are shown. Using multi resolution approximation concept of wavelet transform speech signal can be partitioned in many separate bands as the level of the resolution increases. Using algorithm given file is divided into ten discrete band of frequencies. Then individual band is processed. Increment loudness level in decibel in individual is observed. Around 3db, 5db and 7db increment in each and every band can be seen with simulation graph. For derivation daubechies wavelet with four nonzero coefficient (DB4) is used. During application of different wavelet transforms multiplied numerical coefficient is kept constant and using that same one change in loudness level can be observed. However still noise is present with the signal in the case of loudness increment then scenario is more serious. Always hearing aids can be used in presence of real environment. In real time system noise is always present in any form with speech. If speech is not clean than no use of loudness control for individual frequency.

1.7.1 Voice Activity Detection in the Noisy Speech

Prime requirement of the noise free speech is reduction of noise from speech signal anyhow. Very basic first step is in the presence of background noise first of all identify occurrence of the acoustic signal around evenly distributed surrounding noise. As a consequence noise corrupted speech signal is given to the VAD algorithm which finds
our pauses or silence of the speech. In VAD algorithm mainly focused area is searching our zero crossing of speech occurrence areas and based on that speech signal can be recognized. In the total speech with back ground noise wherever speech signals are there that areas are getting more zero crossing rates. In the space of back ground noise zero crossing rate is very less. In VAD next most important part is searching for the energy through the spread of the signal. Compared to original voice signal, noise occurrence getting very less energy. By deciding specific threshold level it could be possible for searching our signal energy in the given area of the speech signal. For the weak fricatives it can be case where energy would be very less for the specific letter so in that case energy as well as zero crossing rate is giving tolerable decision for VAD.

In the next step filtered speech is given to voice activity detection algorithm which is only developed for the extraction of speech activity in the speech. So that occurrence of the silence is detected and rest of the time speech can be detected. Detected silence can be applied for the wavelet thresholding which makes position of the silence tends to zero.

Using VAD pauses in the speech can be detected with very great extend. By searching these pauses no of samples responsible for the noise can be identified and that samples are make to zero for making silence free form noise in presence of the surrounding areas.

1.7.2 Adaptive Filtering for Speech Signal

Now in the next step main target for cleaning is only from the speech because all the silences are now noise less. Still the speech is with given back ground noise. For that adaptive filters are more suitable. To make successful implementation of work it is necessary to prepare acoustic environment in that two signals are required. One can be considered as a primary input and second can be considered as secondary input. In recent work acquisition of noisy speech is with reference noise signal can be taken. Now the real time noisy speech must be removed with the given algorithm. For taking reference signal and primary signal adaptive environment can be prepared with necessary data.
Basic concept of adaptive filtering says that if any non stationary inputs are present with the desired signal then using adaption of filter coefficient with incoming desired signal the filtering operation can be carried out. Basically in the category of adaptive algorithm mainly used algorithms are LMS, NLMS and RLS linear filters.

LMS are the simplest type of the steepest descend based adaptive algorithm. Algorithm mainly based on transversal filter domain. In algorithm the measurements of the relevant correlation functions and matrix inversion are no required. Because of this simplicity here LMS algorithm is used. In LMS when tap weights supplied by the information source is large then algorithm is suffered from the gradient noise.

To overcome this difficulty in real time speech NLMS helps a lot. Basically in NLMS, mainly coefficients are normalized with respect to the square Euclidean norm of the tap
input adjustments and which can be applied to different tap weight vector at different forward iterations.

Next category of adaptive filter is based on the theory of matrix inversion lemma. That is known as recursive least square algorithm RLS. RLS is more complex algorithm than other two mentioned above with advantage that convergence rate is very faster with incoming signal.

By identifying nature of the noise adaptive algorithm can be applied to calculate filter weight. Derived filter weight can be as per the incoming signal noise characteristics. Continuously updated filter weight mechanism is designed for the noise removal. Filter designing must be arranged through adaptive filter algorithm like LMS, NLMS and RLS. Main task is to extract features for the filter adaptation from speech signal. For that training vector must be developed. By taking training vector filter coefficient is modified and error can be removed. Figure 1.5 shows total flow of implementation.

For the given category of adaptive algorithms as a input noisy speech is given with that as a reference present noise is given and as per the functionality given algorithm. For each category MSE and PSNR is measured for the comparison.

1.8 Motivation of Present Work

In the world there is huge mass which are suffering from the hearing deafness. It is very chronical disability noticed. The problem of deafness calls for the best talent and efforts from the scientists. The hearing impaired subjects especially sensorineural loss patients and aged people experience more difficulty in understanding conversational speech in background noise than normal listeners. Statistics indicates that more than 50% auditory handicaps are sensorineural loss subjects. Many deaf people, especially aged persons, have severe chronic visual impairment as well, and it cannot be rectified. The present analog hearing aids are just sound amplifiers. If some methods are developed to reduce noise without distorting the signal, it will provide significant benefit to sensorineural loss handicaps and aged people. Also one can bring significant changes in the hearing impaired child by providing an appropriate digital hearing aid and by giving the necessary educational training at an early stage. Hence, with digital hearing aid it is
possible to make him/her more self-reliant. However, many scientists agree that the complete absence of auditory response among the deaf population is decidedly rare. There appears to be irrefutable evidence that digital hearing aids could make considerably greater contribution toward alleviating the burden of deafness. Although the concept of a digital hearing aid was anticipated at an early date, because of the problem of developing digital circuitry that is small enough and sufficiently low in power consumption for practical use is yet to be resolved. So far the work that has been carried out based on using FIR filters where in large number of multiplications is required. Hence, challenge of research is that designing of filter bank with optimum condition that can be taking variable way for changing its value and accordingly efficient to improve noise in the incoming disturbed signal with minimum complexity.

1.9 Proposed Contents of Thesis

The thesis has been arranged in nine chapters. A brief description of each chapter is given below.

Chapter 1 provides idea about preamble which includes background of the problem domain in detail. It also includes motivation towards criteria for the defining problem domain. By analyzing background problem definition is described with necessary objectives to resolve the given problem in the definition. Step wise objectives are defined over here to get desired results in the problem statement. Utilization of the solving the problem to the society is also discussed in the detail in given chapter.

Chapter 2 includes characteristics of human hear and reaction to the sound. It includes general terms and condition of the listening of the human ear. What are the basic causes of the deafness which includes sensory neural problem and second deafness due to poor bone conduction. In that next part is included to solve the problem of the deafness in which two techniques are includes: cochlear implant and digital hearing aids with its advantages and disadvantages. It also includes nature of the speech which is required to processed for the digital hearing aids. Generation of the speech, theoretical model of the speech types of different sound characters are described in detail with mathematical modeling which are used for the process of the enhancement.
Chapter 4 covers all the types of noises which are affected to the audio frequency in the natural environment. Mathematical modeling of the noises which is naturally generated in the background and is always with the speech signals. Frequency and time domain characteristics of the noise signal are discussed in detail for the removing non stationary noises.

Chapter 4 includes glimpses of the adaptive filtering algorithm like LMS, NLMS and RLS. Unit describes detail architecture of each adaptive algorithm, adaptation concept, mathematical modeling, convergence parameters, coefficient adaption concept, complexity of the algorithm and comparison. With adaption algorithm concept of voice activity detection is also discussed in detail with feature extraction form the given speech and identifying of the pauses or silences in the speech signal by applying VAD. It also includes all the summary of implementation of adaptive algorithms.

Chapter 5 gives detail idea about investigation towards problem. Most important unit describes how noisy speech can be cleaned out by taking two signal adaptive algorithms. Flow of the implementation is discussed towards problem definition. Each of the adaptive algorithm implementation methods discussed with different varying parameters and signal generation. All simulation results discussed in detail for each method of adaption in terms of PNSR, SNR and MSE.

Chapter 6 explores detail review of wavelet and discrete wavelet transform which is applicable to the speech processing. Multi resolution concept is discussed in detail for getting of time frequency and amplitude relation of the speech signal. How MRA can be utilized and what is necessary mathematical treatment that is listed.

Chapter 7 includes collection of all the results which will prove the definition. All the simulation results discussed in terms of band separation and enhancement in detail. Comments are also discussed in favor to make results more effective. In this unit application of wavelet is considered in terms of multi resolution application. Section describes how using DWT speech signal can be processed and each band can be separated out and modification can be achieved using values of coefficient.
Chapter 8 gives comparison and conclusion of the project definition with future scope and further development criteria. What might be expansion is quite very clearly over here. It describes among all discussed technique which one is giving better performance for speech signal with what value of signal strength and noise strength.