Hearing Aid Devices for Impaired People using MATLAB

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BONAFIDE CERTIFICATE

Certified that this project report entitled "**Hearing Aid Devices for Impaired People using MATLAB"** is a bonafide work of **Anwesh Choudhury-19BEC1344, Divyansh Jain-19BEC1253, Kanupriya-19BEC1267, Utkarsh Maurya-19BEC1308** who carried out the Project work under my supervision and guidance for **ECE1004 – Signals and Systems.**

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ABSTRACT

Traditional analog hearing aids are similar to a simple radio. They can be tuned and adjusted for volume, bass and treble. But hearing loss is not just a technical loss of volume. Rather, hearing deficiency can increase 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, allowing one in while filtering out the other. Approximately 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 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. Through the use of digital signal processing, digital hearing aid now offers what the analog hearing aid cannot offer.

It proposes the possibility of performing signal-to noise enhancement, flexible 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 (DHA) system includes the noise reduction filter, frequency shaper function, and amplitude compression 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.

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TABLE OF CONTENTS

1. INTRODUCTION

OBJECTIVES AND GOALS

- The aim of the hearing aid is to amplify sound signals in such a way that they become audible for the hearing-impaired person.
- Write a MATLAB code for noise reduction.
- Write a MATLAB code for frequency shaping.
- Write a MATLAB code for amplitude compression.
- To compile the MATLAB code and create a proper working model of the Digital Hearing Aid.

APPLICATIONS

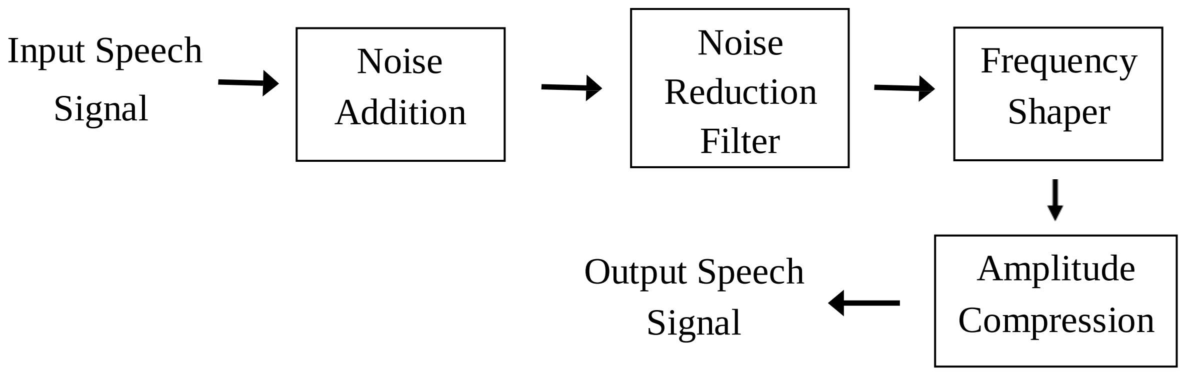
- The aid automatically adjusts the volume and pitch on its own.
- It performs thousands of adjustments per second which results in reduced background noise, improved listening in noisy situations, sound quality and multiple program settings
- The aid can be used by impaired people with normal hearing loss to people with moderate hearing loss.
- Simply turning up the volume to hear more isn’t effective. This can be overpowering or screechy in certain locations or settings. With digital hearing aids you can customise your hearing experience to suit your type of hearing loss, which enhances your comfort.
- Style-wise, digital hearing aids are smart and usually small; the perfect package.
- 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.

FEATURES

- Hearing aid technologies have changed over the years. Before digital hearing aids came on the scene, hearing aids were based on analog technology alone.
- Analog hearing aids' sound signals are continuous and uniform in flow, so the sophisticated nuances or layers of sound that digital hearing aids have are missing. Adapting to different sound environments with analog hearing aids means simply turning up the volume, which is very uncomfortable. Also, with analog hearing aids, you don't have the option to reduce background noises – they would also be amplified. With the help of MATLAB we didn't just cranked up the volume but made adjustments with the audio input.
- It has 3 filters to amplify the sound, namely noise reduction, frequency shaper and amplitude compressor.

2. METHODOLOGY

Below 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.



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 Adaptive White Gaussian Noise (AWGN) and random noise are added to the input speech signal by using MATLAB function. 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.

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.

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 [5]. 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.

AMPLITUDE COMPRESSOR

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.

3. CODE

Denoise

```
function y = denoiseEm(x);
[thr,sorh,keepapp]=ddencmp( 'den' , 'wv' ,x);
y=wdencmp( 'gbl' ,x, 'db3' ,2,thr,sorh,keepapp);
subplot(2,1,1)
plot(x);
subplot(2,1,2)
plot(y);
```

Apply Ski Slope

```
function y = applySkiSlope(x,g,transitionV,fs);
first = transitionV(1);
second = transitionV(2);
third = transitionV(3);
fourth = transitionV(4);
x_length = length(x);
n = nextpow2(x_length);
N = 2^n;
T = 1/fs;
X =  fft(x,N);
gain = zeros(N,1);

% Sets the gain for the first stage of frequencies
firstC = (.3*(g-1))/first;
k=0;
while(k/N <= first/fs)
   gain(k+1) = firstC*k/(N*T) + 1;
   gain(N-k) = gain(k+1);
   k=k+1;
end;

% Sets the gain for the second stage of frequencies
secondC = firstC*first +1;
secondC2 = (second-first)/5;
while(k/N <= second/fs)
   gain(k+1) = 1 + (secondC-1)*exp(-((k/(N*T))-first)/secondC2);
   gain(N-k) = gain(k+1);
   k=k+1;
end;

% Sets the gain for the third stage of frequencies
thirdC = 1 + (secondC-1)*exp(-second/secondC2);
thirdC2 = (third-second)/5;
while(k/N <= third/fs)
   gain(k+1) = g + (thirdC-g)*exp(-((k/(N*T)-second))/thirdC2);
   gain(N-k) = gain(k+1);
   k=k+1;
end;

% Sets the gain for the fourth stage of frequencies
while(k/N <= fourth/fs)
   gain(k+1) = g;
   gain(N-k) = gain(k+1);
   k=k+1;
```

```
end;

% Sets the gain for the fifth stage of frequencies
fifthC = g;
fifthC2 = (fs/2-fourth)/5;
while(k/N <= .5)
   gain(k+1) = 1 + (fifthC-1)*exp(-((k/(N*T))-fourth)/fifthC2);
   gain(N-k) = gain(k+1);
   k=k+1;
end;


k_v = (0:N-1)/N;
plot(k_v,gain);%entire filter transfer function

figure;%non-redundant filter transfer function
k_v = k_v*fs;
k_v = k_v(1:N/2+1);
plot(k_v,gain(1:N/2+1));
title('Frequency Shaper Transfer Function');
xlabel('Frequency  (Hertz)');
ylabel('Gain');

Y = X+gain;
y = real(ifft(Y,N));

y = y(1:x_length);
t=[0:1/fs:(x_length-1)/fs];
figure;
plot(t,y,'r');
%hold;
figure;
plot(t,x);
```

Power Compress

```
function y = powerCompress(input, Psat,Fs);

x=input;
len=Fs*0.1;
iter=floor(length(x)/len);
Plow=0.008;

for rg=0:1:iter;
 start=rg*len+1;
 en= rg*len+len;
 if rg*len+len>length(x)
 en=length(x);
end
clear signal X  X_pow Y_pow Y y z X_phase;
signal=x(start:en);
n = nextpow2(len);
N = 2^n;
X = fft(signal,N);
X_phase=angle(X);                  % Save the old phase information
X_pow = abs(X)/N;
```

```
Y_pow = X_pow;
Y=zeros(N,1);
for k=0:N/2
   if Y_pow(k+1)<Plow              % Take out noise
      Y_pow(k+1)=0;
      Y_pow(N-k)=0;
   elseif Y_pow(k+1)>Psat          % Clip amplitudes higher than Psat
      Y_pow(k+1)=Psat;
      Y_pow(N-k)=Psat;
   end;
   Y(k+1) = Y_pow(k+1)*(cos(X_phase(k+1))+i*sin(X_phase(k+1)));
       Y(N-k) = Y_pow(N-k)*(cos(X_phase(N-k))+i*sin(X_phase(N-k)));
end;

y = real(ifft(Y,N));

z = y(1:en-start+1);

sig_out(start:en)=z;
end;
y = sig_out*2000;
```

Main Code

```
function y = hearingAidF(input,g,Psat,transitionV,newfile);

input='sample.wav'
[x,fs] = audioread(input);
xc = denoiseEm(x);                                  % denoising filter
xf = applySkiSlope(xc,100,[1000 2000 3000 4000],fs);       % frequency 
shaping filter
y = powerCompress(xf,90,fs);                     % amplitude shaping filter
x_length = length(x);
t=[0:1/fs:(x_length-1)/fs];
%sound(y,fs);


% plots for the input and output signals

figure;
subplot(2,1,1);
plot(t,x,'b');
axis tight;
xlabel('Time (sec)');
ylabel('Relative Magnitude');
title('Time Profile for Data in Signal 2');


subplot(2,1,2);
plot(t,y,'r');
axis tight;
xlabel('Time (sec)');
ylabel('Relative Magnitude');
title('Time Profile for Data in Adjusted Signal 2');

figure;
```

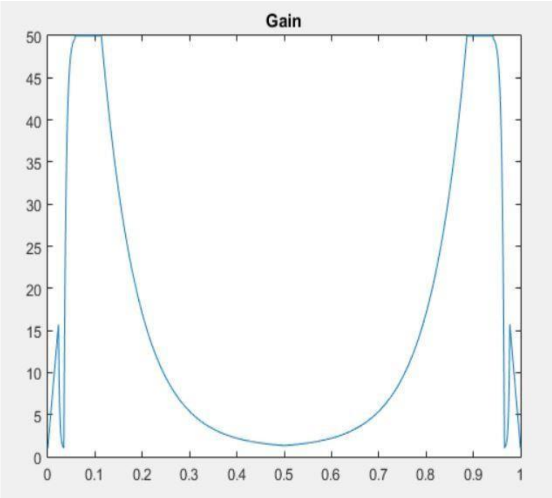
```
subplot(2,1,1);
specgram(x);
title('Spectrogram of Original Signal 2');

subplot(2,1,2);
specgram(y);
title('Spectrogram of Adjusted Signal 2');


%soundsc(input, fs);
sound(y,fs);
%audiowrite(y,fs,nbits,'linear',newfile);
audiowrite('temp_file.wav',y,fs);
```

4. RESULTS

The graphs of the functions use are shown below.



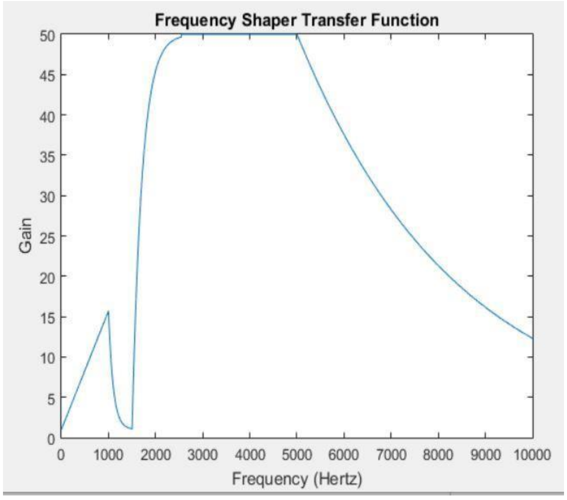
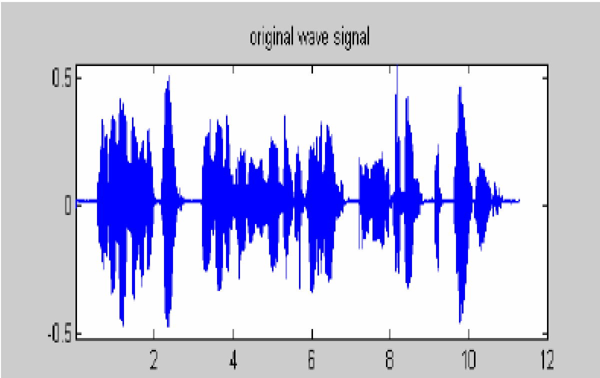
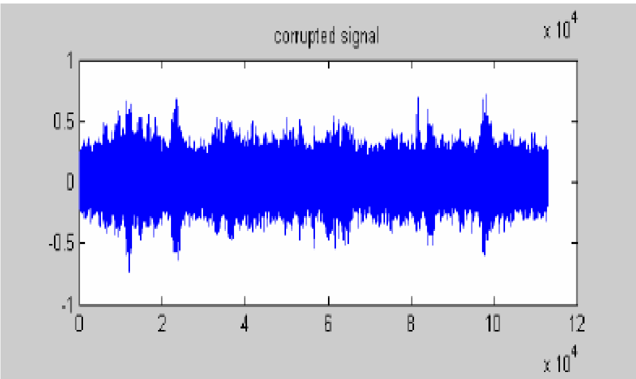


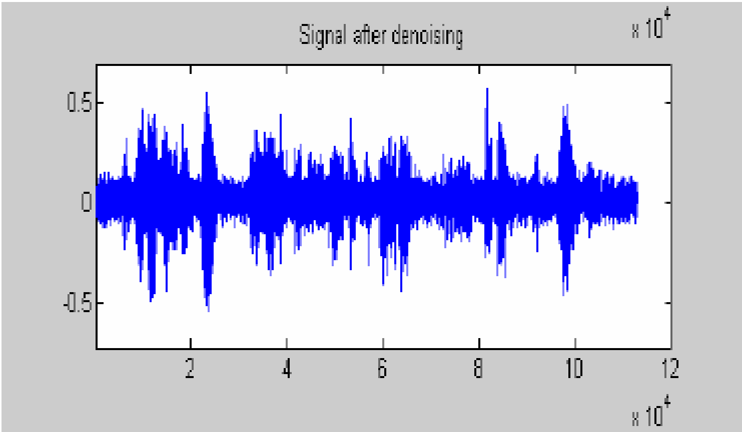
Figure below is the original speech signal which is plot on time versus amplitude axis.

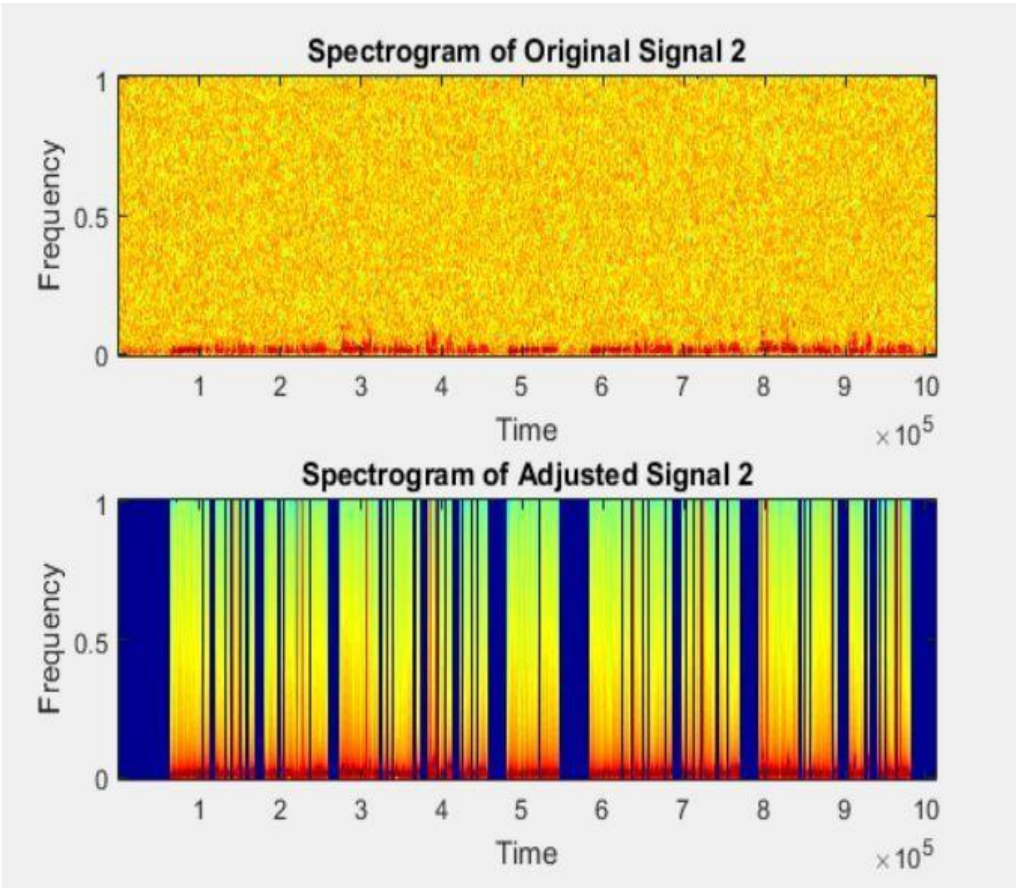


Next, Adaptive White Gaussian Noise is added to the original wave signal. The purpose of this addition just to simulate noises in the real life situation.
Figure below shows the signal after noise addition.



Afterward, the denoising process takes place which removes most of the noise in the signal as shown in 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 as shown in figure.





However the strength of the adjusted signal is not increase as our expectation. Possibly the cause of this error is due to the gain function improperly implied.

5. CONCLUSION

The newer digital aids offer more ability to fine tune the sound without distorting the quality and help the listener. 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 problem with conventional amplifier which amplified the whole signal including noise. In general, digital hearing aids, when the incoming signals are converted 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 could Improve quality of life by improving sound quality, Higher listening comfort, better communication in noisy environment, better speech intelligibility in group conversations and more flexibility in case of progressive hearing less.

FUTURE PLANS

- Can be improved such that it can intelligently detect the required inputs, according to user’s need.
- To improve the quality of the output signal being produced.
- To be able to remove different types of noise and make it easier for the person, suffering from the hearing loss, to hear the voice.
- The process can be automated with the use of AI and constant hearing aid is delivered.

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