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Institute Of Technology, Nirma University

2EC502 - Digital Signal Processing

Special Assignment Report

Audio Equalizer Design

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21BEC045

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Abstract

This report proposes a simple design for a digital audio equalizer. This report discusses about various equalizer types and characteristics. Theoretically, an ideal equalizer is the inverse system of the room transfer function. This report uses graphic equalizer to demonstrate the practical usage of audio equalizers. The approach is implemented using MATLAB software. The proposed equalizer can be widely used in audio signal processing applications. This report also discusses the applications of audio equalizer. A variety of digital signal processing methods are used for adjusting the spectral balance in audio signals and reconstruction of the equalized signal. The audio output is plotted and compared with the original to obtain the results.

Introduction

Equalization is the process of changing the amplitude of the frequency bands in an audio signal. An equalizer is an audio filter that can amplify or diminish certain range of frequency components in a sound signal. Equalizers are an important part of modern audio systems. They are used in sound recording or live music to make vocals or certain instruments more prominent by boosting the corresponding frequency range.

Audio signals can be segmented into multiple frequency bands. Each of these bands has its own slider that adjusts the strength of the incoming audio stream, giving the user control over the frequency ranges. The frequency bands and the revised audio track are then combined to get the equalized audio.

The goal of equalizing is to correct or enhance the performance of the system. This includes, the correction of a speaker response and room's response of the sound, equalization of speakers to assure natural music listening and enhancement of recorded music.

Types of Equalizers

There are many different types of equalizers based on the number of parameters that the user can control.

Graphic Equalizer

The most commonly used equalizers for music systems are graphic equalizers. It operates by assigning different range of frequencies to a set of bands. The user controls the gain for each band using sliders and the frequencies are attenuated or amplified accordingly. The user cannot control any other parameters like centre frequencies and bandwidth of the bands or the design of the bandpass filters. The more bands there are, the more precise control is offered.

Graphic equalizers are routinely used by audio technicians to modify the frequency response of audio. They can be used to correct for speaker bias or to add bass to a song. They are simply a collection of filters that work together to provide a unique overall frequency response. Although the flexibility of a graphic equalizer is not as good as that of a parametric equalizer, it is often a preferred choice in sound enhancement.

Parametric Equalizer

These equalizers are most commonly found in high-end audio systems and recording studios. The Parametric Equalizer allows the user to adjust each frequency band's center frequency, gain, and bandwidth. A parametric equalizer is a cascade of individually adjustable filters used in real time to change the spectrum response characteristics of the original sound reducing or boosting certain frequency bands.

In practical application, to equalize an audio source, many parametric equalisers are cascaded and used at the same time. A parametric equalizer offers more flexibility to the user but it is difficult to use requiring expertise on part of the user. Also, it usually has a limited number of filters that the user can adjust.

Dynamic Equalizers:

This type of equalizer allows users to automatically adjust levels across different frequency bands based on pre-defined settings such as vocal enhancement or bass reduction without having to manually move any sliders themselves. The equalizer can modify one track in relation to the sound of other instruments or vocals. This makes Dynamic equalizing very powerful and useful in instances where you want your track to be more dynamic overall.

Fundamentals of Equalizer

Audio Equalizers use band pass filters to separate bands around a centre frequency, which are commonly in a bell shape. The filter properties of an Equalizer are determined by analyzing the individual bands. These factors determine the spectral range in which an equalizer will function. The filter characteristics include the following categories:

- 1) Center frequency: The center frequency of a band determines the frequency at which the audio signal is affected by a boost or cut in sound energy.
- 2) Filter bandwidth: The range of frequencies around the center frequency that are in the passband of the filter. Bandwidth is usually defined using quality factor.
- 3) Filter Type: The Equalizer band's type of filter includes High Pass Filter (HPF), Low Pass Filter (LPF), Notch Filter, High Shelf Filter (HSF), and a Low Shelf Filter (LSF).
- 4) Filter Slope: The rate of attenuation of sound beyond cut-off frequency is indicated by the slope of a filter.
- 5) Filter Gain: The amount of amplification or attenuation that is achieved through the filter.

Human audible sound signals can be classified into different characteristic frequency bands:

Sub-Bass (16 Hz to 60 Hz) - These are very low bass frequencies. When played by a loudspeaker, these frequencies are felt rather than heard as a rumbling or an earthquake.

Bass (60 Hz to 250 Hz) - This range contains the fundamental notes of the rhythm section. Any equalizer changes will affect the balance of the mix, making it fat or thin. Too much emphasis will make for a boomy mix.

Low Mids (250Hz to 2kHz) - Boosting the range from 250 Hz to 500 Hz will accent ambience in the studio and will add clarity to bass and lower frequency instruments. The range between 500 Hz and 2 kHz can make midrange instruments like guitar, snare, saxophone sound thin.

High Mids (2 kHz to 4 kHz) - The attack portion of percussive and rhythm instruments occurs in this range. High mids are also responsible for the projection of midrange instruments.

Presence (4 kHz to 6 kHz) - This frequency range is partly responsible for the clarity of a mix and provides a measure of control over the perception of distance. Boosting

this frequency range, the mix will be perceived as closer to the listener. Attenuating around 5 kHz will make the mix sound further away but also more transparent.

Brilliance (6 kHz to 16 kHz) - This range controls the brilliance and clarity of your mix. Boosting it too much can cause some clipping.

Pseudo Code

- 1) Import audio file and determine user parameters.
- 2) Convert time domain signal into frequency domain using DFT/FFT to analyze.
- 3) Separate frequencies into bands by using bandpass filter.
- 4) Amplify or attenuate the signal in passband by multiplying by gain factor.
- 5) Reconstruct signal components by adding the adjusted signals bands together.
- 6) Convert frequency domain to time domain by using IDFT.
- 7) Plot magnitude spectrum and waveforms to obtain results.
- 8) Output audio file.

Block Diagram

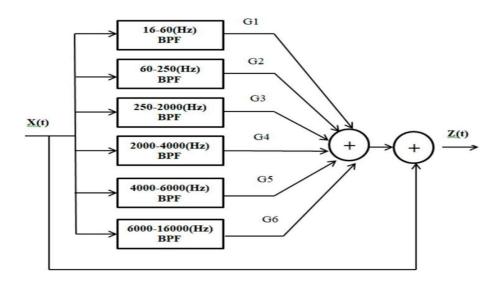


Figure 1. Block Diagram of the Equalizer

Flow Chart Read Audio Signal Determine a Window (Region of Interest) Observe in Frequency Domain Determine Equalization factors Butterworth Band Pass Filter Sub Bass Presence Bass Brilliance Lower Higher Midrange Midrange Midrange Multiplying with a Factor Observing Magnitude Response of each range Summing all the Ranges and original audio

Figure 2 Flowchart of the Process

Comparing the equalized with original audio

Observation

From the results of the audio equalizer design, we can observe that amplitude of the frequencies bands is adjusted according to the given gain parameters. The difference between original signal and equalized signal is plotted and observed to be in correspondence to the gain specified by the user for each frequency band. Sound that is played has an audible equalization effect. The range of frequencies when boosted, sound more prominent in the reconstructed signal.

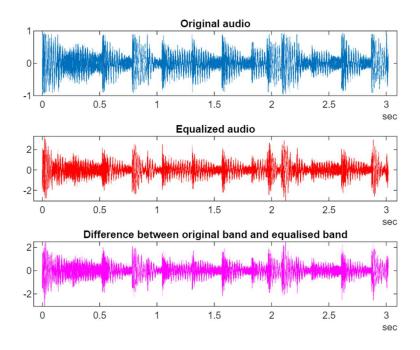


Figure 3 Time Domain representation of original

Applications

Equalizers are designed to correct the frequency loss and gain in a sound system. Equalizing makes a sound system sound more natural and complete. Equalizer can be used to increase loudness of certain instruments or vocals in a sound mix. Modern stereos come equipped with an inbuilt graphic equalizer. High-end audio system more offer precise tuning with more bands and a higher range of gain.

Audio Equalizers assist to compensate for the acoustics in the room. In public address systems, equalizers are employed to sharpen the sound and eliminate echoes. A sound system with equalization will be required in stadiums, sports arenas, and other events. Audio equalizers are used in order to enhance sound output in a variety of locations. Quality audio equipment enhances human voice in loudspeakers to provide crisp and clear announcements. Audio equalizer is very beneficial to bands and other live travelling events, as it is practically hard to build a solid sound system for every location without correcting for frequencies that cause feedback.

Conclusion

The quality of sound can be modified and enhanced using audio equalizer. We have used a simple technique to design a user-friendly equalizer using MATLAB software. Butterworth bandpass filter was used to isolate frequency bands and precisely control the gain of each band. Graphical approaches are useful for observing the features of a signal in a visual manner. Equalizers are a vital component of communications and sound systems as they correct for the frequency loss in sound systems. MATLAB is a very helpful tool in digital signal processing.

References

- 1) J. Rämö, V. Välimäki, and B. Bank. High-Precision Parallel Graphic Equalizer
- 2) Digital Filters for Audio Equalizer Design https://semiwiki.com/semiconductor-services/einfochips/296500-digital-filters-for-audio-equalizer-design/
- 3) S. Cecchi; L. Palestini; E. Moretti; F. Piazza. New Approach to Digital Audio Equalization. IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, 2007.
- 4) Johannes Langelaar, Adam Strömme Mattsson, Filip Natvig. Development of real time audio equalizer application using MATLAB App Designer.

MATLAB CODE



DIGITAL SIGNAL PROCESSING

Special Assignment: Audio Equalizer Design

21BEC045 Hetansh Shah

% Frequency Ranges for Vocals in an audio

Sub bass: 20 – 60Hz.

Bass: 60 – 250Hz.

Lower midrange: 250 – 500Hz.

Midrange: 500Hz – 2kHz.

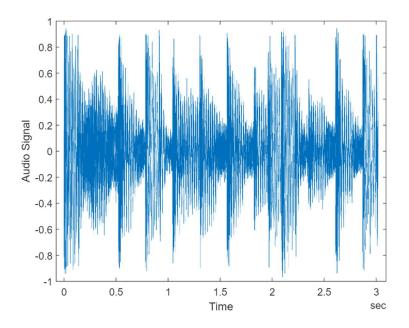
Higher midrange: 2 – 4kHz.

Presence: 4 – 6kHz.

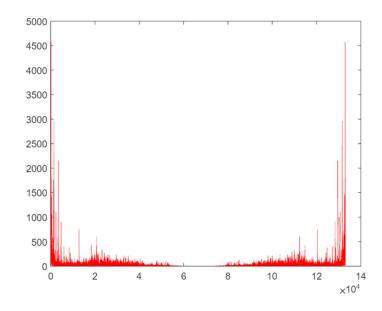
Brilliance: 6 – 20kHz.

```
clc;clear all;close all;
% Selecting the audio input file
[y,Fs]=audioread("D:\Downloads\jinglebells.mp3");
% Getting a particular portion of audio
x=y([(2048*15):(2048*80)],1);
% Getting Information of the audio file
info = audioinfo("D:\Downloads\jinglebells.mp3")
```

```
% Time axis
t = 0:seconds(1/Fs):seconds(length(x)/Fs);
t = t(1:end-1);
% Input audio file
sound(x,Fs);
% Plot the time domain representation of the original audio
plot(t,x);xlabel('Time');ylabel('Audio Signal');
```



```
% Compute the FFT of the original audio
fft=fft(x);
plot(abs(fft),'r')
```



% Define equalization factors for different frequency bands factor1 = 4.2

factor1 = 4.2000

```
factor2 = 4.2
```

```
factor2 = 4.2000
```

```
factor3 = 3.8
```

factor3 = 3.8000

```
factor4 = 4.1
```

factor4 = 4.1000

```
factor5 = 3.8
```

factor5 = 3.8000

```
factor6 = 4.1
```

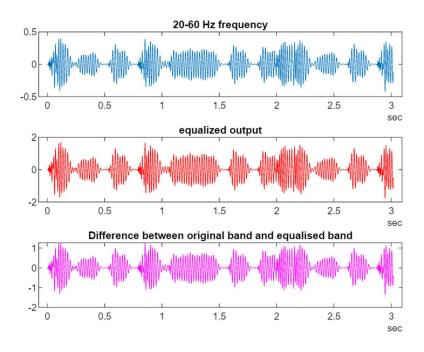
factor6 = 4.1000

```
factor7 = 3.8
```

factor7 = 3.8000

SUB BASS (20-60 Hz)

```
N = 2 ;
[B1,A1] = butter(N,[20 60]/(Fs/2),"bandpass");
g_1 = filter(B1,A1,x);
g1 = g_1*factor1;
% Play the equalized audio for this frequency band
sound(g1,Fs)
% Plot the time domain representation and magnitude spectrum of the original
and equalized audio
figure;subplot(311);plot(t,g_1);title('20-60 Hz frequency');
subplot(312);plot(t,g1,'r');title('equalized output');
subplot(313);plot(t,g1-g_1,'m');title('Difference between original band and
equalised band');figure;
```



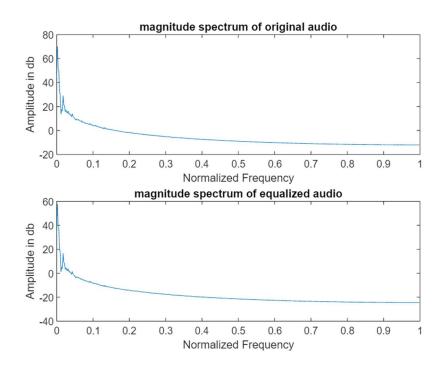
```
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(g1);
[h2 w2]=freqz(g_1);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 70.0030

```
max(20*log10(abs(h2)))
```

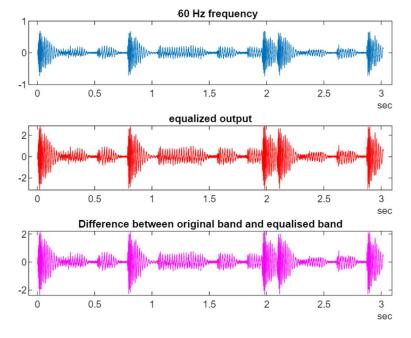
ans = 57.5380

```
subplot(211);plot(w2/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



BASS (60-250 Hz)

```
N = 2 ;
[B2,A2] = butter(N,[60 250]/(Fs/2),"bandpass");
g_2 = filter(B2,A2,x);
g2 = g_2*factor2;
% Play the equalized audio for this frequency band
sound(g2,Fs)
% Plot the time domain representation and magnitude spectrum of the original
and equalized audio
figure; subplot(311); plot(t,g_2); title('60-250 Hz frequency');
subplot(312); plot(t,g2,'r'); title('equalized output');
subplot(313); plot(t,g2-g_2,'m'); title('Difference between original band and
equalised band'); figure;
```



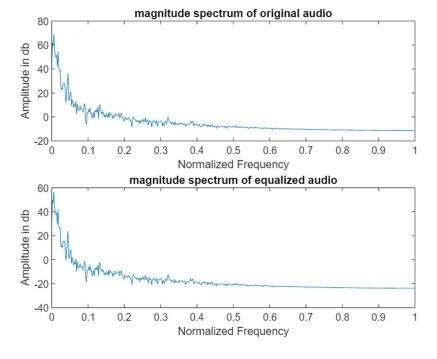
```
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(g2);
[h2 w2]=freqz(g_2);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 68.8113

```
max(20*log10(abs(h2)))
```

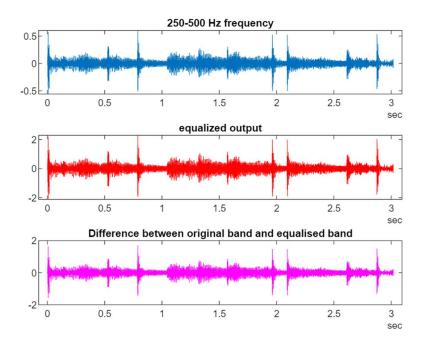
ans = 56.3463

```
subplot(211);plot(w2/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



LOWER MIDRANGE (250-500 Hz)

```
N = 2 ;
[B3,A3] = butter(N,[250 500]/(Fs/2),"bandpass");
g_3 = filter(B3,A3,x);
g3 = g_3*factor3;
% Play the equalized audio for this frequency band
sound(g3,Fs)
% Plot the time domain representation and magnitude spectrum of the original
and equalized audio
figure;subplot(311);plot(t,g_3);title('250-500 Hz frequency');
subplot(312);plot(t,g3,'r');title('equalized output');
subplot(313);plot(t,g3-g_3,'m');title('Difference between original band and
equalised band');figure;
```



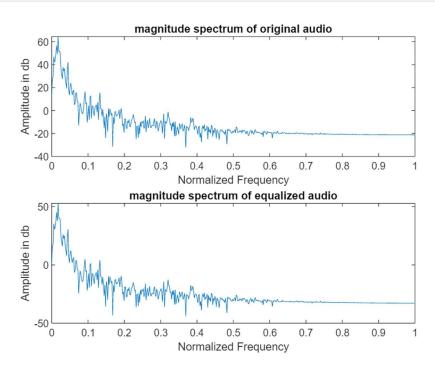
```
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(g3);
[h2 w2]=freqz(g_3);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 64.4038

```
max(20*log10(abs(h2)))
```

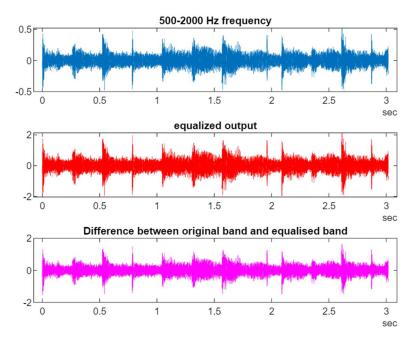
ans = 52.8081

```
subplot(211);plot(w2/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



MIDRANGE (500-2kHz)

```
N = 2 ;
[B4,A4] = butter(N,[500 2000]/(Fs/2),"bandpass");
g_4 = filter(B4,A4,x);
g4 = g_4*factor4;
% Play the equalized audio for this frequency band
sound(g4,Fs)
% Plot the time domain representation and magnitude spectrum of the original
and equalized audio
figure; subplot(311); plot(t,g_4); title('500-2000 Hz frequency');
subplot(312); plot(t,g4,'r'); title('equalized output');
subplot(313); plot(t,g4-g_4,'m'); title('Difference between original band and
equalised band'); figure;
```



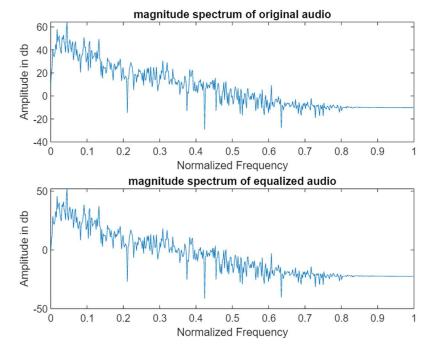
```
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(g4);
[h2 w2]=freqz(g_4);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 64.2653

```
max(20*log10(abs(h2)))
```

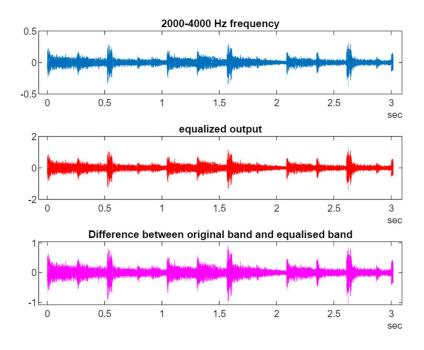
ans = 52.0096

```
subplot(211);plot(w2/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



HIGHER MIDRANGE (2kHz - 4kHz)

```
N = 2 ;
[B5,A5] = butter(N,[2000 4000]/(Fs/2),"bandpass");
g_5 = filter(B5,A5,x);
g5 = g_5*factor5;
% Play the equalized audio for this frequency band
sound(g5,Fs)
% Plot the time domain representation and magnitude spectrum of the original
and equalized audio
figure; subplot(311); plot(t,g_5); title('2000-4000 Hz frequency');
subplot(312); plot(t,g5,'r'); title('equalized output');
subplot(313); plot(t,g5-g_5,'m'); title('Difference between original band and
equalised band'); figure;
```



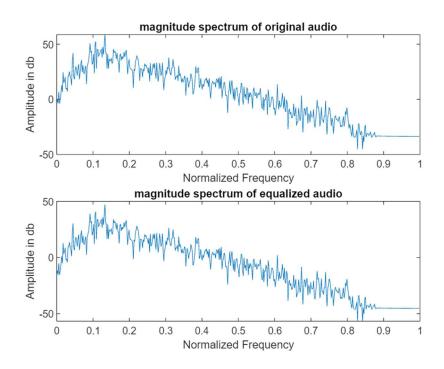
```
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(g5);
[h2 w2]=freqz(g_5);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 58.7008

```
max(20*log10(abs(h2)))
```

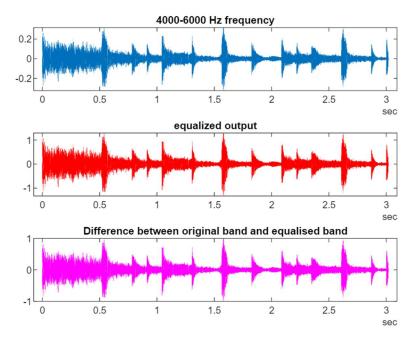
ans = 47.1051

```
subplot(211);plot(w2/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



PRESENCE (4kHz - 6kHz)

```
N = 2 ;
[B6,A6] = butter(N,[4000 6000]/(Fs/2),"bandpass");
g_6 = filter(B6,A6,x);
g6 = g_6*factor6;
% Play the equalized audio for this frequency band
sound(g6,Fs)
% Plot the time domain representation and magnitude spectrum of the original
and equalized audio
figure; subplot(311); plot(t,g_6); title('4000-6000 Hz frequency');
subplot(312); plot(t,g6,'r'); title('equalized output');
subplot(313); plot(t,g6-g_6,'m'); title('Difference between original band and
equalised band'); figure;
```



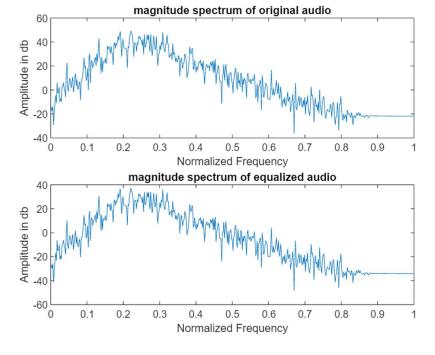
```
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(g6);
[h2 w2]=freqz(g_6);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 49.3486

```
max(20*log10(abs(h2)))
```

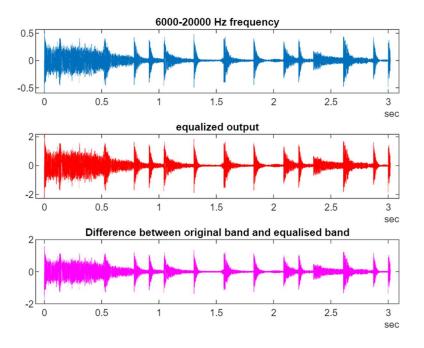
ans = 37.0929

```
subplot(211);plot(w2/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



BRILLIANCE (6kHz - 20kHz)

```
N = 2 ;
[B7,A7] = butter(N,[6000 10000]/(Fs/2),"bandpass");
g_7 = filter(B7,A7,x);
g7 = g_7*factor7;
% Play the equalized audio for this frequency band
sound(g7,Fs)
% Plot the time domain representation and magnitude spectrum of the original
and equalized audio
figure; subplot(311); plot(t,g_7); title('6000-20000 Hz frequency');
subplot(312); plot(t,g7,'r'); title('equalized output');
subplot(313); plot(t,g7-g_7,'m'); title('Difference between original band and
equalised band'); figure;
```



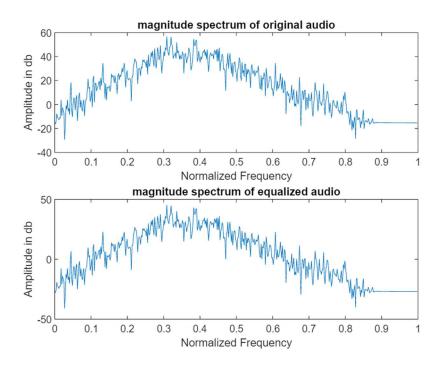
```
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(g7);
[h2 w2]=freqz(g_7);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 56.5627

```
max(20*log10(abs(h2)))
```

ans = 44.9670

```
subplot(211);plot(w2/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



FINAL OUTPUT

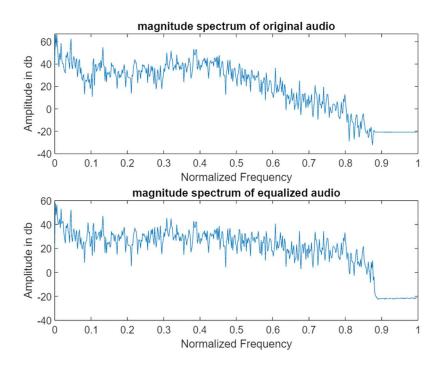
```
% Sum all the equalized audio signals
g=g1+g2+g3+g4+g5+g6+g7;
sound(g,Fs)
figure;
% Plot the magnitude spectrum of original and equalized audio
[h1 w1]=freqz(x);
[h2 w2]=freqz(g);
% Maximum value of Peak Amplitude
max(20*log10(abs(h1)))
```

ans = 57.6173

```
max(20*log10(abs(h2)))
```

ans = 66.8867

```
subplot(211);plot(w2/pi,20*log10(abs(h2)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of original audio')
subplot(212);plot(w1/pi,20*log10(abs(h1)));
xlabel('Normalized Frequency');ylabel('Amplitude in db');
title('magnitude spectrum of equalized audio')
```



mse(x,g)

ans = 0.1948

```
% Plot the time domain representation of the original and isolated components
figure; subplot(311); plot(t,x); title('Original audio');
subplot(312); plot(t,g,'r'); title('Equalized audio');
subplot(313); plot(t,x-g,'m'); title('Difference between original band and
equalised band'); figure;
```

