1. Introduction:

Digital filters are a very important part of Digital signal processing,. In fact, their extraordinary performance is one of the key reasons that DSP has become so popular. Filters have two uses: signal separation and signal restoration. Signal separation is needed when a signal has been contaminated with interference, noise, or other signals. Signal restoration is used when a signal has been distorted in some way. Analog filtering involves physical hardware that alters analog signals before they are passed off to other components to be processed. Digital filtering involves passing analog data to a processor that then runs code to digitally filter the data . The purpose of the filters is to allow some frequencies to pass unaltered, while completely blocking other frequencies. The pass band refers to those frequencies that are passed, while the stop band contains those frequencies that are blocked. The transition band is between pass band and stop band, fast roll-off means that the transition band is very narrow.

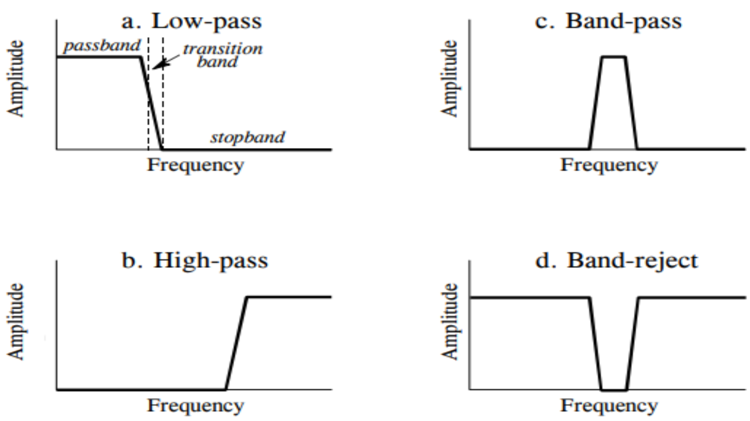


Figure 1 - Response of different types of filters.

- **Advantages of Analog Filter:**

Analog filters have the main advantage of speed.Filtering with hardware means that the signal coming out of the physical filter is the final signal. Analog filters also provide greater dynamic range for frequency.

Disadvantages :

Analog filters require physical space, so they must be used sparingly if space is an issue. As with any physical hardware, if there is an issue with its design, it is much harder to fix once a product is deployed, as hardware cannot be altered.

- **Advantages Digital Filtering :**

The advantages to digital filtering are numerous. The most apparent is that digital filters require less hardware, as they are done on a processor. This makes them very versatile and applicable in any system with a processor. It has low cost. Digital filters are software programmable, which makes them easy to bring up and test.

Disadvantages :

The standard disadvantage of a digital filter is that digital filters are significantly slower than analogfilters . It is also difficult handle large frequency ranges with digital filters.

2. Objectives

a) To stop certain frequencies based on application of band reject filter

A band reject /band stop filter has to be designed using filter designer tool or fda tool, by giving the input sample frequency and two cutoff frequencies, by using filter designer tool we can get the order of the filter, its magnitude spectrum, its phase spectrum, pole-zero plot and also its stability nature, hence we can design a filter according to necessary requirements.

b) To get the results of band reject filter by eliminating higher sharp frequencies component from an input audio.

An input audio file which has high frequency values in it and has a sharp sound of frequencies is taken. Here we are designing a band reject filter which stops the higher sharp frequencies and the output of the signal is a smooth audio. This can also be observed well by using sound function to play the input and output audio files.

c) To compare the graphs of input, output signal, obtained from time and frequency response.

After implication of each step, graph is to be observed, at first frequency response of the input signal and output signal, then time domain response of input and output signal, in this way a comparison has to be made between these graphs and input and output results.

3. Theoretical Background

Filters are mainly classified into two types depending on the duration of their impulse responses. They are known as infinite impulse response (IIR) filters and finite impulse response (FIR) filters.

IIR FILTER :

The IIR filters represent the digital filters that generate infinite impulse response of a dynamic system. The transfer function of the IIR filter contains both the poles and zeros in it. In order to make the filter stable, the poles of the filter must lie inside a unit circle. The sensitivity of the IIR filter is more, hence not easy to control.

FIR FILTER:

The FIR filters represent the digital filters that generate finite impulse response of a dynamic system. In the transfer function of FIR filters, only zeros are present, making the filter more stable. The generation of current output requires the past and present samples of the input. The FIR filter has a comparatively low sensitivity that makes the filter more controllable.

IIR Filter	FIR Filter
IIR filter stands for infinite impulse response filter.	FIR filter stands for finite impulse response filter.
An IIR filter gives impulse responses for an infinite duration of time.	A FIR filter provides impulse responses for a finite duration of time.
IIR filter has poles and zero.	FIR filter has zeros.
IIR filter has closed loop feedback system.	The FIR filter is open loop.
IIR filters are less stable than that of FIR filters due to the presence of poles.	FIR filters do not contain any poles. Hence the FIR filter is always stable and the stability is more than that of IIR filter.
Difference equation of LTI system for IIR filter becomes : $y(n)=-\sum_{k=1}^{N} a_k\, y(n-k)+\sum_{k=0}^{M} b_k\, x(n-k)$	Difference equation of LTI system for FIR filter becomes : $y(n)=\sum_{k=0}^{M} b_k\, x(n-k)$

In this project we have designed a band reject FIR filter. The band stop/ reject filter is a type of frequency selective circuit, The filter that allows above and below the particular range of frequencies and rejects all other frequencies of a given input signal, is known as band stop filter. It is also known as a band-reject filter or band elimination filter or notch filter. The name itself shows that it stops or rejects the particular range of frequencies of a signal. This filter is designed with the low pass filter and high pass filter, which are connected in parallel to allow high and low-frequency components. The band stop filter allows frequency components below the cut-off frequency and above the cut-off frequency. The cut-off frequency of the low pass filter is denoted as fL and the cut-off frequency of the high pass filter is denoted as f_H. When the input signal is applied, the high frequencies are passed through a high pass filter and low frequencies are passed through a low pass filter. The range of frequencies between the fL and fH is attenuated.

The band stop filter theory can be understood by using the block diagram shown below.

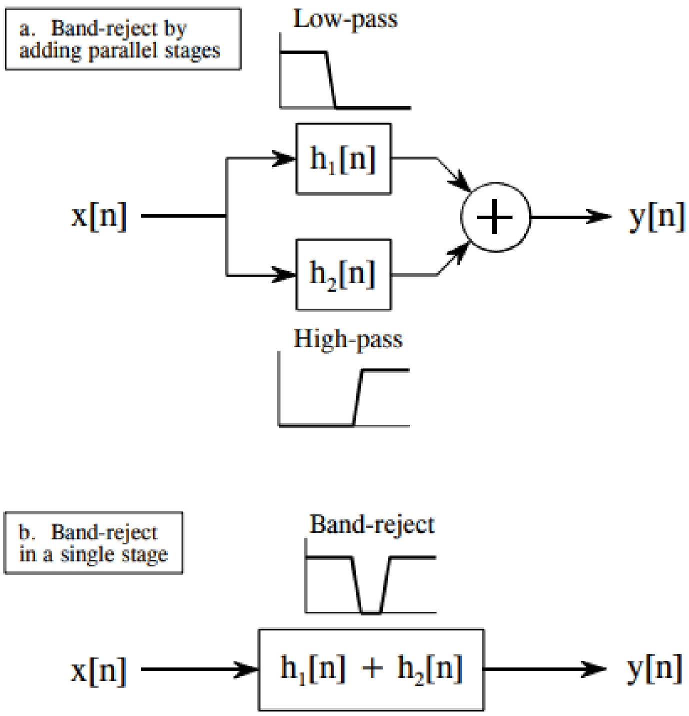


Figure 2 – Band reject filter as parallel combination of LPF and HPF

From the block diagram, we can observe that a band stop filter is a combination of a high pass filter and a low pass filter connected in parallel instead of series. The applied input signal is applied to the filter, the low pass filter allows only the low frequencies whereas the high pass filter allows the high frequencies of the signal. So, the bandstop filter will have two cut-off frequencies named as lower cut-off frequency and upper cut-off frequency.
The low pass filter allows the frequencies below its cut-off frequency while the high pass filter allows the frequency above its cut-off frequency. These two cut-off frequencies are predetermined based on the component values used in the circuit. The bandstop filter doesn't allow the frequencies between these two cut-off frequencies and are attenuated or rejected.

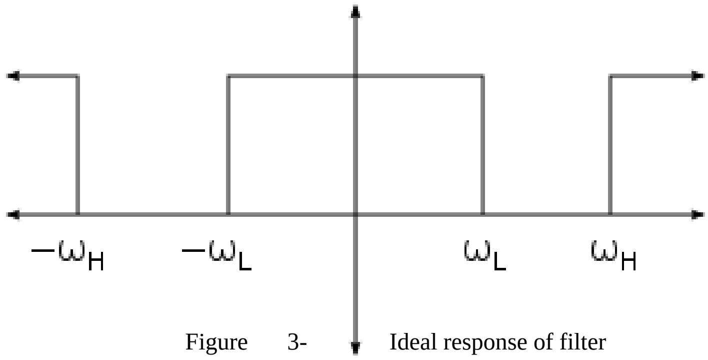


Figure 3- Ideal response of filter

The ideal band stop filter would have infinite attenuation in its stop band and zero attenuation in either pass band. The transition between the two pass bands and the stop band would be vertical (brick wall) as shown in figure a.

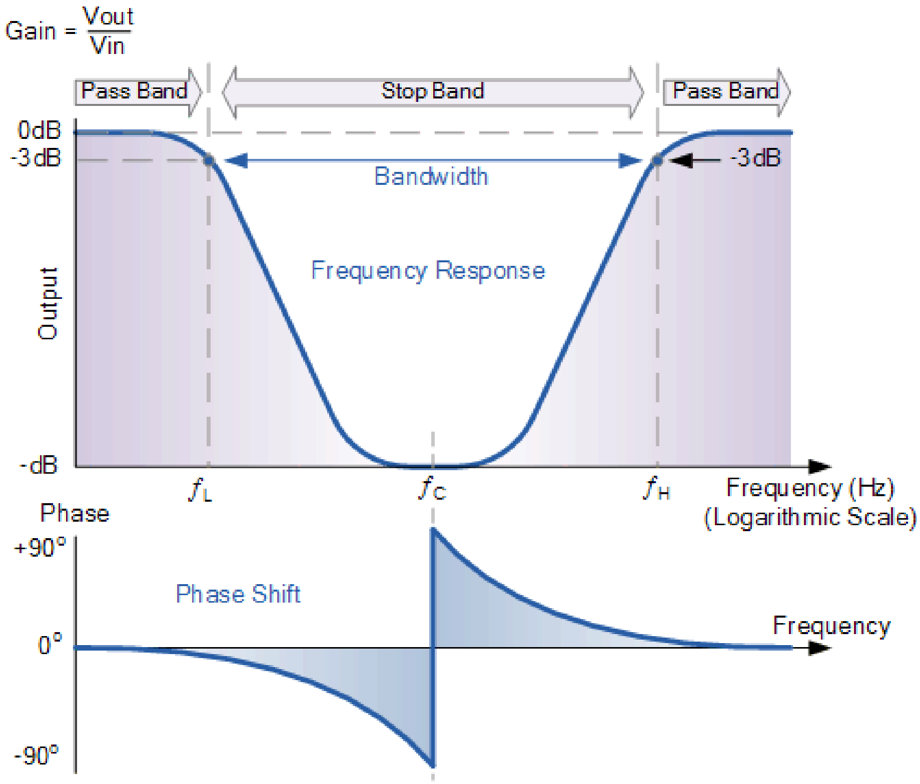
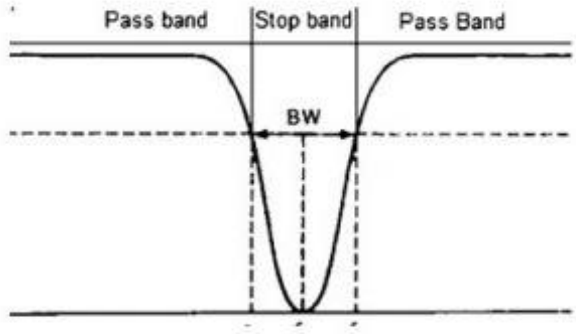


Figure 4 – Magnitude and phase spectrum

As we can see from the amplitude and phase curves above for the band pass circuit, that the quantities f_L, f_H and f_C are the same as those used to describe the behaviour of the band-pass filter. This is because the band stop filter is simply an inverted or complimented form of the standard band-pass filter.

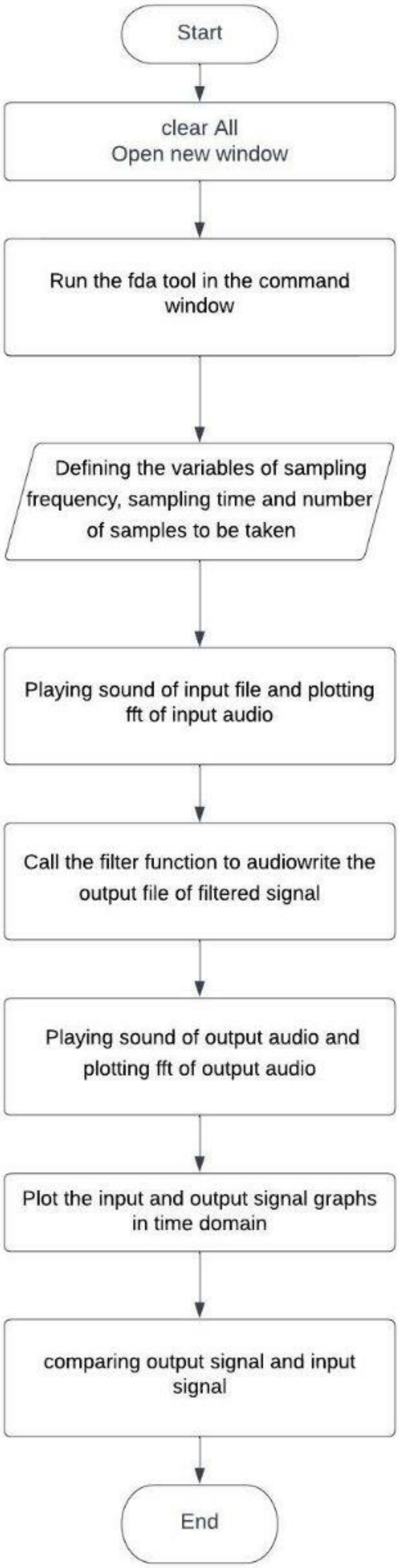


The narrow stop band filter is referred to as the NOTCH filter. For the elimination of single frequency, this notch filter is used. It is also called as twin T network due to its two T shaped networks. At center frequency $f_C = 1/2\pi RC$, maximum elimination takes place.
The transition band is a range of frequencies that allows a transition between a pass band and a stop band of a signal processing filter. When the transition band is narrower then the characteristics approach to ideal characteristics of filter. The width of transition band can be reduced by increasing the order of the filter.

Steps to be followed in the code.

1. Clear the command window
2. Run the fda tool in the command window and set the cutoff frequencies of fir filter according to requirement of which frequencies need to be stopped. Due to this a num variable which stores value of coefficients of numerator of the designed filter is created.
3. Defining the variables of sampling frequency, sampling time and number of samples to be taken
4. Playing sound of input file and plotting fft of input audio
5. Call the filter function to audiowrite the output file of filtered signal
6. Playing sound of output audio and plotting fft of output audio
7. Plot the input and output signal graphs in time domain
8. Comparison: the output signal is having a smooth audio which indicated the higher sharp frequencies of output signal is stopped and hence filter is designed successfully.

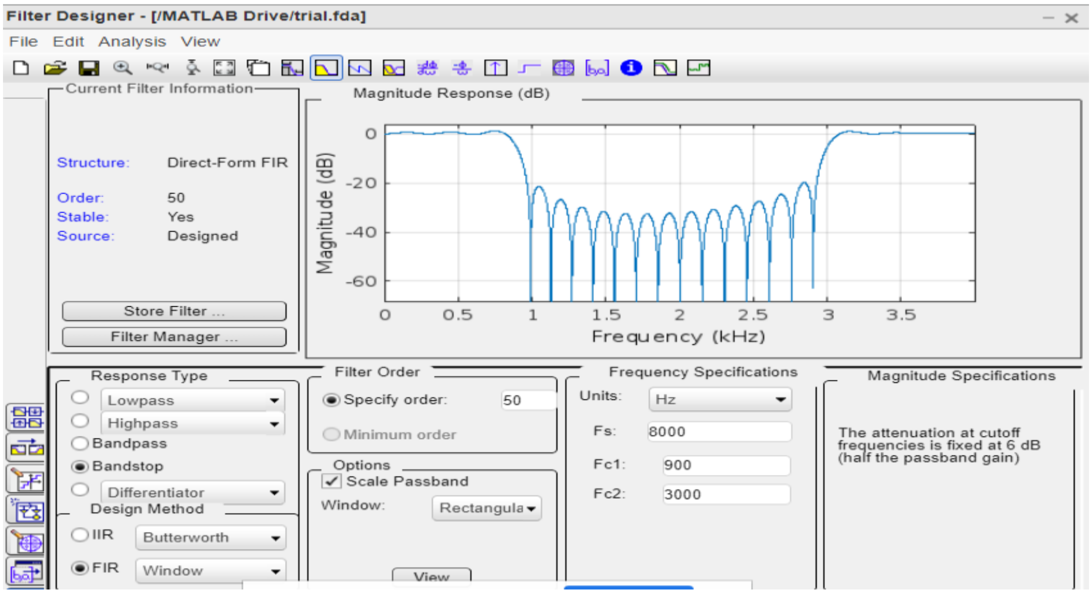
4. Flow chart


Start
clear All
Open new window
Run the fda tool in the command window
Defining the variables of sampling frequency, sampling time and number of samples to be taken
Playing sound of input file and plotting fft of input audio
Call the filter function to audiowrite the output file of filtered signal
Playing sound of output audio and plotting fft of output audio
Plot the input and output signal graphs in time domain
comparing output signal and input signal
End

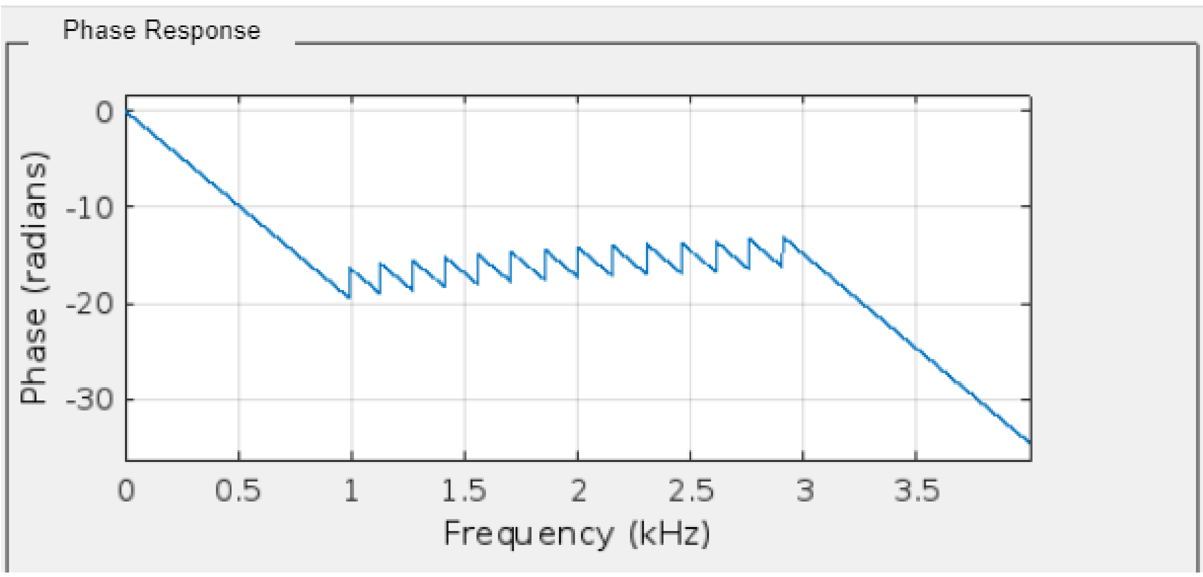
5. Code:

```
%clearvars;
%clc
%close all;
[x,Fs] = audioread('instrumental.mp3');
sound(x,Fs);

N = size(x,1);  %N=number of sanples
f = Fs*(0:N-1)*N;
T = 1/Fs;  %T=sampling time of input signal
t = (0:N-1)*T;

Length_audio=length(x);
df=Fs/Length_audio;
frequency_audio=-Fs/2:df:Fs/2-df;

figure
FFT_x=fftshift(fft(x))/length(fft(x));
plot(frequency_audio,abs(FFT_x));
title('FFT of Input Audio');
xlabel('Frequency(Hz)');
ylabel('Amplitude');
disp(''FFT of input'');
%%

y= filter(Num,1,x);
audiowrite('result.wav',y,Fs);
sound(y,Fs);
disp('Playing sound of output');
Length_audio=length(y);
df=Fs/Length_audio;
frequency_audio=-Fs/2:df:Fs/2-df;

figure
FFT_y=fftshift(fft(y))/length(fft(y));
plot(frequency_audio,abs(FFT_y));
title('FFT of Output Audio');
xlabel('Frequency(Hz)');
ylabel('Amplitude');
disp('Plotting FFT of Output Audio');
%%
figure
subplot(2,1,1)
stem(t , x);title('Original: Time-domain'); xlabel('time(seconds)');
subplot(2,1,2)
stem(t, y);title('After processing: Time-domain'); xlabel('time(seconds)');
disp('graphs in time domain');
```

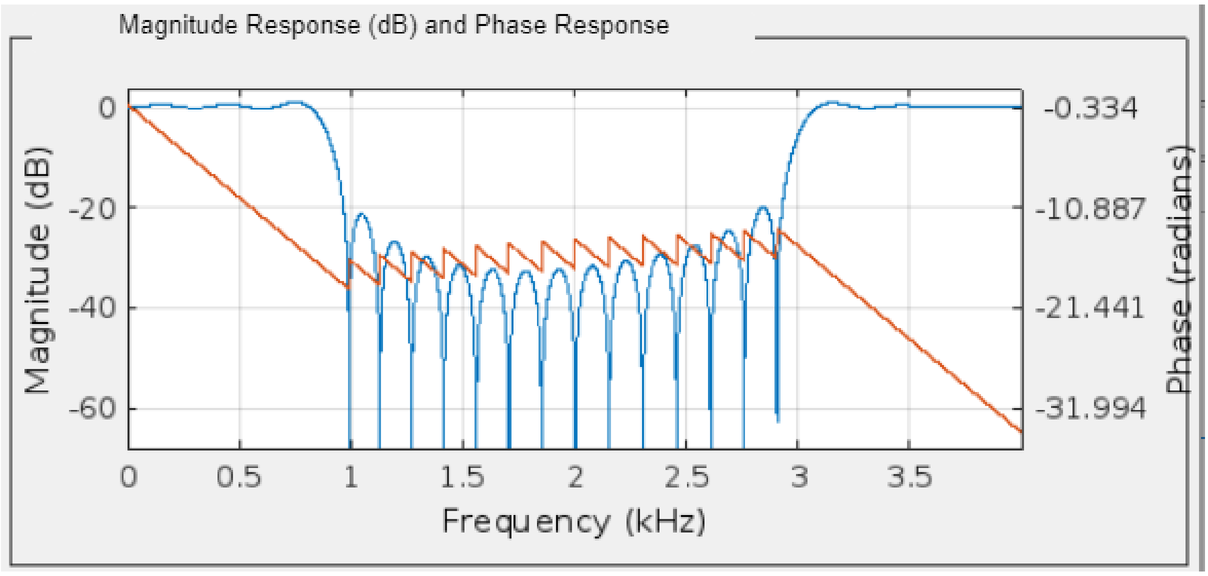
6. Results:



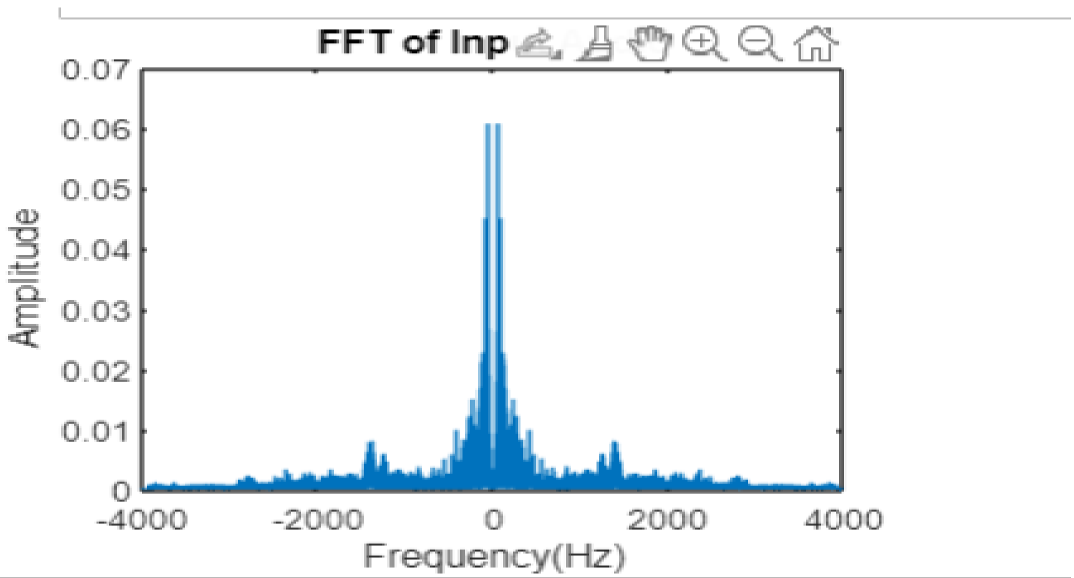
In this figure we have designed a band stop filter using filterDesigner tool. In this we have set the respective stop band and pass band frequency. After designing filter we get its order , stability structure, phase and magnitude spectrum graph.



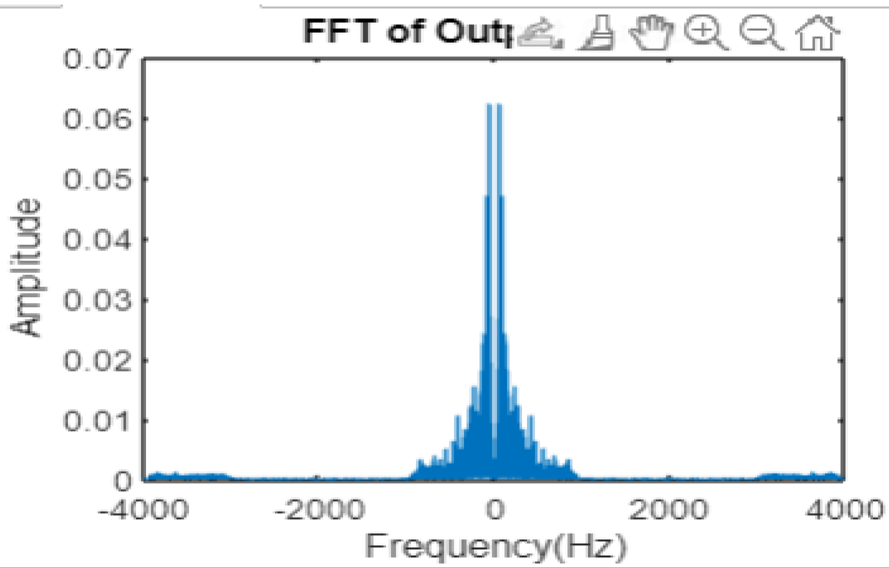
It gives the phase Response plot of the filtered band stop design.



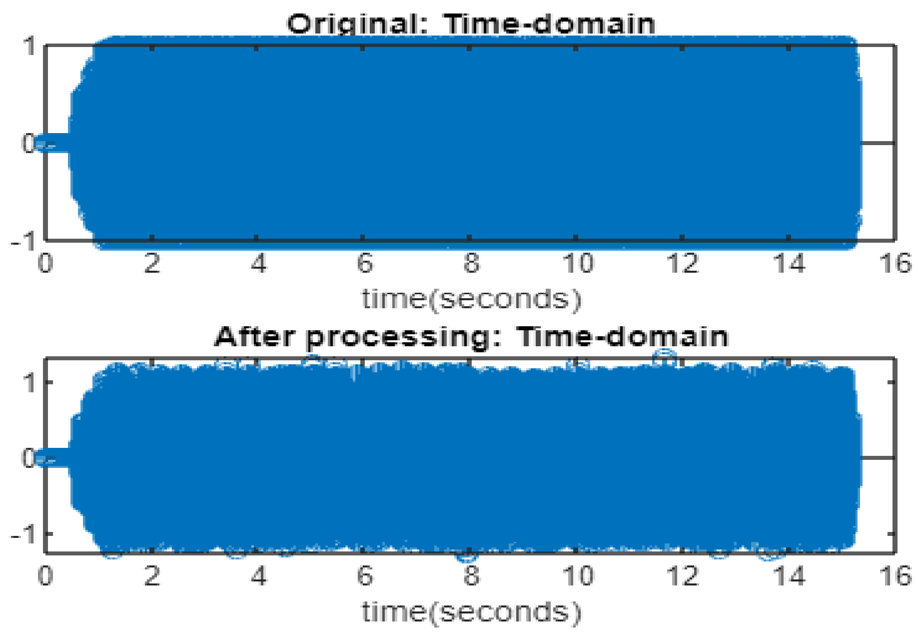
In this the magnitude and phase of the signal is shown after the filtering operation is carried out . It helps to understand that which frequencies are cutoff from the spectrum after using the filter. In this as it is spectrum on x axis we have frequency and on y axis we have magnitude (in dB) and phase(in radians) for magnitude and phase spectrum respectively.



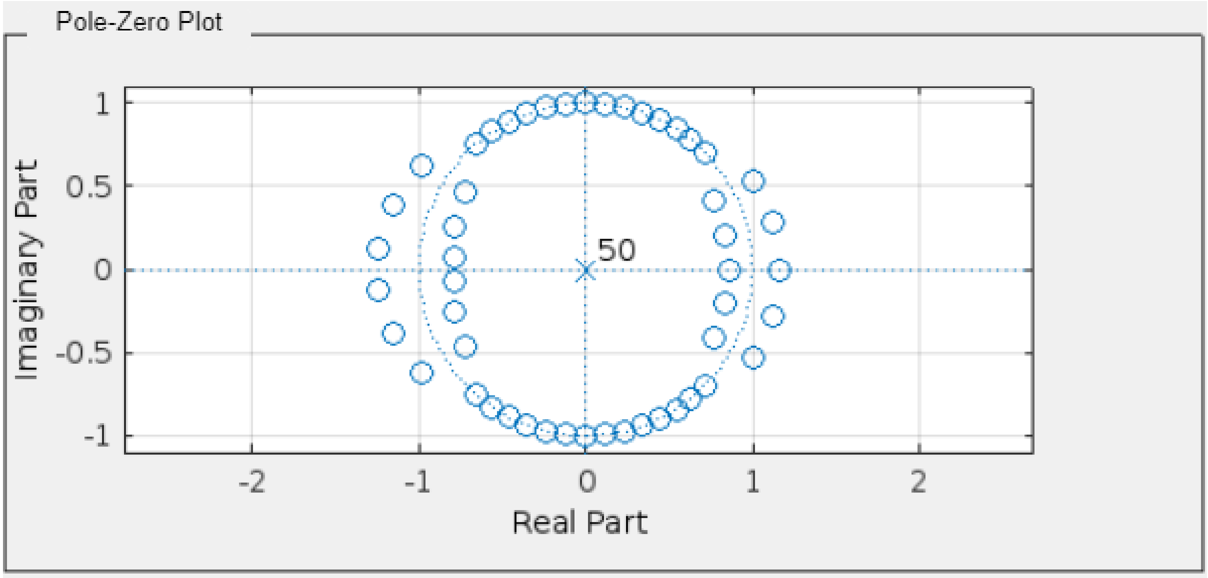
In the graph the fft (fast Fourier transform) of the input signal are plotted. In the x axis frequency is taken in Hz and on y axis amplitude of Signal is plotted.



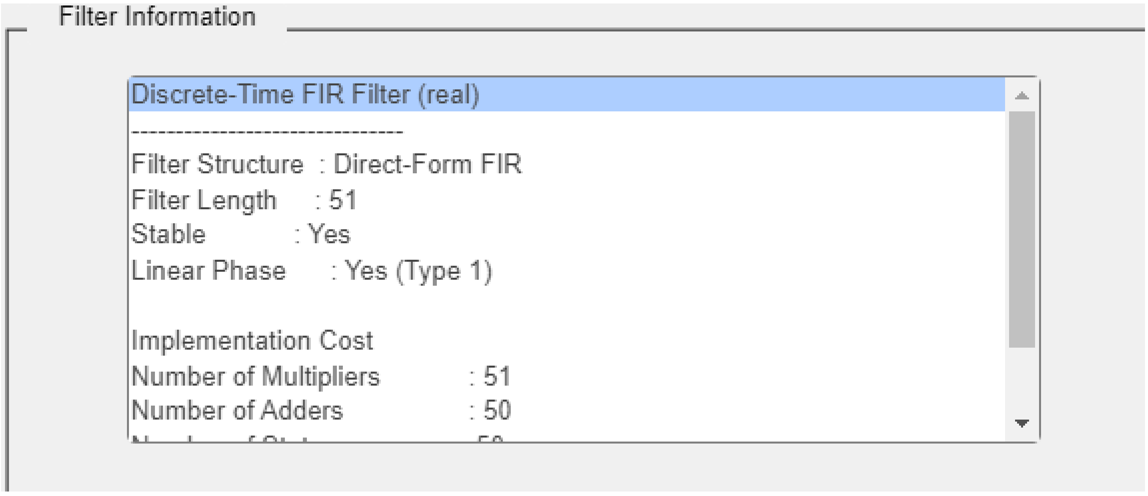
In the graph the fft (fast Fourier transform) of the output signal are plotted. In the x axis frequency is taken in Hz and on y axis amplitude of Signal is plotted. In this the noise cutoff frequency from 900 to 2500 Hz isremoved.



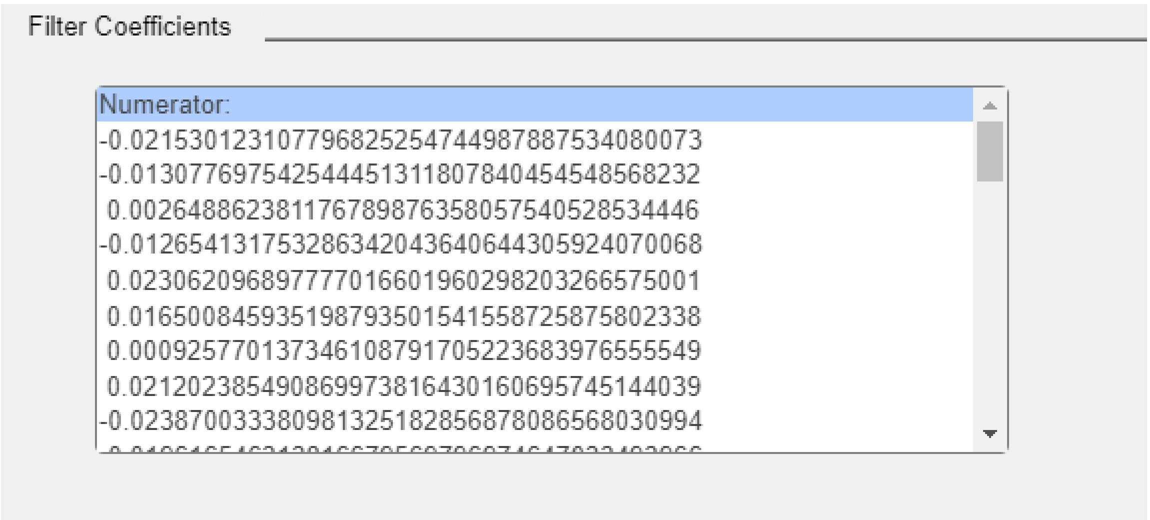
In this the input original signal graph and after filtering output signal graph is plotted.



The pole_zero plot of the designed filter is plotted here.



In this figure the filter information of designed filter is shown.



These are some of the coefficients of the numerator of the designed band-stop filter.

Workspace

Name	Value	Size	Class
df	0.0658	1×1	double
f	1×121536 d...	1×121536	double
FFT_x	121536×1 c...	121536×1	double (co...
FFT_y	121536×1 c...	121536×1	double (co...
frequen...	1×121536 d...	1×121536	double
Fs	8000	1×1	double
Length_...	121536	1×1	double
N	121536	1×1	double
Num	1×51 double	1×51	double
T	1.2500e-04	1×1	double
t	1×121536 d...	1×121536	double
x	121536×1 d...	121536×1	double
y	121536×1 d...	121536×1	double

This is the workspace that specifies the values of the variables used in the code of filter

Command Window

```
Playing sound of output
Plotting FFT of Output Audio
graphs in time domain
```

This is the command window that displays the execution of code

Conclusion :

In this we have designed a band stop which is stopping the required band of frequencies. Basically we have designed band stop filter based on its application in music industry. In this the music audio of sharp high frequencies is used as the input to the band stop filter. After setting the cutoff frequency of fir rectangular window band stop filter we get the output as a smooth music note. Hence the designed filter has removed the high frequency audio which can be even recognized as the output audio is smooth and is not piercing the ears. Hence we have successfully designed a band stop filter application which is used in musical industry to tone the musical audios.

7.Applications:

1. Band In telephone technology, band stop filter are used to reduce telephone line noise.
2. stop filter is used in electric guitar amplifiers, this guitar produces hum at 60 hz. The filter reduces that hum to amplify signal produced by guitar.
3. In communication electronics, band stop filter is used to eliminate the unwanted harmonics (noise).
4. In image and signal processing these filters are preferred to reject noise.
5. Used in medical field , in ECG signal to remove line noise.
6. In optical fibre at the end there is interfering of frequencies of light which makes distortion in light beam. These distortions are eliminated by band stop filter.

8.Limitations and future scope

Limitations

In this filter as we are increasing the stop band frequencies value, that is when we are trying to increase the band width of stop band, more repulse are generated. On increasing the stop band undoubtedly the more frequencies are being stopped and the output audio is smooth as desired but this is increasing the order of filter and the poles, zeroes are getting out of the unit circle which indicates instability of designed filter. To get a stable ideal filter characteristics with minimum filter order is a limitation of the designed filter.

Future scope:

This band stop can be used for wider range of stop band frequencies if we use the specify order option in the filter designer step. As order will increase no doubt the hardware increases but the characteristics reach to ideal characteristics with less ripples in bands. Here we have taken monotonic signal , in future we can implement stereo type signal also.

9.References:

- https://r.search.yahoo.com/_ylt=AwrKC.jD_G9j3QEuKlG7HAx.;_ylu=Y29sbwNzZzMEcG9zAzEEdnRpZAMEc2VjA3Ny/RV=2/RE=1668312387/RO=10/RU=https%3a%2f%2fwww.watelectronics.com%2fwhat-is-a-band-stop-filter-design-its-characteristics%2f/RK=2/RS=sM.J.Tr4eBOlIMl5Ok3Mq7eKQvI-
- https://www.google.com/url?sa=t&source=web&rct=j&url=https://wiki.analog.com/university/labs/band_stop_filters_adalm2000%23:~:text%3DA%2520Band%2520Stop%2520Filter%252C%2520also,to%2520pass%2520with%2520little%2520attenuation.&ved=2ahUKEwjBsZTLmqv7AhXcRmwGHSqxAq8QFnoECBUQBQ&usg=AOvVaw1CHUV_ywNw02Q5EB6oW4bu
- https://www.electronicshub.org/band-stop-filter/
- https://www.electronics-tutorials.ws/filter/band-stop-filter.html
- https://en.m.wikipedia.org/wiki/Band-stop_filter
- https://www.watelectronics.com/what-is-a-band-stop-filter-design-its-characteristics/