DIGITAL SIGNAL PROCESSING LAB REPORT

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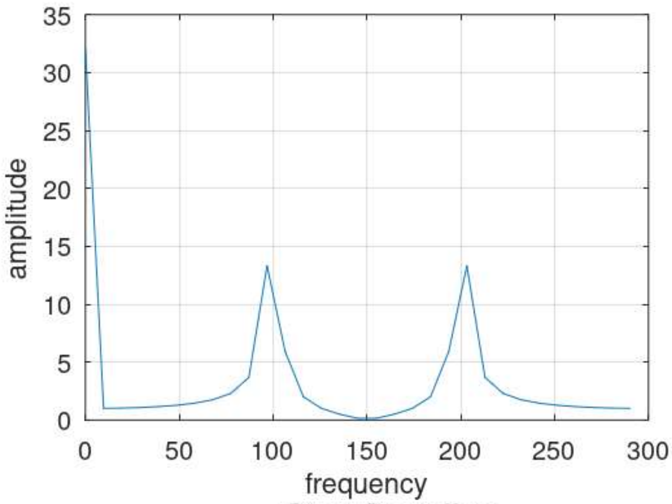
SEC : D
DATE : 06/04/2024

LAB NO. : 1
Sampling_ FD :
CODE :

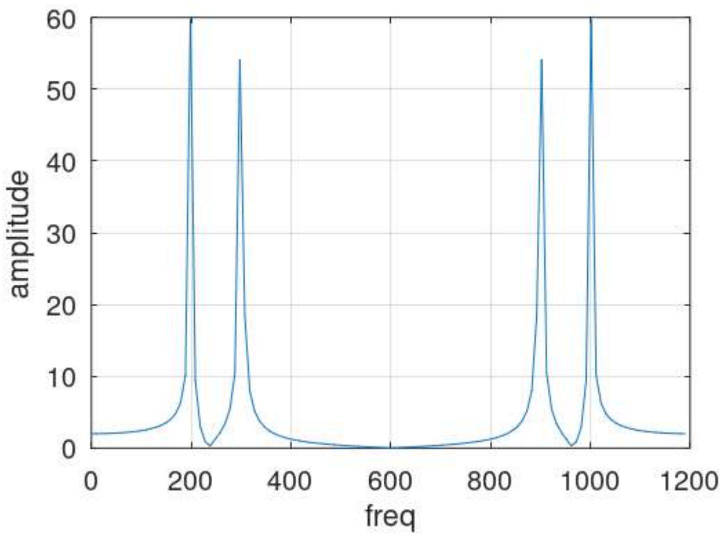
```
clear all;
f1 = input('Enter the first sine wave frequency = ');
f2 = input('Enter the second sine wave frequency = ');
fn = 2*max(f1,f2);
fs = fn/2;
t = [0:1/fs:0.1];
x = cos(2*pi*f1*t)+cos(2*pi*f2*t);
xk = fft(x);
f = [0:length(xk)-1]*fs/length(xk);
figure(1);
plot(f,abs(xk));
xlabel('frequency');
ylabel('amplitude');
title('Under Sampling');
grid;
fs = fn;
t = [0:1/fs:0.1];
x = cos(2*pi*f1*t)+cos(2*pi*f2*t);
xk = fft(x);
f = [0:length(xk)-1]*fs/length(xk);
figure(2);
plot(f,abs(xk));
xlabel('frequency');
ylabel('amplitude');
title('Nyquist Rate Sampling');
grid;
fs = 2*fn;
t = [0:1/fs:0.1];
x = cos(2*pi*f1*t)+cos(2*pi*f2*t);
xk = fft(x);
f = [0:length(xk)-1]*fs/length(xk);
figure(3);
plot(f,abs(xk));
xlabel('freq');
ylabel('amplitude');
title('Over Sampling');
grid;
```

OUTPUT:

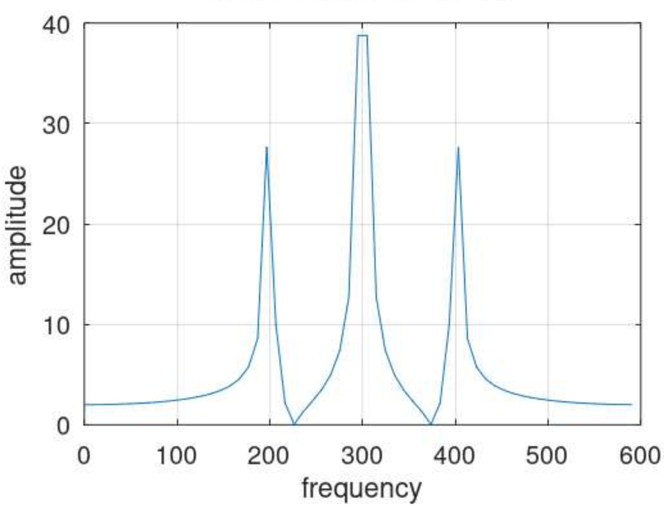
Under Sampling



Over Sampling



Nyquist Rate Sampling

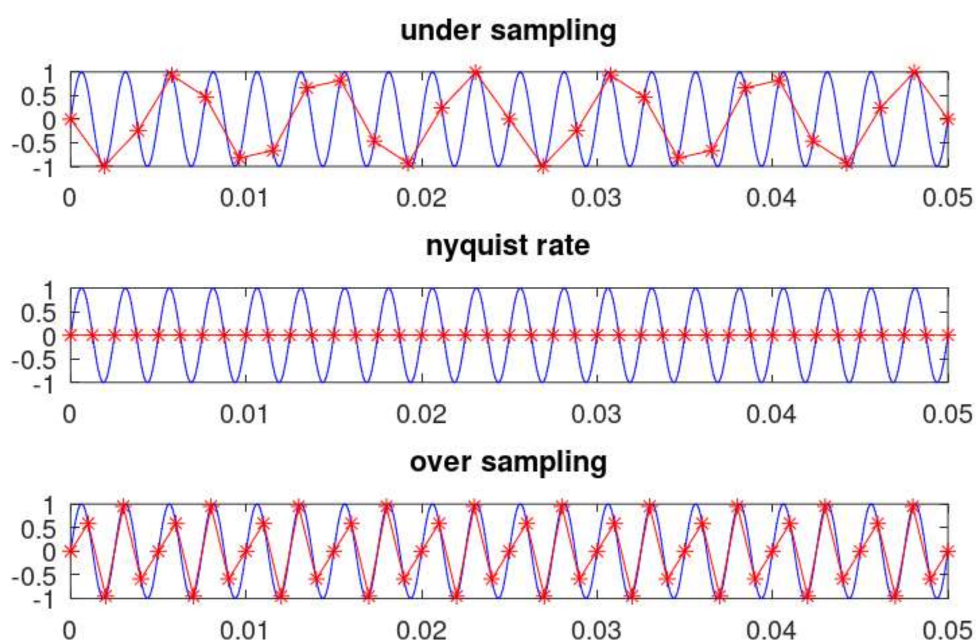


SAMPLING_FD

CODE:

```
clc;
clear all;
close all;
tfinal = 0.05;
t= 0:0.00005: tfinal;
fd= input('enter the analog frequency');
xt = sin(2*pi*fd*t);
fs1 = 1.3*fd;
n1= 0: 1/fs1: tfinal;
xn = sin(2*pi*n1*fd);
subplot(3,1,1);
plot(t,xt,'b',n1,xn,'r*-');
title('under sampling');
fs2= 2*fd;
n2= 0:1/fs2:tfinal;
xn = sin(2*pi*n2*fd);
subplot(3,1,2);
plot(t,xt,'b',n2,xn,'r*-');
title('nyquist rate');
fs3 = 2.5*fd;
n3= 0:1/fs3:tfinal;
xn = sin(2*pi*n3*fd);
subplot(3,1,3);
plot(t,xt,'b',n3,xn,'r*-');
title('over sampling');
```

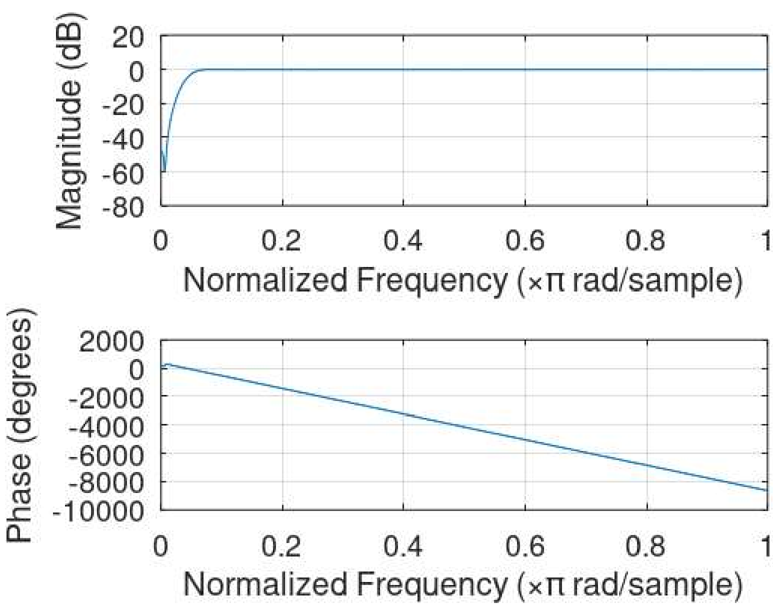
OUTPUT:



LAB NO. : 2 : FIR_1_function

for Nyquist
CODE:

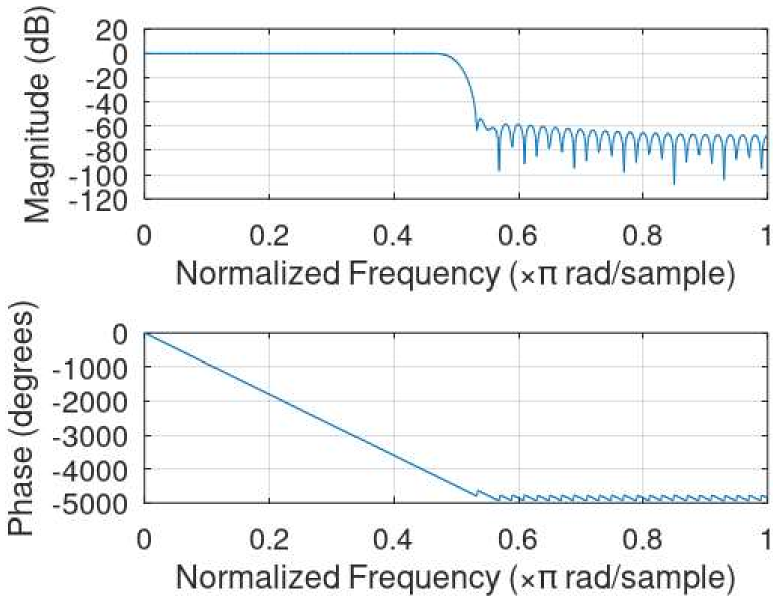
```
clear;
clc;
n = 100
Fs = 48000;
Nyq = Fs/2;
cutoffFreq = 1000;
Wn = cutoffFreq/Nyq;
h = fir1(n,Wn,'high');
freqz(h);
Fs = 48000;
N = 1*Fs;
x = 0.2*randn(N,1);
y = conv(x,h);
sound(x,Fs);
%sound(y,Fs);
```



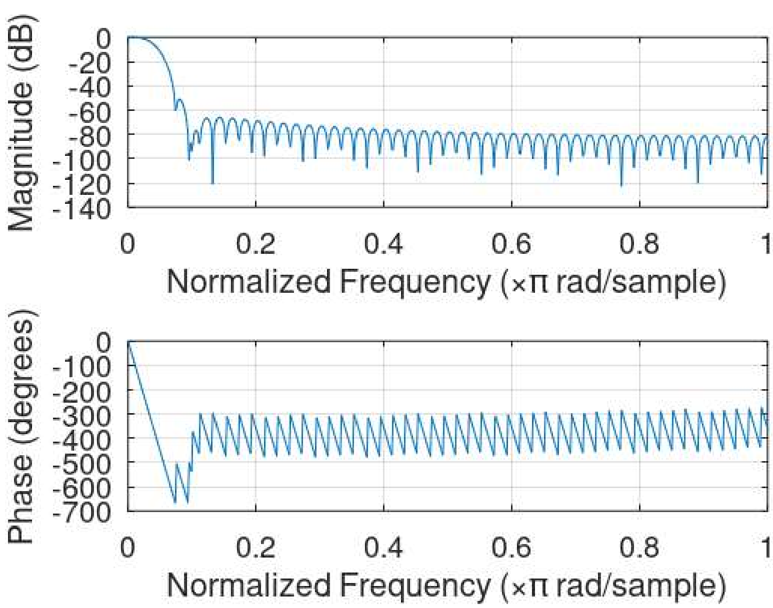
for simple LOW PASS filter
CODE:

```
clear;
clc;
n = 100
Wn = 0.5;
h = fir1(n,Wn);
freqz(h);
Fs = 48000;
N = 1*Fs;
x = 0.2*randn(N,1);
y = conv(x,h);
```

```
sound(x,Fs);
```

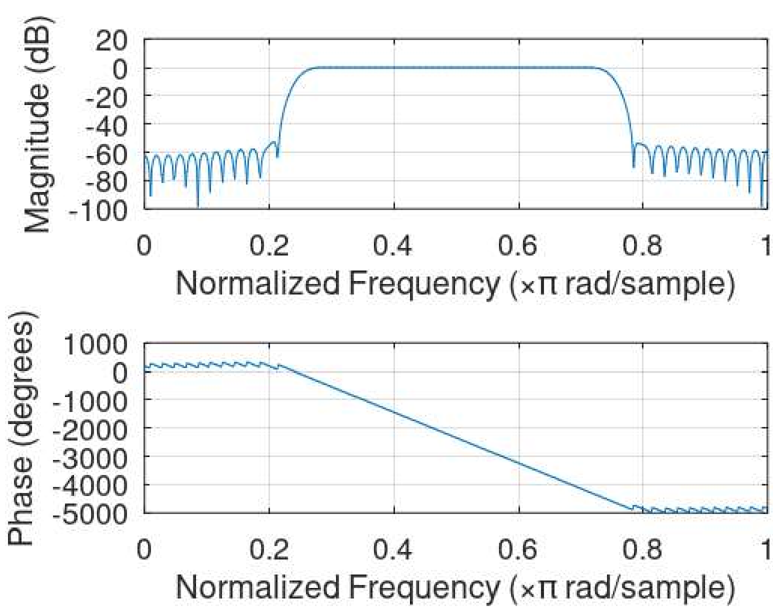



```
% Normalised cut-off frequency set relative to to sampling frequency
n = 100
Fs = 48000;
Nyq = Fs/2;
cutoffFreq = 1000;
Wn = cutoffFreq/Nyq;
h = fir1(n,Wn);
freqz(h);
```



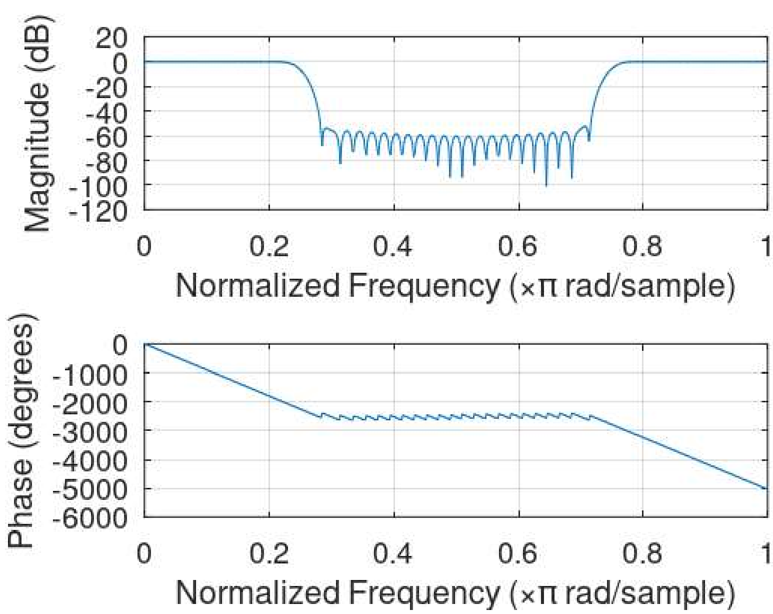
% for simple BAND PASS filter

```
clear;
clc;
n = 100
Fs = 48000;
Nyq = Fs/2;
Wn = [0.25 0.75];
h = fir1(n,Wn,'high');
freqz(h);
```



for simple BAND STOP filter

```
clear;
clc;
n = 100
Fs = 48000;
Nyq = Fs/2;
Wn = [0.25 0.75];
h = fir1(n,Wn,'stop');
freqz(h);
```



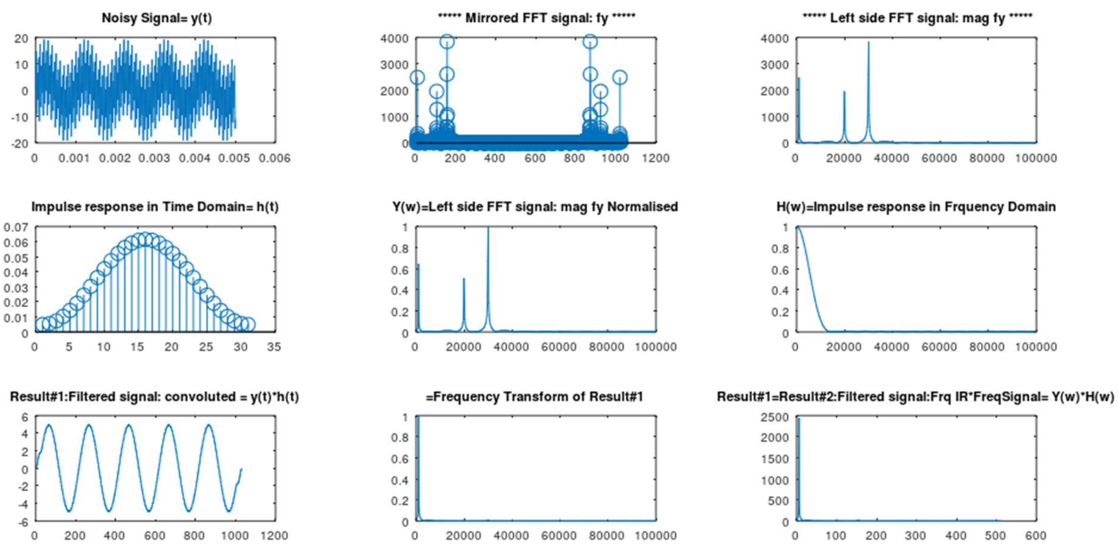
LAB NO. : 3

CODE:

```
clc;
clear all;
Fs=200e3;      %say sampling frequency 200 kHz
Ts=1/Fs;       %set sampling time
dt=0:Ts:5e-3-Ts; %Lets say 5 ms

f1=1e3;  %say 1 kHz
f2=20e3; %say 20 kHz
f3=30e3; %say 30 kHz

y=5*sin(2*pi*f1*dt)+5*sin(2*pi*f2*dt)+10*sin(2*pi*f3*dt);
subplot(3,3,1) ;
plot(dt,y);
title(' Noisy Signal= y(t)');
sound(y,Fs);
nfft=length(y);
nfft2=2.^nextpow2(nfft);
fy=fft(y,nfft2);

subplot(3,3,2) ;
%stem(fy);
stem(abs(fy));
title(' ***** Mirrored FFT signal: fy *****');

fy=fy(1:nfft2/2);
xfft=Fs.*(0:nfft2/2-1)/nfft2;

subplot(3,3,3) ;
plot(xfft,abs(fy));
title(' ***** Left side FFT signal: mag fy *****');

subplot(3,3,5) ;
plot(xfft,abs(fy/max(fy)));
title('Y(w)=Left side FFT signal: mag fy Normalised');

cut_off=1.5e3/Fs/2;
order=30;

h=fir1(order,cut_off);
subplot(3,3,4) ;
stem(h);
title('Impulse response in Time Domain= h(t)');
fh=fft(h,nfft2);
fh=fh(1:nfft2/2);
subplot(3,3,6);
plot(xfft,abs(fh));
title('H(w)=Impulse response in Frquency Domain');
```

```
mul=fh.*fy;
subplot(3,3,7) ;
con=conv(y,h);
plot(con);
title('Result#1:Filtered signal: convoluted = y(t)*h(t)');
subplot(3,3,9);
plot(abs(mul));
title('Result#1=Result#2:Filtered signal:Frq IR*FreqSignal= Y(w)*H(w)');
fcon=fft(con,nfft2);
fcon=fcon(1:nfft2/2);
subplot(3,3,8);
plot(xfft,abs(fcon/max(fcon)));
title('=Frequency Transform of Result#1');
```

OUTPUT:



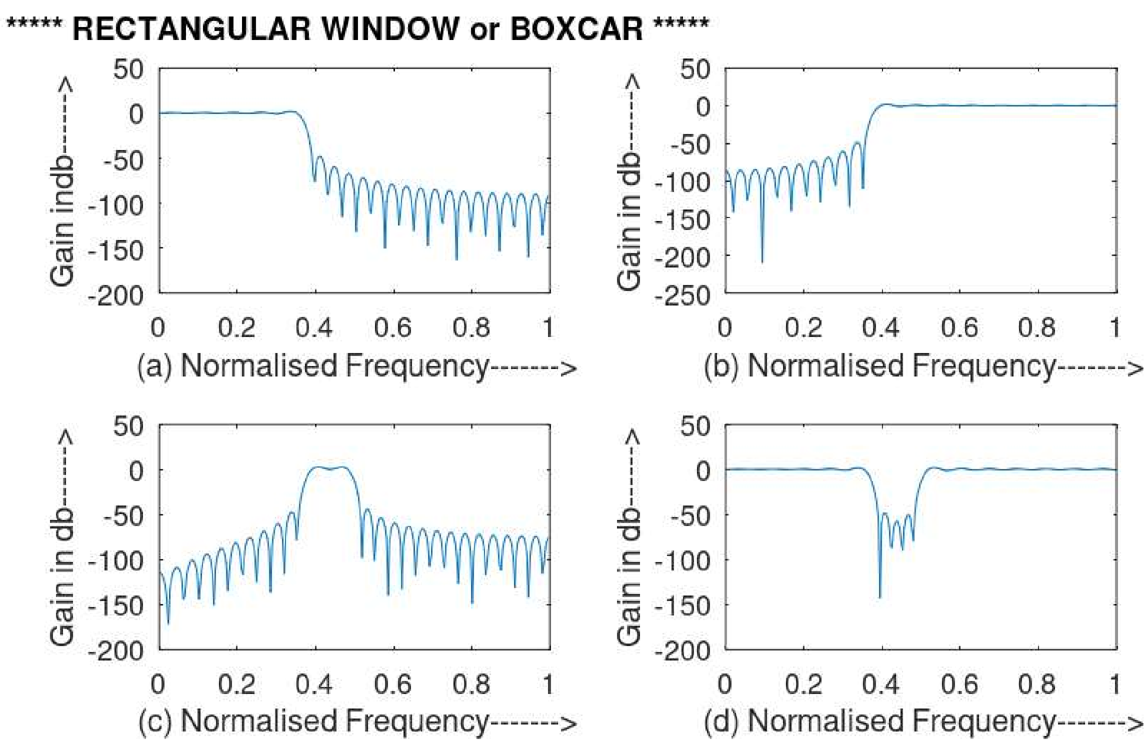
LAB NO. : 4

CODE:

```
clc; close all; clear all;
rp = input('Enter the Pass Band Ripple: ');
rs = input('Enter the Stop Band Ripple: ');
fp = input('Enter the Pass Band Frequency: ');
fs = input('Enter the Stop Band Frequency: ');
f = input('Enter the Sampling Frequency: ');
wp = 2 * fp/f;
ws = 2 * fs/f;
num = - 20 * log(sqrt(rp*rs))- 13;
den = 14.6 * (fs-fp)/f;
n = ceil (num/den) ;
n1 = n+1;
if(rem(n,2)~=0)
n1 = n;
n = n-1;
end
y = boxcar (n1) ;
disp('Filter ordern n= ');n

% LOW PASS FILTER
b = fir1(n,wp,y);
[h,w] = freqz(b,1,256);
m = 20 * log(abs(h));
subplot(2,2,1) ;
plot (w/pi,m) ;
title(' ***** RECTANGULAR WINDOW or BOXCAR *****');
ylabel('Gain indb------>');xlabel(' (a) Normalised Frequency------->');

% HIGH PASS FILTER
b = fir1(n,wp,'high',y);
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(2,2,2);
plot(w/pi,m);
ylabel('Gain in db------>');
xlabel(' (b) Normalised Frequency------->');

% BAND PASS FILTER
wn = [wp ws];
b = fir1(n,wn,y);
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(2,2,3) ;
plot (w/pi, m) ;
ylabel('Gain in db------>');
xlabel(' (c) Normalised Frequency------->');
```

```
% BAND STOP FILTER
b = fir1(n,wn,'stop',y);
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(2,2,4) ;
plot (w/pi, m) ;
ylabel('Gain in db------>');
xlabel(' (d) Normalised Frequency------->');
```

OUTPUT:

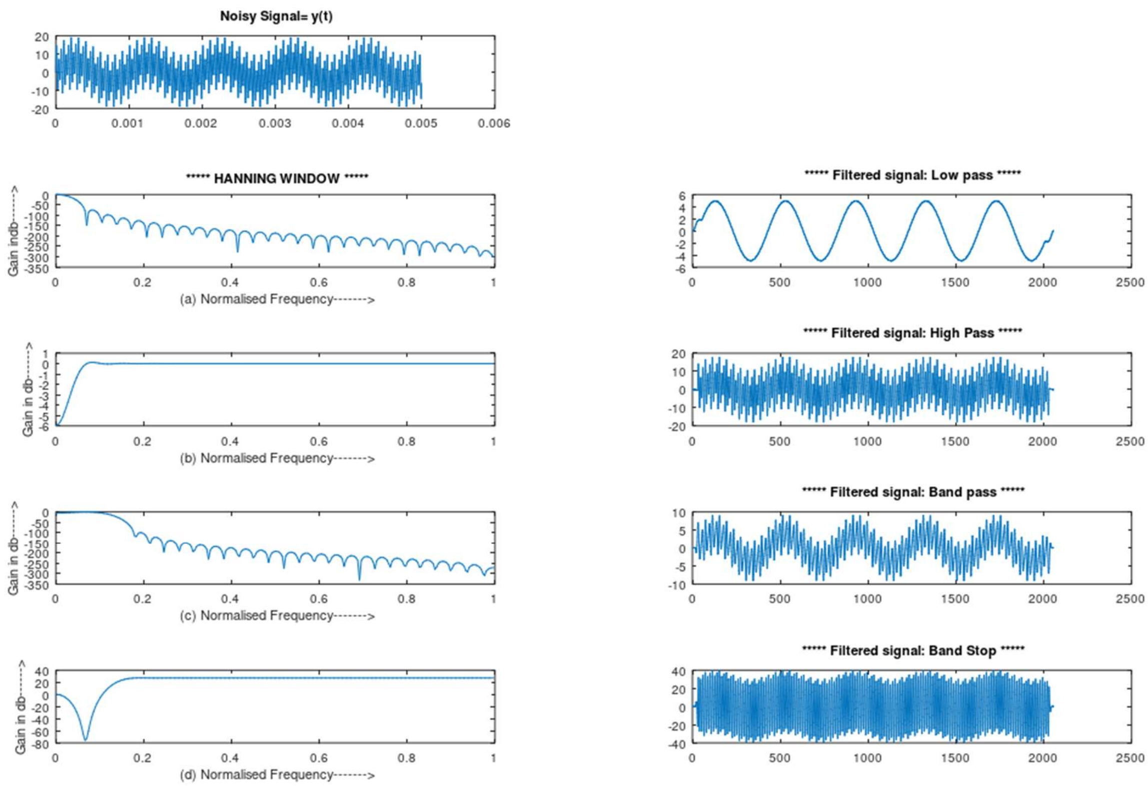


LAB NO. : 5 (hanning)

CODE:

```
rp = .05;
rs = .04;
fp = 2e3;
fs = 25e3;
Fs = 400e3;

Ts=1/Fs;        %set sampling time
dt=0:Ts:5e-3-Ts; %Lets say 5 ms

f1=1e3;  %say 1 kHz
f2=20e3; %say 20 kHz
f3=30e3; %say 30 kHz

%Noisy Signal
y=5*sin(2*pi*f1*dt)+5*sin(2*pi*f2*dt)+10*sin(2*pi*f3*dt);
subplot(5,2,1) ;
plot(dt,y);
title(' Noisy Signal= y(t)');

wp = 2 * fp /Fs;
ws = 2 * fs/Fs;
num = - 20 * log( sqrt(rp*rs))- 13;
den = 14.6 * (fs-fp)/Fs;
n = ceil (num/den) ;
%n=100;
n1 = n+1;
if(rem(n,2)~=0)
n1 = n;
n = n-1;
end
win = hanning(n1);
disp('Filter ordern n= ');n

% LOW PASS FILTER
b = fir1(n,wp,win);
[h,w] = freqz(b,1,256);
%[h,w] = freqz(b,a,n) returns the n-point frequency response vector h and the corresponding angular
frequency vector w for the digital filter with transfer function coefficients stored in b and a.
%Ref: https://in.mathworks.com/help/signal/ref/freqz.html
m = 20 * log(abs(h));
subplot(5,2,3) ;
plot (w/pi,m) ;
title(' ***** HANNING WINDOW *****');
ylabel('Gain indb------>');
xlabel(' (a) Normalised Frequency------->');

con=conv(y,b);
```

```
subplot(5,2,4) ;
plot(con);
title(' ***** Filtered signal: Low pass *****');

% HIGH PASS FILTER
b = fir1(n,wp,'high',win);
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(5,2,5);
plot(w/pi,m) ;
ylabel('Gain in db------>');
xlabel(' (b) Normalised Frequency------->');

con=conv(y,b);

subplot(5,2,6) ;
plot(con);
title(' ***** Filtered signal: High Pass *****');

% BAND PASS FILTER
wn = [wp ws];
b = fir1(n,wn,win);
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(5,2,7) ;
plot (w/pi, m) ;
ylabel('Gain in db------>');
xlabel(' (c) Normalised Frequency------->');

con=conv(y,b);

subplot(5,2,8) ;
plot(con);
title(' ***** Filtered signal: Band pass *****');

% BAND STOP FILTER
b = fir1 (n, wn, 'stop' , win) ;
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(5,2,9) ;
plot (w/pi, m) ;
ylabel('Gain in db------>');
xlabel(' (d) Normalised Frequency------->');

con=conv(y,b);
subplot(5,2,10) ;
plot(con);
title(' ***** Filtered signal: Band Stop *****');
```



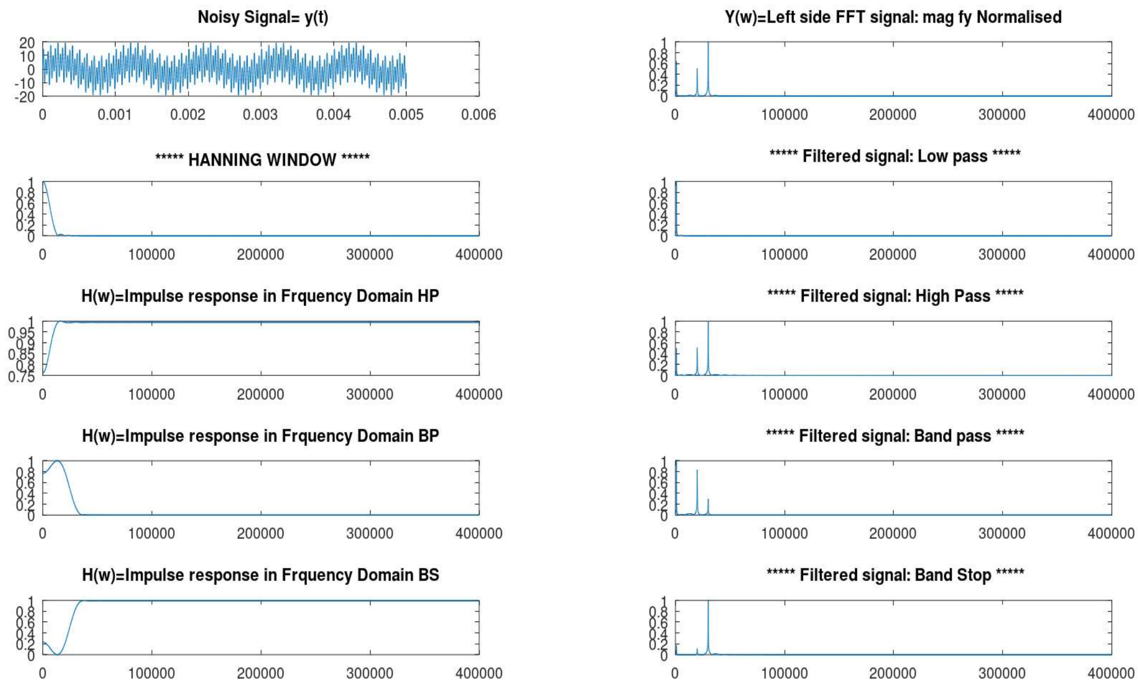
HANNING_2
CODE:

```
rp = .05;
rs = .04;
fp = 2e3;
fs = 25e3;
Fs = 800e3;

Ts=1/Fs;        %set sampling time
dt=0:Ts:5e-3-Ts; %Lets say 5 ms

f1=1e3;  %say 1 kHz
f2=20e3; %say 20 kHz
f3=30e3; %say 30 kHz

%Noisy Signal
y=5*sin(2*pi*f1*dt)+5*sin(2*pi*f2*dt)+10*sin(2*pi*f3*dt);
subplot(5,2,1) ;
plot(dt,y);
title(' Noisy Signal= y(t)');

nfft=length(y);
nfft2=2.^nextpow2(nfft);
```

```
fy=fft(y,nfft2);
fy=fy(1:nfft2/2);
xfft=Fs.*(0:nfft2/2-1)/nfft2;

subplot(5,2,2) ;
plot(xfft,abs(fy/max(fy))); %Normalising Y-axis (magnitude) to 1
title('Y(w)=Left side FFT signal: mag fy Normalised');

wp = 2 * fp /Fs;
ws = 2 * fs/Fs;
num = - 20 * log( sqrt(rp*rs))- 13;
den = 14.6 * (fs-fp)/Fs;
n = ceil (num/den) ;
%n=30000;
n1 = n+1;
if(rem(n,2)~=0)
n1 = n;
n = n-1;
end
win = hanning(n1);
disp('Filter ordern n= ');n

% LOW PASS FILTER
b1 = fir1(n,wp,win);
[h,w] = freqz(b1,1,256);

fh_low=fft(b1,nfft2);
fh_low=fh_low(1:nfft2/2);

m = 20 * log(abs(h));

subplot(5,2,3) ;
%plot(xfft,abs(fh_low));
plot(xfft,abs(fh_low/max(fh_low)));
title('H(w)=Impulse response in Frquency Domain LP');

%plot (w/pi,m) ;
title(' ***** HANNING WINDOW *****');


mul=fh_low.*fy;
subplot(5,2,4) ;
%plot(con);
plot(xfft,abs(mul/max(mul)));
title(' ***** Filtered signal: Low pass *****');

% HIGH PASS FILTER
b2 = fir1(n,wp,'high',win);
[h,w] = freqz(b2,1,256);
```

```
fh_high=fft(b2,nfft2);
fh_high=fh_high(1:nfft2/2);

m = 20*log(abs(h));
subplot(5,2,5);

plot(xfft,abs(fh_high/max(fh_high))); %No
title('H(w)=Impulse response in Frquency Domain HP');

mul=fh_high.*fy;

subplot(5,2,6) ;
%plot(con);
plot(xfft,abs(mul/max(mul)));
title(' ***** Filtered signal: High Pass *****');

% BAND PASS FILTER
wn = [wp ws];
b3 = fir1(n,wn,win);
[h,w] = freqz(b3,1,256);

fh_bp=fft(b3,nfft2);
fh_bp=fh_bp(1:nfft2/2);
m = 20*log(abs(h));
subplot(5,2,7) ;

plot(xfft,abs(fh_bp/max(fh_bp)));
title('H(w)=Impulse response in Frquency Domain BP');


mul=fh_bp.*fy;

subplot(5,2,8) ;
%plot(con);
plot(xfft,abs(mul/max(mul)));
title(' ***** Filtered signal: Band pass *****');

% BAND STOP FILTER
b4 = fir1 (n, wn, 'stop' , win) ;
[h,w] = freqz(b4,1,256);

fh_bs=fft(b4,nfft2);
fh_bs=fh_bs(1:nfft2/2);

m = 20*log(abs(h));
subplot(5,2,9) ;

plot(xfft,abs(fh_bs/max(fh_bs)));
title('H(w)=Impulse response in Frquency Domain BS');
```

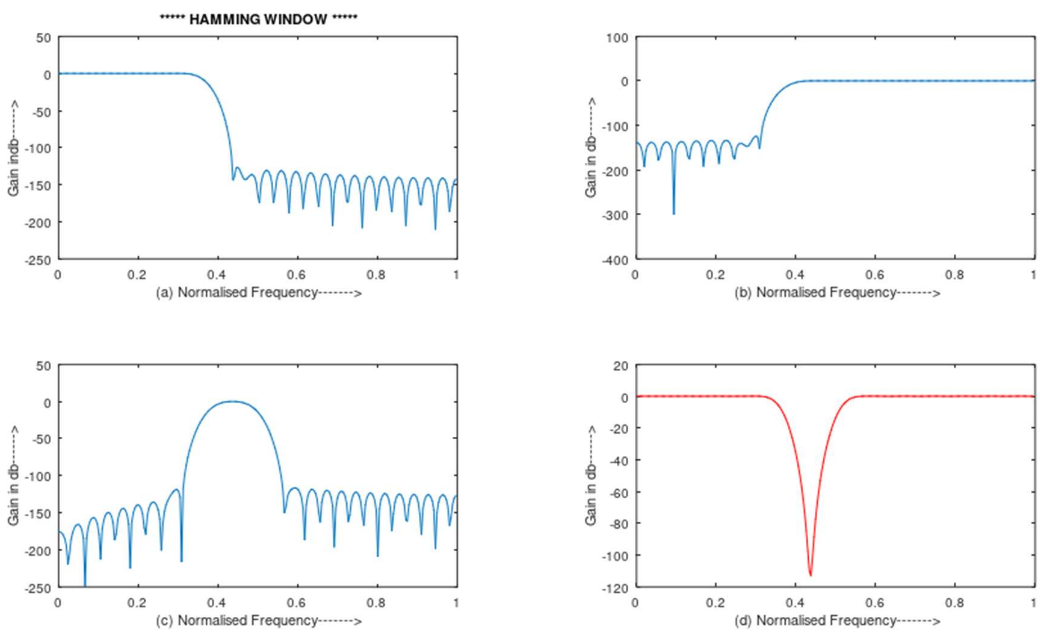
```
mul=fh_bs.*fy;
subplot(5,2,10) ;
%plot(con);
plot(xfft,abs(mul/max(mul)));
title(' ***** Filtered signal: Band Stop *****');
```



LAB NO. : 6

```
clc; close all; clear all;
rp = input('Enter the Pass Band Ripple: ');
rs = input('Enter the Stop Band Ripple: ');
fp = input('Enter the Pass Band Frequency: ');
fs = input('Enter the Stop Band Frequency: ');
f = input('Enter the Sampling Frequency: ');
wp = 2 * fp /f;
ws = 2 * fs/f;
num = - 20 * log( sqrt(rp*rs))- 13;
den = 14.6 * (fs-fp)/f;
n = ceil (num/den) ;
n1 = n+1;
if(rem(n,2)~=0)
n1 = n;
n = n-1;
end
y = hamming(n1);
disp('Filter ordern n= ');
% LOW PASS FILTER
b = fir1(n,wp,y);
[h,w] = freqz(b,1,256);
m = 20 * log(abs(h));
subplot(2,2,1) ;
plot (w/pi,m) ;
title(' ***** HAMMING WINDOW *****');
ylabel('Gain indb------>');
xlabel(' (a) Normalised Frequency------->');
% HIGH PASS FILTER
b = fir1(n,wp,'high',y);
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(2,2,2);
plot(w/pi,m) ;
ylabel('Gain in db------>');
xlabel(' (b) Normalised Frequency------->');
% BAND PASS FILTER
wn = [wp ws];
b = fir1(n,wn,y);
```

```
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(2,2,3) ;
plot (w/pi, m) ;
ylabel('Gain in db------>');
xlabel(' (c) Normalised Frequency------->');
% BAND STOP FILTER
b = fir1 (n, wn, 'stop' , y) ;
[h,w] = freqz(b,1,256);
m = 20*log(abs(h));
subplot(2,2,4) ;
plot (w/pi, m,'r') ;
ylabel('Gain in db------>');
xlabel(' (d) Normalised Frequency------->');
```

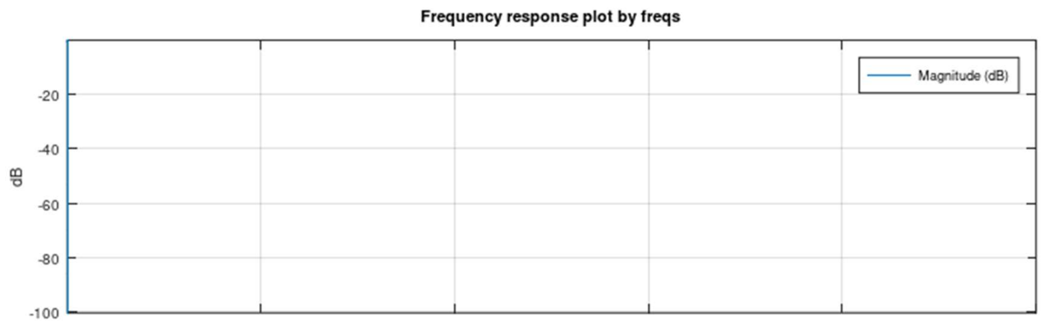


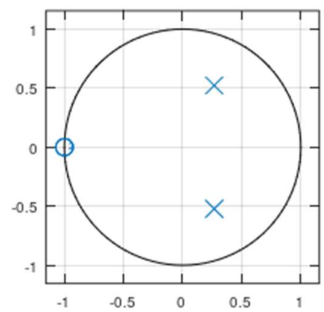
LAB NO. : 7

CODE:

```
clc;
close all;
format long;
T = 1;
fs = 1/T;
wp = 0.35*pi;
ws = 0.7*pi;
rp = 0.9;
rs = 0.275;
op = (2/T)*tan(wp/2);
os = (2/T)*tan(ws/2);
display(op);
display(os);
rP = abs(20*log10(rp));
rS = abs(20*log10(rs));
display(rP);
display(rS);
[n,wc] = cheb1ord(op,os,rP,rS,'s');
display(n);
display(wc);
[z,p,k] = cheb1ap(n,rP);
display(z);
display(p);
display(k);
[b,a] = cheby1(n,rP,1,'low','s');
display(b);
display(a);
[z, p, k] = sftrans (z, p, k,wc, false);
[bt, at] = zp2tf (z, p, k);
display(bt);
display(at);

s = tf(bt,at);
display(s);
w = logspace(-10,10);
freqs(bt,at,w);
[bz,az] = bilinear(bt,at,fs);
display(bz);
display(az);
z = tf(bz,az,T,'variable','z^-1');
display(z);
zplane(bz,az);
```

OUTPUT:

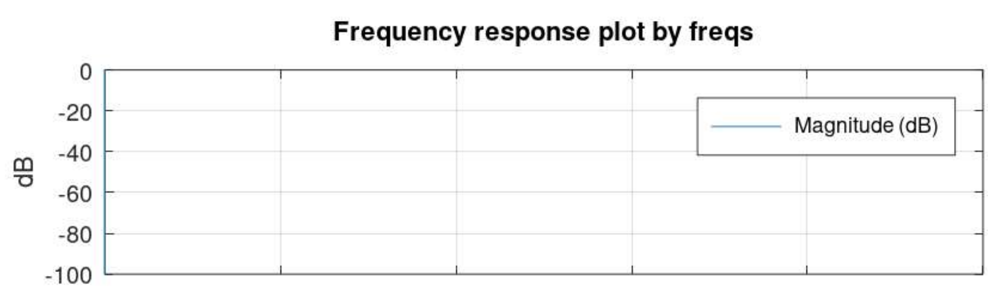


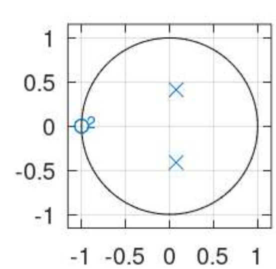


FOR BUTTERWORTH FILTER

```
clc;
close all;
format long;
T = 1;
fs = 1/T;
wp = 0.35*pi;
ws = 0.7*pi;
rp = 0.9;
rs = 0.275;
% Convert analog frequencies to digital domain
op = (2/T)*tan(wp/2);
os = (2/T)*tan(ws/2);
display(op);
display(os);
% Calculate passband and stopband attenuation in dB
rP = abs(20*log10(rp));
rS = abs(20*log10(rs));
display(rP);
display(rS);
[n,wc] = buttord(op,os,rP,rS,'s');
display(n);
```

```
display(wc);
[z,p,k] = buttap(n);
display(z);
display(p);
display(k);
% Design Butterworth analog filter
[b,a] = zp2tf(z, p, k);
display(b);
display(a);
% Change cutoff frequency for lowpass analog filter
[z, p, k] = sftrans (z, p, k,wc, false);
[bt, at] = zp2tf (z, p, k);
display(bt);
display(at);
s = tf(bt,at);
display(s);
w = logspace(-10,10);
freqs(bt,at,w);
[bz,az] = bilinear(bt,at,fs);
display(bz);
display(az);
z = tf(bz,az,T,'variable','z^-1');
display(z);
zplane(bz,az);
fvtool(bz,az);
```





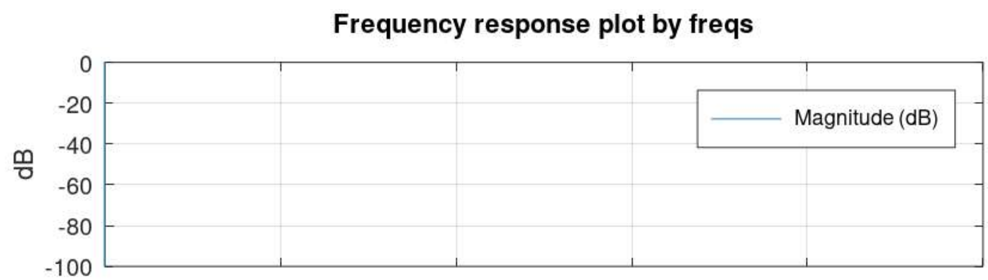
LAB NO. : 8

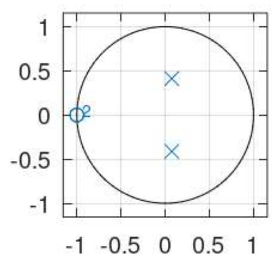
CODE :

```
clc; close all; format long;
T = 1; %Time
fs = 1/T;

wp = 0.35*pi;
ws = 0.7*pi;
rp = 0.9;
rs = 0.275;

op = (2/T)*tan(wp/2);
os = (2/T)*tan(ws/2);
display(op);
display(os);

rP = abs(20*log10(rp));
rS = abs(20*log10(rs));
display(rP);
display(rS);

[n,wc] = cheb1ord(op,os,rP,rS,'s');
display(n);
display(wc);

[z,p,k] = cheb1ap(n,rP);
display(z);
display(p);
display(k);

[b,a] = cheby1(n,rP,1,'low','s');

display(b);
display(a);

[z, p, k] = sftrans (z, p, k,wc, false);
[bt, at] = zp2tf (z, p, k);
display(bt);
display(at);
s = tf(bt,at);
display(s);

w = logspace(-10,10);

freqs(bt,at,w);
```

```
zplane(bz,az); %ZERO-POLE Plot
fvtool(bz,az);
```

OUTPUT:

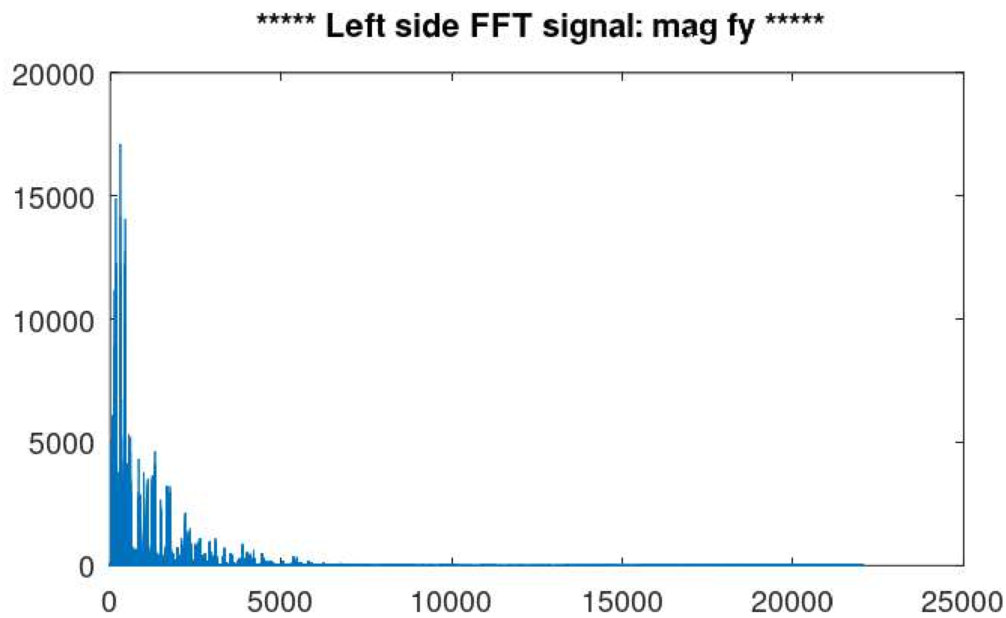




LAB NO. : 9
CODE:

```
clc;
clear all;
info = audioinfo('handel.wav');
[y,Fs] = audioread('handel.wav');
%sound(y,Fs);
Ts=1/Fs;        %set sampling time
t = 0:(1/Fs):8.9249;
t = t(1:end-1);
nfft=length(y);
nfft2=2.^nextpow2(nfft);
fy=fft(y,nfft2);
fy=fy(1:nfft2/2);
xfft=Fs.*(0:nfft2/2-1)/nfft2;
plot(xfft,abs(fy));
title(' ***** Left side FFT signal: mag fy *****');
f_pass = 15000;#input('Enter the Pass Band Frequency: ');
f_stop = 20000; #input('Enter the Stop Band Frequency: ');
rp = 0.05; #input('Enter the Pass Band Ripple: ');
rs = 0.04; #input('Enter the Stop Band Ripple: ');
wp = 2 * f_pass /Fs;
ws = 2 * f_stop/Fs;
%cut_off=1.5e3/Fs/2; % cut off frequency should be normalised to Nyquist Freq, i.e. Fs/2
%order=30;
num = - 20 * log( sqrt(rp*rs))- 13;
den = 14.6 * (f_stop-f_pass)/Fs;
n = ceil (num/den) ;
n1 = n+1;
if(rem(n,2)~=0)
n1 = n;
n = n-1;
end
z = hamming(n1);
% LOW PASS FILTER
h_low = fir1(n,wp,z);
% HIGH PASS FILTER
h_high = fir1(n,wp,'high',z);
% BAND PASS FILTER
wn = [wp ws];
h_band = fir1(n,wn,"bandpass",z);
con_L=conv(y,h_low);
%sound(con_L,Fs);
con_H=50.*conv(y,h_high);
sound(con_H,Fs);
con_B=conv(y,h_band);
%sound(con_B,Fs);
s=input('done  ');
```

OUTPUT:



***** Left side FFT signal: mag fy *****
20000
15000
10000
5000
0
0
5000
10000
15000
20000
25000