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Dynamic Frequency Filtering based on Analysis of Hearing Impairment

[1] Padmaja Kuruba, [2] Tilak Kumar L
[1] Global Academy of Technology, Bangalore, Karnataka

Abstract: - **Worldwide more than 10% of the people are suffering from Hearing impairment. The statistical report shows more than 20 crores of people are subjected to various levels of hearing problems based on various reasons depending on the surrounding environment. Hearing impairment is seen in all age groups from infants to adults. In this paper, we propose a digital hearing system which mimics the biological ear. The system is designed to produce the required sound based on the patient impairment level, using digital signal processing techniques. This involves noise reduction, the frequency selected filtering & amplification and compression such that bouncing of the frequencies is avoided. The proposed work is simulated using MATLAB where the following parameters are tested, namely 1) reduction of white Gaussian Noise 2) increase in the gain of specific frequencies 3) dynamic shaping of the signal amplitude to reduce bouncing of the signal.**

Keywords: **Hearing Impairment, Gaussian Noise, Filtering.**

I. INTRODUCTION

Today, a major part of the human population suffers from hearing loss. The main complaint of people with hearing loss is low ability to deduce speech in noisy environments. So, by using the Digital Signal Processing (DSP), it enhances the possibility of performing signal-to-noise ratio. A hearing aid is an electronic, battery operated device that amplifies and changes sound to allow for improved communication. Hearing aids receive sound through a microphone, which then converts the sound waves into electrical signals. The amplifier increases the loudness of the signals and then sends the sound to the ear through a speaker. A microphone is used to collect the sound and convert it into electrical impulses. Thus, reproduces the rise and fall of pitch of the sound (high or low) and the intensity (loudness measured in decibels). An amplifier, amplifies the electrical impulses, makes sounds louder. It has an integrated circuit comprising of several transistors and integrated circuits. Earphone converts the amplified signal into sound and feeds them into the ear. The audio frequency range is generally between 20 Hz to 20 kHz which is capable to hear. The human ear is only sensible to hear the frequency range between 1 kHz to 4 kHz. So below 1 kHz, ear will not respond and above the 4 kHz, it may damage the hearing capability. The general structure of a Hearing Aid is as shown in the Fig. 1.

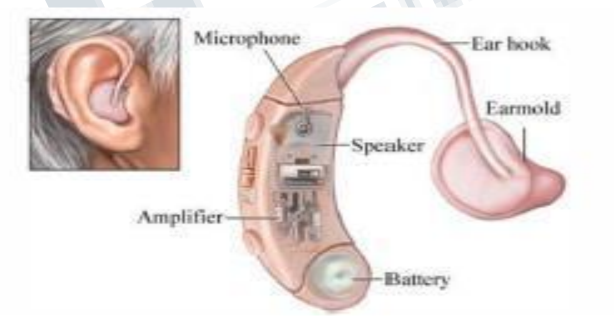

Fig. 1 General Structure of a Hearing Aid

The categories of hearing losses are conductive and sensor neural. Hearing loss can also be attributed to a combination of both types, a mixed hearing loss. Conductive hearing loss is caused by any obstruction that prevents sound waves from reaching the inner ear. Some of the causes of conductive hearing loss include an accumulation of earwax, a collection of fluid in the middle ear or due to infections. Sensorineural hearing loss refers to problems in the cochlea or the auditory nerve. Most are due to deterioration of the tiny inner or outer hair cells. This accounts for 90% of permanent hearing losses and although it may be a natural part of aging. Other causes can include due to head injury, certain medical treatments such as chemo and radiation therapy, genetic predisposition. Sensor neural hearing losses cannot currently be corrected medically. With the help of Audiogram, hearing can be plot on graph or in other words audiogram

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may define as, a graph that shows the audible threshold for standardized frequencies as measured by an audiometer .The horizontal axis of audiogram represent frequency in Hz, vertical axis indicates the amplitude in db. A spectrogram is a representation of how the frequency content of a signal changes with time. Time is displayed along the x-axis, frequency along the y-axis. At the end of a hearing test, hearing levels decide the degree of hearing loss. Hearing loss is measured in decibels hearing level (dB). If person can hear sounds across a range of frequencies at -10dB to 26dB it a will considered as having normal hearing. The threshold for the different types of hearing loss is as shown in the Table I.

Table I. Threshold for different types of Hearing Loss

Types of Hearing Loss	Threshold (Hearing Level)
Normal Hearing	-10dB to 26dB
Mid Hearing Loss	27dB to 40dB
Moderate Hearing Loss	40dB to 70dB
Severe Hearing Loss	70dB to 90dB
Profound Hearing Loss	Greater than 90dB

II. METHODOLOGY

The input speech signal takes the form of human voice. An adjusted output speech signal which can be audible to the hearing impaired person is produced by passing the input speech signal through several functions i.e., noise addition, noise reduction filter, frequency shaper and amplitude compression. Fig. 2 shows the block diagram for the MATLAB implementation of Digital Hearing Aid System.

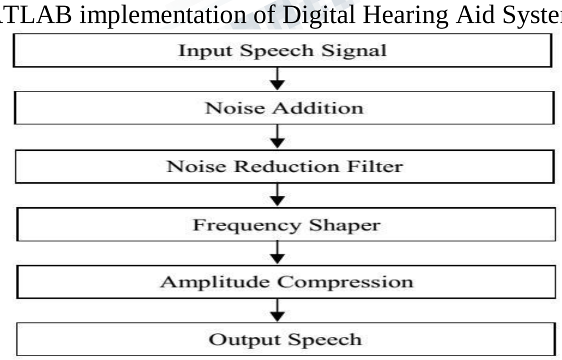

Fig. 2 Block diagram representation of the hearing aid system.

A. Noise Addition

A clean signal is considered as the input speech signal for this system and some noise such as Additive White Gaussian Noise (AWGN) and random noise is added to it to model a real situation. This noise (AWGN) has a continuous and uniform frequency spectrum over a specified frequency band and has equal power per Hertz of this band. It consists of all frequencies at equal intensity and has a normal (Gaussian) probability density function.

B. Noise Reduction Filter

To eliminate the noise, a reduction filter function is used in this design. The noise in the signal is suppressed by using the wavelet filter function.

C. Frequency Shaper

The hearing aid amplifies all signals rather than the significant signal that they desire to hear. Hence, most hearing impaired has difficulty to hear high frequency signals. Therefore, to correct the loss of hearing at certain frequencies, the frequency shaper is used. It is designed in such a way that it applies high gain for higher frequencies and vice versa.

D. Amplitude Compression

The task of controlling the overall gain of a speech amplification system is the function of Amplitude compression. Amplitude compression ensures that the amplified signal should not exceed saturation power.

III. IMPLEMENTATION AND SIMULATION

MATLAB tool is used to write the code. Firstly, the input speech signal is loaded with parameters such as sampling frequency and the number of bits of that signal. Then, Additive White Gaussian Noise (AWGN) and random noise are added to the signal and then processed by various MATLAB functions. Finally, an output which is audible to the hearing impaired person is obtained. To run the demo successfully, it is needed to input all the parameters which include maximum gain to be applied and saturation power.

Equations and Formulae Used For FFT:
The resulting sequence is interpreted as follows:

$$X[K] = \sum_{n=0}^{N-1} x(n) e^{-\frac{j2\pi kn}{N}} \quad k = 0,1,\ldots.\ N-1 \quad (1)$$

The matrix obtained after performing this function contains frames of the original speech signal. This signal is filtered by hamming filter and transformed with FFT. By calculating DFT we can obtain the magnitude spectrum.

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IV. RESULTS AND DISCUSSION

The original input speech signal which is plot on time versus amplitude axis is shown in the Fig. 3.

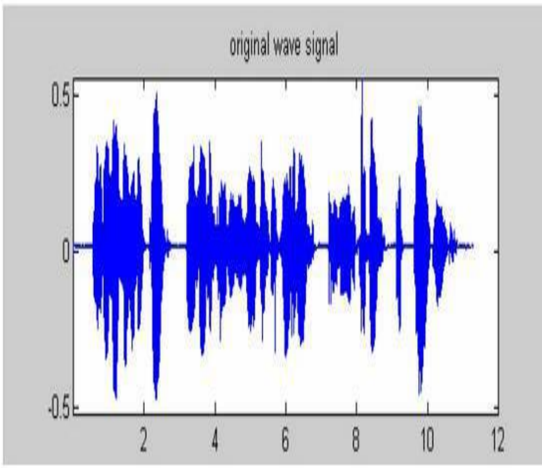

Fig. 3 Original Input Signal

Next, Additive White Gaussian Noise is added to the original wave signal. The purpose of this addition is just to simulate noises in the real life situation. Fig. 4 shows the signal after noise addition.

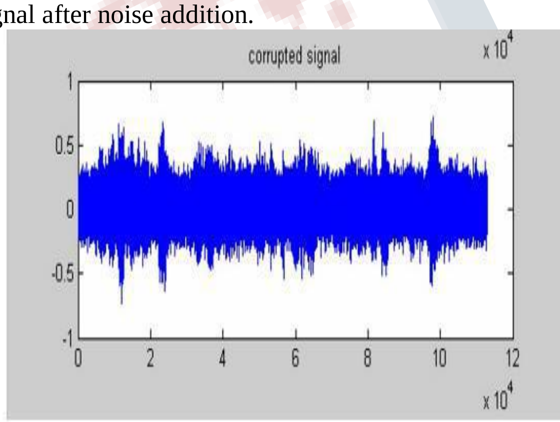

Fig. 4 Corrupted Speech Signal

Afterwards, the de-noising process takes place which removes most of the noise in the signal as shown in the Figure. Comparing the spectrograms of the original signal and the filtered signal, we can see that the amplitude of the noise in the signal was noticeably reduced which is shown in the Fig. 5.

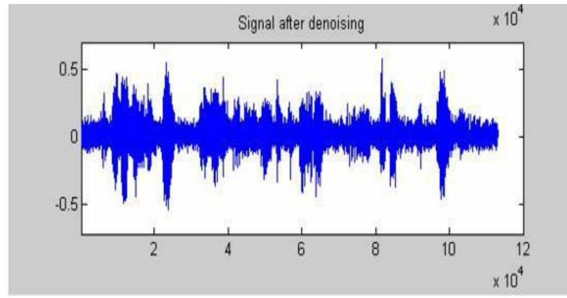

Fig. 5 Signal after Denoising

The output display layout before execution and after execution is shown in the Fig. 6 and Fig. 7 respectively.

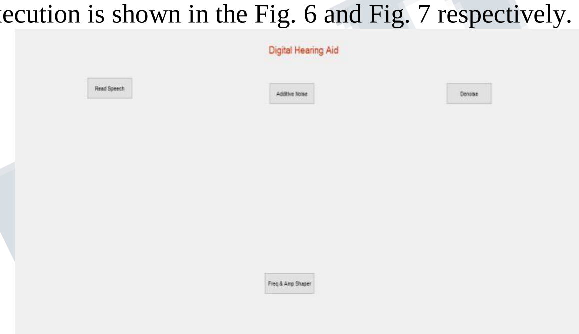

Fig. 6 The Output Display Layout Before Execution

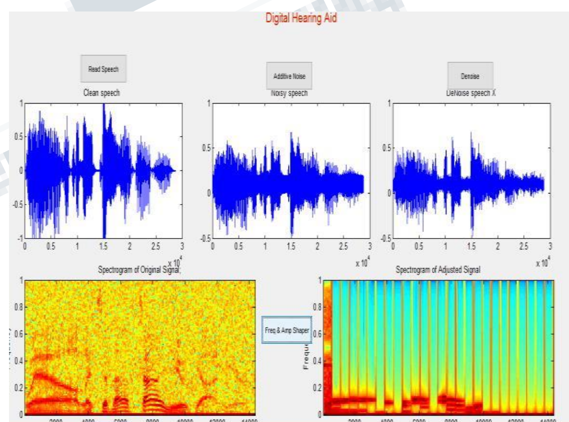

Fig. 7 The Output Display Layout After Execution

V. CONCLUSION

The newer digital aid is more capable of fine-tuning the sound without distorting the quality. In this digital hearing aids system implementation using MATLAB, sound processing is digitalized. Thus, it is possible to refine the sound signal, for instance by reducing noise and improving speech signals. In addition, by using digital technology, the amplification can be done only at the frequencies that the user needs to amplify. This will eliminate the problems with conventional amplifier which amplified the whole signal including the noise. In general, digital hearing aid converts the incoming signals to digital signals. This digitalization makes it possible to precisely analyze & filter the signals. The signals can be processed in one or more frequency channels. At the end, the digital signal is again converted to its analog form. The benefits of using digital aids can improve quality of life by improving sound quality.

VI. FUTURE SCOPE

In recent news it appears that Fuel Cell hearing aids may well be close to ready for the market. It has blown hot and cold since then. Fuel cells are hydrogen powered power sources, because hydrogen needs to be contained in a pressurised container, most of the research has been undertaken on Methanol. Methonal is Hydrogen rich but it does not need a compressed container. In theory you could recharge your hearing aids in the same way you charge a cigarette lighter. It could be done in thirty seconds and you would have up to three days of power.

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