Teachers Assessment Activity

DIGITAL SIGNAL PROCESSING

ECT 354

5th Semester B.Tech. Session-2023-24

<u>Project Report</u>

Title:

“ Applying High pass Filter to audio file with variable cutoff frequency (20Hz-1Khz) input by user ”

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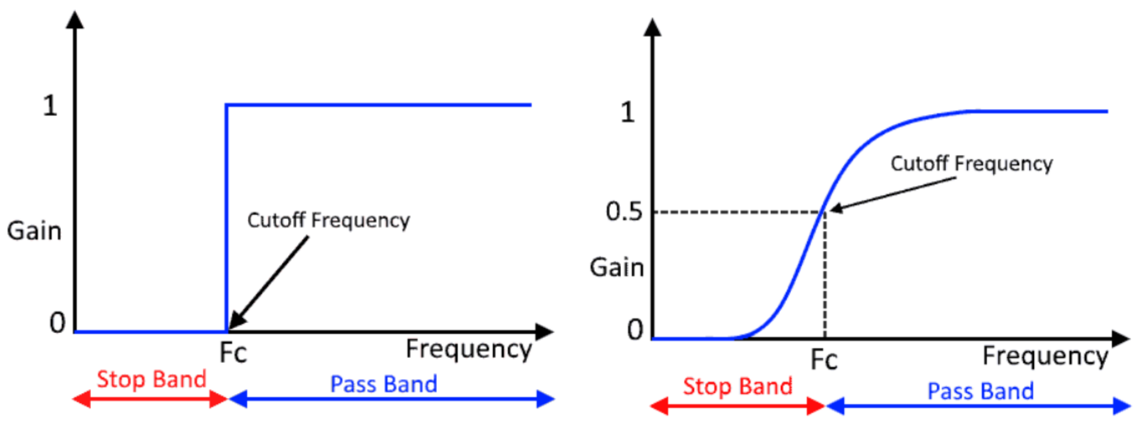
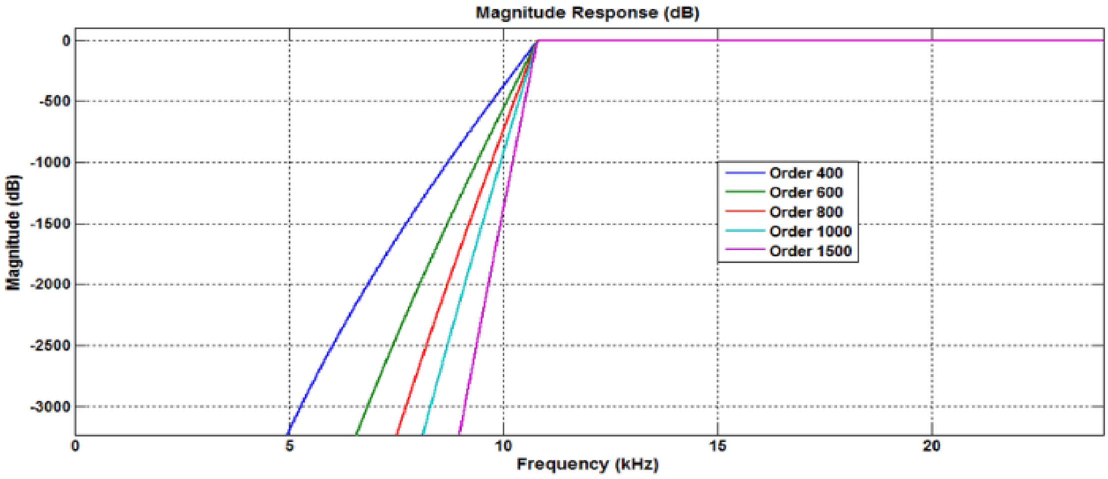
1. Introduction:

Digital Signal Processing (DSP) is a crucial field in electronic and communication engineering, focusing on signal manipulation for various applications. Filters play a crucial role in the domain of signal processing, functioning as indispensable tools for the purpose of shaping and enhancing signals in accordance with specific requirements. Their widespread applications encompass diverse domains, such as telecommunications, audio processing, image processing, and many more. In audio processing, distinguishing between signals and noise is crucial for optimal sound quality. Effective filtering preserves the original signal's integrity while minimizing the impact of unwanted disturbances. Filters are essential for tasks such as noise reduction, equalization, and filtering to tailor audio signals for specific requirements.

Understanding Filters

Filters in DSP act as gatekeepers, selectively allowing or blocking certain frequencies within a signal. They come in various types, each with its unique characteristics suited to specific applications. Broadly, filters can be classified into high-pass, low-pass, band-pass, and band-stop filters. Each type has a distinct role in manipulating the frequency content of a signal. In this project, our emphasis is on implementing a high-pass filter, allowing high-frequency components to pass through while attenuating the lower frequencies. The Ideal and Practical characteristics of High pass filter are as follows:





FIR (Finite Impulse Response) and IIR (Infinite Impulse Response) are two types of digital filters used in signal processing. FIR filters have a finite impulse response, meaning they only respond to a finite duration input. On the other hand, IIR filters have an infinite impulse response, allowing them to have feedback and memory.

This project explores the implementation and analysis of a Butterworth high-pass filter, a widely used filter in audio processing. The main goal is to enhance our understanding of signal filtering techniques and their applications in audio signal processing.

2.Objectives:

a) To Design High Pass Filter with variable cutoff frequency input by user: To develop a user interface that allows the user to input and adjust the cutoff frequency to test the designed high pass filter under various conditions to ensure that it functions as intended.

b) To Remove Beep sound(noise) from the original audio:
Analyzing the noisy input audio signal using Butterworth HPF to get the filtered audio by applying desired cutoff frequency and order of the filter.

c)To Compare the frequency response of input and output audio signals :
Analyzing the graphs derived from the frequency response analysis of input and output audio signals offers valuable insights such as how spectrum is spaced at different cutoff frequencies and order of the filter.

3.Theoretical Background:

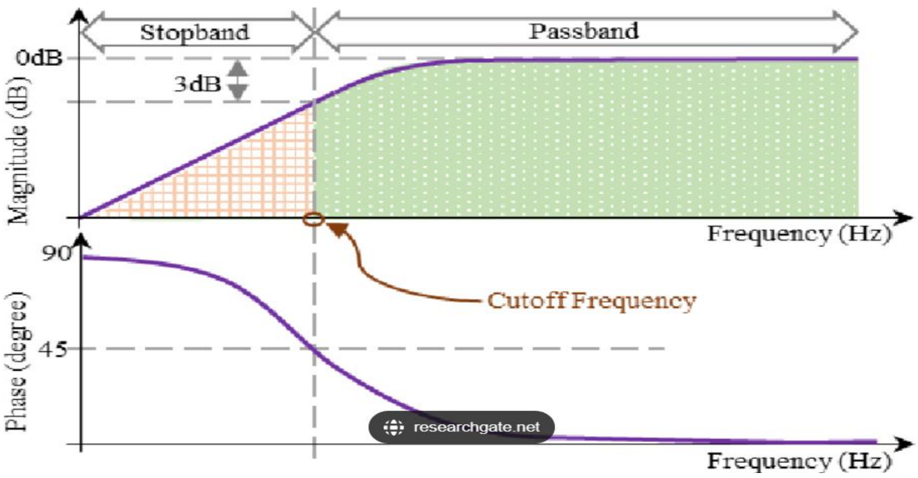
High-Pass Filters

High-pass filters are essential components in signal processing, designed to selectively allow higher-frequency components of a signal to pass through while attenuating lower-frequency elements. In the context of audio engineering, high-pass filters prove valuable for tasks such as eliminating unwanted low-frequency noise, emphasizing specific frequency ranges, and contributing to overall sound equalization. Understanding their operation involves grasping how these filters shape the frequency content of a signal, letting the "high" frequencies proceed unaltered while curbing the "low" ones .

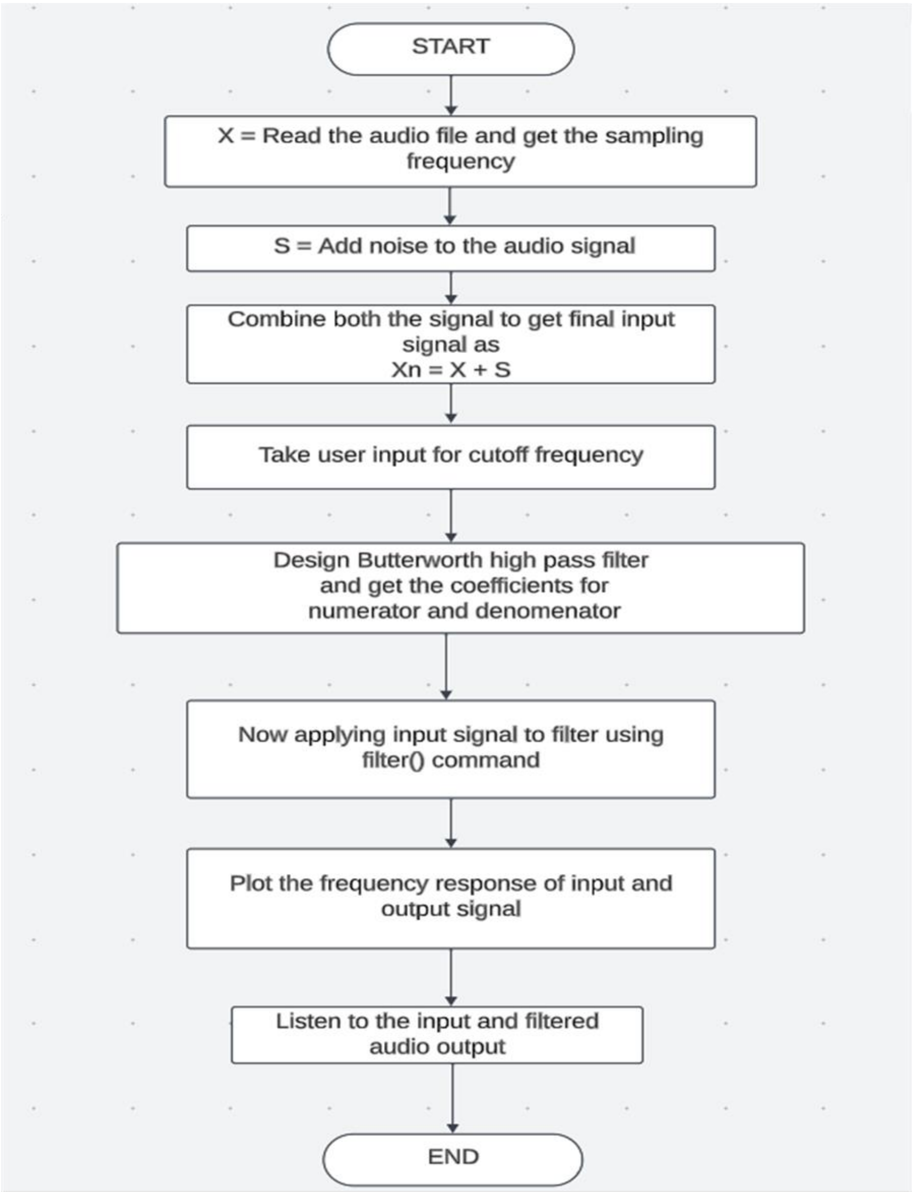
The cutoff frequency in high-pass filters defines the boundary between allowed higher frequencies and attenuated lower frequencies. Its pivotal role lies in shaping the characteristics of the filtered signal by determining which frequencies are permitted or suppressed.
Filter orders play a crucial role in determining the steepness of the frequency response roll-off. Higher-order filters exhibit a more rapid roll-off, effectively suppressing frequencies beyond the cutoff more quickly. However, this increased selectivity comes at the cost of potential phase distortion. Lower-order filters, conversely, maintain a gentler roll-off but preserve phase integrity.

Butterworth High-Pass Filter

Butterworth filter is a type of IIR filter. The reason for choosing an IIR Butterworth filter often lies in its characteristics. Butterworth filters provide a flat frequency response in the passband and have a gradual roll-off in the stopband.
The Butterworth high-pass filter is a specific design within the high-pass filter category. What makes it distinctive is its maximally flat frequency response in the passband, ensuring a uniform gain without ripples or distortion. The roll-off in the stopband is gradual, minimizing phase shifts. The Butterworth filter is favored for its simplicity and predictable response. Its characteristics make it a versatile choice in applications where a flat response is crucial.


Stopband
Passband
0dB
3dB
Magnitude (dB)
Frequency (Hz)
90
45
Phase (degree)
Cutoff Frequency
researchgate.net
Frequency (Hz)

4.Flow Chart


START
X = Read the audio file and get the sampling frequency
S = Add noise to the audio signal
Combine both the signal to get final input signal as
Xn = X + S
Take user input for cutoff frequency
Design Butterworth high pass filter and get the coefficients for numerator and denomenator
Now applying input signal to filter using filter() command
Plot the frequency response of input and output signal
Listen to the input and filtered audio output
END

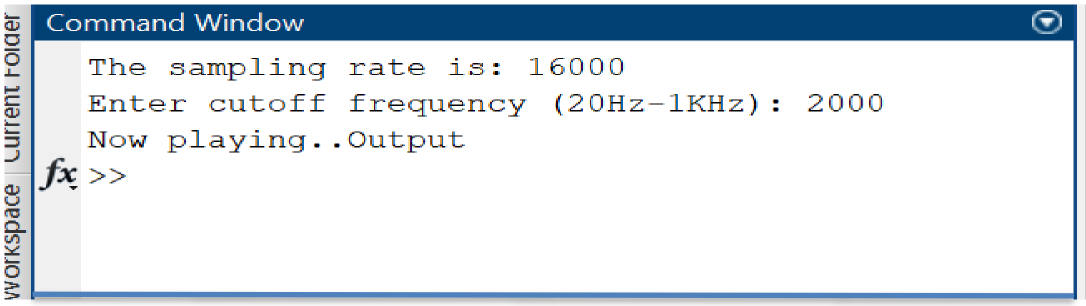
5.Code:

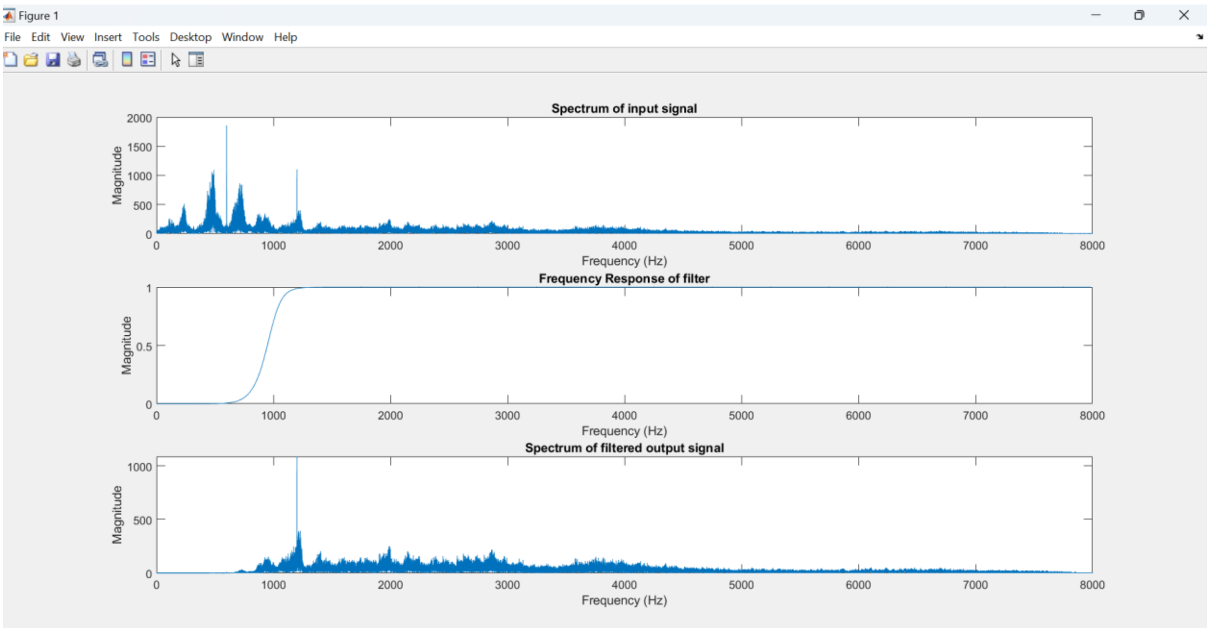
```
%Applying high pass filter to audio file with variable cutoff frequency (1Khz-
20KHz) input by user

clc;
clear;
close all;

% Load audio file
[x, fs] = audioread("C:\Users\hp\Downloads\nishad.mp4");

N = length(x);
t=(0:(N-1))/fs;
s=0.03*sin(2*pi*600*t)+0.02*sin(2*pi*1200*t);
disp("The sampling rate is: " + fs);
xn=x+s';


% User-defined cutoff frequency
fc = input('Enter cutoff frequency (20Hz-1KHz): ');

% Design high-pass filter
[b, a] = butter(10, fc / (fs / 2),"high");
[h,w] = freqz(b,a);

% App,dc.cly the filter to the input signal
y = filter(b, a, xn);

figure(1);%time domain response
%plot input signal in time domain
subplot(4,1,1);
plot(t, x), title('input signal in time domain'), xlabel('time(sec)'),
ylabel('Magnitude');
%plot input noise signal in time domain
subplot(4,1,2);
plot(t, s), title('input noise signal in time domain'), xlabel('time(sec)'),
ylabel('Magnitude');
%plot total input signal in time domain
subplot(4,1,3);
plot(t, xn), title('total input signal in time domain'), xlabel('time(sec)'),
ylabel('Magnitude');
% plot the filtered output signal in time domain
subplot(4,1,4);
plot(t, y), title('filtered output signal in time domain'), xlabel('time(sec)'),
ylabel('Magnitude');

% Plot input and output spectra
f = linspace(0, fs/2, N/2); % Positive frequencies up to Nyquist
X1=fft(x);
X2=fft(s);
X =fft(xn);
Y =fft(y);
%figure;

figure(2);%magnitude spectrum in frequency domain
%plot magnitude response of input signal
subplot(3,2,1);
plot(f, abs(X1(1:N/2))), title('Spectrum of input signal'), xlabel('Frequency
(Hz)'), ylabel('Magnitude');
```

```
subplot(3,2,3);
plot(f, abs(X2(1:N/2))), title('Spectrum of input noise signal'),
xlabel('Frequency (Hz)'), ylabel('Magnitude');

subplot(3,2,2);
plot(f, abs(X(1:N/2))), title('Spectrum of total input signal'), xlabel('Frequency
(Hz)'), ylabel('Magnitude');

% Plot magnitude response of the filter
subplot(3,2,4);
plot(w*(fs/(2*pi)), abs(h)), title('Magnitude Response of filter'),
xlabel('Frequency (Hz)'), ylabel('Magnitude');

subplot(3,2,5);
plot(w*(fs/(2*pi)), angle(h)), title('phase Response of filter'),
xlabel('Frequency (Hz)'), ylabel('Phase');

% plot the filtered output signal
subplot(3,2,6);
plot(f, abs(Y(1:N/2))), title('Spectrum of filtered output signal'),
xlabel('Frequency (Hz)'), ylabel('Magnitude');


figure(3);%power spectrum in frequency domain

subplot(4,1,1);
plot(f, (abs(X1(1:N/2))).^2), title('Power Spectrum of input signal'),
xlabel('Frequency (Hz)'), ylabel('Power');

subplot(4,1,2);
plot(f, (abs(X2(1:N/2))).^2), title('Power Spectrum of input noise signal'),
xlabel('Frequency (Hz)'), ylabel('Power');

subplot(4,1,3);
plot(f, (abs(X(1:N/2))).^2), title('Power Spectrum of total input signal'),
xlabel('Frequency (Hz)'), ylabel('Power');

% plot the filter to the input signal
subplot(4,1,4);
plot(f, (abs(Y(1:N/2))).^2), title('Spectrum of filtered output signal'),
xlabel('Frequency (Hz)'), ylabel('Power');


 % Listen to the signals
 %soundsc(x, fs);
 %pause(20);
 %soundsc(y,fs);
 x_input = audioplayer(xn, fs);
 play(x_input);
 playblocking(x_input);

 pause(2);

 disp("Now playing..Output");
 y_output = audioplayer(y, fs);
 play(y_output);
 playblocking(y_output);
```

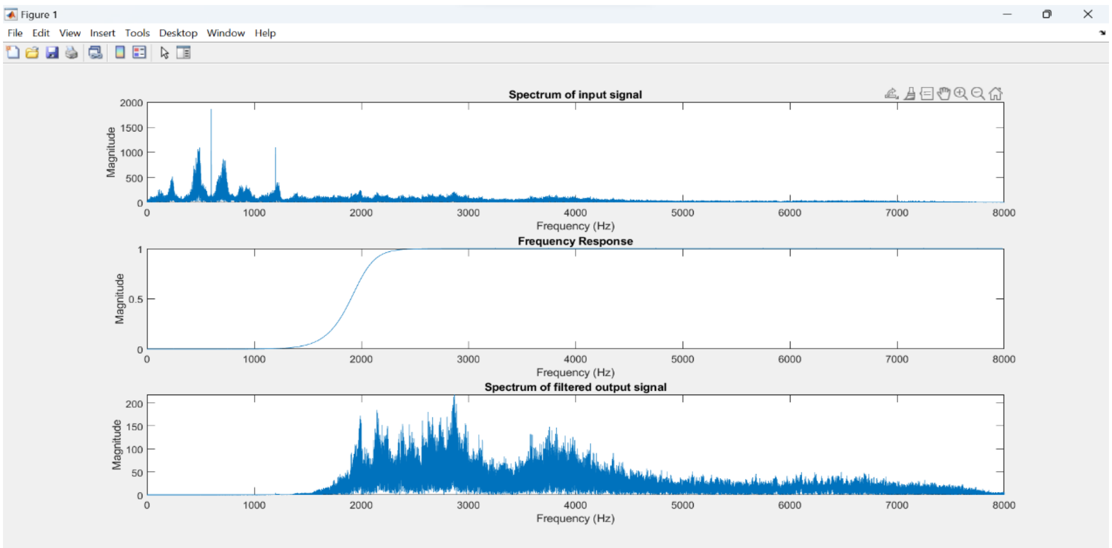
6.Results

The following is the command window for the program. It asks from the user for a cutoff frequency in the variable range of (20 Hz to 1k Hz) as an input. After choosing the cutoff frequency the output is generated from the filter. The frequency response is shown for the following cases:





Order=10 fc=1000

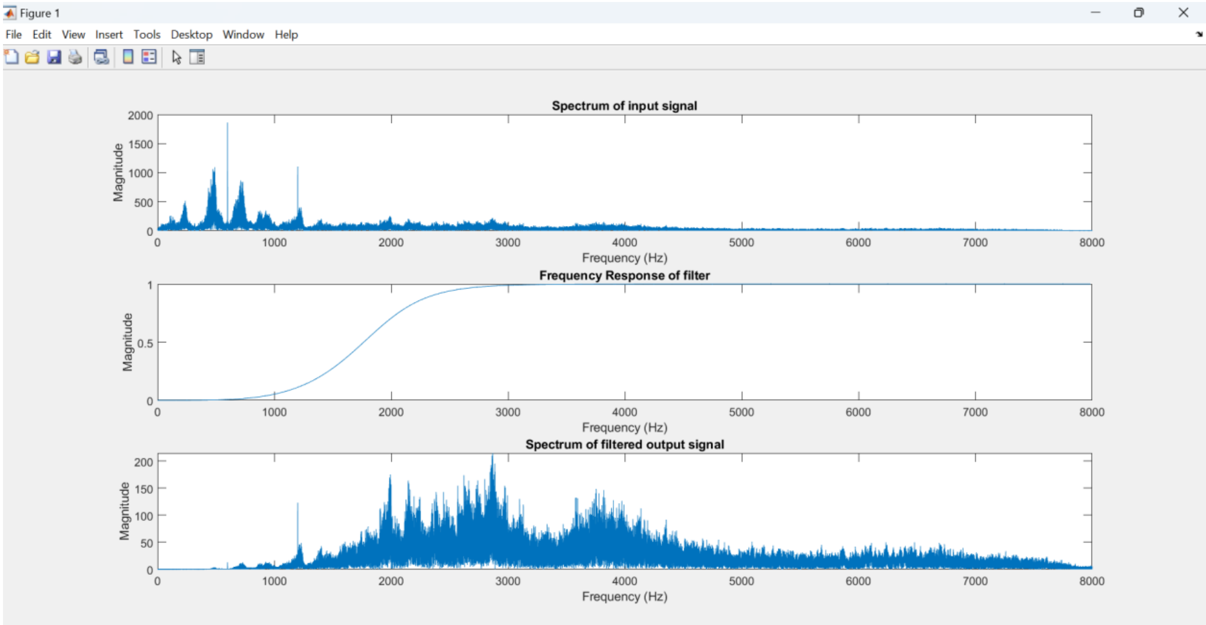


Order = 10 fc=2000

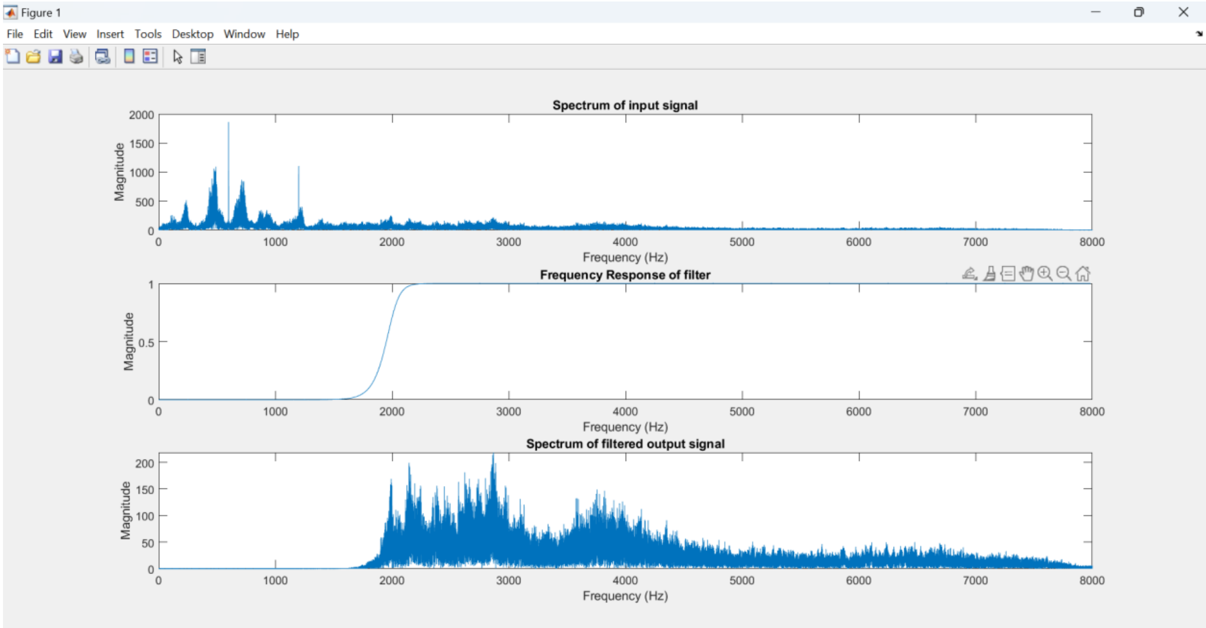
Observations : In this particular scenario, a filter with an order of 10 and a cutoff frequency 1000 Hz and 2000 Hz was employed separately to an input signal containing both audio and noise(specifically ,a beep sound).

The resulting output for:

CASE1](fc=1k) It has some noise as it is present in the pass band. **CASE2]**(fc=2k) In this case the beep sound was effectively eliminated. The improved clarity in the output indicates that the frequency component responsible for noise should be present in the stop band of frequency response, enhancing the overall quality of the filtered audio signal.



Order = 4 fc=2000



Order = 20 fc=2000

Observations: In this particular scenario case we are changing the order of the filter by keeping the same cutoff frequency(fc=2000)

CASE 1] Here order of the filter = 4th and fc=2000 .The 4th-order filter has a less steep roll-off .It allows more of the lower frequencies to pass through before the cutoff frequency, providing a relatively smoother transition between the passband and stopband.

As a result noise is also present in the filtered audio.

CASE 2] Here order of the filter = 20th and fc=2000.The 8th-order filter exhibits a steeper roll-off, resulting in more aggressive attenuation of frequencies below the cutoff. It will provide a sharper transition between the passband and stopband, effectively suppressing lower frequencies. As a result the beep sound was effectively eliminated.

The improved clarity in the output indicates that increasing the order of the filter played a significant role in removing undesired frequencies, enhancing the overall quality of the filtered audio signal.

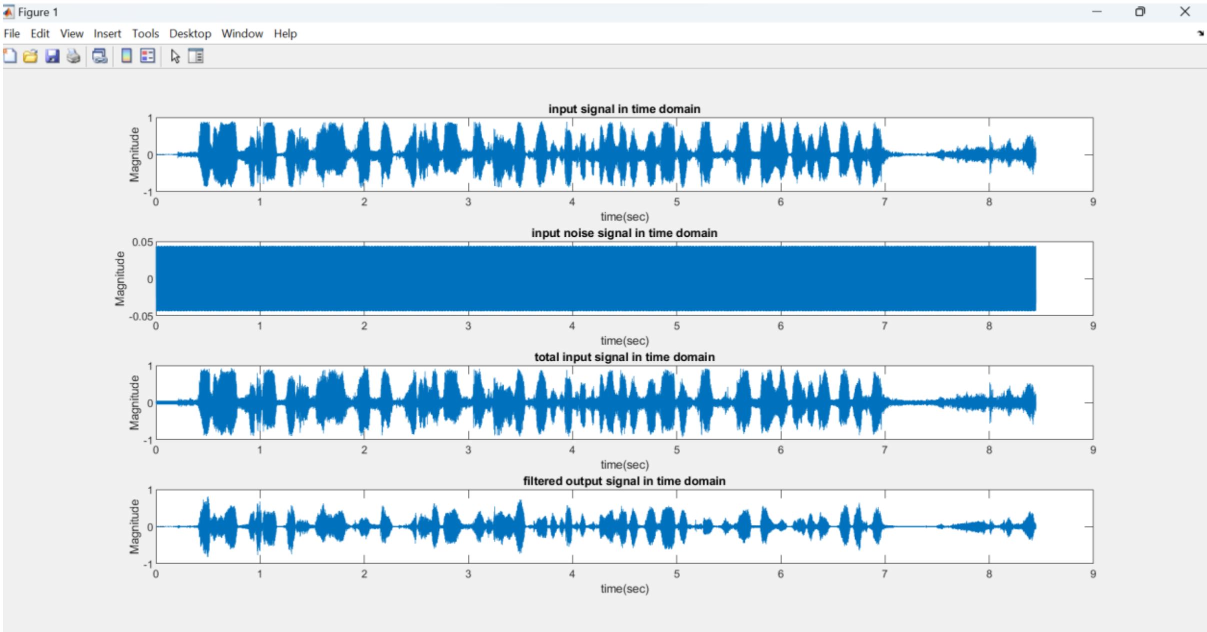


Figure shows the time domain plots of input audio, noisy audio, combination of audio and noise , filtered audio.

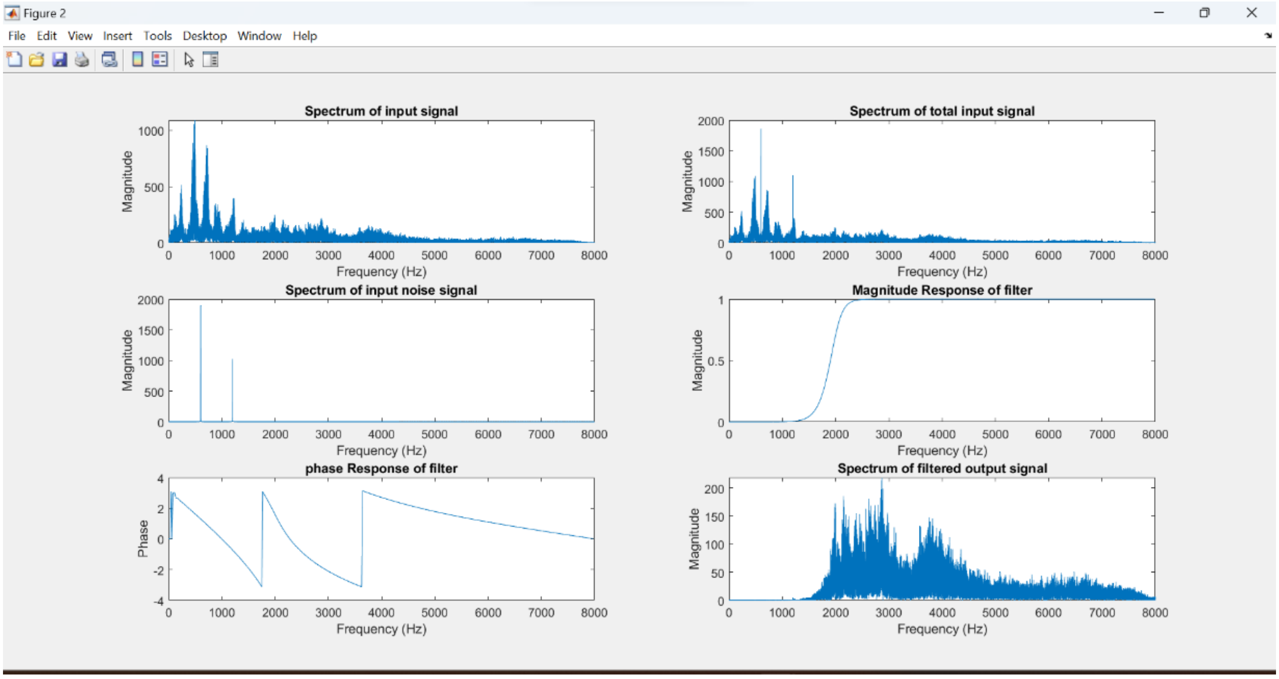


Figure shows magnitude spectrum of input audio , noisy audio , combination of audio and noise, filtered audio, magnitude and phase spectrum of filter.

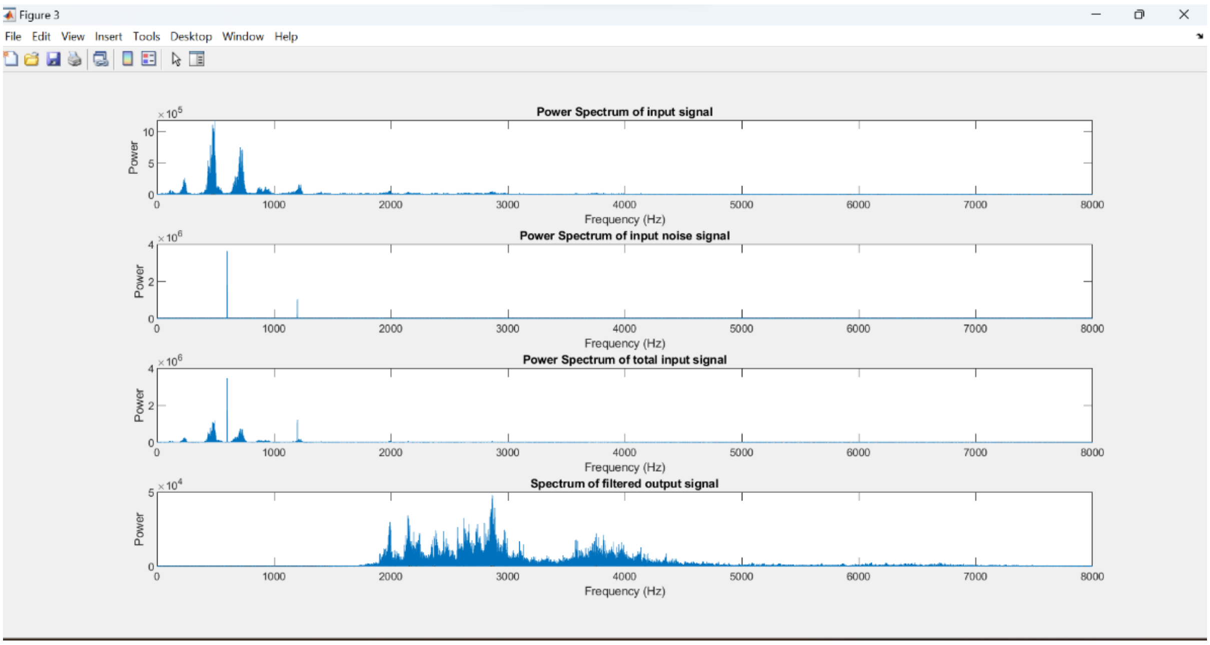
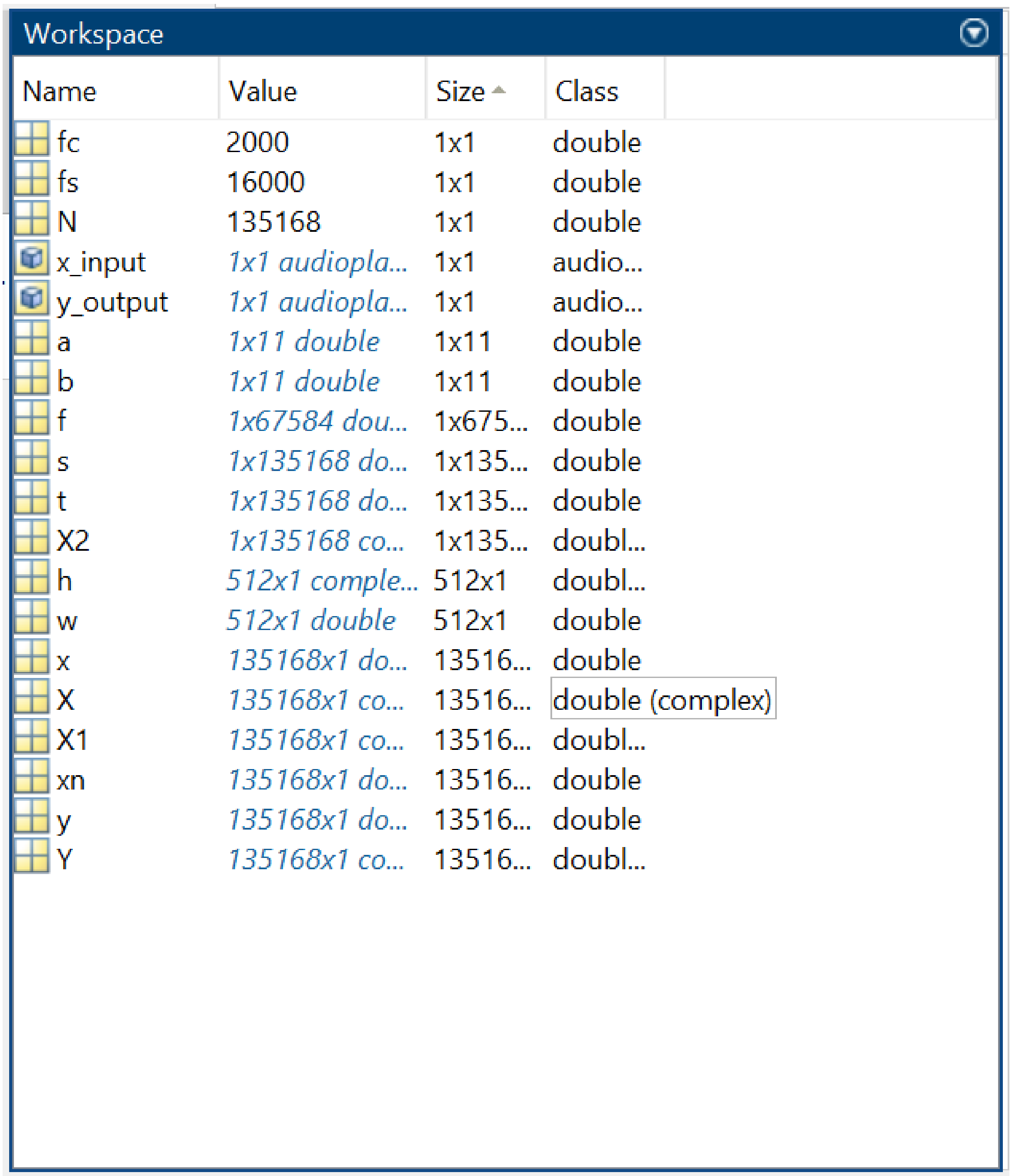


Figure shows power spectrum of input audio , noisy audio , combination of audio and noise, filtered audio



Workspace

Name	Value	Size	Class
fc	2000	1x1	double
fs	16000	1x1	double
N	135168	1x1	double
x_input	1x1 audiopla...	1x1	audio...
y_output	1x1 audiopla...	1x1	audio...
a	1x11 double	1x11	double
b	1x11 double	1x11	double
f	1x67584 dou...	1x675...	double
s	1x135168 do...	1x135...	double
t	1x135168 do...	1x135...	double
X2	1x135168 co...	1x135...	doubl...
h	512x1 comple...	512x1	doubl...
w	512x1 double	512x1	double
x	135168x1 do...	13516...	double
X	135168x1 co...	13516...	double (complex)
X1	135168x1 co...	13516...	doubl...
xn	135168x1 do...	13516...	double
y	135168x1 do...	13516...	double
Y	135168x1 co...	13516...	doubl...

Figure shows the workspace after executing the program

7.Applications:

1. Audio Equalizers in Music Production:

Real-Life Application: Audio engineers use high-pass filters in equalizer units during music production.

Explanation: In mixing and mastering music, high-pass filters are employed to remove low-frequency rumble or unwanted bass from tracks. For instance, in a recording with vocal tracks, engineers may use high-pass filters to eliminate low-frequency background noise without affecting the vocal clarity.

2. Image Enhancement in Photography:

Real-Life Application: Photographers use high-pass filters for image sharpening and enhancement.

Explanation: High-pass filters are used in photo editing software to sharpen images by emphasizing edges and fine details. For instance, when sharpening a portrait photograph, photographers selectively apply high-pass filters to enhance facial features without affecting the background.

3. Sensor Signal Processing in Biomedical Devices:

Real-Life Application: High-pass filters are utilized in biomedical devices for signal conditioning.

Explanation: Biomedical devices like electrocardiogram (ECG) machines use high-pass filters to eliminate baseline wander or slow-varying components in heart rate signals. By removing low-frequency interference, accurate detection of rapid changes in heart rhythm is ensured.

4.Speech Quality Enhancement:

Explanation*:* High-pass filters can be employed in communication systems to improve speech quality. By selectively allowing higher-frequency components associated with speech to pass through, these filters attenuate undesirable low-frequency noise or interference, thereby enhancing the overall clarity and quality of conversation.

8.Limitations and Future scope:

Limitations

1]As the order of the filter is pre decided and it is of higher order. Higher-order filters generally require more complex computations and involve more components in their implementation. This increases the computational complexity.

2]As the cutoff frequency is user defined, filtered output is highly dependent on user cutoff frequency to eliminate noise in audio signal so the stop band should include the unwanted low frequency.

3]As this is designed as high pass filter to remove unwanted low frequency noise but if the noise is present in entire range of frequencies then this filter would be insufficient to do it.

Future Scope:

As the order is fixed in the avoid filter ,this can be also taken as an input from the user. Allowing the user to specify the filter order in a digital filter, such as a Butterworth filter which will provide flexibility and customization based on the specific requirements of the signal processing task

9.References:

www.eeeguide.com
https://en.wikipedia.org/
mathworks.com
analog.com