IJCRT.ORG ISSN : 2320-2882
INTERNATIONAL JOURNAL OF CREATIVE RESEARCH THOUGHTS (IJCRT)
An International Open Access, Peer-reviewed, Refereed Journal

DIGITAL HEARING AID SYSTEM

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Abstract: Traditional analog hearing aids is same as simple radio. They can be calibrated and adjusted for volume, bass and treble. But hearing loss is not just a technical loss of volume. Rather, hearing scantiness can intensify sensitivity and reduce tolerance to certain sounds while diminishing sensitivity to others. For instance, digital technology can tell the difference between speech and background noise, permitting one in while filtering out the other. More or less 10% of the world's population suffers from some type of hearing loss, yet only a small percentage of this statistic use a hearing aid. The stigma connected with wearing a hearing aid, customer disappointment with hearing aid performance, and the cost related with a high-performance solution are all causes of low market penetration. Through the use of digital signal processing, digital hearing aid now gives what the analog hearing aid cannot give. It suggests the possibility of performing signal-to noises strengthen, pliable gain-processing, digital feedback reduction, etc. In this paper, the simulation of simple digital hearing aid was developed using MATLAB programming language. The implementation of this configurable digital hearing aid system includes the noise reduction filter, frequency shaper function, and amplitude compacting function. This digital hearing aid system is design to adapt for mild and moderate hearing loss patient since different gain can be set to map different levels of hearing loss.

Index Terms **- Hearing aid, Gaussian Noise, Signal Processing, Fourier transform, Signal to Noise Ratio.**

1. INTRODUCTION

Hearing Aids systems are one of the most important issues for human being. They are a small electronic instrument which makes sound louder and makes speech easier to hear and understand. The hearing aid is designed to pick up sound waves with a tiny microphone, change weaker sounds into louder sounds and send them to the ear through a tiny speaker. With the microchips available today, hearing aids have gotten smaller and smaller and have significantly improved quality. Roughly 10% of the world population bears from some hearing loss. However, only a portion uses hearing aid. This is due several factors which include the stigma associated with wearing a hearing aid, customer dissatisfaction with the devices not meeting their expectations, and the cost associated with the new digital 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. This is why there are many types of hearing aids with a wide range of functions and features to address individual needs. Table 1 shows the classification of degrees of Hearing Loss. A hearing aid is an electronic device that makes sounds louder and can help to offset hearing loss. The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing-impaired person.

Basically, all hearing aids were using 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[1]. 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 been growth and now a small programmable computer that are capable in amplifying millions of different sound signals had been constructed in the devices, thus improving the hearing ability of hearing-impaired people. The first digital hearing aids were launched in the mid 80's, but these early models were slightly unpractical. After ten years later, the digital hearing aids really became successful, with small digital devices placed either inside or discreetly behind the ear. 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[2]. Hearing aids also was benefited from the emergence of digital technology. Among the advantages of digital Signal Processing that allows hands free operation. The aid automatically adjusts the volume and pitch on its own. 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.

Classification of Hearing Loss	Hearing Level
Normal Hearing	-10 dB to 26 dB
Mild Hearing Loss	27 dB to 40 dB
Moderate Hearing Loss	40 dB to 70 dB
Severe Hearing Loss	70 dB to 90 dB
Profound Hearing Loss	Greater than 90 dB

Table 1: Different Degrees of Hearing Losses

2. 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.

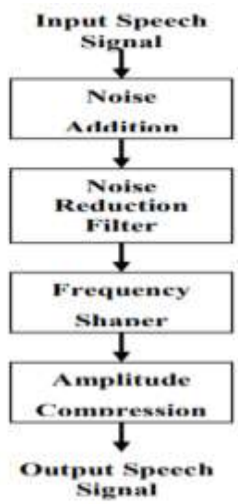

Fig.1: Block Diagram

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. It consists of all frequencies at equal intensity and has a normal (Gaussian) probability density function[3].

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 has difficulties to hear high frequency signal. Therefore, the frequency shaper is designed to correct for loss of hearing at certain frequencies. It applies high gain for higher frequencies and vice versa. The typical frequency shaper transfer function is shown in fig.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. Saturation power is where the sound signal begins to become uncomfortable.

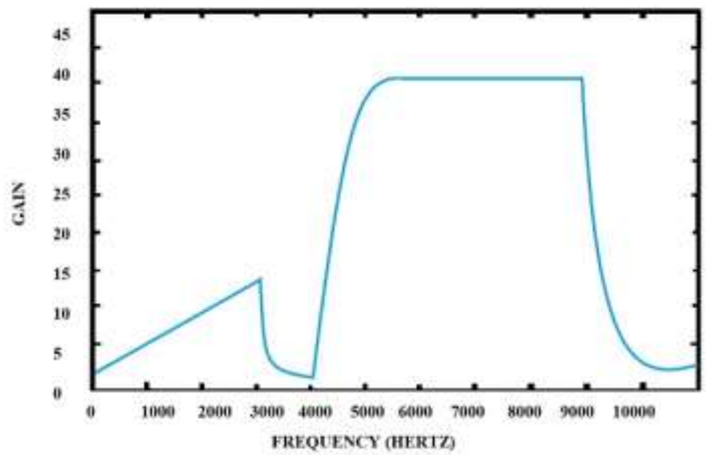

Fig.2: Typical Frequency Shaper

3. 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 Vishnu". 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 figure 3 shows the flow char of the program. The patient suffers moderate hearing loss characterized by:

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

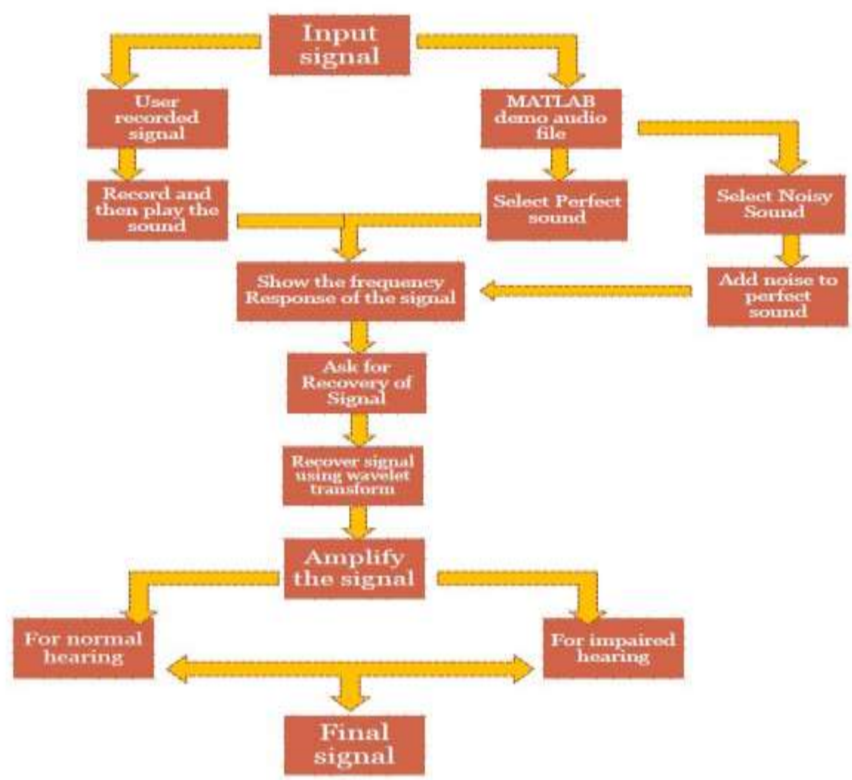

Fig.3: Flow Chart

4. 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. 4 is the original speech signal which is plot on time versus amplitude axis.

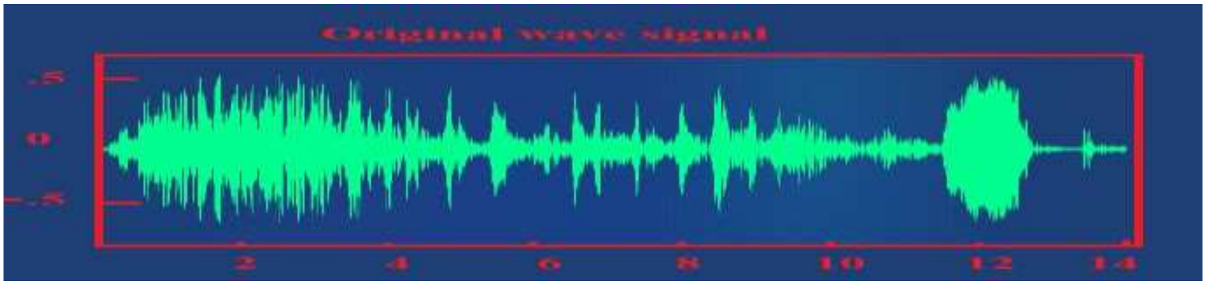

Fig.4: Input Speech Signal

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. 5 shows the signal after noise addition.

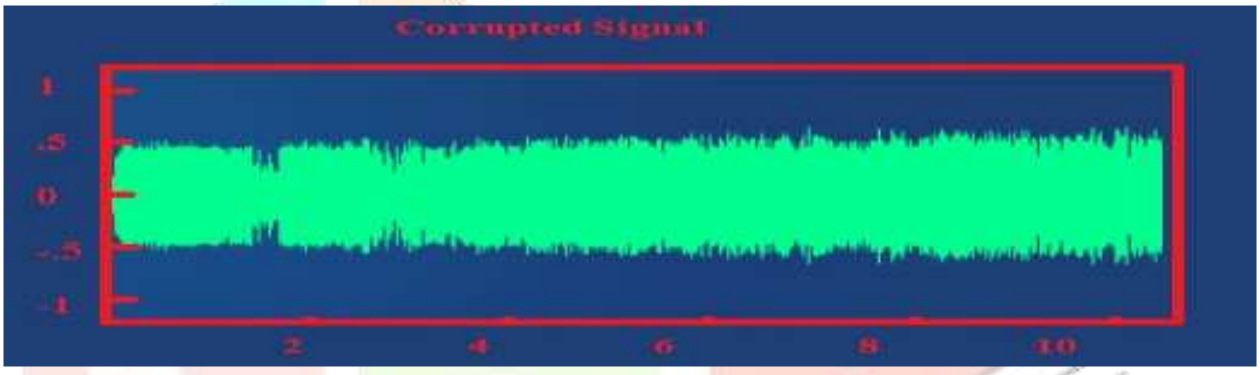

Fig.5: Corrupted Signal

Afterward, the de-noising process takes place which removes most of the noise in the signal as shown in Fig. 6. Fig.7 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. 8.

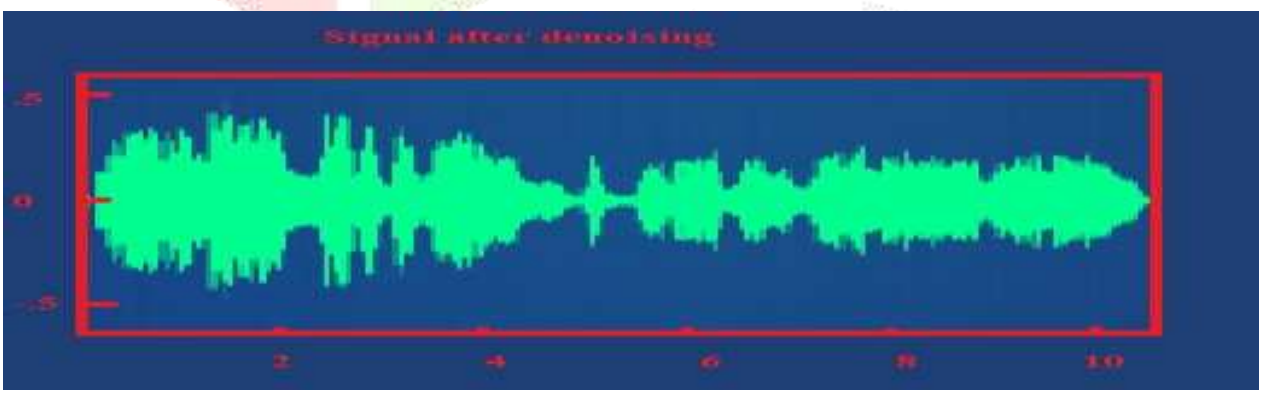

Fig.6: Signal After Denoising

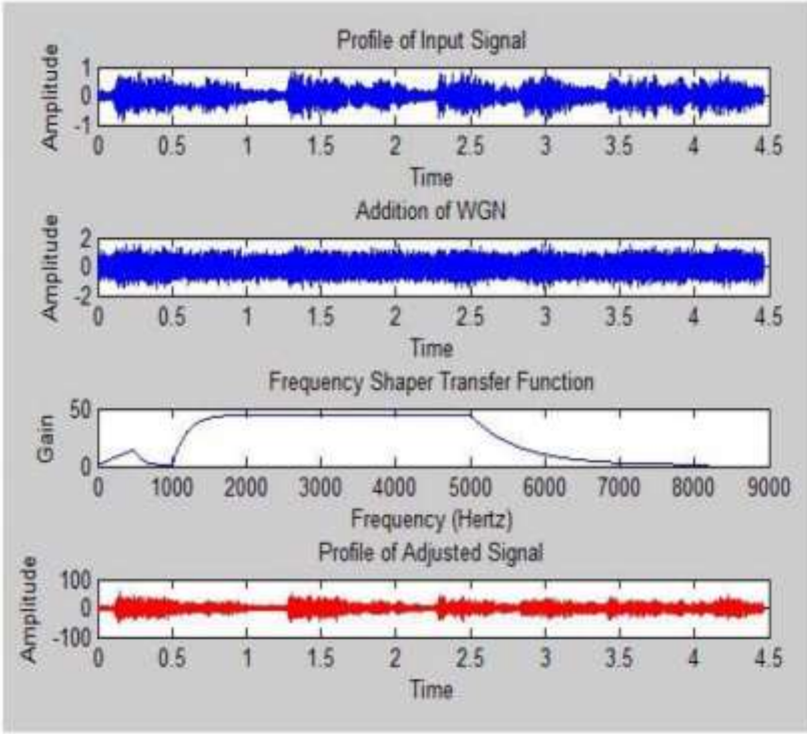

Fig.7: Representation of Frequency Shaper Transfer Function

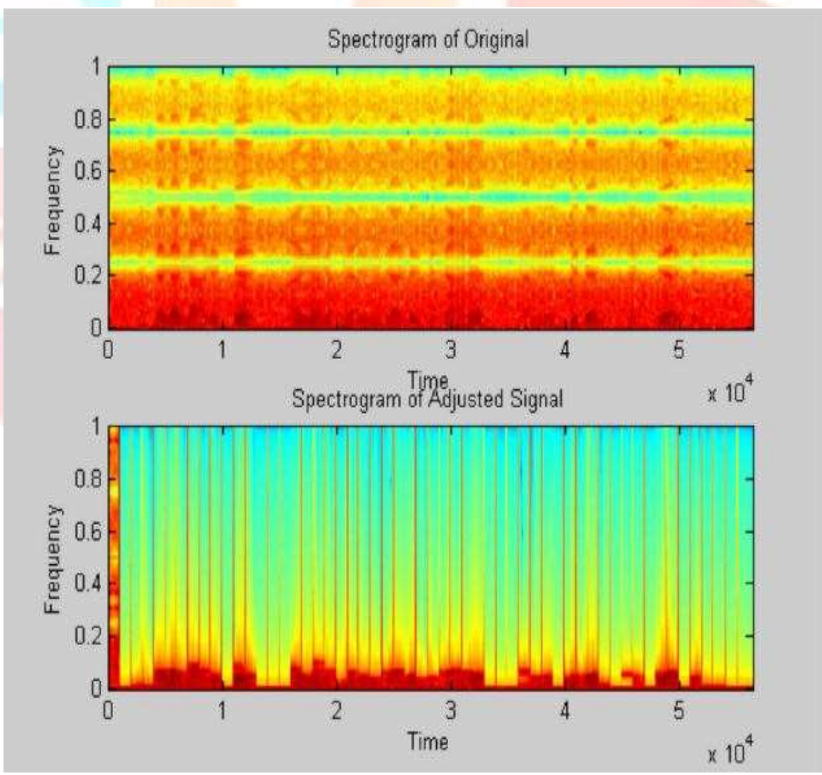

Fig.8: Spectrogram of Original and Adjusted Signals

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