Contents

1. Introduction:

Digital filters play a crucial role in digital signal processing, driving its popularity due to their exceptional performance. They serve two primary purposes: signal separation, necessary when a signal encounters interference, noise, or other distortions, and signal restoration, used to rectify distorted signals. While analog filtering involves physical hardware altering analog signals before processing, digital filtering entails passing analog data to a processor for code-based filtration. These filters selectively permit certain frequencies to pass unchanged while entirely blocking others. The passband encompasses permitted frequencies, the stop band includes blocked frequencies, and the transition band lies between them. A fast roll-off indicates a narrow transition band.

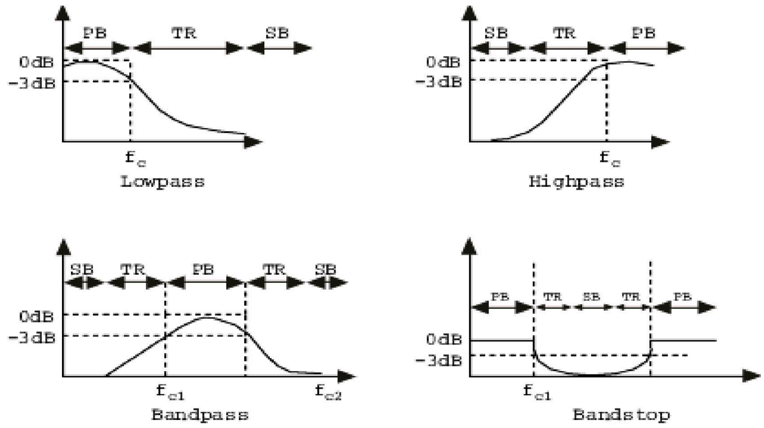


Figure 1 Four types of Filters and their Characteristics

Advantages of Analog Filters:	Disadvantages of Analog Filters:
• Smooth, continuous signal processing • Low latency, immediate manipulation • Simple design for basic tasks • Ideal for real-time applications	• Limited flexibility, requiring physical adjustments • Prone to noise, impacting accuracy • Signal degradation over time or distance • Costly and larger in complex setups
Advantages of Digital Filters: • Precise control for high accuracy • Strong immunity to noise interference • Consistent reproducibility • Highly adaptable and adjustable	**Disadvantages of Digital Filters:** • Processing demands may introduce delays • Potential aliasing effects • Quantization errors affect accuracy • Complexity in implementation requires specialized knowledge

2. Objectives:

a) To get the results of a Band-reject filter by eliminating higher frequency components from an input audio.

An input audio file that has a Jazz Music audio sample with high frequency components thus giving sharper sounding noise with the input. Here we have designed a band reject filter that stops the higher sharp frequencies and the output of the signal is smooth audio. This can also be observed well by using the audioplayer function to play the input and output audio files.

b) To understand how the band-reject filter shapes the spectrum of the input signal based on the filter characteristics.

Here we used uniform white noise as the input signal, thus all the frequencies in the spectrum contain equal magnitude. Thus, we can observe the attenuation or gain changes in the signal after passing through the filter.

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

We can compare the input and output audio waveforms to notice any significant changes that the filter has caused. Also, to understand the input and output frequency response and compare their magnitude spectrums to understand the rejection or thus suppression of a frequency band.

3. Theoretical Background:

The Band Rejection Filter is a frequency-selective filter that attenuates a specific range of frequencies while allowing others to pass through unaffected. The Band Stop filter has two passbands and one stopband. It removes or notches out frequencies between the two cut-off frequencies while passing frequencies outside the cut-off frequencies. One typical application of a band stop filter is in Audio Signal Processing, for removing a specific range of undesirable frequencies like noise, while not attenuating the rest. Another application is in the rejection of a specific signal from a range of signals in communication systems. The band-stop filter is formed when a low pass filter and a high pass filter are connected in parallel with each other. When the signal is given an input, a low pass filter allows the low frequencies to pass through the circuit and a high pass filter allows the high frequencies to pass through the circuit. 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 fH.

The range of frequencies between the fL and fH is attenuated. The summing of the high-pass and low-pass filters means that their frequency responses do not overlap. The difference in the starting and ending frequency points causes the two filters to connect effectively without any overlapping.

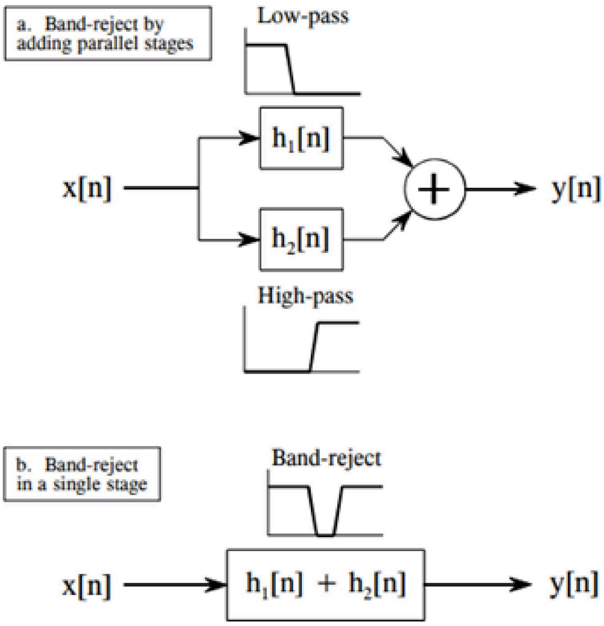


Figure 2 Construction of Band-Stop Filter using High Pass Filter and Low Pass Filter

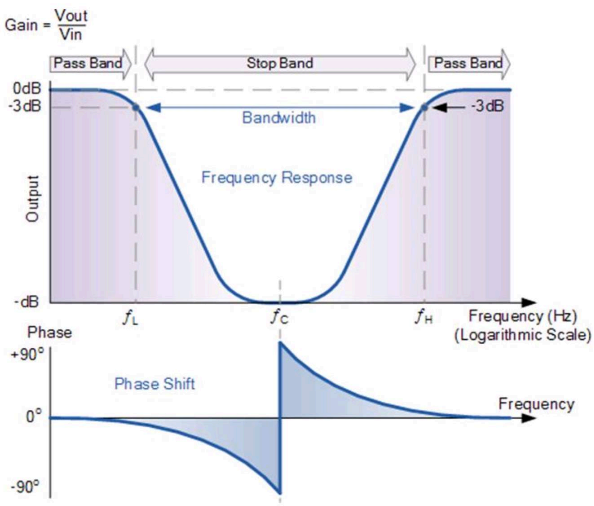


Figure 3 Magnitude and Phase Response of Band-Stop Filter

As we can see from the magnitude and phase curves above for the band-stop digital circuit, the quantities fL, fH, and fC are the same as those used to describe the behavior 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.

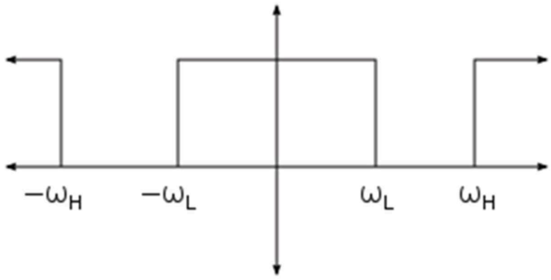


Figure 4 Ideal Band-Stop Filter Characteristics

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 passbands and the stop band would be vertical (brick wall) as shown in the figure. The narrow stop band filter is referred to as the Notch Filter. For the elimination of a single frequency, this notch filter is used. It is also called the twin T network due to its two T-shaped networks. At center frequency fC = 1/2πRC, maximum elimination takes place.

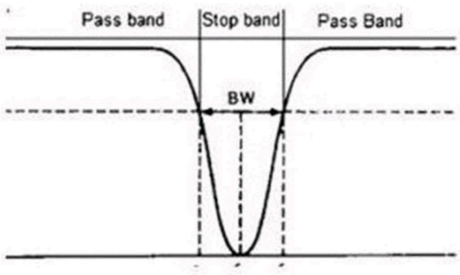


Figure 5 Band-Stop Filter with Narrow Bandwidth

Bilinear Transform: The Bilinear transform is a mathematical relationship that can be used to convert the transfer function of a particular filter in the complex Laplace domain into the z-domain, and vice-versa. The resulting filter will have the same characteristics as the original filter but can be implemented using different techniques. The bilinear transform can be used to produce a piecewise constant magnitude response that approximates the magnitude response of an equivalent analog filter.

$$\omega_p = \frac{2}{T}\tan\left(\frac{\omega T}{2}\right)$$

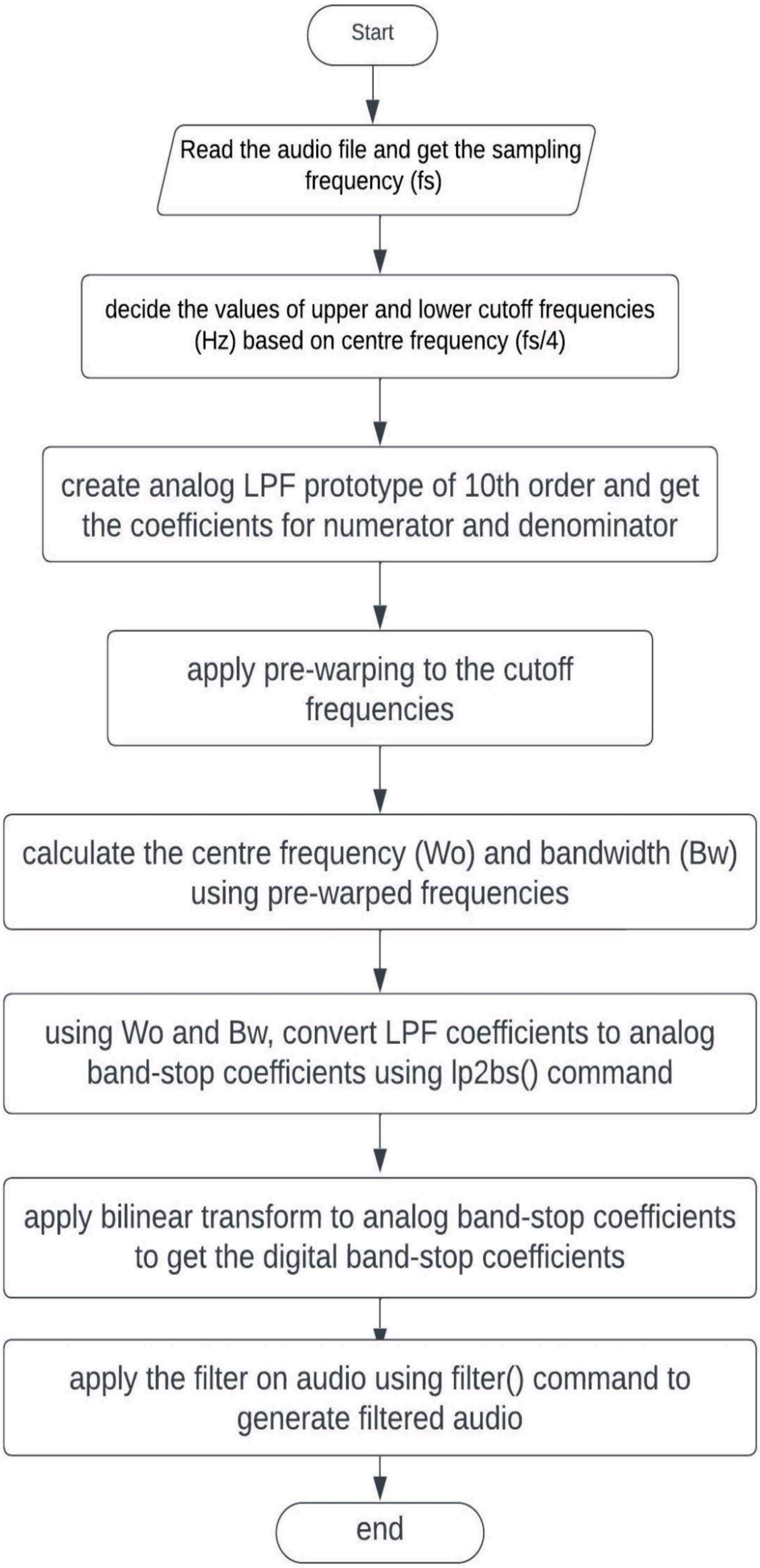
Pre-warping of analog frequencies (Wp) according to desired digital frequencies.

$$s = \frac{2(1 - z^{-1})}{T(1 + z^{-1})}$$

Mapping of s-domain (analog) poles and zeros into z-domain (digital)

Pre-warping in Bilinear Transform: Frequency warping follows a known pattern, and there is a known relationship between the warped frequency and the known frequency. We can use a technique called pre-warping to account for the nonlinearity and produce a more faithful mapping. The bilinear transform does not maintain the phase characteristics of the analog filter, and there is no way to correct the phase response to match.

4. Flow chart:



5. Code:

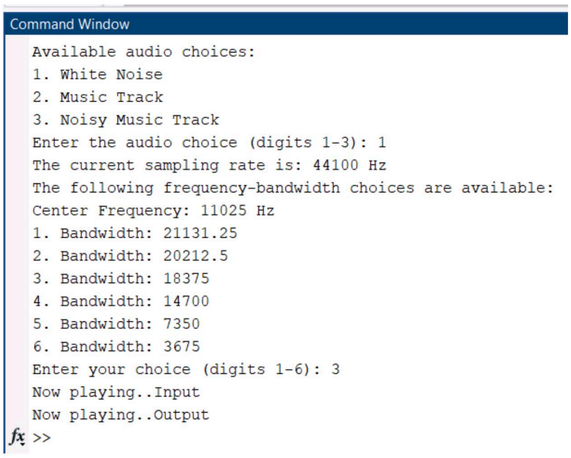
The following is the MATLAB program that implements the Band-Stop Filter design using bilinear transform using the analog low pass filter as a prototype. Here we designed the 20th order Band Stop Filter (thus the low pass filter has an order of 10).

```
clc; clear; close all;

% Display audio choices
disp('Available audio choices:');
disp('1. White Noise');
disp('2. Music Track');
disp('3. Noisy Music Track');

audio_choice = input("Enter the audio choice (digits 1-3): ");
switch audio_choice
    case 1
        file = "White_Noise.wav";
    case 2
        file = "Harpsichord_Melody.mp3";
    case 3
        file = "JazzTrioNoisy.wav";
    otherwise
        error('Error. Select from the available choices.');
end

[x, fs] = audioread(file);
disp("The current sampling rate is: " + fs + " Hz");
fl_vector = fs.*[1/96, 1/48, 1/24, 1/12, 1/6, 1/4.8];

disp("The following frequency-bandwidth choices are available: ");
disp("Center Frequency: " + fs/4 + " Hz");
for i = 1:length(fl_vector)
    disp(i + ". Bandwidth: " + ((fs/2 - 2*fl_vector(i))));
end

choice = input("Enter your choice (digits 1-6): ");
if choice > length(fl_vector) || choice < 1
    error('Error. Select from the available choices.');
end

% define the stopband frequencies
fl = fl_vector(choice);
fh = fs/2 - fl;
```

```
% design the analog lpf
order = 10;
[z,p,k] = buttap(order);
[b,a] = zp2tf(z,p,k);

% pre-warping of frequencies:
u1 = 2*fs*tan((2*pi*fl)*(1/fs)/2);
u2 = 2*fs*tan((2*pi*fh)*(1/fs)/2);

Wo = sqrt(u1 * u2);                   % center frequency
Bw = abs(u2-u1);                      % bandwidth

[bs,as] = lp2bs(b,a,Wo,Bw);
[bd, ad] = bilinear(bs, as, fs);
y = filter(bd, ad, x);

[Hz, f] = freqz(bd, ad, 4096, fs);

N = length(x);
X = fft(x);
Y = fft(y);

freq = (0:(N-1)/2) * (fs / N);
figure(1)
subplot(4, 1, 1)
plot(freq, abs(X(1:length(freq)))),
title('spectrum of i/p audio signal'),
xlabel('frequency (Hz)'), ylabel('magnitude')
subplot(4, 1, 3)
plot(freq, abs(Y(1:length(freq)))),
title(['frequency response of filter: fc1 = ',num2str(fl),' Hz and fc2 = '
,num2str(fh), ' Hz']),
xlabel('frequency (Hz)'), ylabel('magnitude')
subplot(4, 1, 2)
plot(f, abs(Hz)), title('spectrum of filtered audio signal'),
xlabel('frequency (Hz)'), ylabel('magnitude')
subplot(4, 1, 4)
plot(f, angle(Hz)), title('phase response of filter'),
xlabel('frequency (Hz)'), ylabel('magnitude')

time = (0:N-1)/fs;
figure(2)
subplot(2, 1, 1)
plot(time, x),
title('Time domain input signal'),
xlabel('time in seconds'), ylabel('Magnitude')
subplot(2, 1, 2)
plot(time, y),
title('Time domain output signal'),
xlabel('time in seconds'), ylabel('Magnitude')
```

```
time = (0:N-1)/fs;
figure(2)
subplot(2, 1, 1)
plot(time, x),
title('Time domain input signal'),
xlabel('time in seconds'), ylabel('Magnitude')
subplot(2, 1, 2)
plot(time, y),
title('Time domain output signal'),
xlabel('time in seconds'), ylabel('Magnitude')
```

6. Results:



```
Command Window
Available audio choices:
1. White Noise
2. Music Track
3. Noisy Music Track
Enter the audio choice (digits 1-3): 1
The current sampling rate is: 44100 Hz
The following frequency-bandwidth choices are available:
Center Frequency: 11025 Hz
1. Bandwidth: 21131.25
2. Bandwidth: 20212.5
3. Bandwidth: 18375
4. Bandwidth: 14700
5. Bandwidth: 7350
6. Bandwidth: 3675
Enter your choice (digits 1-6): 3
Now playing..Input
Now playing..Output
fx >>
```

Figure 6 Command window program takes user input and executes the filtering operation based on the chosen bandwidth and audio file

1. Response of the filter on White Noise:

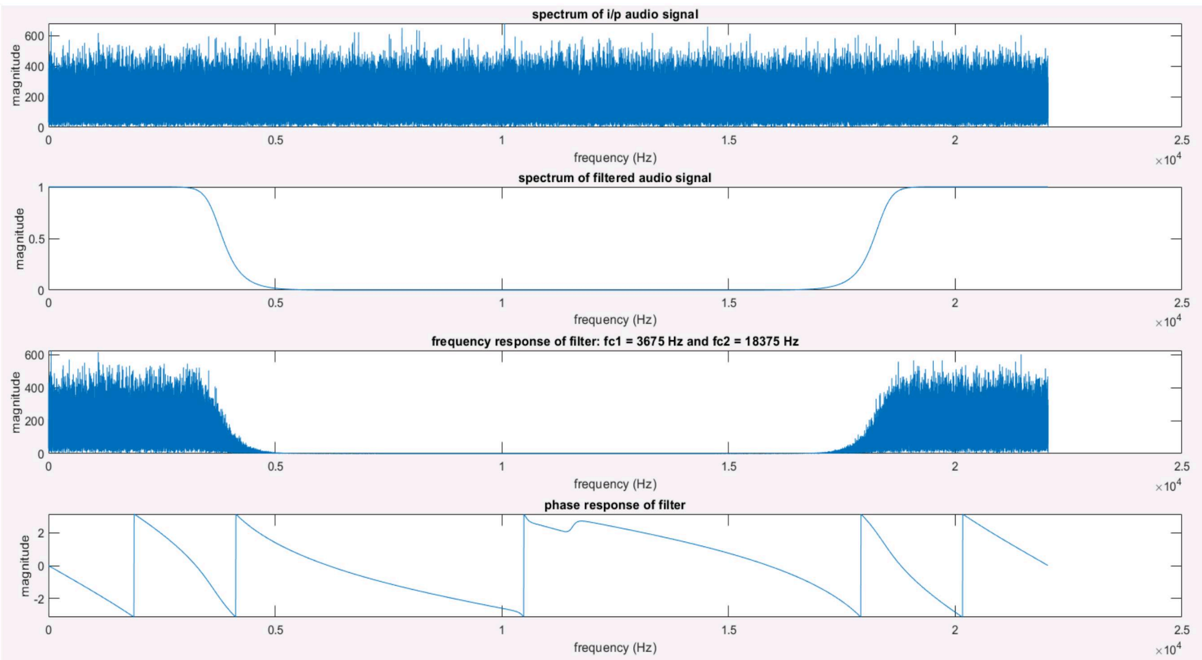


Figure 7 Magnitude and Phase Spectrum of the Band-Stop Filter, Input and Output Signal's Spectrums

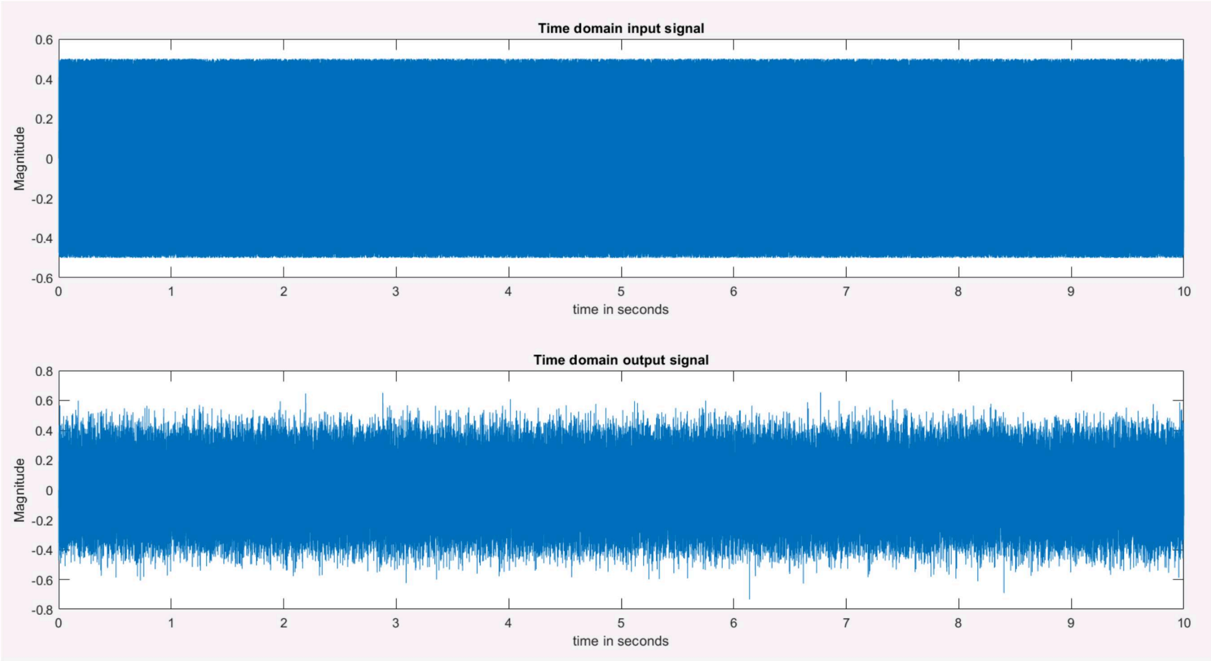


Figure 8 Time Domain Analysis of the Input and Output Audio Signals

From the above figures, we can see how the filter's characteristics shape the spectrum of the input signal. We can see how the pass band frequencies remain unchanged while the transition band and stop band frequencies are being suppressed.

2. **Elimination of Higher Frequency components from an audio signal:**

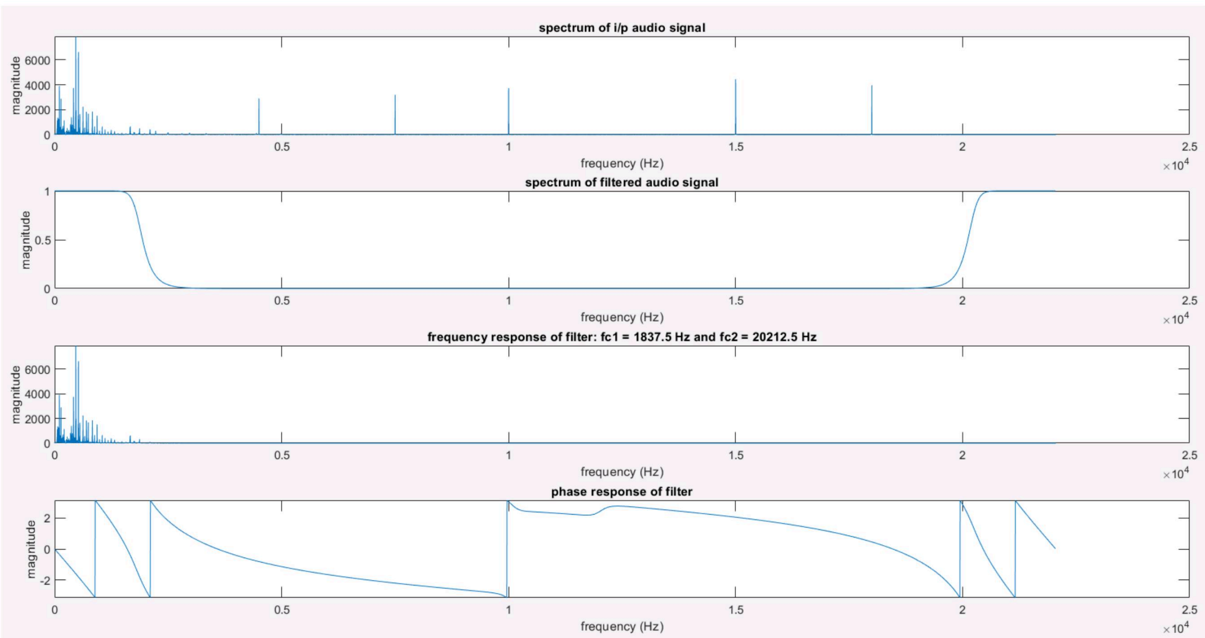


Figure 9 Spectrums of Input and Output audio signals

From the above figure, we can observe the suppression of higher-frequency components in the audio signal. frequencies being suppressed are 4500 Hz, 7500 Hz, 10000 Hz, 15000 Hz, and 18000 Hz.

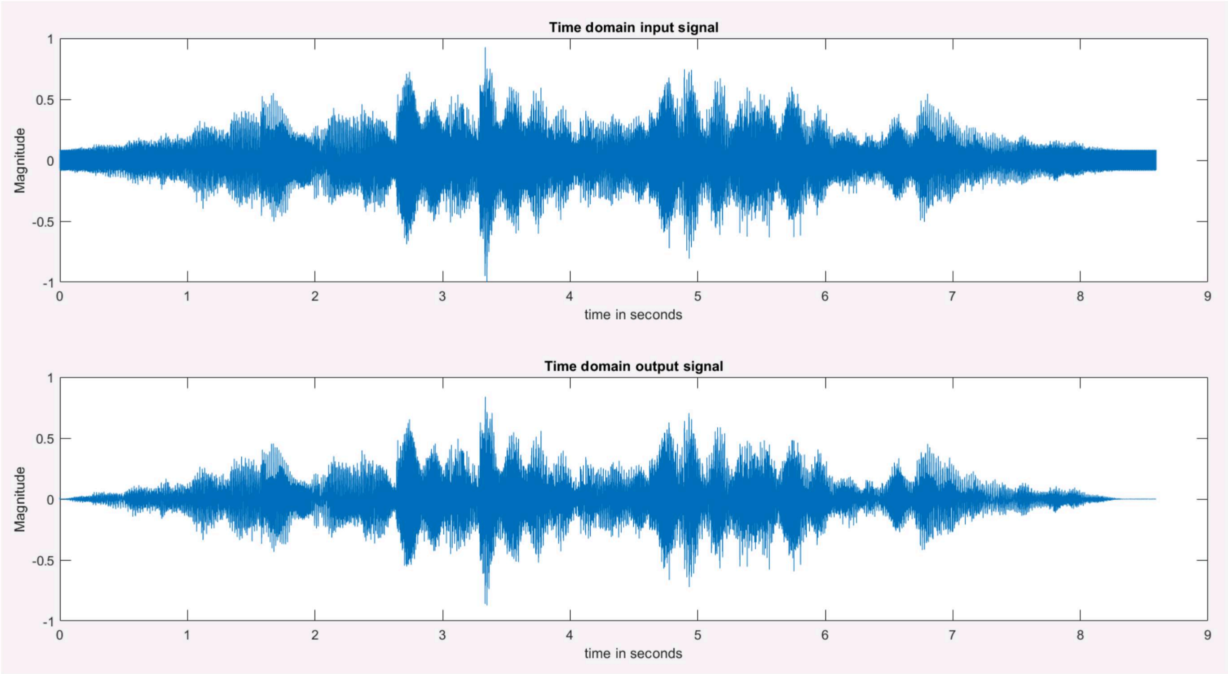


Figure 10 Time Domain Analysis of the Audio Signal

From the above figure, we can observe the changes in the output waveform. The time domain displays how a signal changes over time. By comparing the above two plots, we can see the reduction of louder components or thus amplitudes which were present due to higher frequencies while the lower frequencies are observed to be preserved.

In conclusion, we have designed a 20th Butterworth IIR band stop filter that stops the required band of frequencies according to the chosen bandwidth. We have designed a band stop filter based on its application in the music industry. We applied the filter to an audio file that has high-frequency noise components and thus removed those noise components. Hence, we have successfully designed a band-stop filter application that can be used in audio processing.

7. Applications:

1. In communication electronics the signal is distorted due to noise which makes the original signal interfere with other signals and lead to errors in the output. Band-stop filters are used to eliminate these unwanted noises.

2. Band stop filter is used in electric guitar amplifiers, the guitar produces hum at 60 Hz. The filter reduces that hum to amplify the signal produced by the guitar.

3. They are used in biomedical instruments like ECG to remove unwanted line noise. This noise can sneak into medical signals from electrical sources. The band-stop filter precisely targets and eliminates this noise, ensuring accurate and reliable reading.
4. Band stop filters are used to remove unwanted frequencies like electrical noise, and interference that affects the quality of conversation by selectively blocking out specific signals to improve overall speech quality.

8. Limitations and Future scope:

Limitations:

The band-stop filter technique employed in this project has certain limitations. The reliance on analog low-pass to band-stop transformation introduces sensitivity to filter order, potentially affecting performance. The fixed frequency choices may limit adaptability to varying audio characteristics as the center frequency is fixed at fs/4. Changing the center frequency may cause distortion in the characteristic as the order requirement also needs to be adjusted.

Future Scope:

To address these limitations, future work could explore the dynamic adaptation of various filter parameters like cutoff frequency and filter order, allowing users to define custom frequency ranges. Thus, allowing a better design for a narrow band-stop filter with variable center frequency.

9. References:

- https://in.mathworks.com/help/signal/ref/lp2bs.html
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- https://en.m.wikipedia.org/wiki/Band-stop_filter
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