Digital Filtering In Hearing Aid System For The Hearing Impaired

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***Abstract*—A great proportion of human population suffers from hearing loss. Hearing loss is a measure of shift in auditory system compared to that of a normal ear for detection of a pure tone. But with the availability of modern day technologies and the recent developments in signal processing area, sophisticated artificial hearing aid systems can be designed that relax the job of damaged auditory systems to a great extent and make much of the sound available to the hearing impaired. In this project, the simulation of the simple digital hearing aid was developed using MATLAB programming language. The implementation of this configurable digital hearing aid (DHA) system includes noise reduction filter, frequency-dependent amplification and amplitude compression. We tested our filters on a mock patient and successfully reduced white Gaussian noise, increased the gain for frequencies which were difficult to hear, and shaped the amplitude to prevent any of the frequencies from becoming too loud. Finally, future trends and expected innovations in the hearing aid industry are discussed.**

***Keywords—Hearing aid, Gaussian Noise, Signal Processing, Fourier transform, Signal to Noise Ratio*.**

I. INTRODUCTION

One of the most important issues for human being is aid in hearing. They are actually small electronic instruments which make sound louder and make speech easier to hear and understand. It has been designed to pick up sound waves with a microphone, change weaker sounds into louder sounds and send them to the ear through a speaker. With the microchips available today, hearing aids have become smaller and smaller and have significantly improved in quality. Roughly 10% of the world population bears from some hearing loss [1]. However, only a portion uses hearing aid. This is due several factors which include the stigma associated with wearing a hearing aid, customer dissatisfaction and the cost associated with the new versions of hearing aids. Hearing loss is typically measured as the shift in auditory threshold relative to that of a normal ear for detection of a pure tone [2]. There are many types of hearing aids with a wide range of functions and features to address individual needs. The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing impaired person. *Table 1* shows different levels of hearing loss for a person [2]. Basically, all hearing aids use the analogue technology for the treatment of sound. Improvements have been made by using the development of digital sound treatment for the efficiency of hearing aids. Nowadays, the digital hearing aids are small, which can be hidden inside the ear and have an almost perfect sound reproduction. The research of digital hearing aids has begin its growth and now a small programmable computer that is capable in amplifying millions of different sound signals has been fitted in the devices, thus improving the hearing ability of hearing impaired people [3]. The first digital hearing aids were launched in the mid 80's, but these early models were slightly impractical. After ten years, the digital hearing aids really became successful, with small digital devices placed either inside or discreetly behind the ear [4]. Today, digital technology is very much a part of daily life. Most households have a variety of digital products, such as telephones, video recorders and personal computers. Hearing aids was also benefited by the emergence of digital technology. Among the advantages of digital signal processing is that it allows hands free operation. The hearing aid automatically self adjusts the volume and pitch. It performs thousands of adjustments per second which results in reduced background noise, improved listening in noisy situations, sound quality and multiple program settings. The user can switch between varieties of programs for different listening situations.

TABLE I. DIFFERENT DEGREES OF HEARING LOSS [2]

Level	Different Degrees of Hearing Loss		
	Classification of hearing loss	***Lower limit***	***Upper limit***
1	Normal hearing	-10dB	26dB
2	Mild hearing loss	27dB	40dB
3	Moderate hearing loss	40dB	70dB
4	Severe hearing loss	70dB	90dB
5	Profound hearing loss	90dB	_

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II. METHODOLOGY

Fig. 1 is a block diagram for the MATLAB implementation of Digital Hearing Aid System. The input speech signal takes the form of human voice. The input speech signal will pass through several functions i.e., noise addition, noise reduction filter, frequency shaper and amplitude compression before producing an adjusted output speech signal which is audible to the hearing impaired person.

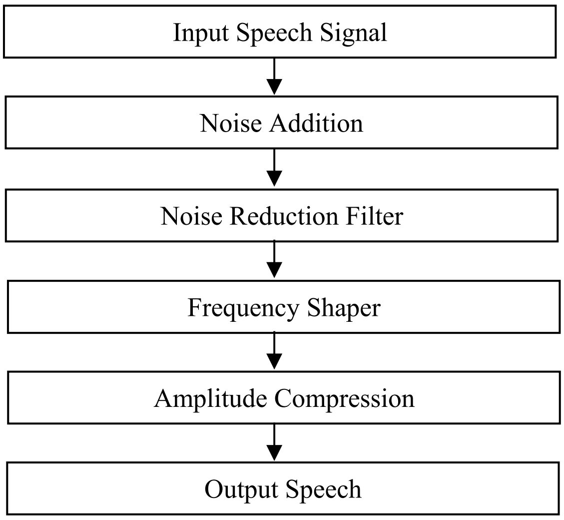

Fig. 1. Block diagram representation of the hearing aid system.

A. Noise Addition

Since the input speech signal for this system is a clean signal, some noise is added in order to simulate a real situation. In this system, the Additive White Gaussian Noise (AWGN) and random noise are added to the input speech signal by using MATLAB function. The noise (AWGN) has a continuous and uniform frequency spectrum over a specified frequency band and has equal power per Hertz of this band [5]. It consists of all frequencies at equal intensity and has a normal (Gaussian) probability density function.

B. Noise Reduction Filter

A major anxiety for the people with hearing loss is the capability of hearing aid to differentiate intended speech signal in a noisy environment. Hence, to eliminate the noise, a reduction filter function is used in this design. To suppress the noise in the signal, the wavelet filter function is used.

C. Frequency Shaper

One major complaint of hearing aid users is that the hearing aid amplifies all signals rather than the significant signal that they desire to hear. Most hearing impaired have difficulty to hear high frequency signals. Therefore, the frequency shaper is designed to correct for the loss of hearing at certain frequencies. It applies high gain for higher frequencies and vice versa [2].

D. Amplitude Compression

Fundamentally, amplitude compression function is the task of controlling the overall gain of a speech amplification system. Amplitude compression will ensure that the amplified signal will not exceed saturation power [6].

III. IMPLEMENTATION AND SIMULATION

The code, written in MATLAB, loads the input speech signal, takes the sampling frequency and the number of bits of that signal. Then, Additive White Gaussian Noise (AWGN) and random noise are added to the signal before they are processed by various MATLAB functions to get an output which is audible to the hearing impaired person. For the analysis purpose, a sample of speech signal is selected. The sample is "I am Ritwik Dhawan". This signal is added by Additive White Gaussian Noise (AWGN) and random noise. To run the demo successfully, it is needed to input all the parameters which include maximum gain to be applied, saturation power and four frequency values where the gain changes. The patient suffers moderate hearing loss characterized by [2]:

- Threshold of Hearing at 40 dB
- Threshold of Pain at 110dB
- Saturation level(P_{sat}) of 90dB
- Difficulty to hear high frequencies

Equations and Formulae Used

For FFT:

The resulting sequence is interpreted as follows:

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-j2\pi kn/N}, k = 0, 1, \dots. N-1 \qquad (1)$$

The zero frequency corresponds to n=0, positive frequencies $0<f<Fs/2$ correspond to values $1 \le n \le N/2-1$ while negative frequencies correspond to $N/2+1 \le n \le N-1$ [7].

Here, Fs denotes sampling frequency which is 22050 Hz here. The matrix obtained after performing this function contains frames of the original speech signal filtered by hamming filter and transformed with FFT. The elements of the matrix are complex numbers and symmetrical because FFT was used to transform. By calculating DFT we can obtain the magnitude spectrum.

For AWGN:

$$Y=s+10^{\wedge}(-Eb\_N_0\_dB/20)*n; \qquad (2)$$

where 's' is the transmitted sequence, *Eb_N*$_0$*_dB* is the SNR and 'n' is the Additive White Gaussian Noise.

IV. RESULTS AND DISCUSSION

First we make a frequency shaper function which has gain of the speech signal been modified on specific frequency range as per the user requirements as in *Fig. 2*.

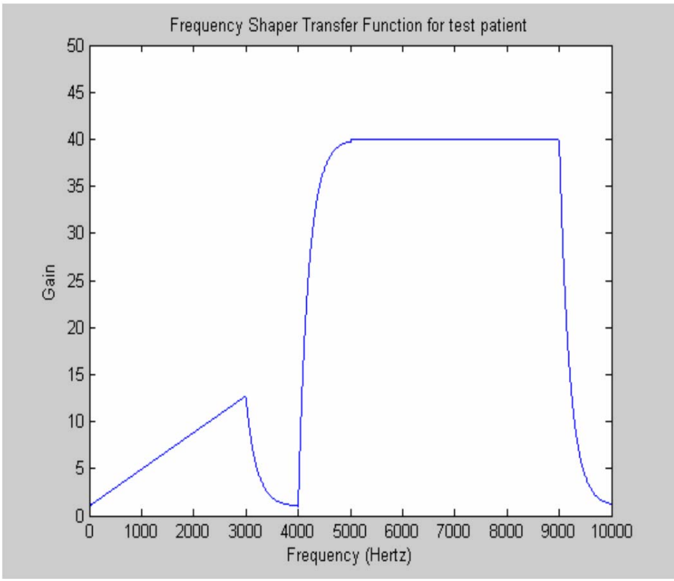

Fig. 2. Frequency Shaper Transfer Function

Fig. 3 is the original speech signal which is plot on time versus amplitude axis.

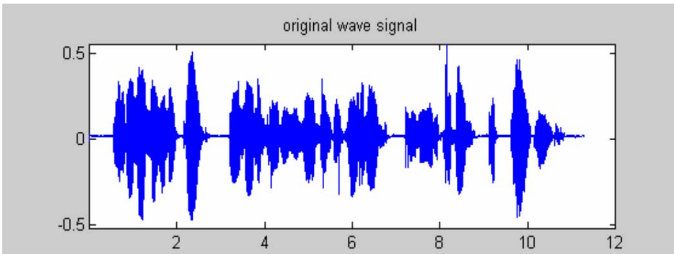

Fig. 3. Input speech signal

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.

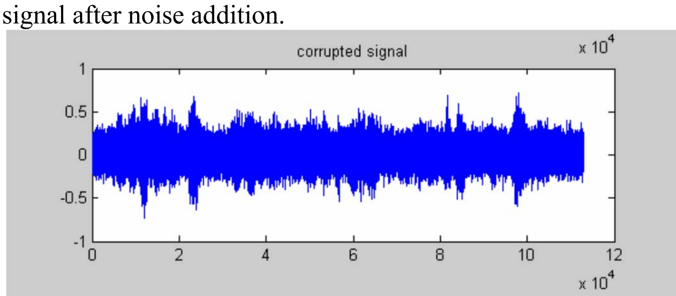

Fig. 4. Corrupted speech signal

Afterward, the de-noising process takes place which removes most of the noise [8] in the signal as shown in *Fig. 5*. *Fig.6* shows all the outcomes plotted in a single window. 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 in *Fig. 7*.

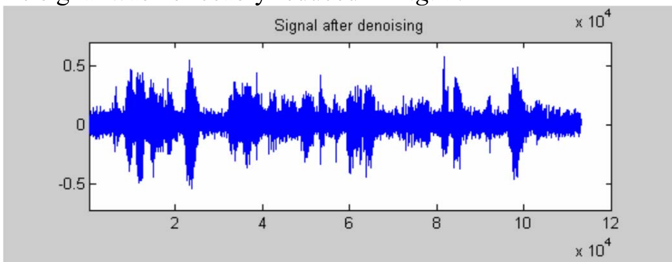

Fig. 5. Signal after denoising

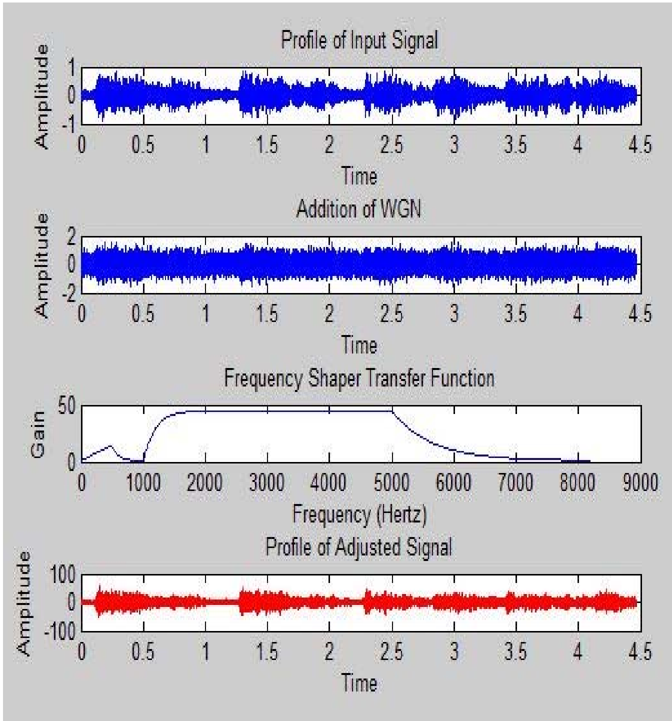

Fig. 6. Representation of Frequency Shaper Transfer Function

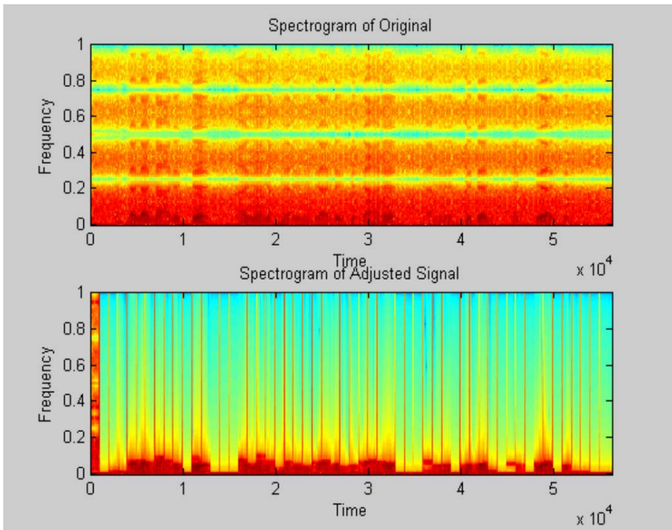

Fig. 7. Spectrogram of Original and Adjusted Signals

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 its 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.

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