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Improving Speech Perception for Hearing Impaired

Swapnil Aswale[1], ManasSawant[2], ShreyasPathak[3], Jagdish Shinde[4], Prof.PravinDhulekar[5]

Department of Electronics & Telecommunication, Sandip Foundation's Sandip Institute of Technology and Research Centre, Nashik, Savitribai Phule Pune University, India[1,2,3,4,5]

Abstract: India has a substantial number of hearing impaired persons, especially in the rural sector. It is estimated that as many as 50% of inner city children under the age of seven have transient hearing loss of up to 10%, recurrent or persistent enough to affect their educational and social progress. The number of hearing impaired persons per lakh of population in rural sector is 600 for males and 500 for females. Conventional analog hearing aids presently have a number of functional deficiencies. These include fixed frequency and I/O response, lack of flexibility, inability to distinguish speech from noise, etc. These problems can be solved by developing effective speech processing algorithm for hearing aids. This paper is focused on types of hearing impairment and role of inner ear hearing impairment in maximizing the audibility of impaired peoples.

Keywords: Hearing Impaired; Speech processing algorithm; inner ear hearing impairment; Audibility.

I. INTRODUCTION

People who suffer from significant hearing impairment can be classified according to the region of the ear that is affected. The ear is divided into three sections, the outer ear and the middle ear, which are concerned with the conduction of the sound and the inner ear, which has cochlea and the auditory nerves. The conductive disorders can be corrected with surgery and instrumentation. The conventional hearing aids provide the amplification of speech, as the hearing loss is independent of frequency [1]. Many such aids are provided with filtering and amplitude compression. These aids are not designed to improve the speech presentation for the person suffering from sensorineural impairment (arising due to loss of hair cells in the inner ear and auditory nerve). Sensorineural hearing loss causes reduction in frequency resolving capacity of the ear (auditory system). Sensorineural hearing loss can be divided into two categories, when there is damage to the inner ear (cochlea) or to the nerve pathways from inner ear (retrocochlear) to brain. Sensorineural hearing impairment, which is associated with loss of hair cells in the cochlea. The speech signal gets affected in sensorineural hearing impairment while perceiving its intensity, time and frequency, etc. Sensorineural hearing impairments are frequency dependent shifts in hearing threshold, loudness recruitment, reduced frequency and temporal resolution, and increased spectral and temporal masking [2].

II. SYSTEM DEVELOPMENT

A. Basics about Human Auditory System:

Signal processing is used in cochlear implant development to convert sound to electrical pulses.

The pulses can bypass the damaged parts of a deaf person's ear and be transmitted to the brain to provide partial hearing.

Converting sound into something the human brain can understand involves the inner, middle, and outer ear, hair cells, neurons, and the central nervous system. When a sound is made, the outer ear picks up acoustic waves, which are converted into mechanical vibrations by tiny bones in the middle ear. The vibrations move to the inner ear, where they travel through fluid in a snail-shaped structure called the cochlea. The fluid displaces different points along the basilar membrane of the cochlea. Displacements along the basilar membrane contain the frequency information of the acoustic signal. A schematic of the membrane is shown here (not drawn to scale) [3].

B. Alleviating Deafness with Cochlear Implants

Deafness is most often caused by degeneration or loss of hair cells in the inner ear, rather than a problem with the associated neurons. This means that if the neurons can be stimulated by a means other than hair cells, some hearing can be restored. A cochlear implant does just that. The implant electrically stimulates neurons directly to provide information about sound to the brain [4].

The problem of how to convert acoustic waves to electrical impulses is one that Signal Processing helps to solve. Multichannel cochlear implants have the following components in common:

- A microphone to pick up sound
- A signal processor to convert acoustic waves to electrical signals
- A transmitter
- A bank of electrodes that receive the electrical signals from the transmitter, and then stimulate auditory nerves.

Just as the basilar membrane of the cochlea resolves a wave into its component frequencies, so does the signal processor in a cochlear implant divide an acoustic signal into component frequencies, that are each then transmitted to an electrode. The electrodes are surgically implanted into the cochlea of the deaf person in such a way that they each stimulate the appropriate regions in the cochlea for the frequency they are transmitting. Electrodes transmitting high-frequency (high-pitched) signals are placed near the base, while those transmitting low-frequency (low-pitched) signals are placed near the apex. Nerve fibers in the vicinity of the electrodes are stimulated and relay the information to the brain. Loud sounds produce high-amplitude electrical pulses that excite a greater number of nerve fibers, while quiet ones excite less. In this way, information both about the frequencies and amplitudes of the components making up a sound can be transmitted to the brain of a deaf person by a cochlear implant.

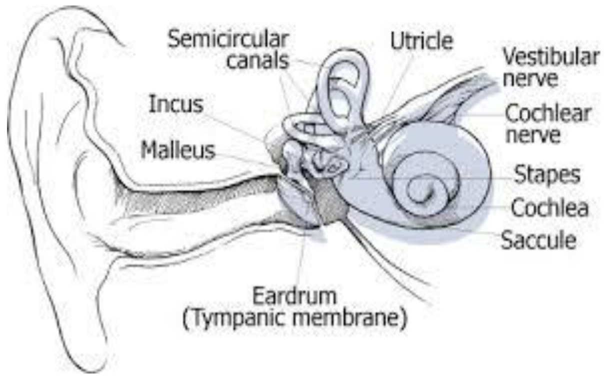

Fig1. : Anatomy of Human Ear

C. Function of the Auditory System

Peripheral auditory system can be broadly divided as outer ear, middle ear, and the inner ear. The pinna (auricle) and the auditory canal (meatus) together form the outer ear. The pinna modifies the incoming signal at higher frequencies and thus helps in sound localization. The auditory canal protects the ear from distant bodies. It exhibit filter transfer function and introduces an increase in sound pressure of about 17 dB at frequency of 2.7 kHz. The outer ear is separated from the middle ear by tympanic membrane.The middle ear, an air filled cavity, is consist of tympanic membrane, three ossicles and middle ear muscles. Ossicles are the malleus, the incus and the stapes. Malleus is attached to the tympanic membrane at one end and to the incus at the other end. Incus makes contact with the stapes, the footplates of which are attached to the membrane of oval window (starting point of the inner ear). The middle ear provides impedance matching and reduces the amount of reflecting sound [5].

III. HEARING IMPAIRMENT

Hearing impairment is broadly categorized as conductive and sensorineural depending upon the location of damage in the auditory system. Conductive impairment takes place due to damage in the outer ear, ear drum or middle ear, blocking the transmission of the sound to the inner ear. It generates attenuation of the stimulus and results in increase in both the hearing threshold levels and discomfort able loudness levels. Conductive impairment can be treated medically or surgically in most cases [5].

Sensorineural hearing impairment takes place due to damage to the transduction mechanism of the inner ear. The impairment due to defect in the cochlea is known as cochlear (sensory) impairment and that occurring due to defects in the auditory nerve is retrocochlear hearing impairment. The cochlear hearing impairment is associated with loss of cochlear hair cells. Exposure to noise causes loss of sensitivity at high frequencies. The amount of the loss increases with the frequencies [6].

Age related auditory dysfunction is commonly termed as presbycusis, which include degenerative changes in the auditory system such as hardening of the basilar membrane, of the cochlea leading to loss of hair cells in the organ of Corti and degeneration of neurons in the entire auditory system. Impairment is bilaterally symmetrical, with greater hearing impairment at high frequencies, and increased difficulty in understanding speech. Initially the hearing loss may be mild with a loss of sensitivity in the high frequencies that continues to increase with increasing age [6].

A. Masking

It is matter of everyday experience that one sound may be obscured, or rendered inaudible; in the presence of other sounds. Thus music from a car radio may mask the sound of the car's engine, provided the music is somewhat more intense. Masking has been defined as:

1. The process by which the threshold of audibility for one sound is raised by the presence of another (masking) sound.
2. The amount by which the threshold of audibility of a sound is raised by the presence of another (masking) sound. The unit customarily used is the decibel.

It has been known for many years that a signal is most likely masked by a sound having frequency components close to, or the same as, those of the signal. This led to the idea that our ability to separate components of a complex sounds depends, at least in part, on the frequency resolving power of the basilar membrane (BM).It also led to the idea that masking reflects the limits of frequency selectivity; if the selectivity of the ear is insufficient to separate the signal and the masker, then masking occurs. Thus masking can be used to quantity frequency selectivity. An important physical parameter which affects masking is time. Most of this chapter is devoted to simultaneous masking, in which the signal is presented at the same time as the masker [7].

B. Dichotic Presentation

Listening with two ears is known as binaural hearing. Presenting same signal to both the ears is known as diotic presentation and presenting two different signals to the two ears is referred to as dichotic presentation. Researches have reported the benefit of binaural dichotic mode of hearing in improving speech perception by persons with sensorineural hearing impairment. Sensorineural hearing impairment is considered by increased threshold of hearing, reduced dynamic hearing range, degraded frequency selectivity and temporal resolution, and

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increased spectral and temporal masking. Some of the studies to solve the problem of senso rineural hearing impairment are discussed in the previous section. To reduce the effect of increased spectral and temporal masking, studies that are based on consonant enhancement technique are also discussed in the previous section. The ability to perceptually combine binaurally received signals from two ears improves speech perception under adverse listening conditions. Binaural listening offers better overall sound quality and intelligibility, more relaxed listening, and it also helps in source localization. Many studies have shown the benefit of binaural hearing over the monaural hearing. Here we discuss the use of binaural dichotic presentation as a means of reducing the effects of masking for persons with moderate bilateral sensorineural impairment. The splitting of information in speech signal for presenting signals to the two ears, in some sort of a complementary fashion, to provide relaxation for sensory cells of the basilar membrane, may help in reducing the effect of increased masking and thereby improve the speech reception in cases of bilateral sensorineural hearing impairment with some residual hearing [7], [8].

IV. PROPOSED ALGORITHM

The work flow of the proposed algorithm is as follows:

1. Acquisition of Speech signals (Prerecorded Speech Materials)
2. Decimation by factor D (if required)
3. Design of Dyadic Analysis Filter bank with
   a) Wavelet transform with different families
   b) Tree Structure is Asymmetric
   c) Constant order with multiple level of decomposition
4. Spectral splitting of recorded speech signal by filter designed in step 3.
5. Obtain a desired number of critical bands with varying bandwidth.
6. Design of Dyadic Synthesis Filter bank with
   d) Wavelet transform with different families
   e) Tree Structure is Asymmetric
   f) Constant order with single level of reconstruction
7. Reconstruction of obtained bands in Even-Odd Index
8. Interpolation by factor I where I=D
9. Convert reconstructed bands to Right and left speech signal
10. Dichotic presentation of processed right and left speech signal to Right-Left ear.
11. Obtain the Hearing test results with Parameters
    a) Recognition score
    b) Recognition rate
    c) Recognition time

A. Overview of Wavelets

The wavelet transform uses a variable width window (wide at low frequencies and narrow at high frequencies) which allows focusing on very short duration high frequency phenomena like transients in signals (Burrus *et al*., 1998). In this chapter, wavelet properties that influence the type of wavelet basis functions that is appropriate for a particular application is examined. Some of the uses of wavelets in speech processing are reviewed.

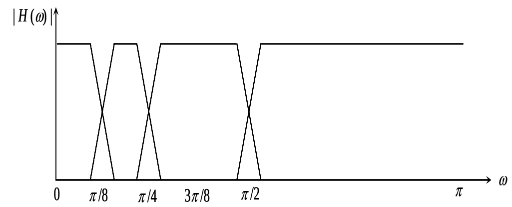

Fig.2 Magnitude Spectrum of Spitted Bands

The discrete wavelets transform (DWT) results in a logarithmic frequency resolution. High frequencies have wide bandwidth whereas low frequencies have narrow bandwidth. Wavelet packets (WP) allow for the segmentation of the higher frequencies into narrower bands (Burrus *et al*., 1998). This section discusses the full wavelet packet decomposition, and the coarsest level will be designated by the highest numerical level [9].

B. Full wavelet packet decomposition

In the DWT decomposition, to obtain the next level coefficients, scaling coefficients (low pass branch in the binary tree) of the current level are split by filtering and down sampling. In the decomposition scheme, the first stage splits the spectrum into two equal bands: one high pass and the other low pass. In the second stage, a pair of filters splits the low pass spectrum into lower low pass and band pass spectra. This splitting results in a logarithmic set of bandwidth shown in Figure 2.

V. CONCLUSION

This project aims at improving speech perception for hearing impaired individuals by using the technique of masking of two sound frequency. The results of the project is judged on the basis of three factors; Recognition score, Recognition rate, Recognition rate.

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