

DIGITAL HEARING AID SYSTEM USING MATLAB

Submitted in partial fulfilment of the requirements for the award of

Bachelor of Engineering Degree in

Electrical and Electronics Engineering

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SATHYABAMA

INSTITUTE OF SCIENCE AND TECHNOLOGY

(DEEMED TO BE UNIVERSITY)

Accredited with Grade "A" by NAAC

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APRIL 2023

DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

BONAFIDE CERTIFICATE

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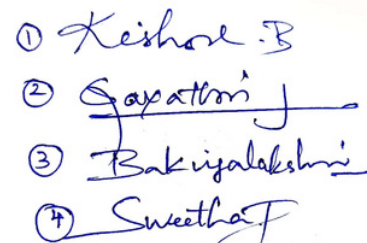
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ACKNOWLEDGEMENT

We are pleased to acknowledge our sincere thanks to Board of Management of **SATHYABAMA** for their kind encouragement in doing this project and for completing it successfully. We are grateful to them.

We convey our thanks to **Dr. N. M. Nandhita** Dean, School of Electrical and Electronics Engineering and **Dr.V.Sivachidambaranathan** , Head of the Department, Dept. of Electrical and Electronics Engineering for providing us necessary support and details at the right time during the progressive reviews.

We would like to express our sincere and deep sense of gratitude to our Project Guide **Dr.V.Meenakshi** for her valuable guidance, suggestions and constant encouragement paved way for the successful completion of our project work.

We wish to express our thanks to all Teaching and Non-teaching staff members of the Department of Electrical and Electronics Engineering who were helpful in many ways for the completion of the project

ABSTRACT

A great proportion of the human population suffers from hearing loss. Traditional analog hearing aid is same as simple radio. They can be calibrated and adjusted for volume, bass and treble. Hearing loss is not just a technical loss of volume. Hearing deficiency increases sensitivity and reduces tolerance to certain sounds. Hearing loss is a measure of shift in the 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, 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 is 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. Through this project there is- reduction of white Gaussian noise, increase in the gain for frequencies which were difficult to hear, and shape the amplitude to prevent any of the frequencies from becoming too loud.

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CHAPTER 1

INTRODUCTION

1.1 GENERAL BACKGROUND

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.

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. 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. 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.

1.2 LITERATURE REVIEW

McCormack et al, 2013 and Chisolm et al, 2007 Suggest that 80% of adults aged 55-74 years who would benefit from wearing a hearing aid do not acquire them, and that many who do acquire them do not wear them. Recent research in Australia indicates growth in hearing aid sales, but a decrease in usage. 15% of over 55 year olds in Australia own a hearing aid, but only 1 in 3 actually uses them. Hogan, 2016 suggest that aural rehabilitation that addresses the psychosocial impact of hearing impairment (e.g. through hearing therapy), as well as the functional impact (through fitting and use of a hearing aid /or other assistive technology), can reduce the perception of hearing problems and increase the use of communication strategies and use of the hearing aid.

Abrams et al, 2002 suggest that Participation in an aural rehabilitation programme, plus use of a hearing aid, provides more benefit and is more cost effective than a hearing aid alone. Kelly et al,2012 suggest that 154 older people already fitted with hearing aids, only 52% of hearing aid users reported receiving enough practical help post fitting, and only 41% reported receiving enough support. Approximately 40% reported not feeling confident in the use of their aids or their controls. Older people wanted more information than they received both before and after hearing aid fitting. Information provision and attention to the psychosocial aspects of care are key to enabling older people to adjust and optimise hearing aid benefit.”

Wolfe et al,2011 suggest that a typical audiological consultation often does not extend beyond hearing aid fitting and orientation often overlooking other useful and critical components of aural rehabilitation including adjustment to amplification; listening/communication strategies; self-management of hearing loss; environmental issues; issues related to work/educational settings; speechreading; emotional factors concerning hearing loss; impact on significant others; hearing protection - all topics which should be included in counselling of hearing aid candidates. Approximately 10% of the world's population suffer from some type of hearing loss, yet only a small percentage of this statistic use a hearing aid The stigma associated with wearing a hearing aid, customer dissatisfaction with hearing aid performance, and the cost associated with a high performance solution are all causes of low market penetration.

Current analog hearing aids yield significant limitations due to their inadequate spectral shaping, narrow operating bandwidth, and only partial noise-reduction capability. This leads to sub-optimal clarity and audibility restoration,

as well as sub-optimal speech perception in noisy environments. Analog hearing aids are hardware-driven and thus are difficult to customize to specific hearing problems. Digital hearing aids can solve these problems. They provide full bandwidth, fine grain spectral shaping, and enhanced noise reduction. As software-driven devices, they are very flexible and easily customizable to a user's needs.

1.3 SCOPE OF THE PROJECT

To provide digital technologies for auditory rehabilitation purposes . The number of research efforts that have direct linkages into the realization of commercially viable digital hearing aids far exceed the space available for this manuscript.

1.4 ADVANTAGES

- You may hear sounds that you have not previously heard.
- You may hear speech over the telephone more clearly.
- You may communicate more easily with family and friends.
- You may improve communication in noisy listening situations, such as at a restaurant or in a large group of people.

CHAPTER 2

THEORITICAL BACKGROUND OF HEARING AID SYSTEM

2.1 DSP METHODOLOGY

Digital signal processing (DSP) is a type of hearing aid that takes sounds that enter through the microphone and converts them into digitized codes. Sound is regularly analyzed with DSP. The sampling rate is the number of times the hearing aids analyze incoming sounds each second. The digitized codes are more accurate the more often they are analyzed each second.

Once sounds are broken down into digitized codes, they are further broken down into what is called bits. If codes weren't broken down into bits, similar sounds may be categorized as the same sound. For instance, two male voices may be categorized as identical sounds, even if they're just similar and not the same. The more bits, the more precise the hearing aids can differentiate between different, but similar, sounds.

From there, bits are separated even further into frequency channels. Each frequency channel can be fine-tuned to the user's particular form of hearing loss and their precise hearing needs. Once the process of converting sounds into codes and then breaking them into bits and channels is completed, the codes are combined again and put together as one. This allows the user to hear sounds from the receiver.

2.2 TYPES OF FILTERS USED

BAND PASS FILTER CIRCUIT

The band pass filter is a combination of low pass and high pass filters. Therefore, the circuit diagram contains the circuit of high pass and low pass filters. The circuit diagram of the passive RC band pass filter is as shown in the below figure.

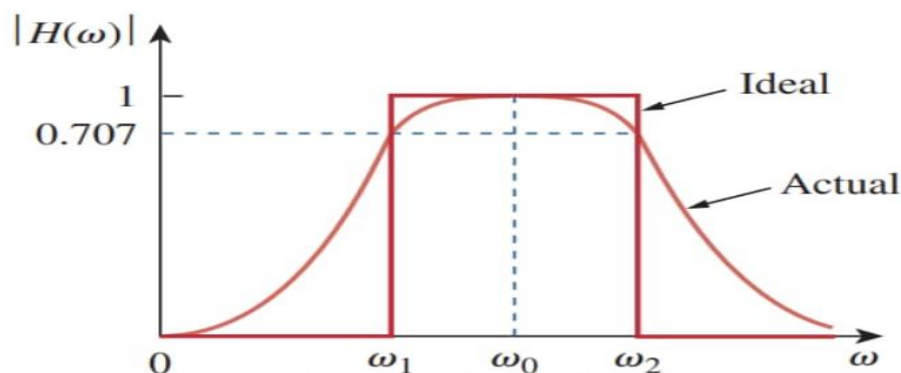


Figure 2.1. Band Pass Filter Waveform

LOW PASS FILTER

A low-pass filter is a filter that passes signals with a frequency lower than a selected cutoff frequency and attenuates signals with frequencies higher than the cutoff frequency. The exact frequency response of the filter depends on the filter design. The filter is sometimes called a high-cut filter, or treble-cut filter in audio applications. A low-pass filter is the complement of a high-pass filter.

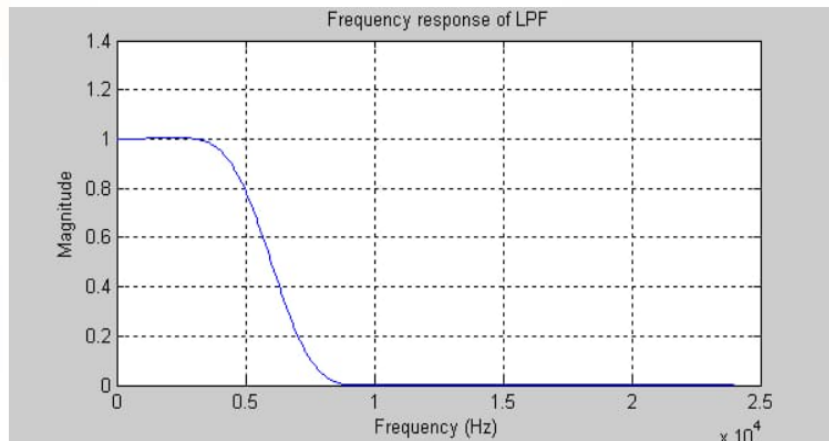


Figure 2.2.Low Pass Filter Waveform

CHAPTER 3

PROJECT DESCRIPTION

3.1 BLOCK DIAGRAM

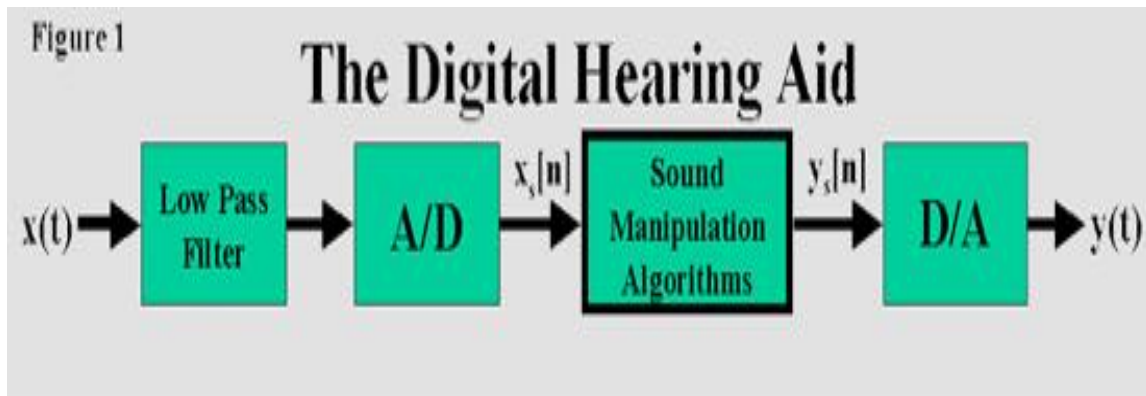


Figure 3.1 The Digital Hearing Aid

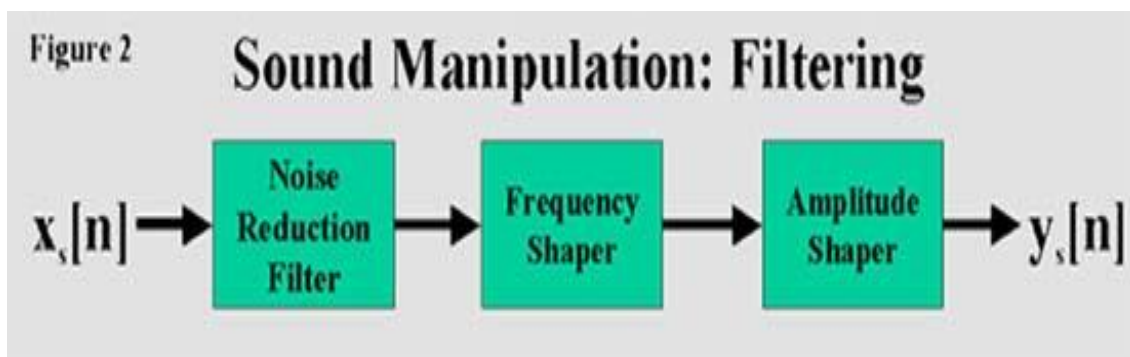


Figure 3.2 Sound Manipulation

3.2 PROJECT CONCEPT

Digital hearing aids are more common. They have all the features of analog programmable aids, but they convert sound waves into digital signals and produce an exact duplication of sound. Microchips in digital hearing aids analyze speech and other environmental sounds and store multiple program settings.

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

3.3 EXISTING MODEL

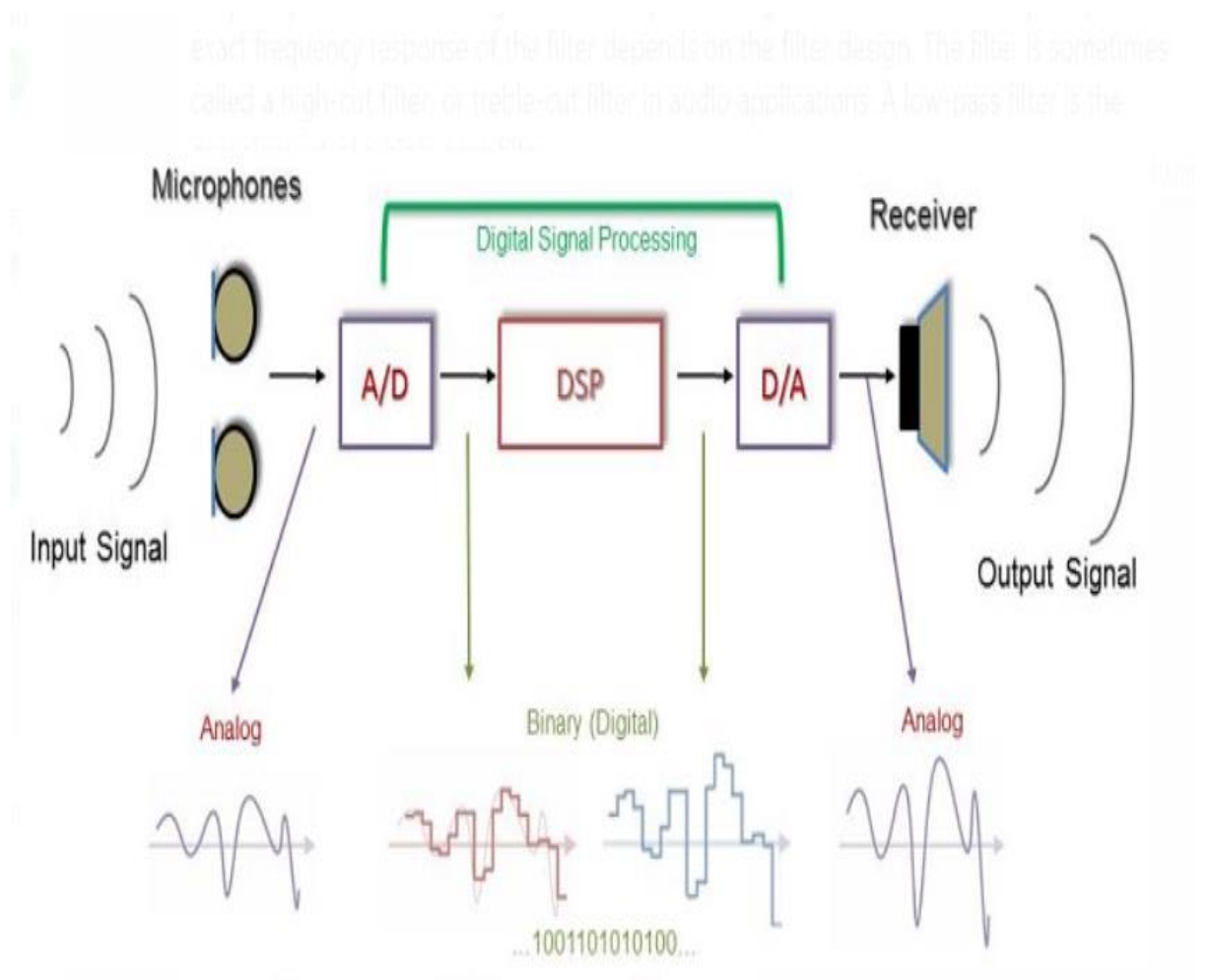


Figure 3.3 Existing Model

CHAPTER 4

PROPOSED SYSTEM

4.1 PROJECT DESIGN

The analog sound signal is converted into digital domain. The digital signal processor at the heart of a digital hearing aid manipulates the signal without causing any distortion, so sounds come through more clearly and speech is easier to hear and understand. The DHP combines crisp digital sound with totally hands-free operation, making it a logical choice compared to many of the other, more traditional solutions available.

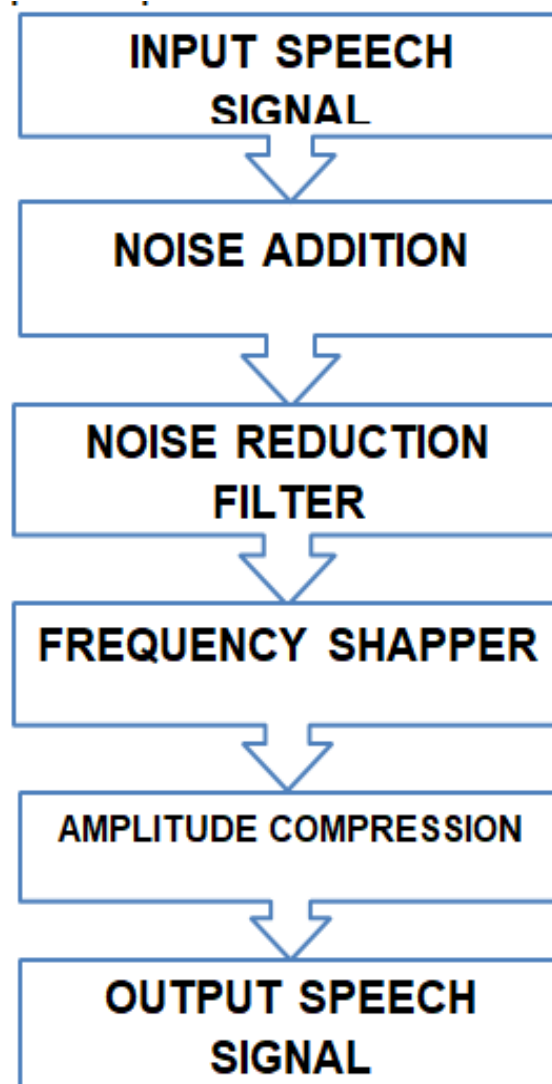


Figure 4.1 Project Design

Above 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.

4.2 WORKING

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. White Gaussian noise (WGN) 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. For example, a hiss or the sound of many people talking can be modeled as WGN. Because white Gaussian noise is random, we can generate it in MATLAB using the random number generator function, `randn`.

NOISE REDUCTION

In everyday situations, there are always external signals that may interfere with the sounds that the hearing aid user actually wants to hear. This ability to distinguish a single sound in a noisy environment is a major concern for the hearing impaired. For people with hearing loss, background noise degrades speech intelligibility more than for people with normal hearing, because there is less redundancy that allows them to recognize the speech signal. Often the problem lies not only in being able to hear the speech, but in understanding speech signals due to the effects of masking. To adjust for this loss, we developed a noise reduction filter in MATLAB for our hearing aid.

ADVANTAGES OF USING WAVELETS

Wavelets are nonlinear functions and do not remove noise by low-pass filtering like many traditional methods. Low-pass filtering approaches, which are linear time invariant, can blur the sharp features in a signal and sometimes it is difficult to separate noise from the signal where their Fourier spectra overlap. For wavelets the amplitude, instead of the location of the Fourier spectra, differ from that of the noise.

This allows for thresholding of the wavelet coefficients to remove the noise. If a signal has energy concentrated in a small number of wavelet coefficients, their values will be large in comparison to the noise that has its energy spread over a large number of coefficients. These localizing properties of the wavelet transform allow the filtering of noise from a signal to be very effective .

While linear methods trade-off suppression of noise for broadening of the signal features, noise reduction using wavelets allows features in the original signal to remain sharp. A problem with wavelet denoising is the lack of shift-invariance, which means the wavelet coefficients do not move by the same amount that the signal is shifted.

This can be overcome by averaging the denoising result over all possible shifts of the signal. This works very well and even overcomes pseudo-Gibbs phenomena that is often seen due to lack of shift invariance.

FREQUENCY SHAPER

The frequency shaper is designed to correct for loss of hearing at certain frequencies. We completely designed this filter ourselves. The filter applies a gain greater than one to the frequencies that the user has difficulty hearing. As one of its parameters, the filter takes in a vector of frequencies, determined by an audiologist, that define the user's hearing characteristics. For each range, the frequency shaper applies a certain gain based on the user's specific hearing loss. Thus, our frequency shaper is completely configurable to any user. We implemented our frequency shaper in MATLAB

AMPLITUDE SHAPER

Once the signal has been passed through the Noise Reduction Filter and the Frequency Shaper, it will be passed through our Amplitude Shaper. The dynamic range of hearing is measured in terms of sound pressure, in decibels. A normal hearing range extends from approximately 0 dB to 120 dB, where 0 dB is the Threshold of Hearing and 120 dB is the Threshold of Pain. Discomfort usually begins to occur around a saturation level of about 90 dB of sound.

Hearing loss compresses the range of hearing, raising the Threshold of Hearing and typically lowering the Threshold of Pain. For example, a person with moderate hearing loss would have a Threshold of Hearing around 40 - 70 dB and a Threshold of Pain around 100 dB. We assume that the Frequency Shaper raises the frequencies that the user has difficulty hearing to sound pressure levels within his dynamic range of hearing.

Therefore, all that our Amplitude Shaper has to do is check, bit by bit, that output power does not exceed a given saturation level, P_{sat} . Since noise is concentrated in the low power levels as well, the filter also removes a significant amount of noise. Output power is equal to zero for levels below P_{sat} . To implement this, algorithm, we created amplitude shaper in MATLAB.

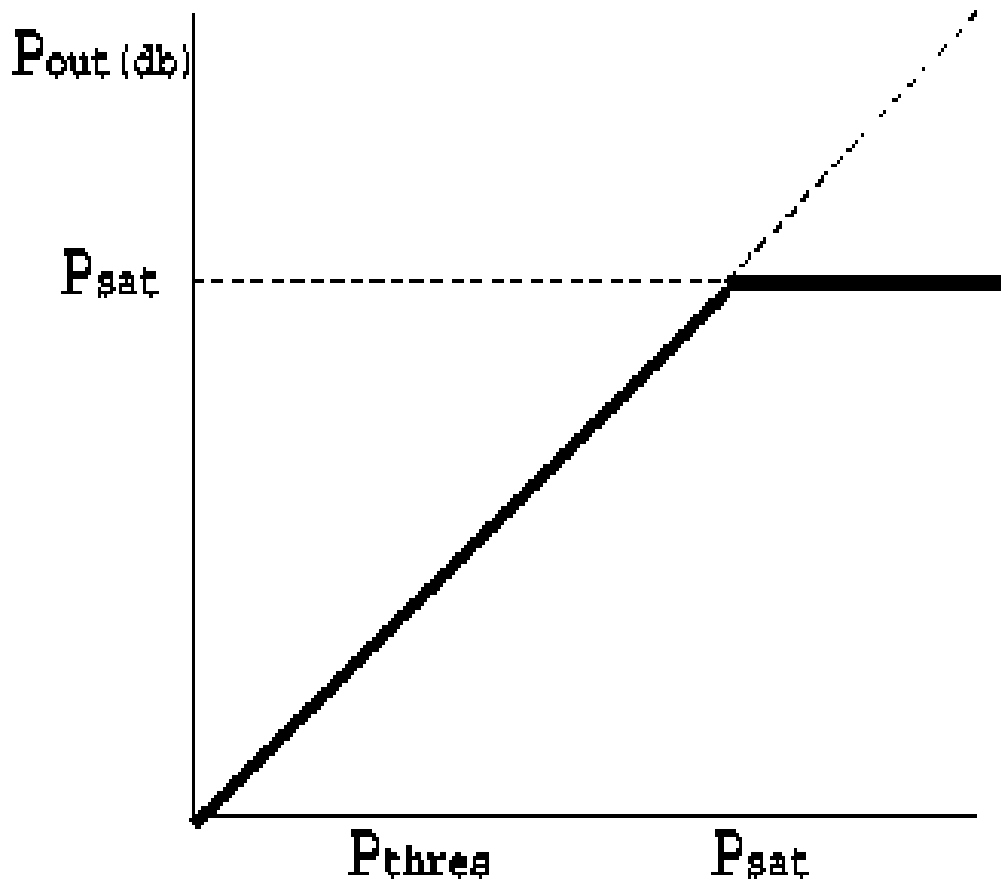


Figure 4.2 Amplitude Shaper

AMPLITUDE COMPRESSION

Amplitude compression will ensure that the amplified signal will not exceed saturation power. Fundamentally, amplitude compression function is the task of controlling the overall gain of a speech amplification. Saturation power is where the sound signal begins to become uncomfortable.

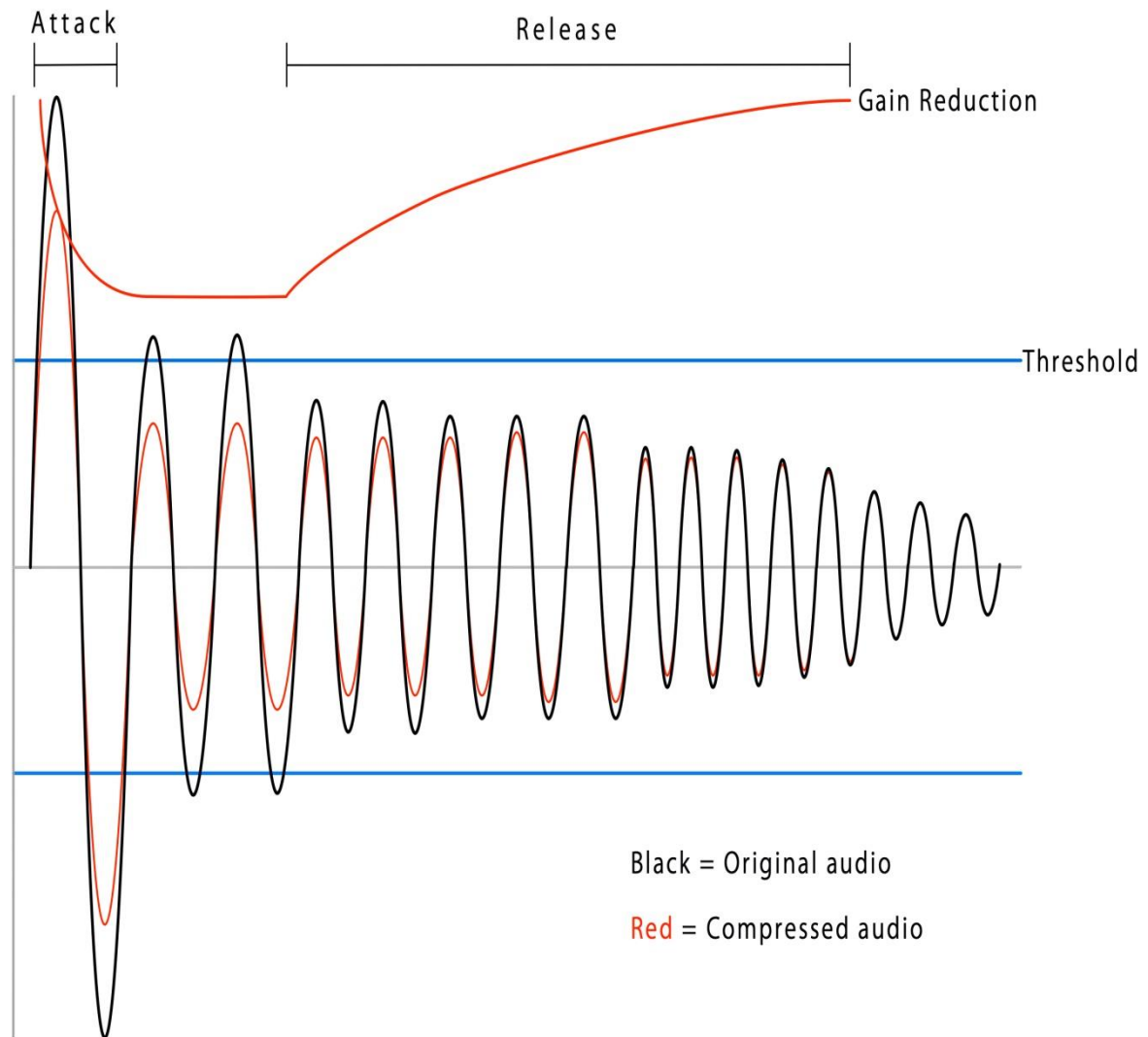


Figure 4.3 Amplitude Compression

TECHNIQUE USED

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.

Denoise: To eliminate the noise, a reduction filter function is used in this design. `ddencmp`: It gives default values for all the general procedures related to de-noising.

wdencomp: It returns a denoised or compressed version of the input data X obtained by wavelet coefficients thresholding using the global positive threshold THR

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



Figure 4.4 Implementation and Simulation

4.3.CODE APPROACH

➔FOR THE INPUT

```
clc
clear all
close all
disp('input sound')
load handel.mat
fs=Fs;
y=y(:,1);
% info = auditionfo(input);
sound(y);
pause(10);
```

➔FOR ADDING NOISE

```
figure,plot(y);
title('input');
xlabel('samples');
ylabel('amplitude');
y = awgn(y,40);
noi = y;
figure,plot(y);
xlabel('samples');
ylabel('amplitude');
title('awgn');
disp('playing added noise...');
sound(y);
pause(10)
```

➔ NOISE REDUCTION

%'Fp,Fst,Ap,Ast' (passband frequency, stopband frequency ,passband

%ripple,stop band attenuation

hlpf = fdesign.lowpass('Fp,Fst,Ap,Ast',3.0e3,3.5e3,0.5,50,fs);

D = design(hlpf);

freqz(D);

x = filter(D,y);

disp('playing denoised sound');

figure,plot(x);

title('denoise')

sound(x,fs);

xlabel('samples')

ylabel('amplitude')

pause(10)

➔ FREQ SHAPER USING BAND PASS

T = 1/fs;

len = length(x);

p = log2(len);

p = ceil(p);

N = 2^p;

f1=fdesign.bandpass('Fst1,Fp1,Fp2,Fst2,Ast1,Ap,Ast2',2000,3000,4000,5000,60,2,60,2*fs
)

hd=design(f1,'equiripple');

y = filter(hd,x);

freqz(hd);

y = y*100;

```
disp('playing frequency shaped...');  
sound(y,fs);  
pause(10)
```

➔ AMPLITUDE SHAPER

```
disp('amplitude shaper');  
out1=fft(y);  
phse = angle(out1);  
mag = abs(out1)/N;  
[magsig,~]=size(mag);  
threshold=1000;  
out=zeros(magsig,1);  
  
for i=1:magsig/2  
    if(mag(i)>threshold)  
        mag(i) = threshold;mag(magsig-i)=threshold;  
    end  
  
    out(i)=mag(i)*exp(j*phse(i));  
    out(magsig-i)=out(i);  
end  
  
outfinal=real(ifft(out))*10000;  
disp('playing amplitude shaped...');  
sound(outfinal,fs);  
pause(10);
```

```
load handle.mat  
  
figure;  
subplot(2,1,1);  
specgram(noi);  
title('Spectrogram of Original Signal');  
  
subplot(2,1,2);  
specgram(outfinal);  
title('Spectrogram of Adjusted Signal');
```

4.4 RESULT ANALYSIS

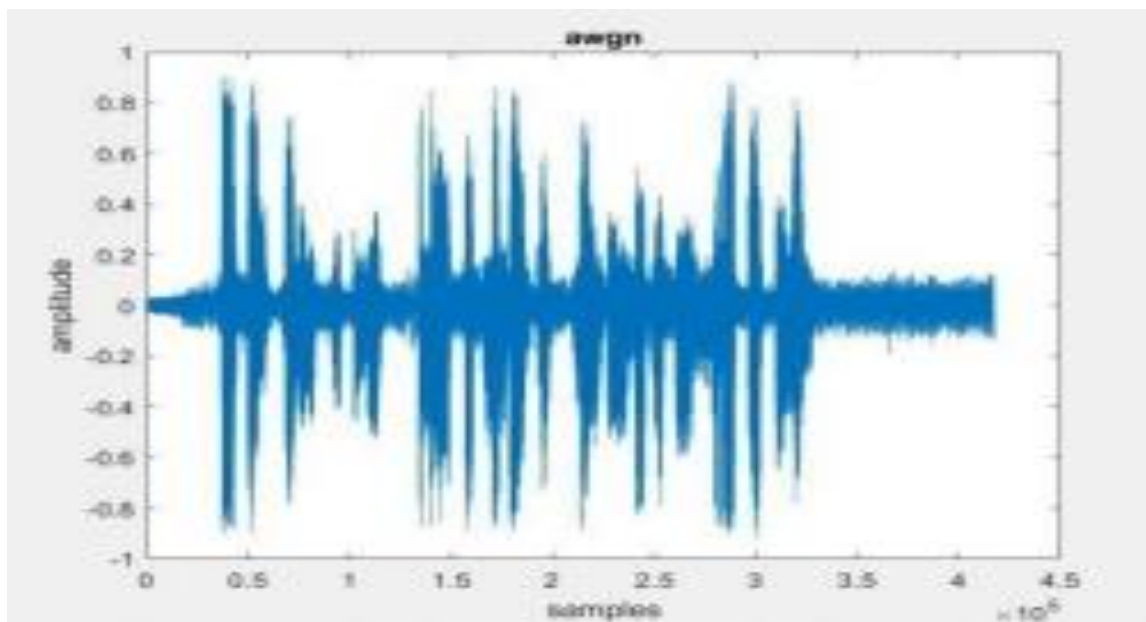


Figure 4.5 Awgn Signal

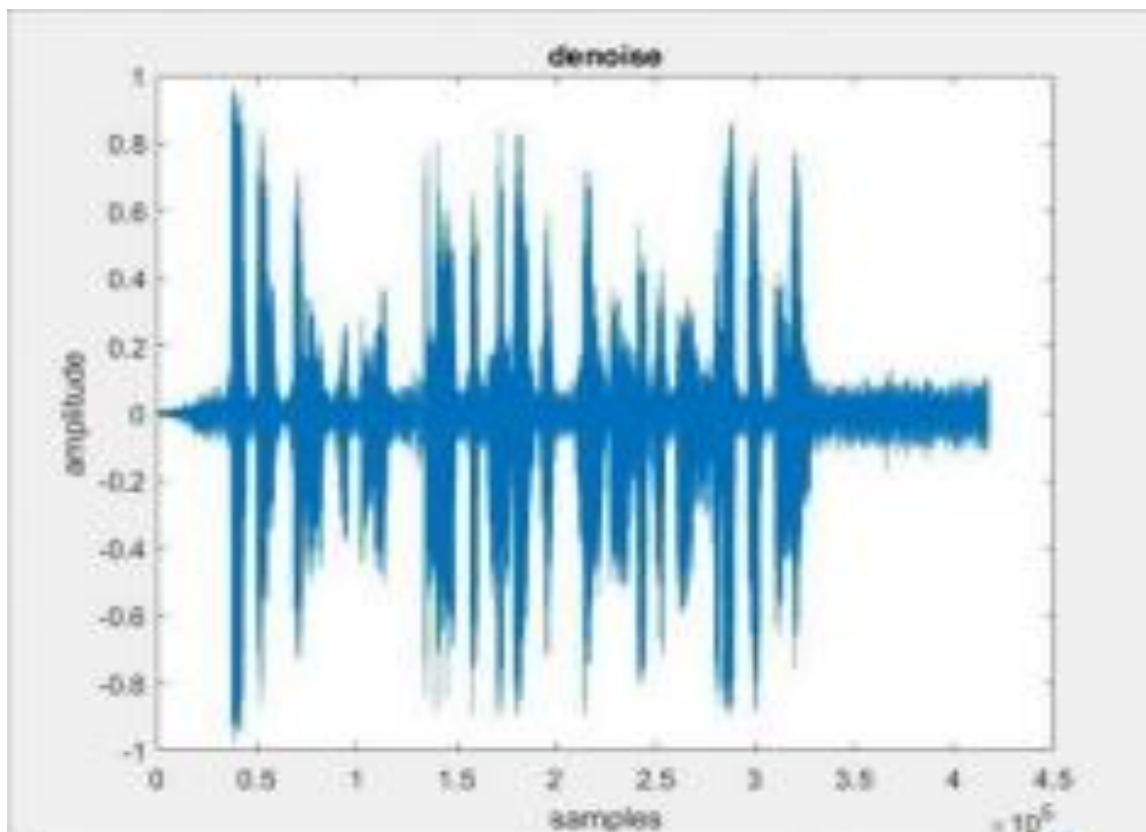


Figure 4.6 Denoise Signal

SPECTRUM OF ORIGINAL SIGNAL AND ADJUSTED SIGNAL

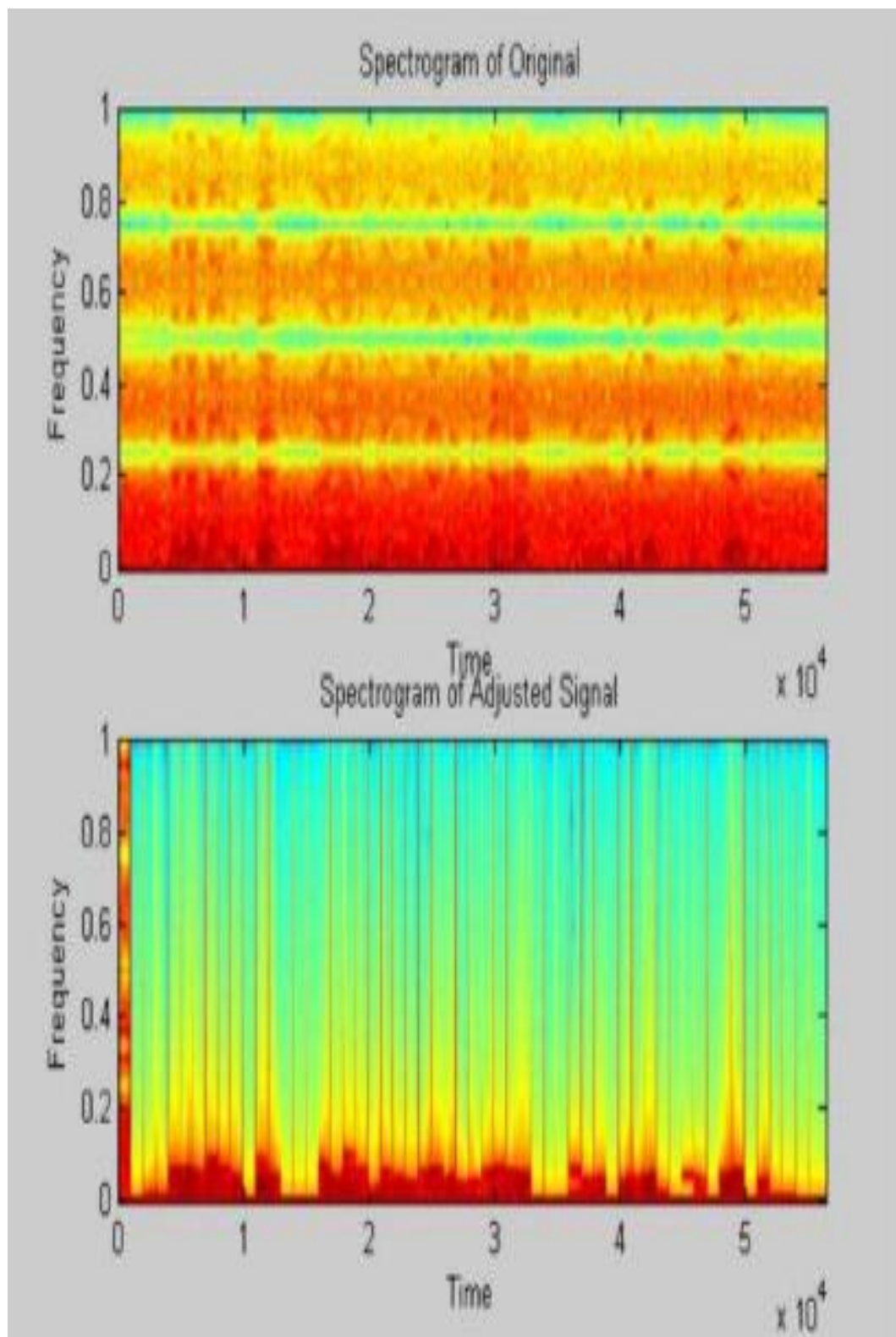


Figure 4.7 Spectrum of Original and Adjusted Signal

REPRESENTATION OF FREQUENCY SHAPER TRANSFER FUNCTION

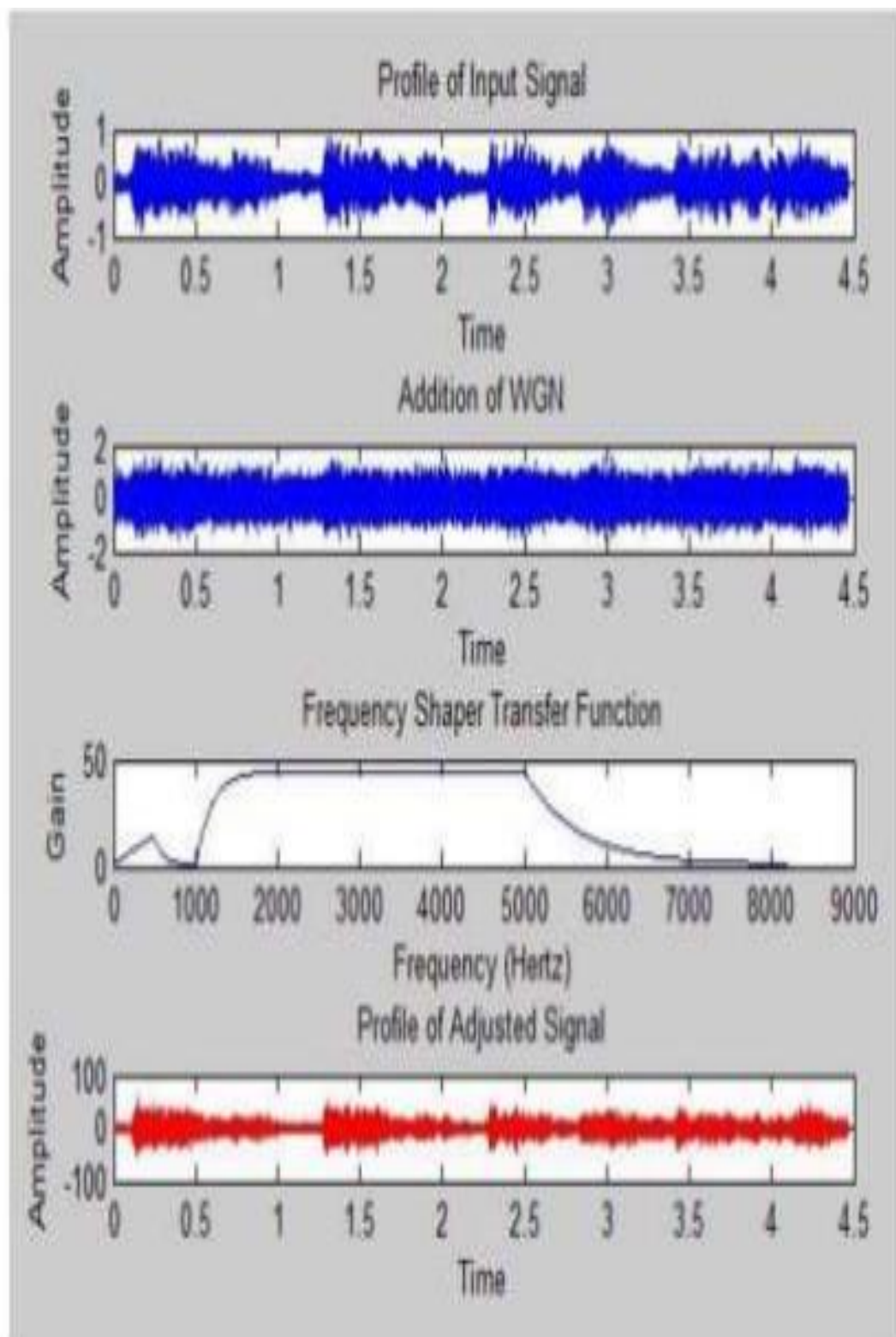


Figure 4.8 Signals from the input to the output

CHAPTER 5

CONCLUSION AND FUTURE SCOPE

5.1 SUMMERY

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 amplifiers 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 analyse & 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

5.2 FUTURE SCOPE

The future of hearing aids may be much more than just a hearable. The introduction to wireless and Bluetooth technology has provided a foundation for the next wave of advancement. Imagine a hearing aid that not only helps you hear, but also tracks and monitors your heart rate. There are many devices on the market currently that help track your health but they require regular charging, and some are obtrusive and obvious, as they are worn on the wrist.

5.3 References

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