FILTER SYSTEMS

(Part of EE202 Course Project)

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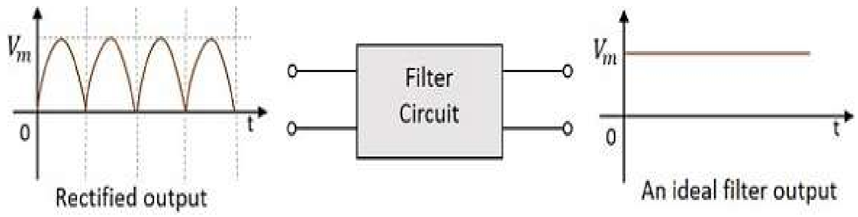
INTRODUCTION

A filter is an electric circuit that can remove a certain band of frequencies (noise) from the signal. The ripples in a signal due to the presence of an AC component generally constitutes a noise. This AC component has to be removed completely from a signal in order to get a pure DC output. This is done with the help of filters.



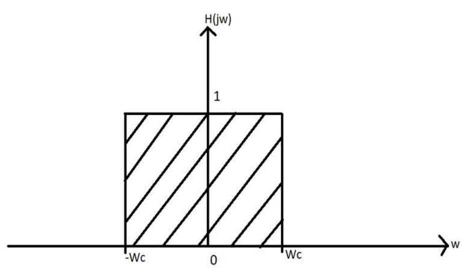
TYPES OF FILTERS

Filters can be categorized under two categories:

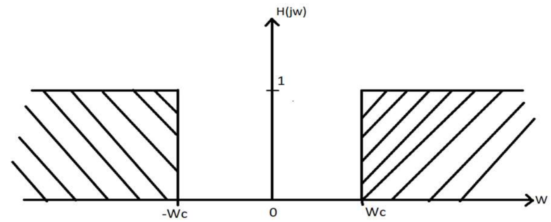
1. **On the basis of nature:**

(i) Analog Filters: Analog filters are circuits made of analog components such as resistors, capacitors, inductors, and op amps. They can be further classified on the basis of their frequency response.

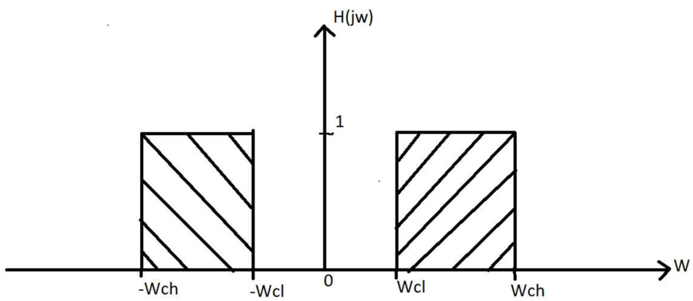
(a.) Low Pass Filters: Low pass filters allow low-frequency signals (frequencies less than cut-off frequency) to pass without any attenuation but it rejects any high frequency signals to pass through.



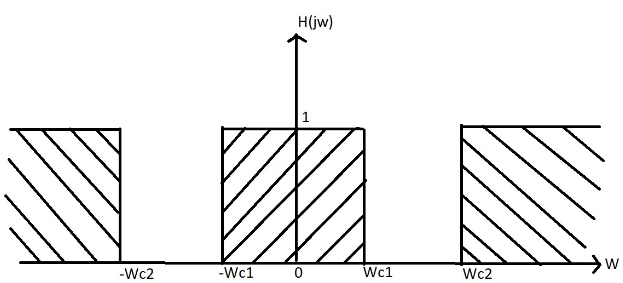
(b.) High Pass Filters: HPF signals allow high-frequency signals (frequencies greater than cut-off frequency) to pass without any attenuation in its amplitude but blocks any low-frequency signal.



(c.) Band Pass Filters: This type of filter allows a specific band of frequencies to pass and blocks any other frequencies lower or higher than its passband frequencies. It has two cut-off frequencies, i.e., upper and lower cut-off frequencies. A band pass filter can be realized by a cascaded arrangement of a high pass and a low pass filter with the condition that *cutoff frequency of LPF>cutoff frequency of HPF*.



(d.) Band Stop Filter/ Band Reject Filter: A band reject filter attenuates the signal whose frequency lies in a fixed band of frequencies. Hence, it works completely opposite to the band pass filter. It also has two cut-off frequencies, i.e., lower and upper cut-off frequencies. It can also be realized with a cascaded connection of a high pass and a low pass filter with the condition that *cutoff frequency of LPF<cutoff frequency of HPF*



(ii) Digital Filters: A digital filter is a circuit made of digital hardware such as Flip Flops, Shift Registers, ALU etc, used for filtering digital sequences . It employs the use of A2D and D2A converters to work with analog signals.

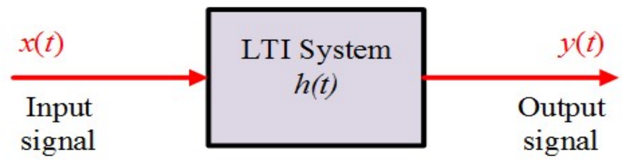
2. **On the basis of implementation:**

   (i) **Passive Filters:** Uses passive circuit elements like resistors, inductors, and capacitors, etc. in its construction. Amplification Gain of these filters is less than or equal to 1.

   (ii) **Active Filters:** Uses Operational Amplifiers in its construction. Amplification gain of these filters is greater than 1.

USING FOURIER TRANSFORM TO OBTAIN SYSTEM RESPONSE

1. Convolution of two signal x(t) and g(t) is defined as $\int_{-\infty}^{\infty} x(\tau).\, g(t-\tau).\, d\tau$
2. In LTI System the output is the convolution of input signal and impulse response of the system.
3. Let x(t) is the input signal and h(t) be the impulse response of the LTI system then y(t) =



4. **FOURIER TRANSFORM**

   Fourier transform of a function f(t) is defined if f(t) is
   (i) Defined on $(-\infty, \infty)$
   (ii) Piece-wise continuous in each finite interval
   (iii) Absolutely integrable in $(-\infty, \infty)$

$$F(jw) = \int_{-\infty}^{\infty} f(t).\, e^{-jwt} dt$$

   The function *F(jw)* is called the Fourier Transform of the function f(t).

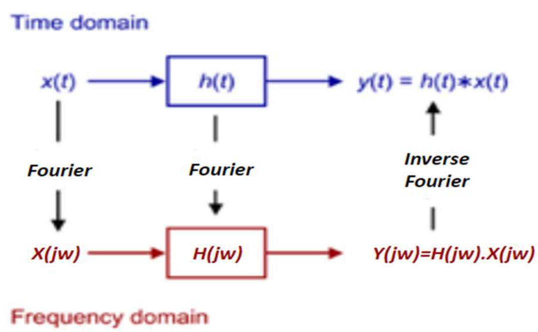
5. According to Convolution theorem in Fourier Transform, the Fourier Transform of convolution of two signals $x(t)$ *and* $g(t)$ is simply the multiplication of Fourier Transform of both the signals.

   Mathematically,

$$x(t) \leftrightarrow X(jw)$$
$$g(t) \leftrightarrow G(jw)$$

   Where *X(jw)* and *G(jw)* are the Fourier Transforms of x(t) and g(t) respectively, then

$$\int_{-\infty}^{\infty} x(\tau).\, g(t-\tau) d\tau \quad \leftrightarrow X(jw).\, G(jw)$$



6.) As the Fourier transform of convolution of two function results in multiplication, the analysis becomes mathematically convenient.

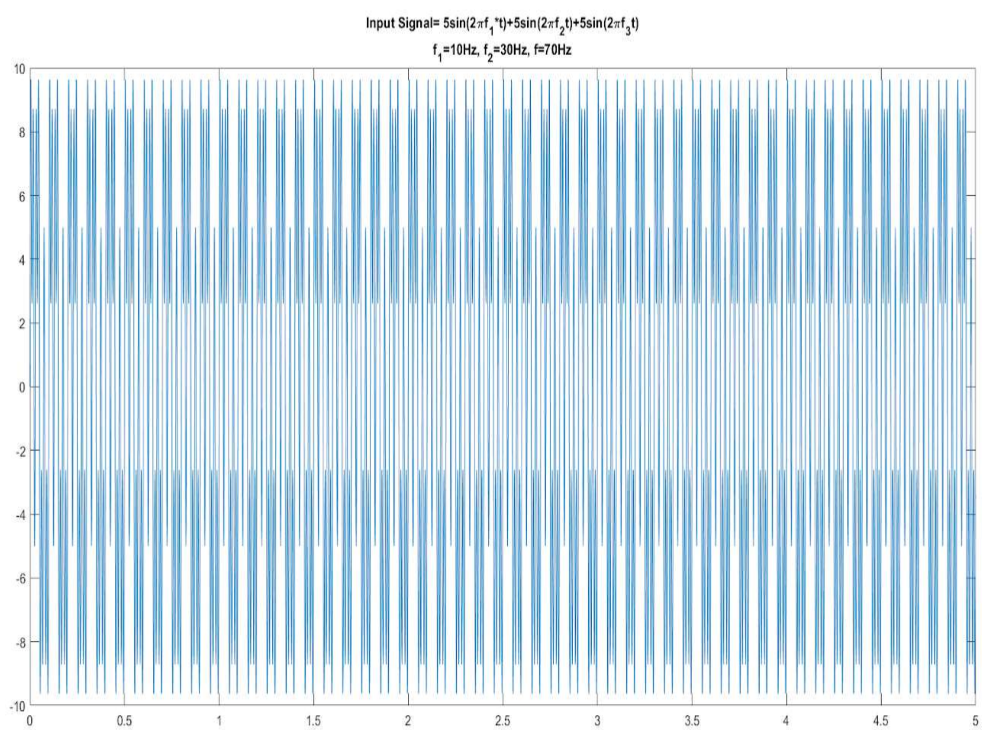
7.) Then after analysing the output, which is in frequency domain, we take inverse Fourier Transform to get back in time domain.

EXAMPLE

We implement the four kinds of Ideal Filters using the discussed theory on MATLAB® (Code is attached at the end). The following graph plots have been obtained for the same.

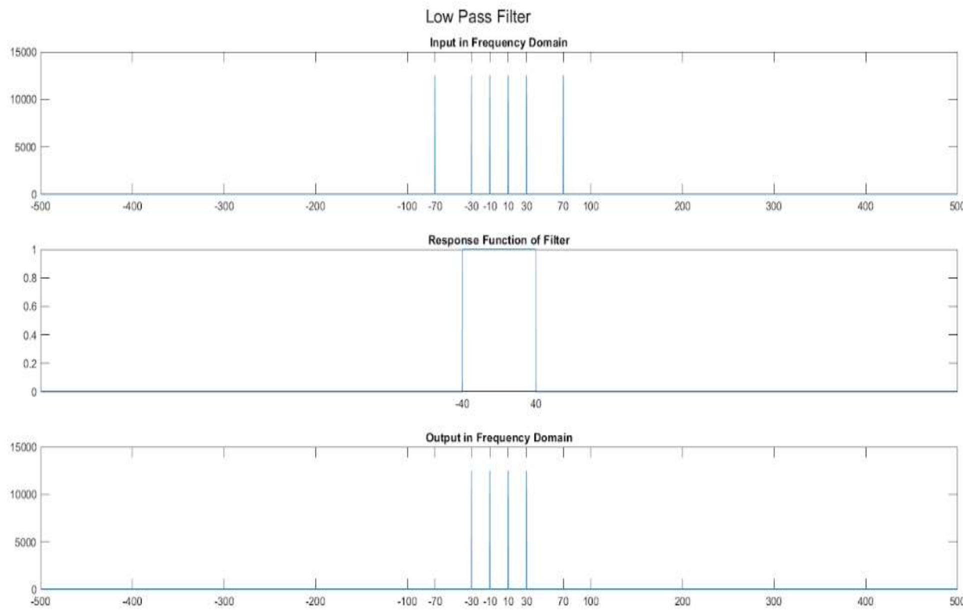
Computation is done using a Sampling Frequency of 1000Hz and a 5 sec time interval.

- Input Signal



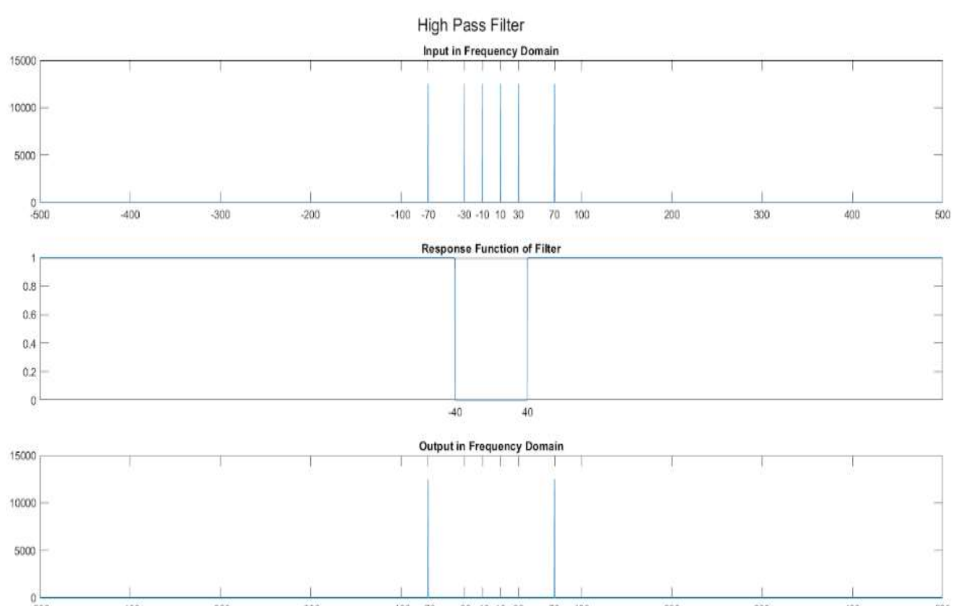
- <u>Low Pass Filter (Frequency Domain)</u>

  Pass Band Region= 0Hz to 40Hz



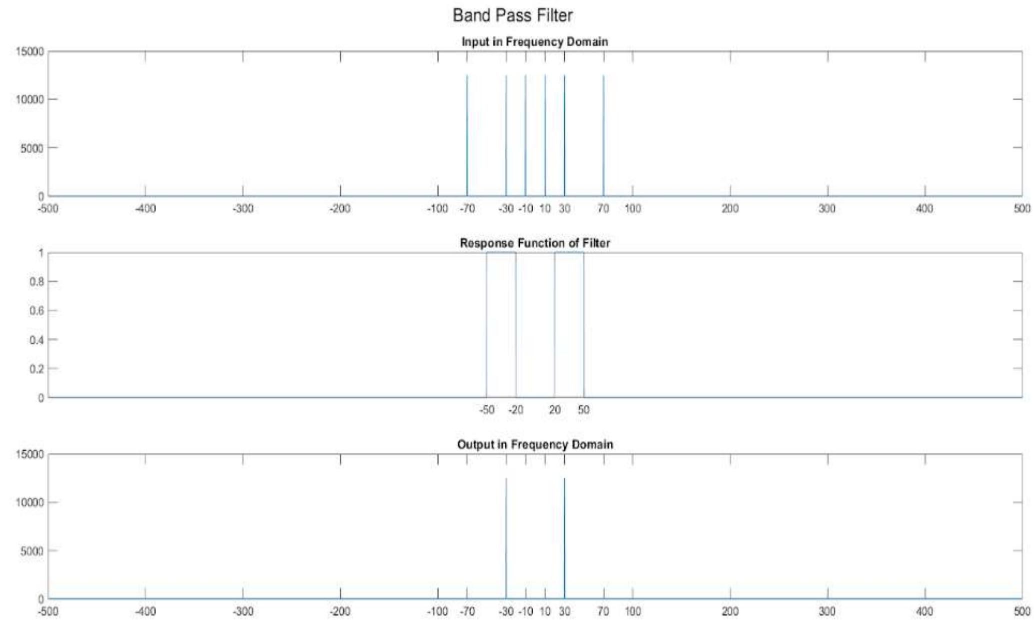
- <u>High Pass Filter (Frequency Domain)</u>

  Pass Band Region = 40Hz to ∞



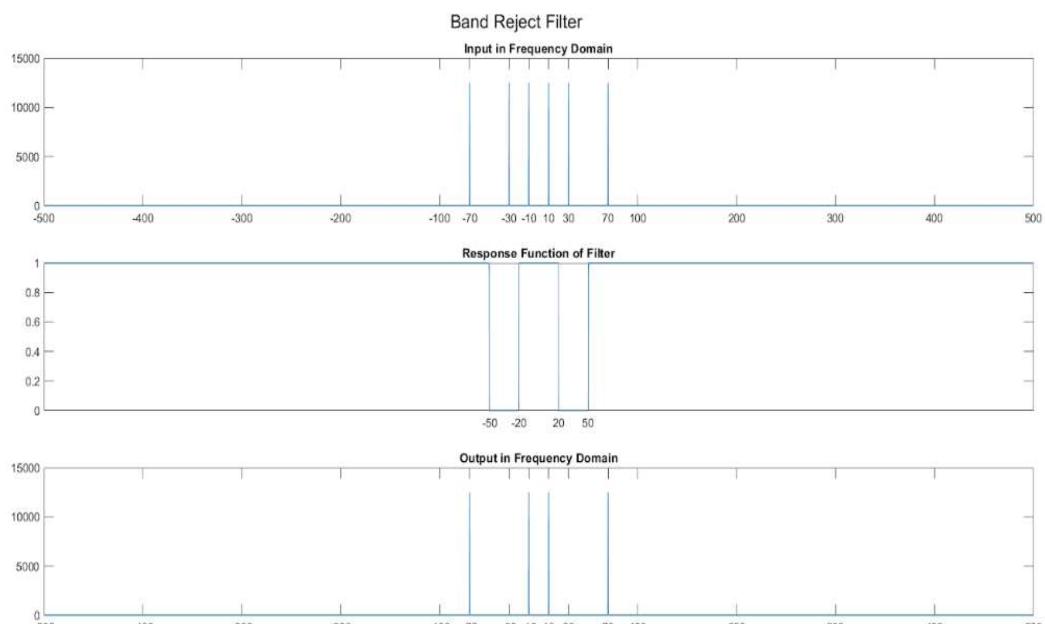
- <u>Band Pass Filter (Frequency Domain)</u>

  Pass Band Region= 20Hz to 50Hz

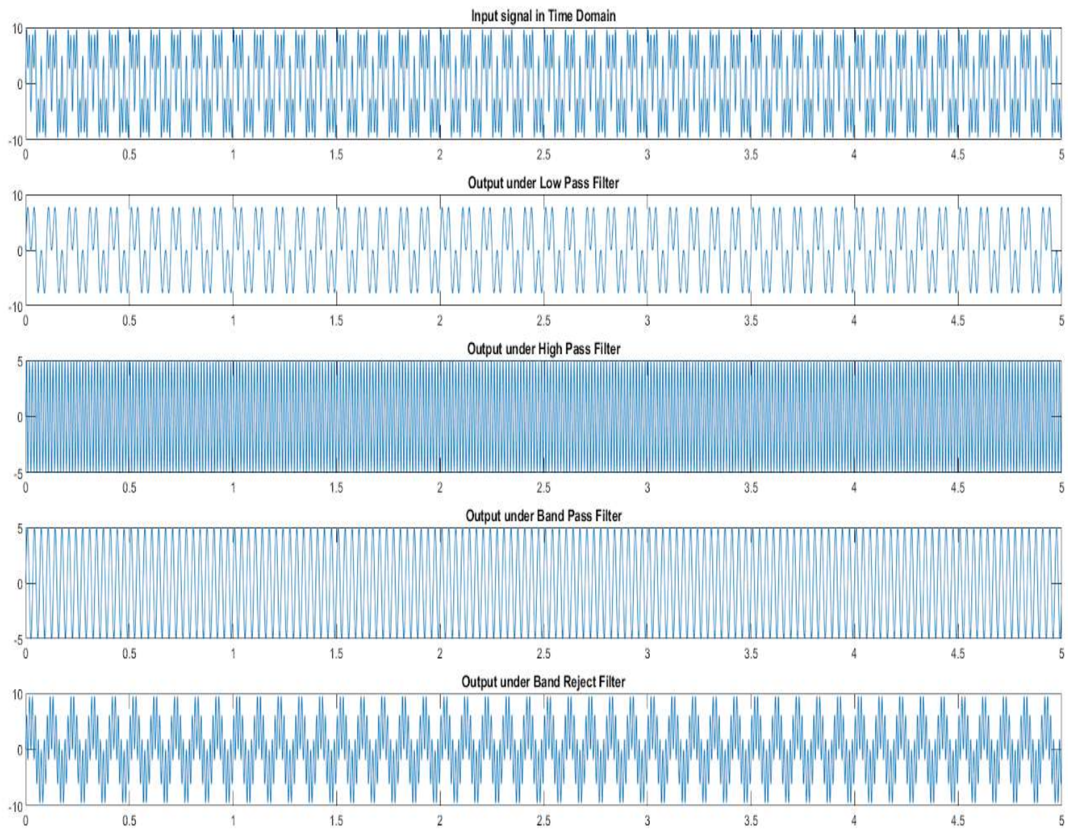


- <u>Band Reject Filter (Frequency Domain)</u>

  Stop Band Region= 20Hz to 50Hz



- <u>Final Outputs</u>



APPLICATIONS OF FILTER CIRCUITS

Filter circuits possess a wide variety of applications. The bandpass filter in the tuner of a radio allows a fixed frequency to the output signal. Filters are also used in speakers. The bass of a speaker have lower frequencies and the treble has higher frequencies. They are separated using high pass and low pass filters and are separately routed to corresponding bass speaker and treble speaker for clear music. Filters also have an important role in Power Supply Smoothing. The output of a power supply which is a rectifier has an AC ripple in it. These frequencies are filtered out using a low pass filter which results in a smoothened output signal. They are also used in various denoising systems and in Speech and Image Processing applications.

REFERENCES

1. MATLAB® Documentation
2. Signals and Systems-: Alan V. Oppenheim, Alan S. Willsky

MATLAB® CODE

```
%    FILTERS     %
%%Part of EE202 Course Project by Aryan, Vardhan and Gautam%%
clc
clear all

Fs=1000; % sampling frequency
Ts=1/Fs; % Sampling period or time step
dt=0:Ts:5-Ts; % 5 second signal duration

f1=10;
f2=30;
f3=70;

%% Input Signal
y_in=5*sin(2*pi*f1*dt)+5*sin(2*pi*f2*dt)+5*sin(2*pi*f3*dt);
plot(dt,y_in)
title(["Input Signal= 5sin(2\pif_1*t)+5sin(2\pif_2t)+5sin(2\pif_3t)","f_1=10Hz, f_2=30Hz, f=70Hz"])
%% Fourier Transform of Input
nfft=length(y_in);
ff=fft(y_in,nfft);
fffshift=fftshift(ff);    %% Scaling
x=[-(length(ff)/2):((length(ff)/2)-1)]*(Fs/length(ff));

%% IDEAL LPF
Stop_freq=40;
H_lpf=rectangularPulse(-Stop_freq,Stop_freq,x);    %Transfer function
Y_lpf=H_lpf.*fffshift;
figure;
subplot(3,1,1);plot(x,abs(fffshift));title("Input in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])
```

```
subplot(3,1,2);plot(x,abs(H_lpf));title("Response Function of Filter");xticks([-40,40])

subplot(3,1,3);plot(x,abs(Y_lpf));title("Output in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])

sgtitle('Low Pass Filter');


y_out_lpf=ifft(ifftshift(Y_lpf));          %Output


%% IDEAL BPF

Lower_f=20;

Higher_f=50;

H_bpf=rectangularPulse(Lower_f,Higher_f,x)+rectangularPulse(-Higher_f,-Lower_f,x);   %Transfer function

Y_bpf=H_bpf.*fffshift;

figure;

subplot(3,1,1);plot(x,abs(fffshift));title("Input in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])

subplot(3,1,2);plot(x,abs(H_bpf));title("Response Function of Filter");xticks([-50,-20,20,50])

subplot(3,1,3);plot(x,abs(Y_bpf));title("Output in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])

sgtitle('Band Pass Filter');


y_out_bpf=ifft(ifftshift(Y_bpf));          %Output


%% IDEAL HPF

Pass_freq=40;

H_hpf= x<-40 | x>40;       %Transfer function

Y_hpf=H_hpf.*fffshift;

figure;

subplot(3,1,1);plot(x,abs(fffshift));title("Input in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])

subplot(3,1,2);plot(x,abs(H_hpf));title("Response Function of Filter");xticks([-40,40])

subplot(3,1,3);plot(x,abs(Y_hpf));title("Output in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])

sgtitle('High Pass Filter');
```

```
y_out_hpf=ifft(ifftshift(Y_hpf));          %Output

%% IDEAL BRF
Low_stop=20;
High_stop=50;
H_brf1=x<Low_stop | x>High_stop ;
H_brf2= x>-Low_stop | x<-High_stop;
H_brf=H_brf1.*H_brf2;        %Transfer function
Y_brf=H_brf.*fffshift;
figure;
subplot(3,1,1);plot(x,abs(fffshift));title("Input in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])
subplot(3,1,2);plot(x,abs(H_brf));title("Response Function of Filter");xticks([-50,-20,20,50])
subplot(3,1,3);plot(x,abs(Y_brf));title("Output in Frequency Domain");xticks([-500,-400,-300,-200,-100,-70,-30,-10,10,30,70,100,200,300,400,500])
sgtitle('Band Reject Filter');

y_out_brf=ifft(ifftshift(Y_brf));          %Output

%% Filtering Results
figure;
subplot(5,1,1);plot(dt,y_in);title("Input signal in Time Domain");
subplot(5,1,2);plot(dt,y_out_lpf);title("Output under Low Pass Filter");
subplot(5,1,3);plot(dt,y_out_hpf);title("Output under High Pass Filter");
subplot(5,1,4);plot(dt,y_out_bpf);title("Output under Band Pass Filter");
subplot(5,1,5);plot(dt,y_out_brf);title("Output under Band Reject Filter");
```

Thank You
