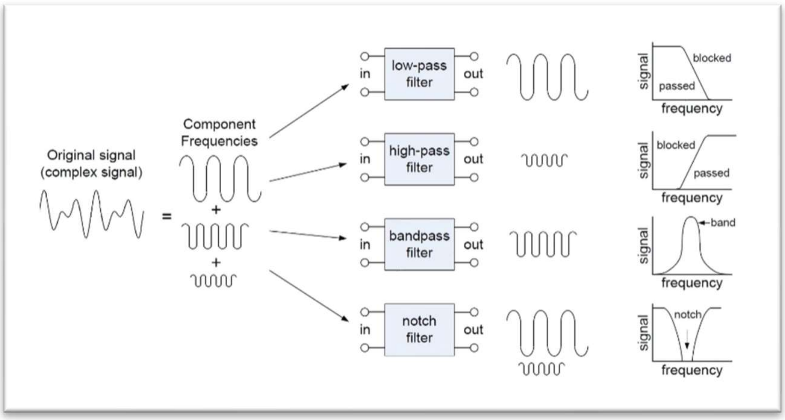
Contents

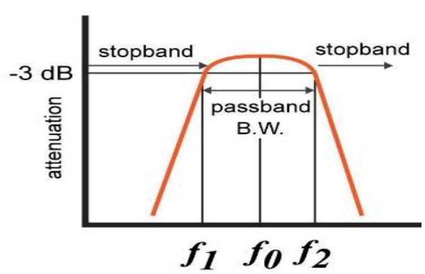
1) Introduction:

<u>Filters</u>: A filter is a circuit capable of passing or amplifying certain frequencies while suppressing or attenuating the rest, that is undesirable or irrelevant.

The four primary types of filters are low-pass, high-pass, band-pass, and band-reject or band-stop filter. In this project, we are concerned with the use of Band Pass Filters.



Band Pass Filter is a device that lets only those frequency components pass through it which are within a particular band or range of frequencies. All other frequencies lying outside this range, i.e., the ones that lie in the stop band are suppressed. The original input signal does not undergo any distortion, and there is no introduction of extra noise involved. Band Pass filters are commonly used in wireless transmitters and receivers to eliminate the signals of undesired frequencies that cause interference.



In this project, we are applying a band pass filter to an input audio file with variable cutoff frequency given by the user.

For this, we first provide an input audio signal to the Matlab software, which comprises of various frequency components. We then define appropriate variables and functions to formulate the code. Following this, we specify the passband frequencies and stopband frequencies.

The sampling frequency is the number of samples per second taken from a continuous signal to make a discrete or digital signal. As per the Nyquist criteria, the sampling frequency must be greater than or equal to twice the maximum frequency of the input signal.

$$Fs \geq 2 \times Fm$$

In this project, we first take audio as input with the help of 'audioread' function. After that, we assigned its length in variable 'N' with the help of 'size' function. Then with the help of 'FilterDesigner', an inbuilt tool in matlab, prepared a band pass filter with above mentioned specifications. We pass our input audio signal into filter and the save the filtered output in a variable with the help of 'audiowrite' function. Then we plot its Frequency Response Graph, Pole Zero plot and Impulse Response plot with the help of 'freqz','zpalne' and 'impz' functions. The Frequency Response plot is important as the magnitude of the analytic signal gives the magnitude spectrum, and phase angle of the analytic signal gives phase spectrum. From these spectrums, features, such as standard deviation of amplitude, standard deviation of phase, and signal energy, are extracted. The Impulse response tells us how the system will behave for inputs at all frequencies and observation of pole zero plot helps us to get the stability of system graphically.

Following this, we take out the Fast Fourier Transform of the input and output signal with the help of 'fft' and 'fftshift' function and plot the respective graphs. Also, we plot the input and filtered output signals in time domain to see the change in characteristics of input signal. Then we calculated the power of the input and output signals using the formula, and subsequently plotted the power spectrum of input signal and filtered output signal.

At last, we hear the input audio and filtered output audio with the help of 'soundsc' function.

2) Objectives:

i. **To use a bandpass filter to remove noise from the input audio file.**

We design a bandpass filter to get a clear output i.e. noise-free audio. We had designed the filter for such frequency ranges which will obtain the best result.

ii. **Observing and comparing input audio and output filtered audio.**

Notice the input and filtered output audio file and interpret the result with the help of graphs. Here, two conditions has been taken by us, first is the condition in which order is not specified by us and in second condition, order of the system is specified by us manually.

iii. **Plotting of graphs to determine the characteristics of input and output audio for both the cases.**

The following graphs help us to interpret the results:

a. Frequency Response
b. Pole Zero plot
c. Impulse Response
d. Fast Fourier Transform signals
e. Signal in Time Domain
f. Power Spectrum plot

3) Theoretical Background:

When it comes to switching circuits or audio amplifiers or frequency signal circuits there is a very good chance for the circuit to be affected by noise signals. Dealing with noise is every engineer's nightmare. The most common way to remove noise from a circuit is by applying a filter circuit. This circuit will filter out the unwanted noise from the actual signal. The most commonly used filter circuit is the bandpass filter which can be constructed using a pair of resistors and capacitors. In signal processing, a finite impulse response (FIR) filter is a filter whose impulse response (or response to any finite length input) is of finite duration because it settles to zero in finite time. It is a time-continuous filter that is invariant with time. FIR systems operate with normalized frequencies, so they do not require the system sampling rate as an extra input argument. Normalized frequency is the frequency in units of cycles/samples commonly used as the frequency axis for the representation of digital signals.

A bandpass filter circuit is used to allow only a pre-defined set of frequencies to pass through it. It will filter all the frequency that is below the set value and above the set value. It is a combination of a high-pass filter and a low-pass filter. A filter that allows only the frequencies that are higher than it is called a high pass filter and the filter that allows the frequencies that are only lower than it is called a low pass filter. A bandpass filter can be obtained by cascading both high and low-pass filters.

The first half of the circuit is a High-Pass filter which filters the low frequencies and allows only the frequency that is higher than the set high cut-off frequency (fc_{HIGH}). The second half of the circuit is the Low-Pass filter circuit which filters the higher frequencies and allows only the frequency that is lower than the set low cut-off frequency (fc_{LOW}). The value of these cut-off frequencies can be determined with the help of formulas. The frequency that lies between fc_{HIGH} and fc_{LOW} is bandwidth. Bandwidth can be calculated by:

Bandwidth = fc_{HIGH}- fc_{LOW}

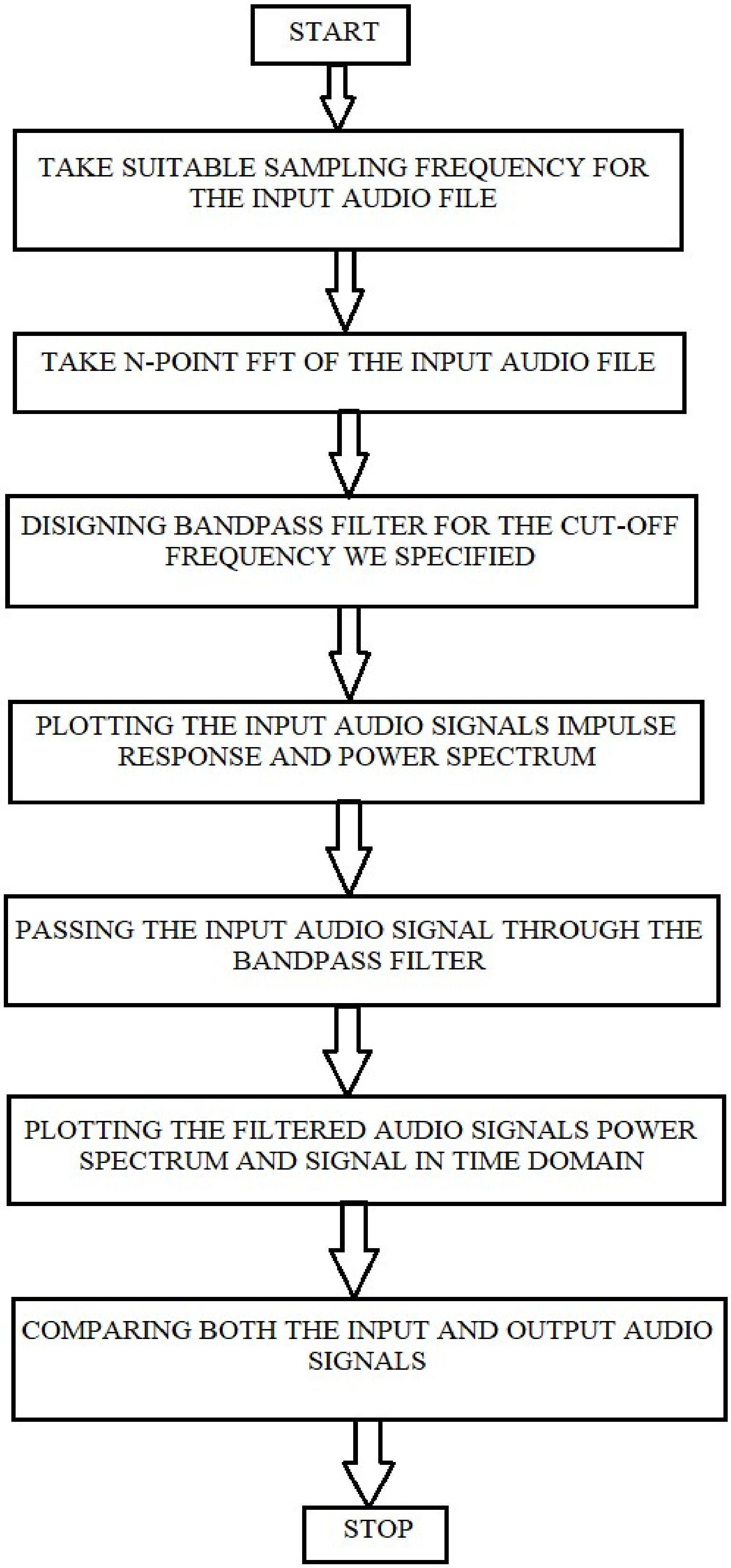
So the input frequency will be allowed to pass through only if it is within the limit of the bandwidth.

We have recorded an audio file and applied it as input to the bandpass filter. The passband frequency allows signals between two specific frequencies to pass, but it discriminates against signals at other frequencies. The passband frequencies are 300 Hz and 1700 Hz. The bandwidth of the signal is 1400 Hz. Stop band frequencies are the one over which the filter is intended not to transmit its input. Stop band frequencies are 300 Hz and 1800 Hz. In this case, the minimum order of the filter comes to be 203; which is too large. Higher order of the filter means the number of elements such as delays, adders will be more. This will tend to increase the hardware of the system. So, order is set in a way that the system will require less hardware which will directly affect on the cost which will require to make the system.

So, in this project, we had made one more filter whose order is set by us manually; which will result is less requirement of hardware and directly the system will be in the budget. We had set the order of system to be equal to 20.

Since the order of the system is large which increases the delay and further increases the hardware, we have used the windowing technique so that the order of the system is optimized.

Flowchart:


START
TAKE SUITABLE SAMPLING FREQUENCY FOR THE INPUT AUDIO FILE
TAKE N-POINT FFT OF THE INPUT AUDIO FILE
DISIGNING BANDPASS FILTER FOR THE CUT-OFF FREQUENCY WE SPECIFIED
PLOTTING THE INPUT AUDIO SIGNALS IMPULSE RESPONSE AND POWER SPECTRUM
PASSING THE INPUT AUDIO SIGNAL THROUGH THE BANDPASS FILTER
PLOTTING THE FILTERED AUDIO SIGNALS POWER SPECTRUM AND SIGNAL IN TIME DOMAIN
COMPARING BOTH THE INPUT AND OUTPUT AUDIO SIGNALS
STOP

4) Code:

```
clearvars;
clc
close all;
[x,Fs] = audioread('fvoice.wav');
N = size(x,1);     %x-rows and 1-column
f = Fs*(0:N-1)*N;
T = 1/Fs;
L = N;
t = (0:L-1)*T;

%filterDesigner
%fstop1=200
%fpass1=400
%fpass2=1800
%fstop2=2000

load('coefficients_pre.mat')
filtered_signal = filter(coefficients_pre,1,x);
audiowrite('foutput.wav',filtered_signal,Fs);

figure
 title('Frequency Response');
 freqz(coefficients_pre,1,N)
 figure
 title('Pole Zero Plot');
 zplane(coefficients_pre,1)
 figure
 title('Impulse Response');
 impz(coefficients_pre,1)

 [x,Fs]=audioread('fvoice.wav');
 L=length(x);
 df=Fs/L;
 frequency_audio=-Fs/2:df:Fs/2-df;
 X=fftshift(fft(x))/length(fft(x));
 figure
 subplot(2,1,1)
 plot(frequency_audio,abs(X));
 axis([-4000 4000 0 0.005]);
 title('FFT of Input Audio');
 xlabel('Frequency(Hz)');
 ylabel('Amplitude');

 [y,audio_freq_sampl]=audioread('foutput.wav');
 L=length(y);
 df=audio_freq_sampl/L;
 frequency_audio=-Fs/2:df:Fs/2-df;
 Y=fftshift(fft(y))/length(fft(y));
 subplot(2,1,2)
 plot(frequency_audio,abs(Y));
```

```
axis([-4000 4000 0 0.005]);
title('FFT of Output');
xlabel('Frequency(Hz)');
ylabel('Amplitude');

figure
subplot(2,1,1)
plot(t, x);axis([0 12 -0.4 0.4]);title('Original: Time-domain'); xlabel('time(seconds)');
subplot(2,1,2)
plot(t, filtered_signal , 'g');axis([0 12 -0.4 0.4]);title('Output'); xlabel('time(seconds)');

Pow= (abs (X).^2)/N^2;
figure
subplot(2,1,1)
plot((frequency_audio), Pow*4);
axis([-4000 4000 0 10*10^(-15)]);
xlabel('frequency')
ylabel('power spectral')
title('power spectrum of input signal')

Pow= (abs (Y).^2)/N^2;
subplot(2,1,2)
plot((frequency_audio), Pow*4);
axis([-4000 4000 0 10*10^(-15)]);
xlabel('frequency')
ylabel('power spectral')
title('power spectrum of output filtered signal ')


soundsc(x,Fs);

soundsc(y,Fs);
```

6) Results:

Band pass filters are used for various applications with excellent performance, reliability and environmental stability.

The sampling frequency F_s is 8000 Hz. The minimum order of the system for equiripple is come out to be 203 for fstop1 = 300, fpass1= 400, fpass2= 1700, fstop2 = 1800 and for same specifications we fixed the order to be equal to 20 because the output order of filter comes out to be very large. Due to large order, there is increase in the hardware elements which directly affects the cost of the system. But the output varies dominantly as we reduce the order of filter. By doing comparison, we found that as following:

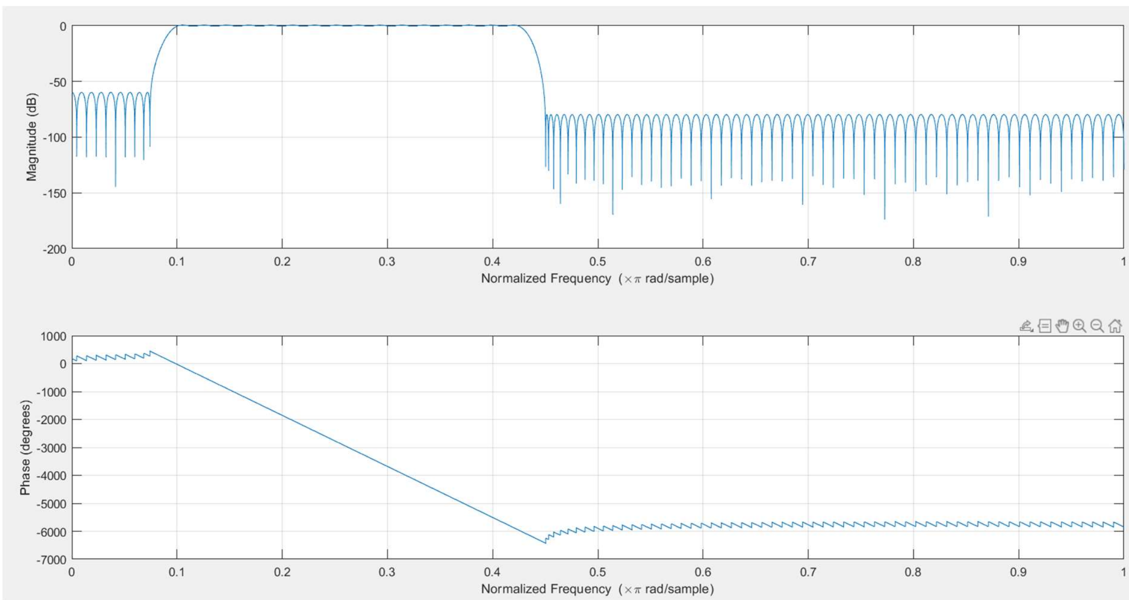


Fig1. a . Frequency Response when order is 203

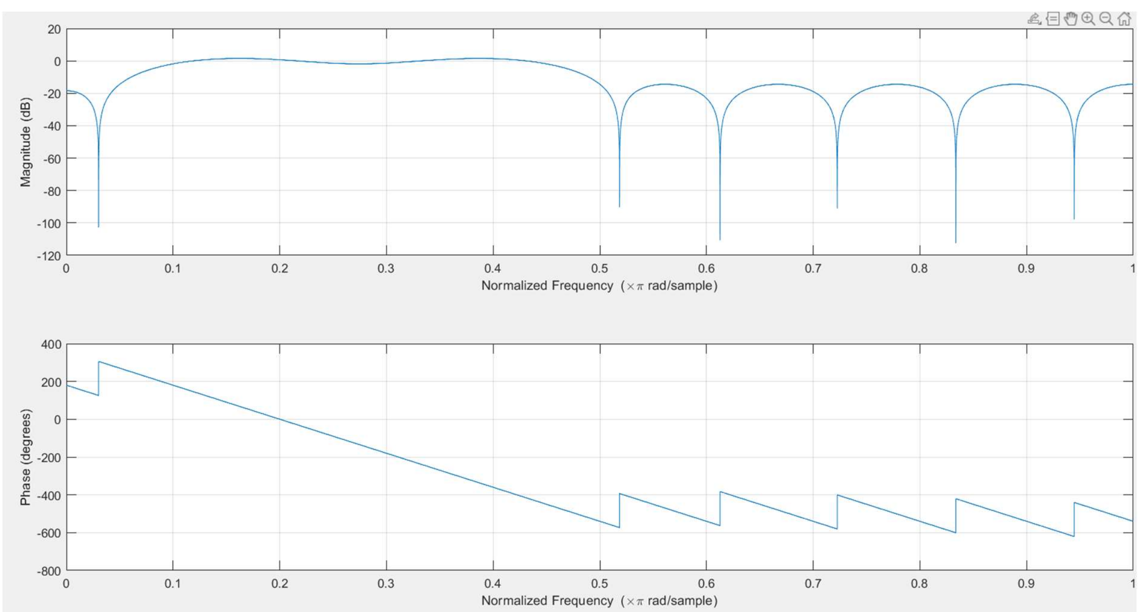


Fig1. b. Frequency Response when order is 20

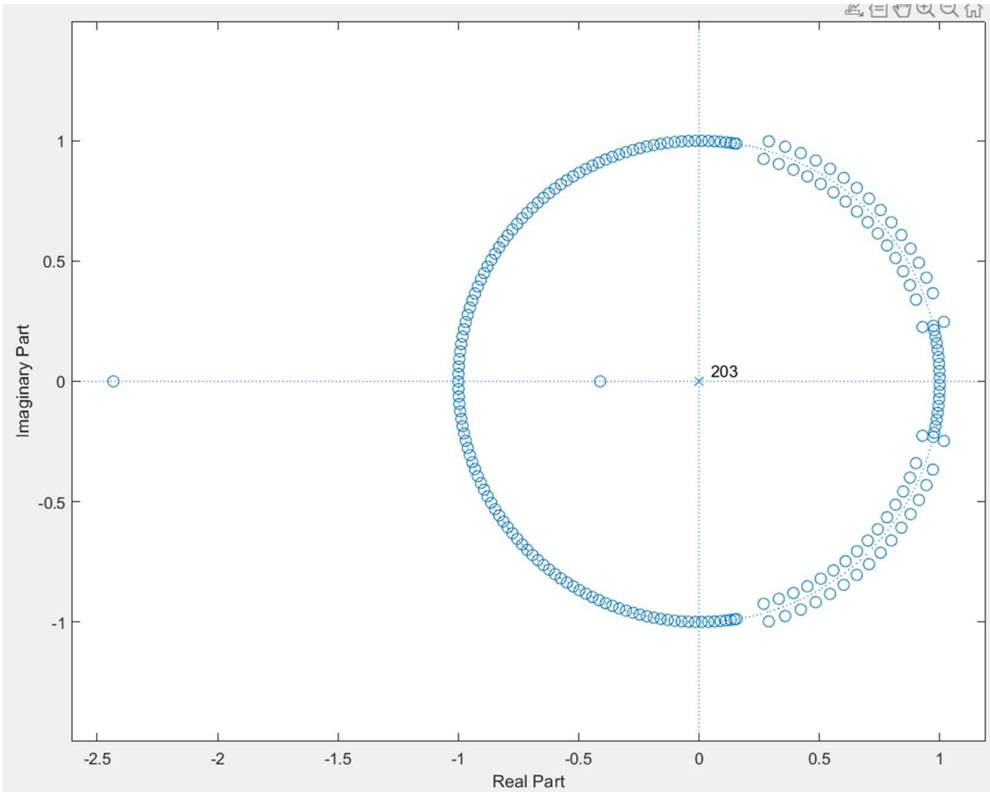


Fig2. a. Pole-Zero plot when order is 203

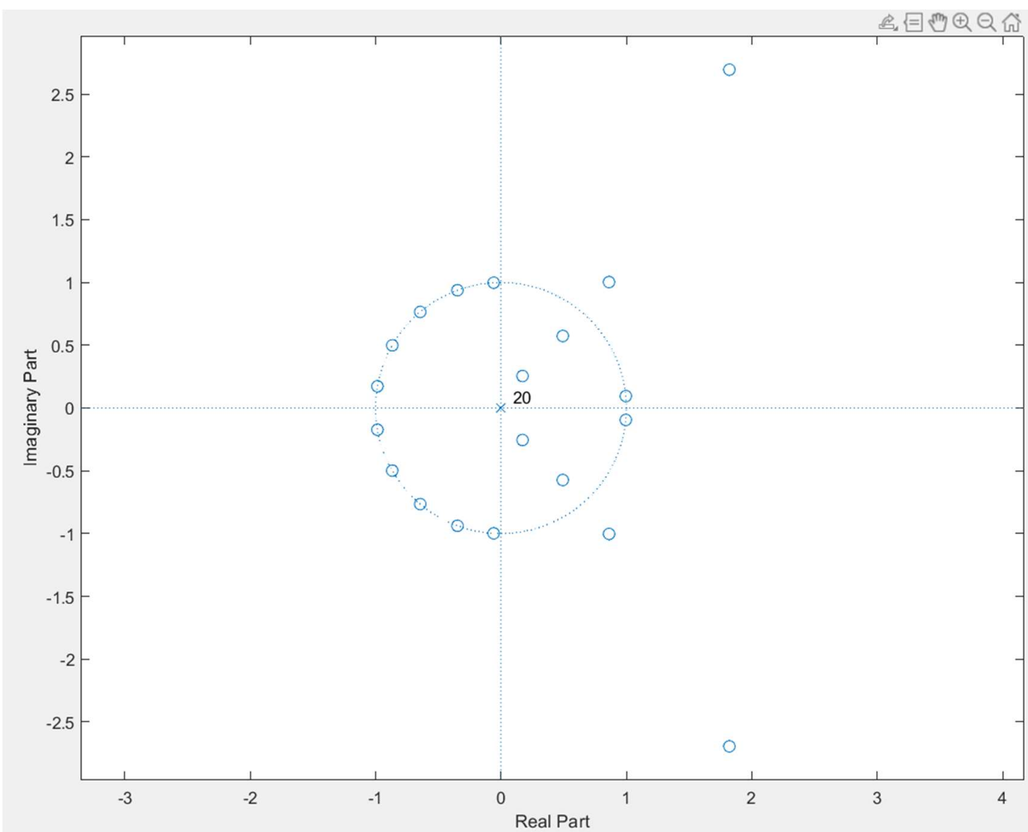


Fig2. b. Pole-Zero plot when order is 20

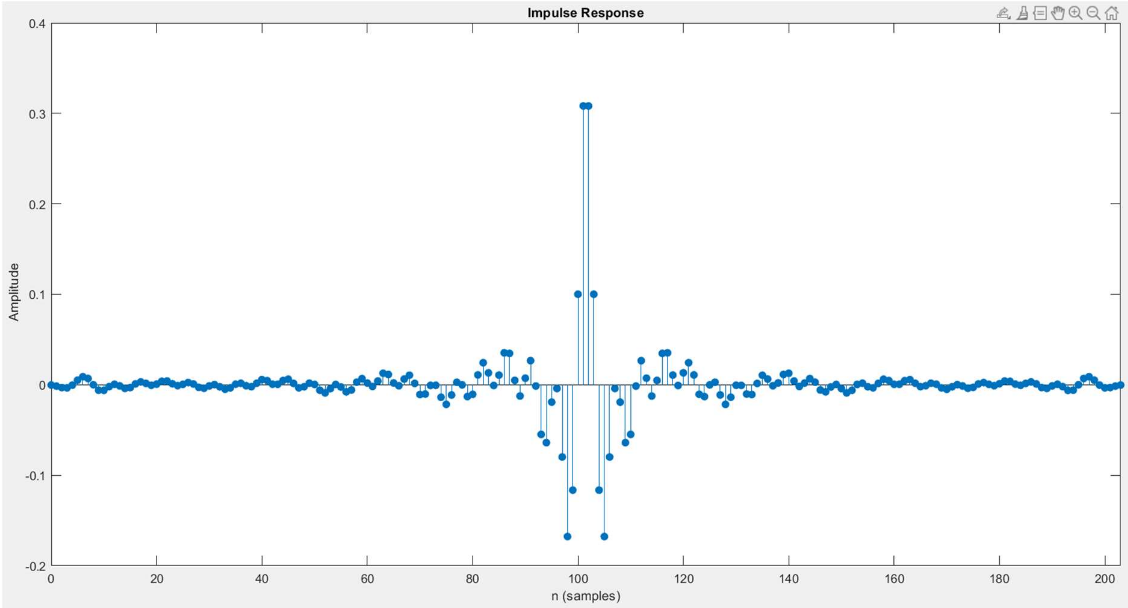


Fig3. a. Impulse Response when order is 203

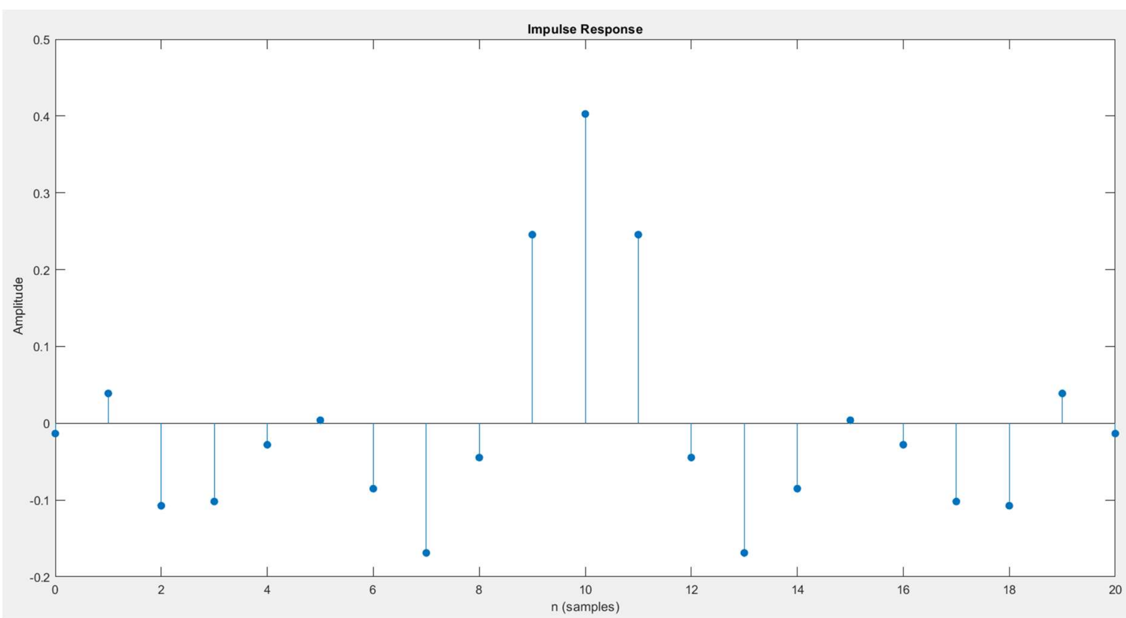


Fig3. b. Impulse Response when order is 20

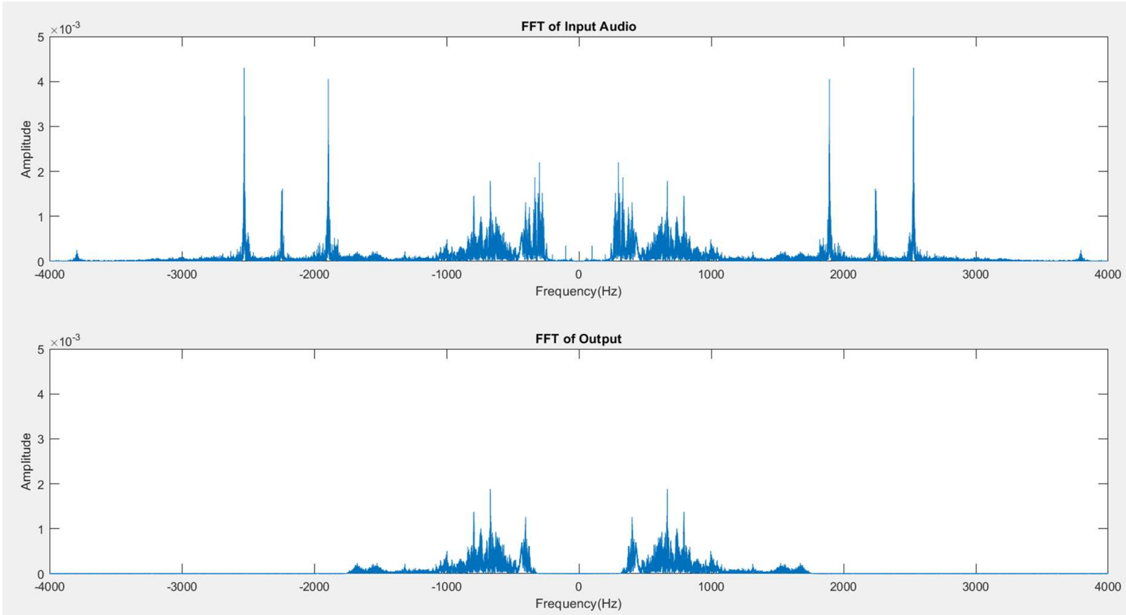


Fig4. a. Fourier Transform Plot when order is 203

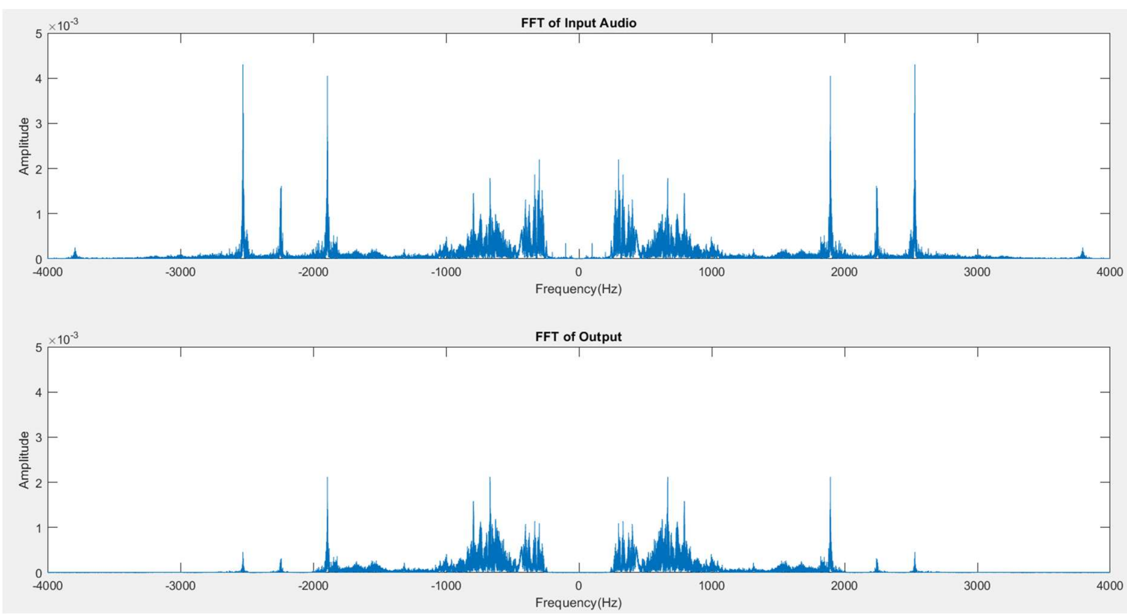


Fig4. b. Fourier Transform Plot when order is 20

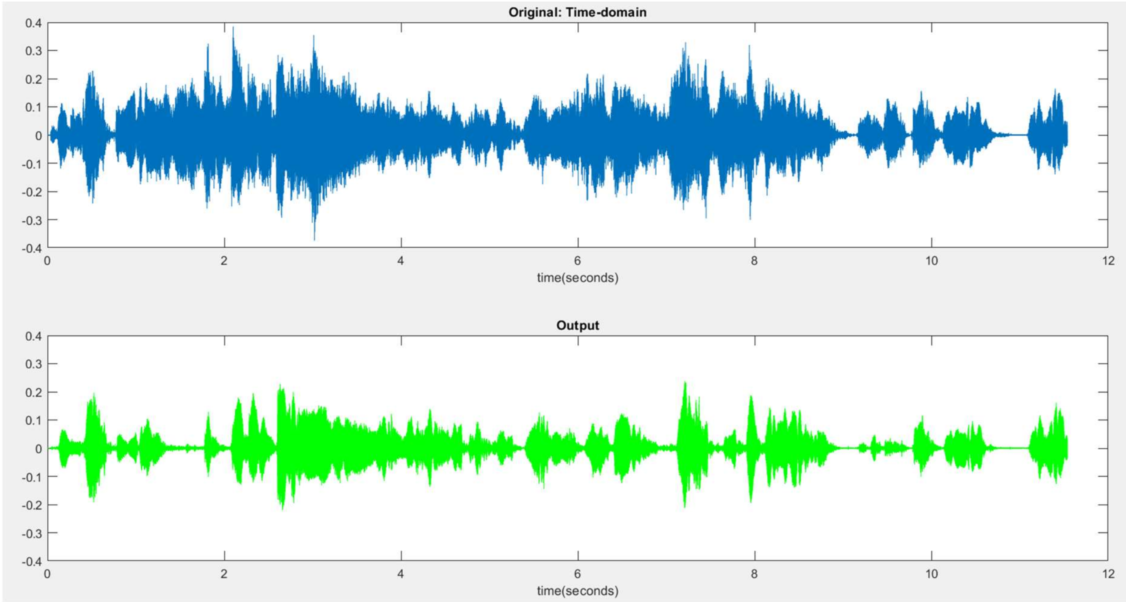


Fig5. a. Signal in Time Domain when order is 203

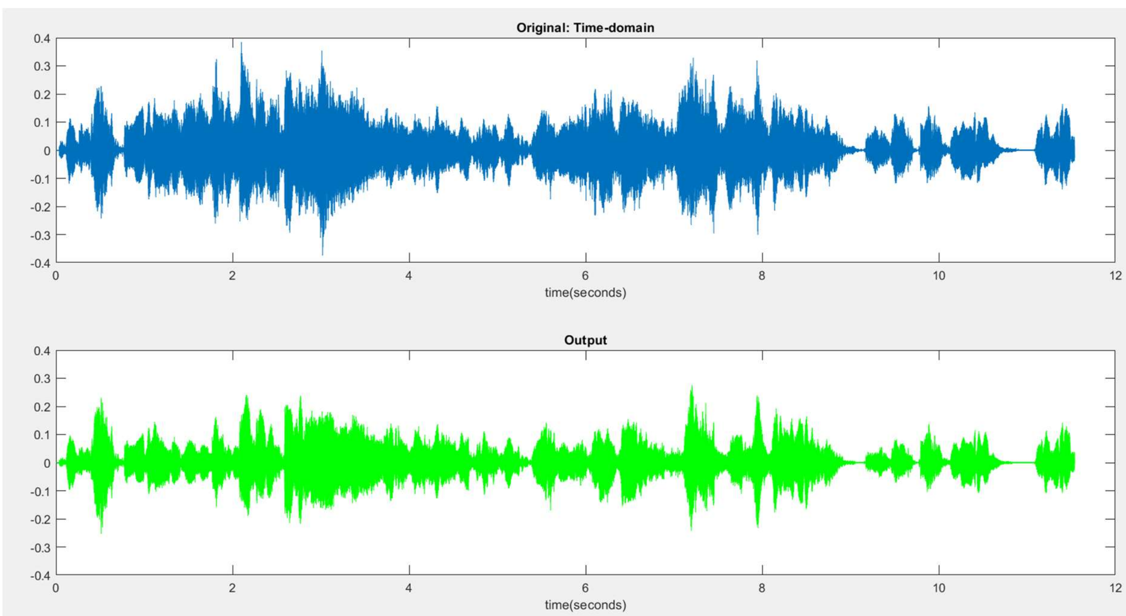


Fig5. b. Signal in Time Domain when order is 20

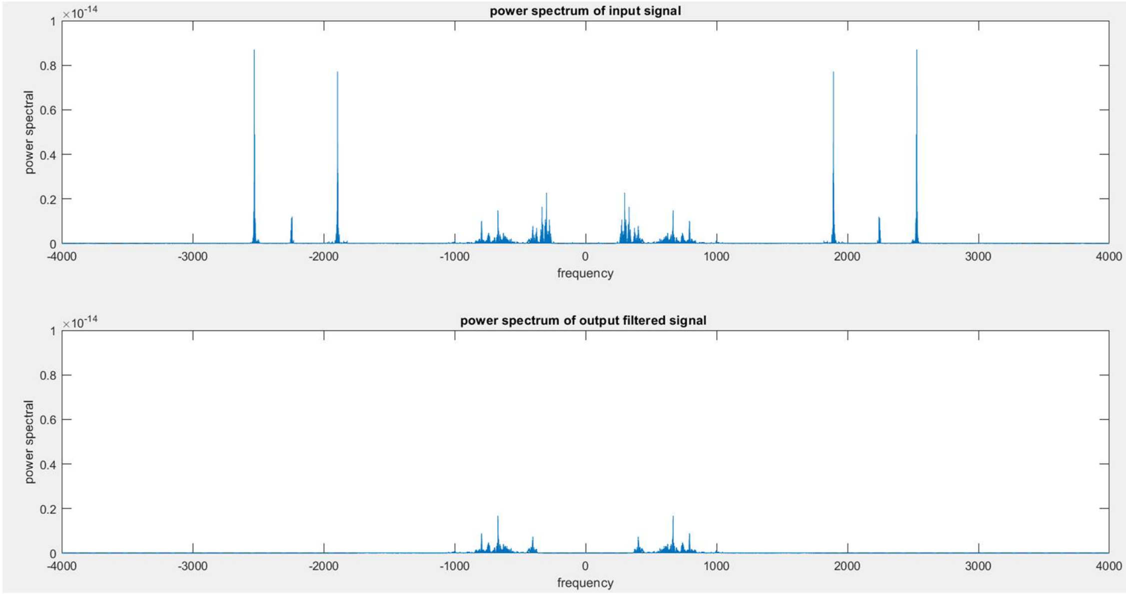


Fig6. a. Power Spectrum when order is 203

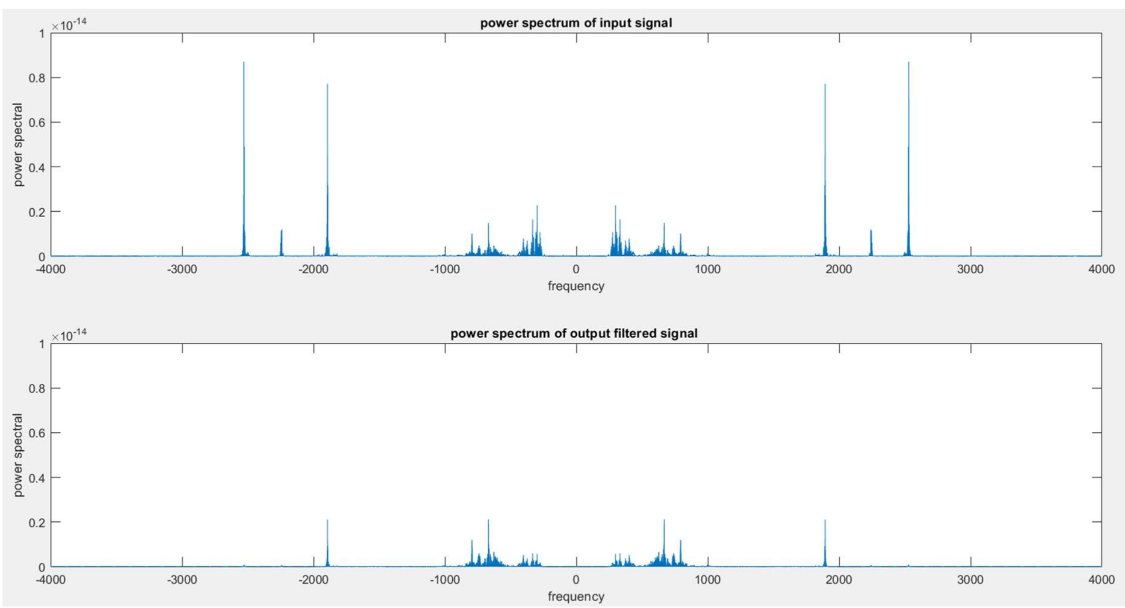


Fig6. b. Power Spectrum when order is 20

So, we can conclude that,

I. We are able to observe from graphs that the noise is getting more annetuated when the order of system is more.
II. As order of system increases, the output gets more precise.
III. But increase in order of the system has leads to the increase in the size of hardware; which affects the cost of system directly.

7) Applications:

BPFs (Bandpass filters) are extensively used in wireless transmitters and receivers.

• They are used in all types of instruments as well as in Sonar, Seismology and even medical applications like EEGs and Electrocardiograms. These filters are also extensively used in optics like lasers, LIDARS, etc.

• A good application of a BPF is in Audio Signal Processing, where a particular range of frequencies of sound is required while removing the rest.

• They are extensively used in satellite communication systems where in the receiver, a bandpass filter allows signals within a selected range of frequencies to be heard or decoded, while preventing signals at unwanted frequencies from getting through.

• When the passive filters are used in conjunction with switches, it offers low insertion loss and performance increases.

• Some car audio systems add a fourth set of speakers in the form of a small midrange driver.

8) Limitations and Future scope:

Limitations:

1) We cannot change the individual transition bands of the bandpass filter as compared to the bandpass filter made from the cascading of low pass filter and high pass filter in which both the transition band is changeable.

2) If we design a bandpass filter of order N, we can make it by cascading the low pass filter and high pass filter. But in such condition, we will have to determine the order of the high pass and low pass filter of order N1 and N2 respectively such that N will be equal to the product of N1 and N2; which is difficult for odd ordered systems.

Future Scope:

We can implement various windowing techniques such as Rectangular, Hamming, Kaiser technique to further improve the filter for better output in less order of the filter system.

9) References:

1) https://www.allaboutcircuits.com/technical-articles/an-introduction-to-filters/#:~:text=A%20filter%20is%20a%20circuit,many%20practical%20applications%20for%20filters.
2) https://www.mathworks.com/help/signal/ref/zplane.html
3) https://www.mathworks.com/help/signal/ref/freqz.html
4) https://www.elprocus.com/what-is-a-band-pass-filter-circuit-diagram-types-and-applications/
5) https://www.watelectronics.com/what-is-band-pass-filter-circuit-its-working/
6) https://www.digitalsurf.com/blog/what-is-a-bandpass-filter-and-should-i-use-it/
7) https://www.mathworks.com/help/signal/ref/bandpass.html
8) https://www.allaboutcircuits.com/textbook/alternating-current/chpt-8/band-pass-filters/
9) https://www.researchgate.net/figure/Flow-chart-for-digital-signal-processing-SNR-Signal-to-noise-ratio_fig2_225781492
10) https://circuitdigest.com/electronic-circuits/band-pass-filter-circuit-diagram