SOFTWARE EXPERIMENTS (USING MATLAB)

Exp .No:01 **Date:**

GENERATION OF DISCRETE TIME SIGNALS

AIM: -To write a "MATLAB" Program to generate of discrete time signals like unit impulse, unit step, unit ramp, exponential signal and sawtooth signals.

SOFTWARE REQURIED :-

- PC and MATLAB Software (2019b 9.7version)

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

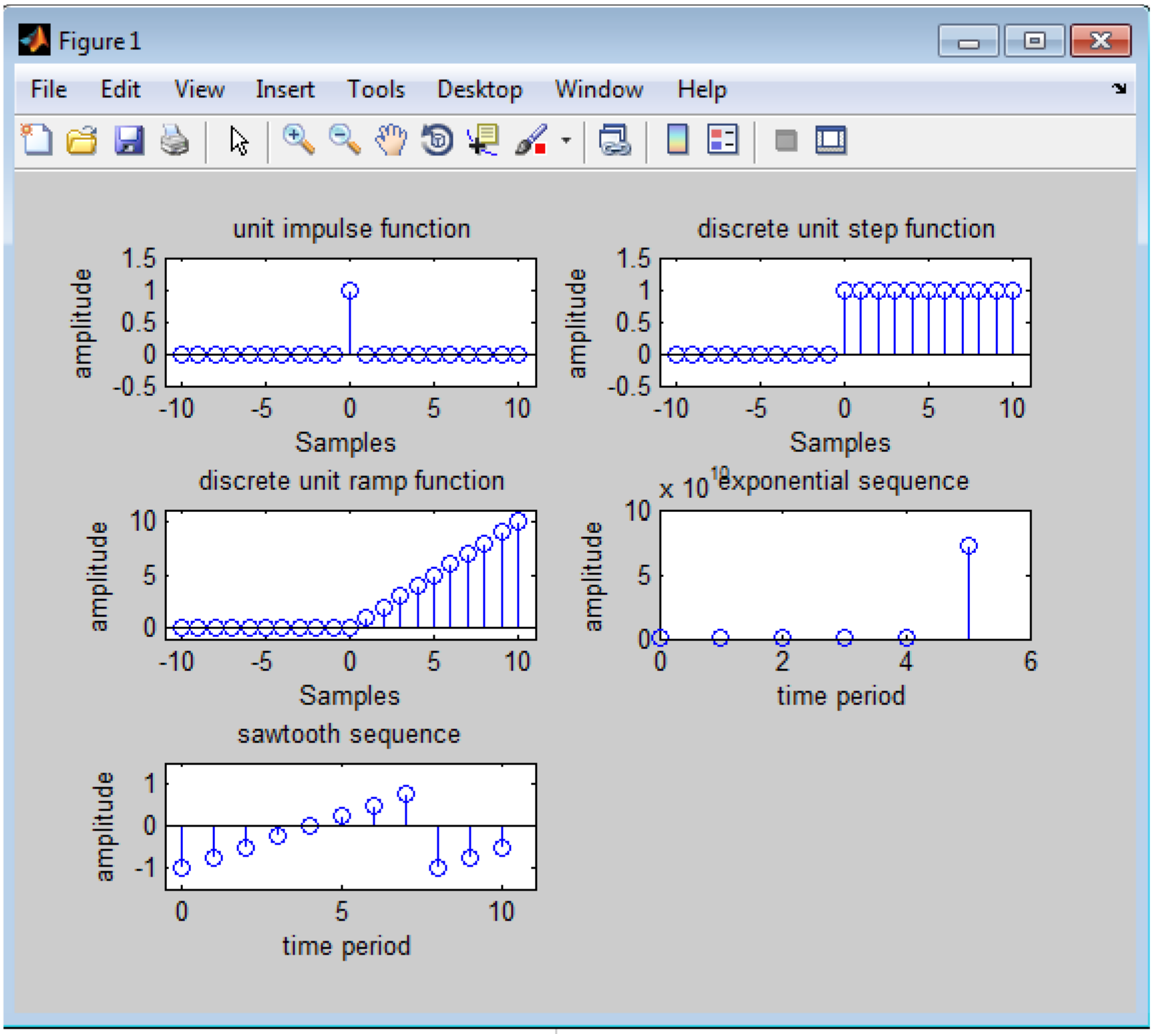
```
%unit impulse function%
%Discrete%
n=-10:10;
Xn=(n==0);
subplot(3,2,1);
stem(n,Xn);
axis([-11 11 -0.5 1.5]);
xlabel('Samples');
ylabel('amplitude');
title(' unit impulse function');
%unit step function%
%Discrete%
n=-10:10;
Xn=(n>=0);
subplot(3,2,2);
stem(n,Xn);
axis([-11 11 -0.5 1.5]);
xlabel('Samples');
ylabel('amplitude');
title(' discrete unit step function');

%unit ramp function%
%Discrete%
n=-10:10;
Xn=(n>=0).*n;
subplot(3,2,3);
stem(n,Xn);
axis([-11 11 -1 11]);
xlabel(' Samples');
ylabel('amplitude');
title(' discrete unit ramp function');
```

```
% exponential signal:
%Discrete%
n2=input('enter the  length of the exponential sequence');
t=0:n2;
a=input('enter the a value');
y2=exp(a*t);
subplot(3,2,4);
stem(t,y2);
ylabel('amplitude');
xlabel('time period');
title('exponential sequence')

%sawtooth signal
%Discrete%
n=0:10;
Xn=sawtooth(pi*n/4);
subplot(3,2,5);
stem(n,Xn);
axis([-0.5 11 -1.5 1.5])
xlabel('time period');
ylabel('amplitude');
title('sawtooth sequence');
```

OUTPUT:-



RESULT:-

CONCLUSIONS:

VIVA QUESTIONS:

1. Define impulse, unit step, ramp signals and write their expressions?
2. Define exponential and sinusoidal signals and write their expressions?
3. Express unit step signal in terms of unit impulse?
4. Express ramp signal in terms of unit step signal?
5. Represent the signal x[n]={1,2,-1,3,2}using impulse signal?

Exp .No:02 **Date:**

SUM OF TWO SINUSOIDAL SIGNALS

AIM: - To write a MATLAB program to find the sum of two sinusoidal signals and to find frequency response (magnitude and phase).

SOFTWARE REQURIED :-

- PC and MATLAB Software (2019b 9.7version)

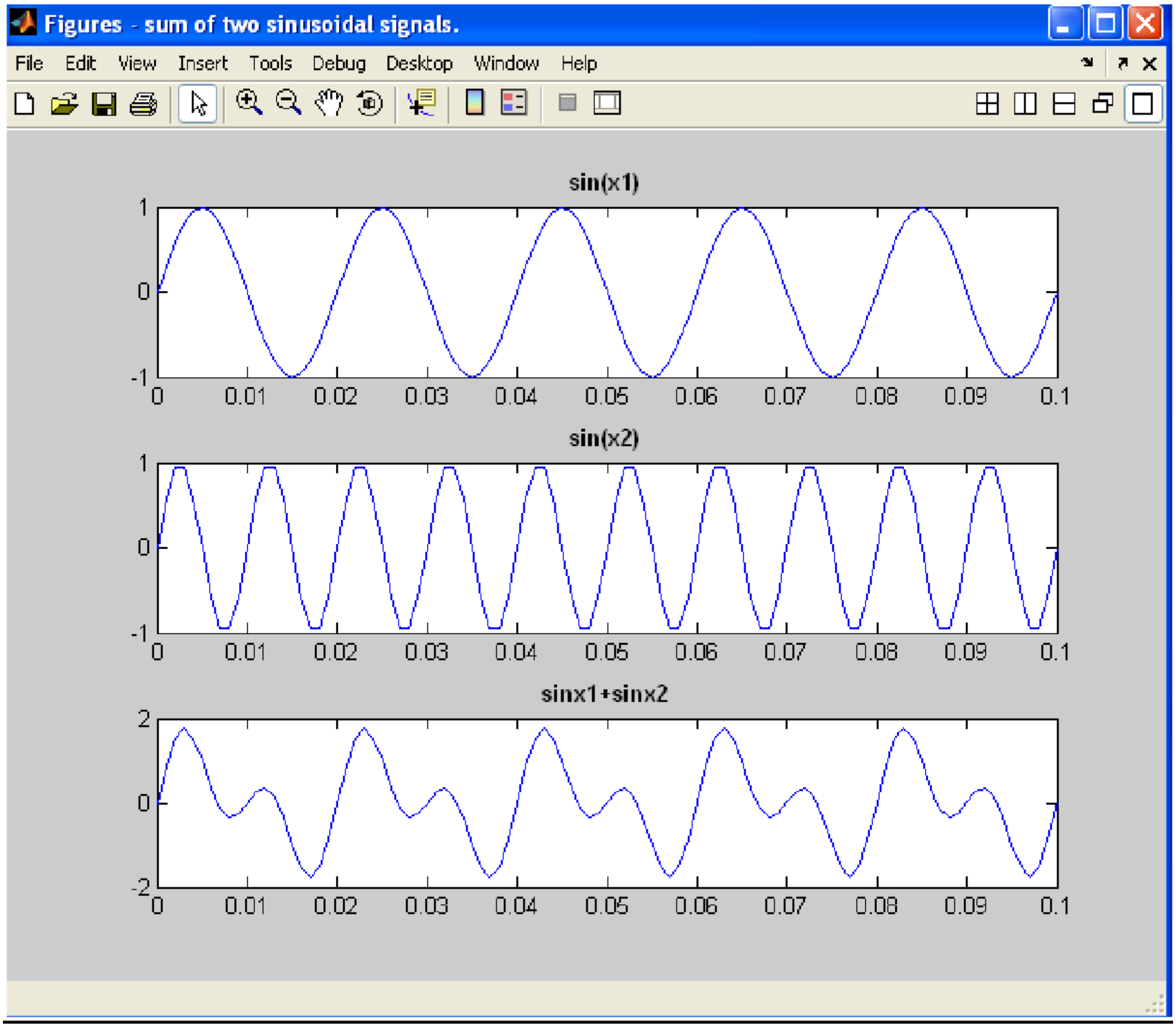
PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

```
clc;
Clear all;
Close all;
t=0:0.001:0.1;
f1=50;
x1=2*pi*f1*t;
y1=sin(x1);
figure;
subplot (3,1,1);
plot (t,y1);
title('sin(x1');
f2=100;
x2=2*pi*f2*t;
y2=sin(x2);
subplot(3,1,2);
plot(t,y2);
title('sin(x2)');
y=y1+y2;
subplot(3,1,3);
plot(t,y);
title('sinx1=sinx2')
```

OUTPUT:



RESULT:

CONCLUSIONS:

VIVA QUESTIONS

1. How do you find sum of sinusoid?

2. How do you add two sinusoidal currents?

3. What is amplitude of sinusoidal function?

4. What is meant by sinusoidal signal?

5. What is phase of sinusoidal function?

Exp .No:03 **Date:**

LINEAR AND CIRCULAR CONVOLUTIONS

AIM: - To write MATLAB programs to find out the linear convolution and Circular convolution of two sequences.

SOFTWARE REQURIED :-

- PC and MATLAB Software (2019b 9.7version)

PROCEDURE:-

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:-

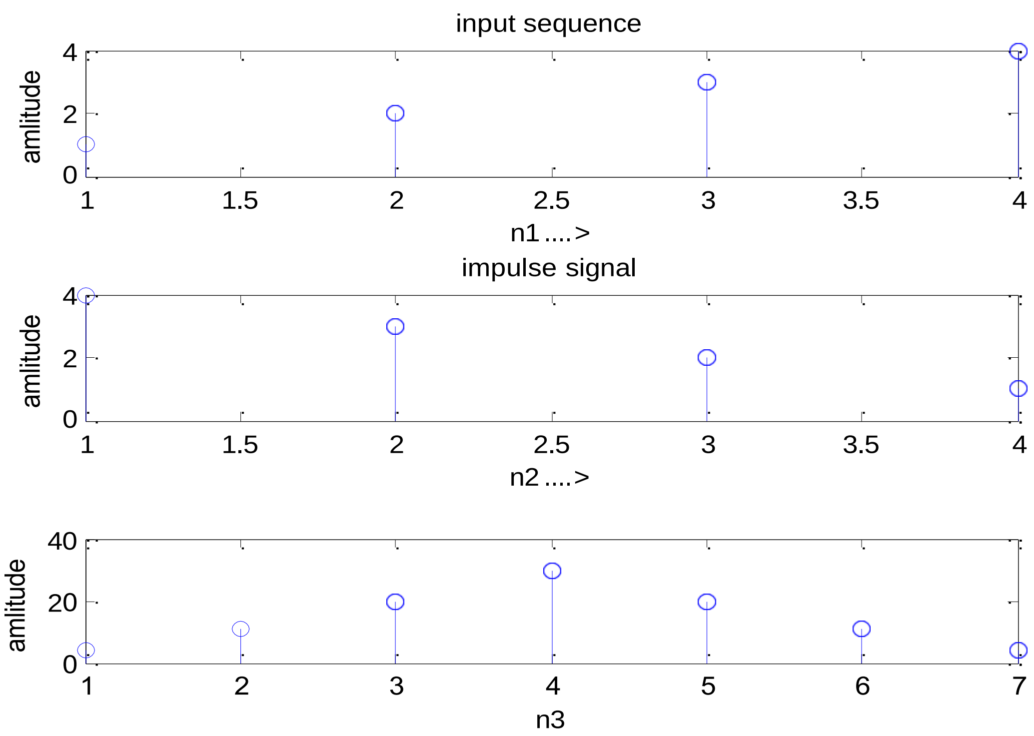
```
%Program for linear convolution
%to get the input sequence
n1=input('enter the length of input sequence');
n2=input('enter the length of impulse sequence');
x=input('enter the input sequence');
h=input('enter the impulse sequence');

%convolution operation
y=conv(x,h);
%to plot the signal
subplot(3,1,1);
stem(x);
ylabel('amplitude');
xlabel('n1... >');
title('input sequence')
subplot(3,1,2);
stem(h);
ylabel('amplitude');
xlabel('n2... >');
title('impulse signal')
subplot(3,1,3);
stem(y);
ylabel('amplitude');
xlabel('n3');
disp('the resultant signal is');y
```

OUTPUT : LINEAR CONVOLUTION

Enter the length of input sequence 4
Enter the length of impulse sequence 4
Enter the input sequence [1 2 3 4]
Enter the impulse sequence [4 3 2 1]

The resultant signal is
y= 4 11 20 30 20 11 4



```
%%Program for circular convolution
clc;
clear all;
close all;
%to get the input sequence
g=input('enter the input sequence');
h=input('enter the impulse sequence');
N1=length(g);
N2=length(h);
N=max(N1,N2);
N3=N1-N2
%loop for getting equal length sequence
if(N3>=0)
h=[h,zeros(1,N3)];
else
g=[g,zeros(1,-N3)];
end
%computation of circular convoluted sequence
for n=1:N;
y(n)=0;
for i=1:N;
j=n-i+1;
if(j<=0)
j=N+j;
end
y(n)=y(n)+g(i)*h(j);
end
end
figure;
subplot(3,1,1);
stem(g);
ylabel('amplitude');
xlabel('n1..>');
title('input sequence')
subplot(3,1,2);
stem(h);
ylabel('amplitude');
xlabel('n2');
title('impulse sequence')
subplot(3,1,3);
stem(y);
ylabel('amplitude');
xlabel('n3');
disp('the resultant signal is');
```

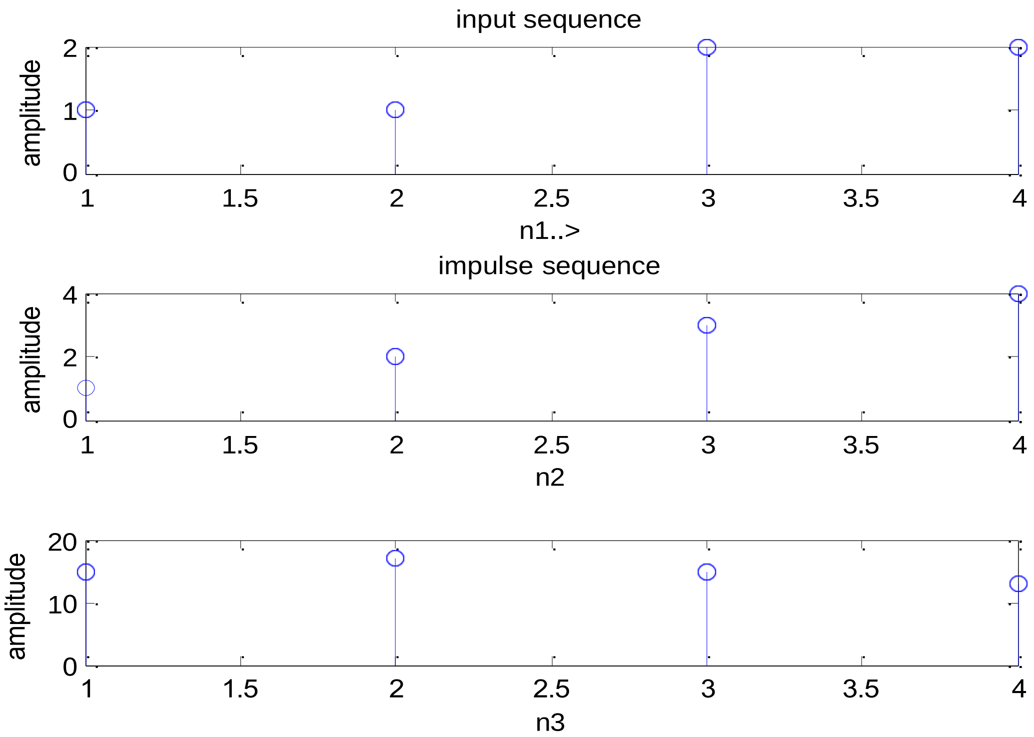
OUTPUT : CIRCULAR CONVOLUTION

Enter the input sequence [1 2 2 1]

Enter the impulse sequence [4 3 2 1]

The resultant signal is

y= 15 17 15 13



RESULT:

CONCLUSIONS:

VIVA QUESTIONS:

1.) Define Convolution?

2.) What are the types of Convolution?

3.) What is Linear Convolution & Circular Convolution?

4.) State the methods available to compute Convolution Sum?

5.) What is the significance of convolution?

Exp .No: 04 **Date:**

AUTO-CORRELATION AND CROSS-CORRELATION

AIM: - To write a Matlab program to compute autocorrelation for the given sequence and cross correlation between two signals.

SOFTWARE REQURIED :-

- PC and MATLAB Software (2019b 9.7version)

.PROCEDURE:

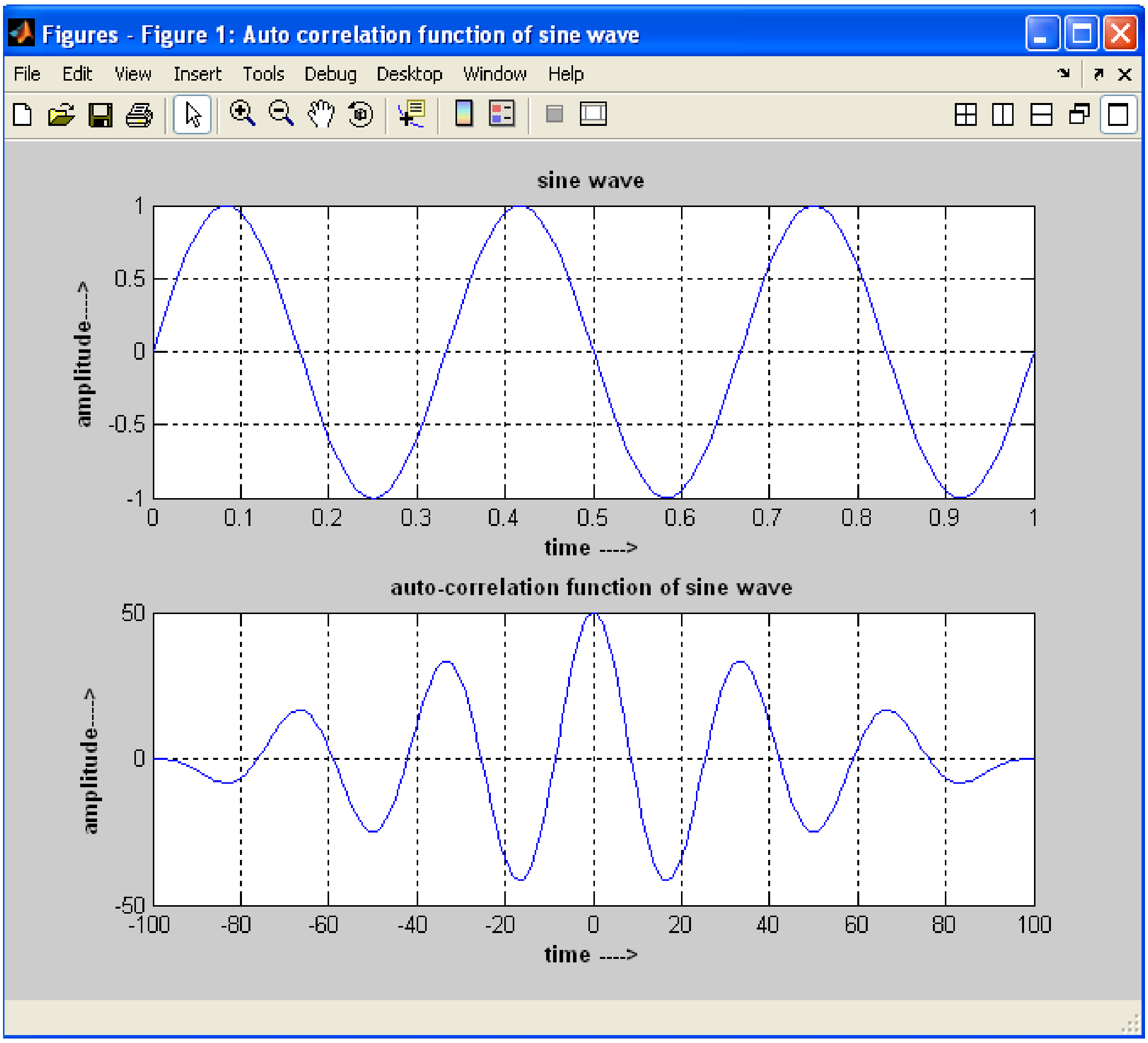
- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

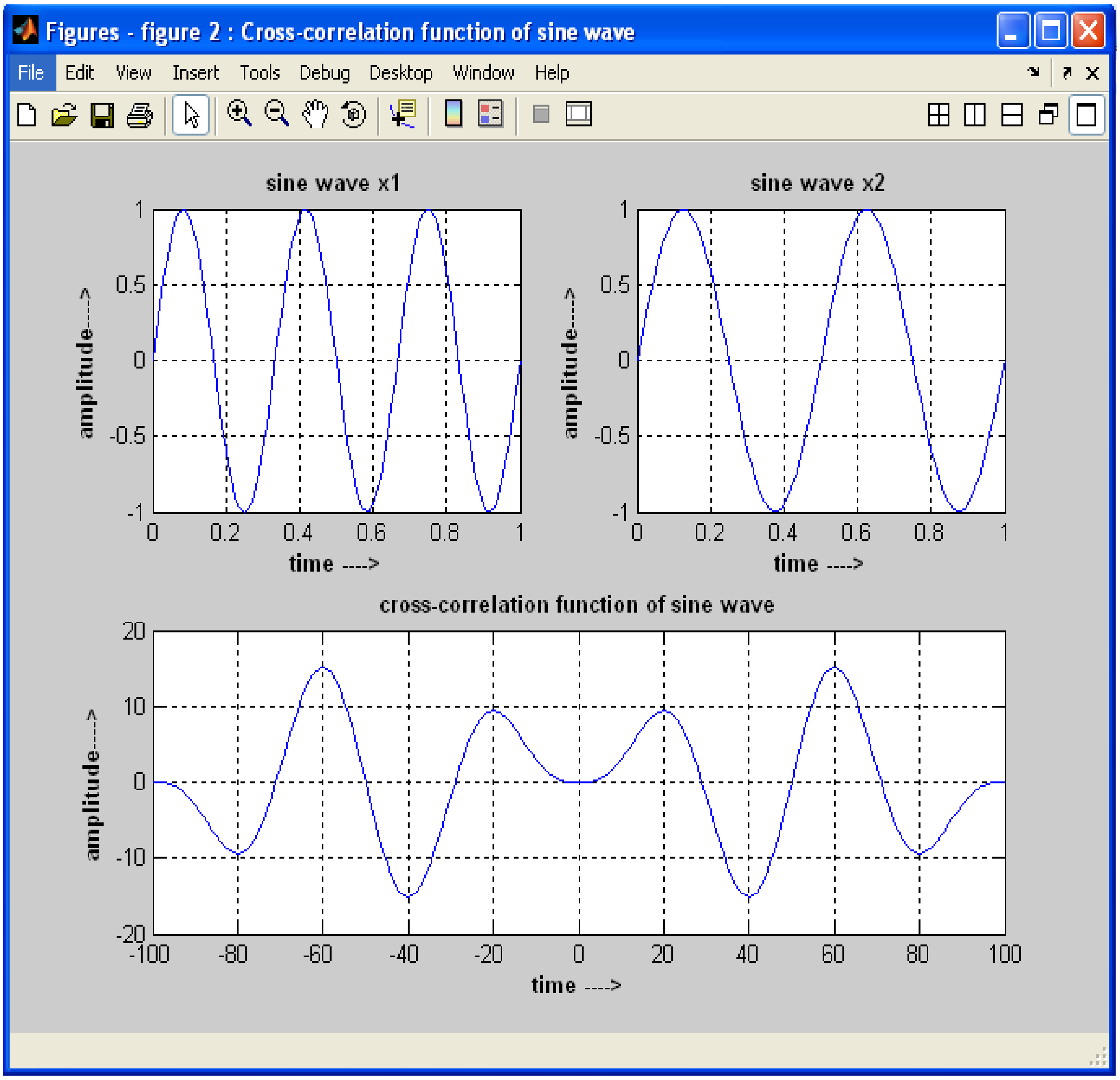
PROGRAM:

```
clc; clear all; close all;
t=0:0.01:1;
f1=3;
x1=sin(2*pi*f1*t);
figure;
subplot(2,1,1);
plot(t,x1);
title('sine wave');
xlabel('time ---->');
 ylabel('amplitude---->');
grid;
[rxx lag1]=xcorr(x1);
subplot(2,1,2);
plot(lag1,rxx);
grid;
title('auto-correlation function of sine wave');
figure;
subplot(2,2,1);
plot(t,x1);
title('sine wave x1');
xlabel('time ---->');
ylabel('amplitude---->');
grid;
 f2=2;
x2=sin(2*pi*f2*t);
subplot(2,2,2);
plot(t,x2);
```

```
title('sine wave x2');
xlabel('time ---->');,ylabel('amplitude---->');
grid;
[cxx lag2]=xcorr(x1,x2);
subplot(2,2,[3,4]);
plot(lag2,cxx);
grid;
title('cross-correlation function of sine wave');
```

OUTPUT:




Figures - figure 2 : Cross-correlation function of sine wave
File Edit View Insert Tools Debug Desktop Window Help
sine wave x1
sine wave x2
amplitude---->
time ---->
cross-correlation function of sine wave
amplitude---->
time ---->

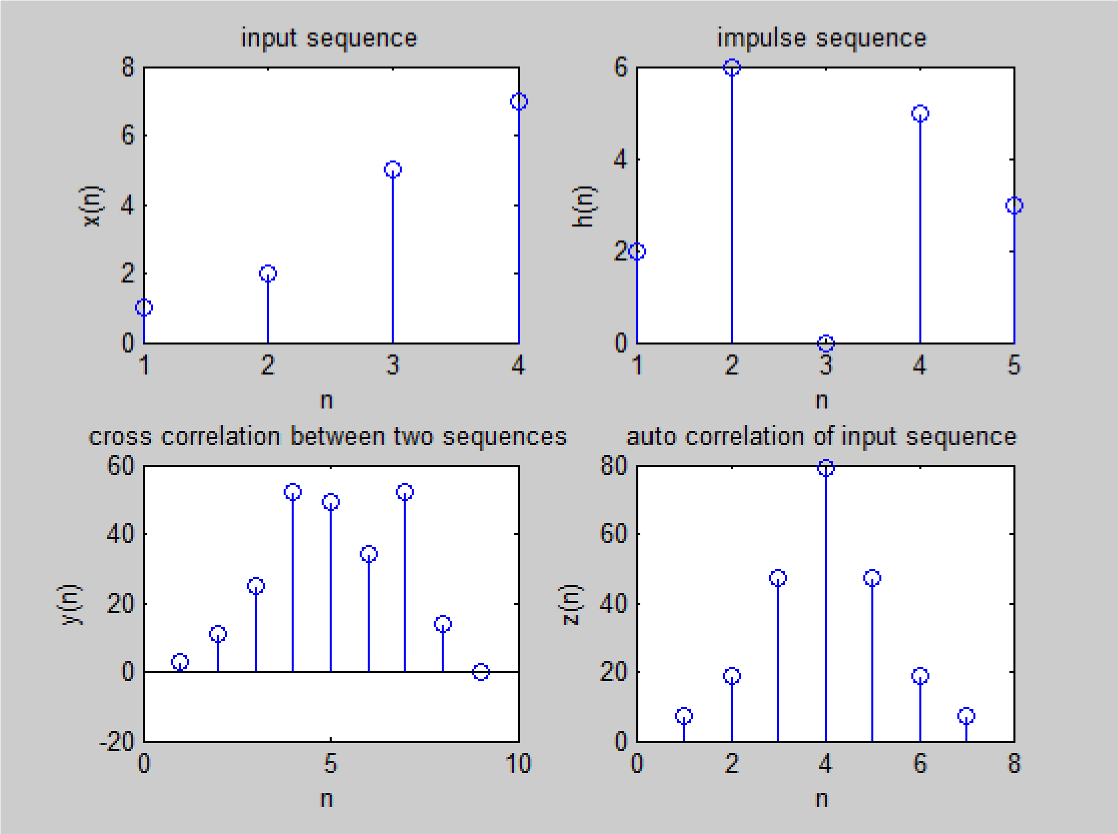
PROGRAM:

```
clc;
close all;
clear all;
% two input sequences
x=input('enter input sequence');
h=input('enter the impulse suquence');
subplot(2,2,1);
stem(x);
xlabel('n');
ylabel('x(n)');
title('input sequence');
subplot(2,2,2);
stem(h);
xlabel('n');
ylabel('h(n)');
title('impulse sequence');
% cross correlation between two sequence
y=xcorr(x,h);
subplot(2,2,3);
stem(y);
xlabel('n');
ylabel('y(n)');
title(' cross correlation between two sequences ');
% auto correlation of input sequence
z=xcorr(x,x);
subplot(2,2,4);
stem(z);
xlabel('n');
ylabel('z(n)');
title('auto correlation of input sequence');
```

INPUT:

enter input sequence [1 2 5 7]
enter the impulse sequence [2 6 0 5 3]

OUTPUT:



RESULT:

CONCLUSIONS:

VIVA QUESTIONS:

1.) Define Cross Correlation?

2.) What are the applications of Cross Correlation in signal de noising?

3.) Define Cross Power Spectral Density?

4.) How Cross Correlation is different Auto Correlation?

5.) Describe the formula to compute Cross Correlation?

Exp .No: 05 **Date:**

DISCRETE FOURIER TRANSFORM

AIM: To find DFT of a given sequence and compute the power density spectrum of the sequence.

SOFTWARE REQUIRED:

- PC and MATLAB Software (2019b 9.7version)

PROCEDURE:

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

THEORY:

Basic equation to find the DFT of a sequence is given below.

$$X(k) = \sum_{n=0}^{N-1} x(n) W_N^{nk}$$

$$where \;\; W_N^{nk} = e^{-j\frac{2\pi nk}{N}} \;\; \text{[TWIDDLE FACTOR]}$$

Basic equation to find the IDFT of a sequence is given below.

$$x_n = \frac{1}{N}\sum_{k=0}^{N-1} X_k e^{\frac{2\pi i}{N}kn} \qquad n = 0, \dots, N-1.$$

4. PROGRAM:

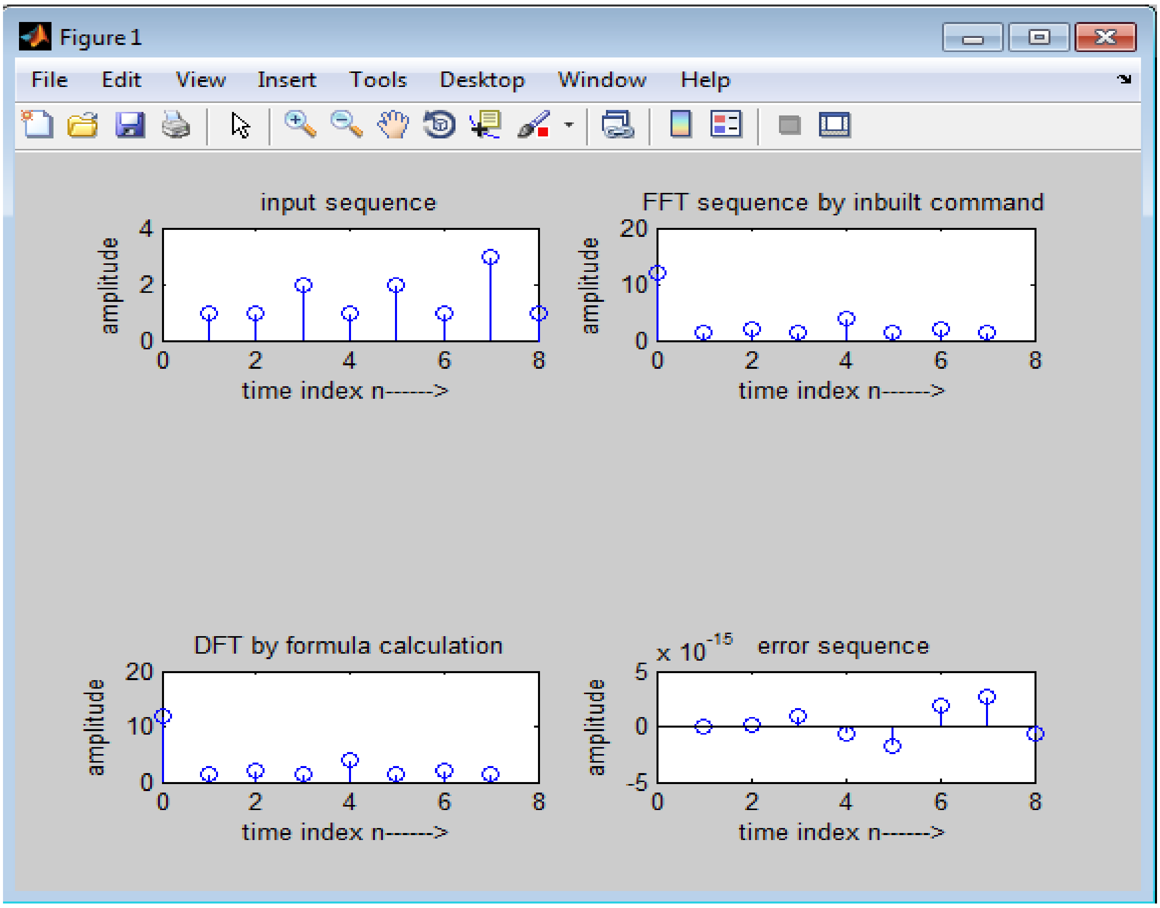
```
clc;
close all;
clear all;
tic;
fprintf('Date & Time:');
Date= datestr(now);
disp(Date);
%DFT
disp('D.F.T');
a=input('Enter the input sequence:');
n=length(a);
x=fft(a,n);
for  k=1:n;
y(k)=0;
for i=1:n
y(k)=y(k)+a(i)*exp((-j)*2*pi*(i-1)*(k-1)*(1/n));
end
end
error=x-y;
disp(x);
disp(y);
disp(error);
subplot(3,2,1);
stem(a);
xlabel('time index n ----->');
ylabel('amplitude');
title('input sequence');
subplot(3,2,2);
stem(0:n-1,abs(x));
xlabel('time index n ----->');
ylabel('amplitude');
title('FFT sequence by inbuilt command');
subplot(3,2,5);
stem(0:n-1,abs(y));
xlabel('time index n ----->');
ylabel('amplitude');
title('DFT by formula calculation');
subplot(3,2,6);
stem(error);
xlabel('time index n ----->');
ylabel('amplitude');
title('error sequence');
% Power Spectral Density
P = x.* conj(x) / 512;
f = 1000*(0:256)/512;
figure,plot(f,P(1:257))
title('Frequency content of y');
xlabel('frequency (Hz)');
```

INPUT:

D.F.T
Enter the input sequence:[1 1 2 1 2 1 3 1]

OUTPUT

Columns 1 through 4
12.0000 -1.0000 + 1.0000i -2.0000 -1.0000 - 1.0000i
Columns 5 through 8
4.0000 -1.0000 + 1.0000i -2.0000 -1.0000 - 1.0000i
Columns 1 through 4
12.0000 -1.0000 + 1.0000i -2.0000 - 0.0000i -1.0000 - 1.0000i
Columns 5 through 8
4.0000 + 0.0000i -1.0000 + 1.0000i -2.0000 - 0.0000i -1.0000 - 1.0000i
1.0e-014 *
Columns 1 through 4
0 0.0444 + 0.0222i 0.0444 + 0.0888i -0.0666 - 0.0666i
Columns 5 through 8
0 - 0.1715i 0.7105 + 0.1887i 0.2665 + 0.2665i -0.1887 - 0.0666i
Elapsed time is 27.683359 seconds



RESULT:

CONCLUSIONS:

VIVA QUESTIONS:

1. What is the difference between DTFT and DFT?

2. Write any Two Properties of DFT?

3. What is zero Padding and Explain the effect of it on magnitude Spectrum?

4. Write the Two properties of Twiddle Factor?

5. How many no. of Complex Multiplications and Additions are required to compute N-Point DFT?

Exp .No: 06 **Date:**

FFT USING Decimation-In-Time (DIT) ALGORITHM

1. AIM: Write a MATLAB program to perform N-point DIT- FFT of a given signal and to plot its magnitude and phase spectrum.

2. SOFTWARE REQUIRED:

- PC and MATLAB Software (2019b 9.7version)

3. PROCEDURE:

- Open MATLAB
- Open new M-file
- Type the program
- .Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window\

PROGRAM:

```
clc;
close all;
clear all;
tic;
fprintf('Date & Time:');
Date= Datestr(now);
disp(Date);
fprintf('DIT-FFT');
fprintf('\n\n');
N=input('Enter the length of the input sequence:');
for i=1:N
re(i)=input('Enter the real part of the time domain sequence:');
im(i)=input('Enter the imaginary part of the time domain sequence:');
end
[re1,im1]=DITFFT(re,im,N);
subplot(3,2,1);
stem(0:N-1,re);
xlabel('time index n ----->');
ylabel('amplitude');
title('input real sequence');
subplot(3,2,2);
stem(0:N-1,im);
xlabel('time index n ----->');
ylabel('amplitude');
title('input imaginary sequence');
subplot(3,2,5);
stem(0:N-1,re1);
xlabel('frequency------>');
```

```
ylabel('amplitude');
title('output real sequence');
subplot(3,2,6);
stem(0:N-1,im1);
xlabel('frequency------>');
ylabel('amplitude');
title('output imaginary sequence');
disp(re1);
disp(im1);
```

DIT-FFT

```
Enter the length of the input sequence:8
Enter the real part of the time domain sequence:1
Enter the imaginary part of the time domain sequence:2
Enter the real part of the time domain sequence:1
Enter the imaginary part of the time domain sequence:3
Enter the real part of the time domain sequence:1
Enter the imaginary part of the time domain sequence:2
Enter the real part of the time domain sequence:2
Enter the imaginary part of the time domain sequence:1
Enter the real part of the time domain sequence:2
Enter the imaginary part of the time domain sequence:1
Enter the real part of the time domain sequence:3
Enter the imaginary part of the time domain sequence:1
Enter the real part of the time domain sequence:2
Enter the imaginary part of the time domain sequence:1
Enter the real part of the time domain sequence:3
Enter the imaginary part of the time domain sequence:1
15.0000 0.7071 2.0000 0.1213 -3.0000 -0.7071 -2.0000 -4.1213
12.0000 5.5355 1.0000 0.7071 0 -1.5355 -1.0000 -0.7071
Elapsed time is 23.565254 seconds.
```

6. RESULT:

7. CONCLUSION:

VIVA QUESTIONS:

1. Write the Block diagram of 8-Point DIT FFT & DIF DFFT radix 2 Algorithm?

2. Explain using convolution the effects of taking an FFT of a sample with no windowing

3. What's the difference between FFT and DFT?

4. Why do we need Fourier transform in DSP?

5. What is "decimation-in-time" versus "decimation-in-frequency"?

Exp .No: 07 **Date:**

INVERSE FAST FOURIER TRANSFORM (IFFT)

AIM: Write a MATLAB program to perform N-point IFFT of a given sequence.

SOFTWARE REQUIRED:

- PC and MATLAB Software (2019b 9.7version)

PROCEDURE:

- Open MATLAB
- Open new M-file
- Type the program
- .Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window\

4. PROGRAM:

```
clc;
clear all;
close all;
n=input('enter value of n=');
x=input('enter input sequence=');
a=1:1:n;
y=fft(x,n);
disp('fft of input sequence');
disp(y);
z=ifft(y);
disp('ifft of input sequence');
disp(z);
```

5. OUTPUT:

```
enter value of n=8
enter input sequence=[1 2 3 4 5 6 7 8]
fft of input sequence
 Columns 1 through 6
 36.0000 + 0.0000i  -4.0000 + 9.6569i  -4.0000 + 4.0000i  -4.0000 + 1.6569i  -4.0000 + 0.0000i  -
4.0000 - 1.6569i
 Columns 7 through 8
 -4.0000 - 4.0000i  -4.0000 - 9.6569i
ifft of input sequence
     1     2     3     4     5     6     7     8
```

RESULT:

CONCLUSION:

VIVA QUESTIONS:

1.) What is the need of FFT?

2.) Write the Applications of FFT?

3.) FFT is in complex domain how to use it in real life signals optimally?

4.) What is the difference between FFT and IFFT?

5.) How many no. of Complex Multiplications and Additions are required to compute N-Point DFT using FFT?

Exp .No: 08 **Date:**

DESIGN OF IIR BUTTERWORTH (LP/HP) FILTERS

AIM: To write a MATLAB program to design Butterworth IIR LP\HP filters.

SOFTWARE REQUIRED:

- PC and MATLAB Software (2019b 9.7version)

PROCEDURE:

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

PROGRAM:

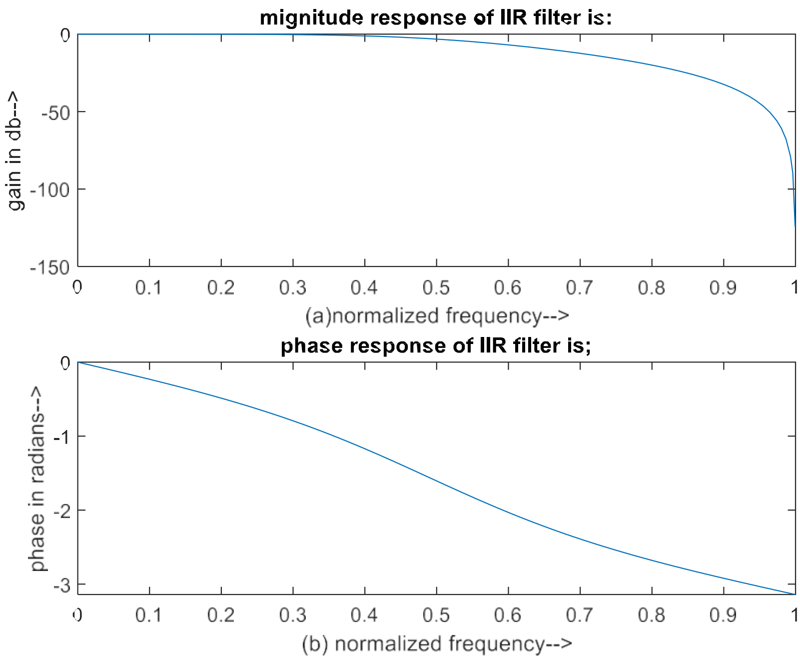
```
clc;
clear all;
close all;
display('enter the iir filter design specifications');
        rp=input('enter the pass band ripple:');
        rs=input('enter the stop band ripple:');
        wp=input('enter the pass band freq:');
ws=input('enter the stop band freq:');
fs=input('enter the sampling freq:');
w1=2*wp/fs;
w2=2*ws/fs;
[n,wn]=buttord(w1,w2,rp,rs);
c=input('enter choice filter 1.lpf 2.hpf /n');
if(c==1)
display('frequency response of IIR lpf is:');
[b,a]=butter(n,wn,'low');
end
if(c==2)
display('freq response of IIR hpf IS:');
[b,a]=butter(n,wn,'high');
end
w=0:0.01:pi;
h=freqz(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure;
subplot(2,1,1);
plot(w/pi,m);
title('mignitude response of IIR filter is:');
xlabel('(a)normalized frequency-->');
ylabel('gain in db-->');
subplot(2,1,2);
plot(w/pi,an);
titel('phase response of IIR filter is;');
```

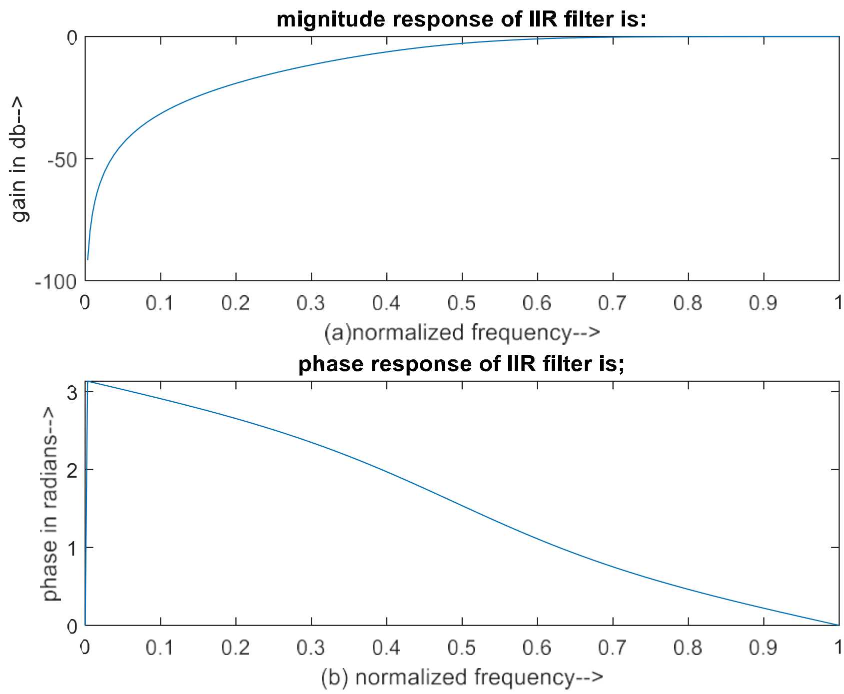
```
xlabel('(b) normalized frequency-->');
ylabel('phase in radians-->');
```

INPUT:

enter the iir filter design specifications
enter the pass band ripple:2
enter the stop band ripple:20
enter the pass band freq:1000
enter the stop band freq:2000
enter the sampling freq:5000
enter choice filter 1.lpf 2.hpf /n

5. OUTPUT:





RESULT:

CONCLUSION:

VIVA QUESTIONS:

1.) What do you mean by cut-off frequency?

2.) Give the differences between analog and digital filters?

3.) What is the difference between type 1 and type 2 filter structures?

4.) What is the role of delay element in filter design?

5.) List the Differences between Butterworth and chebyshev filters?

Exp .No: 09 **Date:**

DESIGN OF IIR CHEBYSHEV (LP/HP) FILTERS

AIM: To write a MATLAB program to design Chebyshev IIR LP\HP filters.

SOFTWARE REQUIRED:

- PC and MATLAB Software (2019b 9.7version)

PROCEDURE:

- Open MATLAB
- Open new M-file
- Type the program
- Save in current directory
- Compile and Run the program
- For the output see command window\ Figure window

4. PROGRAM:

% Program for the design of Chebyshev Type-1 low-pass filter

```
clc;
close all;clear all;
format long
rp=input(„enter the passband ripple...‟);
rs=input(„enter the stopband ripple...‟);
wp=input(„enter the passband freq...‟);
ws=input(„enter the stopband freq...‟);
fs=input(„enter the sampling freq...‟);
w1=2*wp/fs;w2=2*ws/fs;
[n,wn]=cheb1ord(w1,w2,rp,rs,‟s‟);
[b,a]=cheby1(n,rp,wn,‟s‟);
W=0:.01:pi;
[h,om]=freqs(b,a,w);
M=20*log10(abs(h));
An=angle(h);
subplot(2,1,1);
plot(om/pi,m);
ylabel(„Gain in dB --.‟);
xlabel(„(a) Normalised frequency --.‟);
subplot(2,1,2);
plot(om/pi,an);
xlabel(„(b) Normalised frequency --.‟);
ylabel(„Phase in radians --.‟);
```

% Program for the design of Chebyshev Type-2 High pass analog filter

```
clc;
close all;clear all;
format long
rp=input('enter the passband ripple...');
rs=input('enter the stopband ripple...');
wp=input('enter the passband freq...');
ws=input('enter the stopband freq...');
```

```
fs=input('enter the sampling freq...');
w1=2*wp/fs;
w2=2*ws/fs;
[n,wn]=cheb2ord(w1,w2,rp,rs,'s');
[b,a]=cheby2(n,rs,wn,'high','s');
w=0:.01:pi;
[h,om]=freqs(b,a,w);
m=20*log10(abs(h));
an=angle(h);
subplot(2,1,1);
plot(om/pi,m);
ylabel('Gain in dB --.');
xlabel('(a) Normalised frequency --.');
subplot(2,1,2);
plot(om/pi,an);
xlabel('(b) Normalised frequency --.');
ylabel('Phase in radians --.');
```

INPUT:

LPF

enter the passband ripple... 0.23

enter the stopband ripple... 47

enter the passband freq... 1300

enter the stopband freq... 1550

enter the sampling freq... 7800

HPF

enter the passband ripple... 0.34

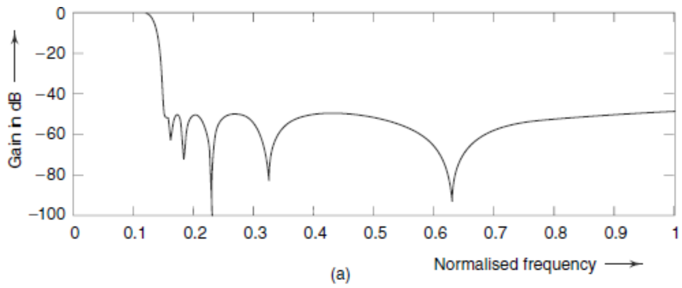
enter the stopband ripple... 34

enter the passband freq... 1400

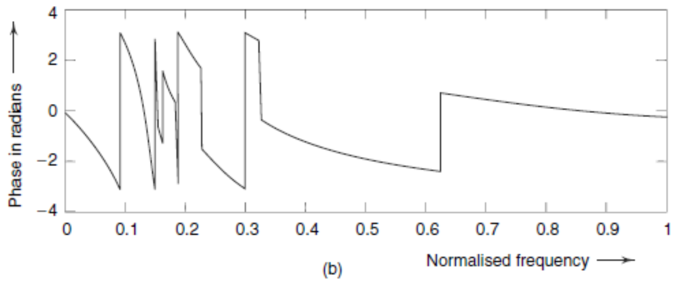
enter the stopband freq... 1600

enter the sampling freq... 10000

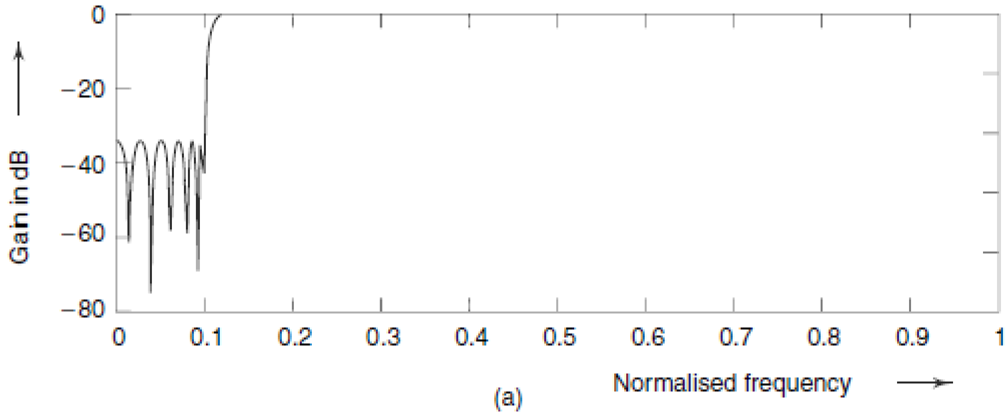
OUTPUT:

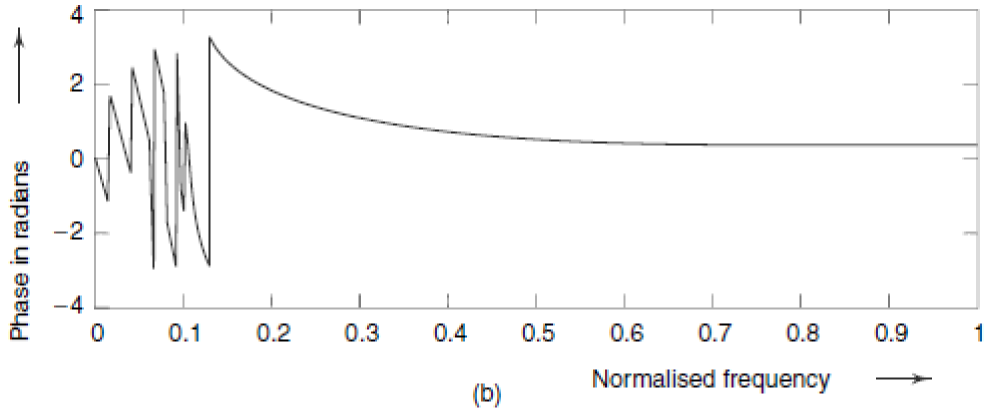


(a)



(b)





Chebyshev Type - 2 High-pass Analog Filter
(a) Amplitude Response and (b) Phase Response

RESULT:

CONCLUSION:

VIVA QUESTIONS:

1.) What is the difference between type 1 and type 2 filter structures?

2.) IIR Filters are non linear phase filters. Why?

3) IIR filters are not always stable. Why?

4.) Write the realization Techniques for IIR Filter?

5.) Compare all realization techniques vailable for IIR Filter?

Exp .No: 10 **Date:**

DESIGN OF FIR (LP/HP) USING WINDOWING TECHNIQUE

AIM: To write a MATLAB program to design FIR with Low pass filter and High Pass filter using any three Windowing techniques.

SOFTWARE REQUIRED:

- PC and MATLAB Software (2019b 9.7version) **PROCEDURE:**

  - Open MATLAB
  - Open new M-file
  - Type the program
  - Save in current directory
  - Compile and Run the program
  - For the output see command window\ Figure window

PROGRAM:

```
%FIR Low Pass/High pass filter design using Rectangular/Hamming/Kaiser window
clc; clear all; close all;
rp=input('enter passband ripple');
rs=input('enter the stopband ripple');
wp=input('enter passband freq');
ws=input('enter stopband freq');
fs=input('enter sampling freq ');
beta=input('enter beta value');
w1=2*wp/fs;
w2=2*ws/fs;
num=-20*log10(sqrt(rp*rs))-13;
dem=14.6*(ws-wp)/fs;
n=ceil(num/dem);
n1=n+1;
if(rem(n,2)~=0)
n1=n; n=n-1;
end
c=input('enter your choice of window function 1. rectangular
2. Hamming 3.kaiser: \n ');
if(c==1)
y=rectwin(n1);
disp('Rectangular window filter response');
end
if (c==2)
y=hamming(n1);
disp('Hamming window filter response');
end
if(c==3)
y=kaiser(n1,beta);
disp('kaiser window filter response');
```

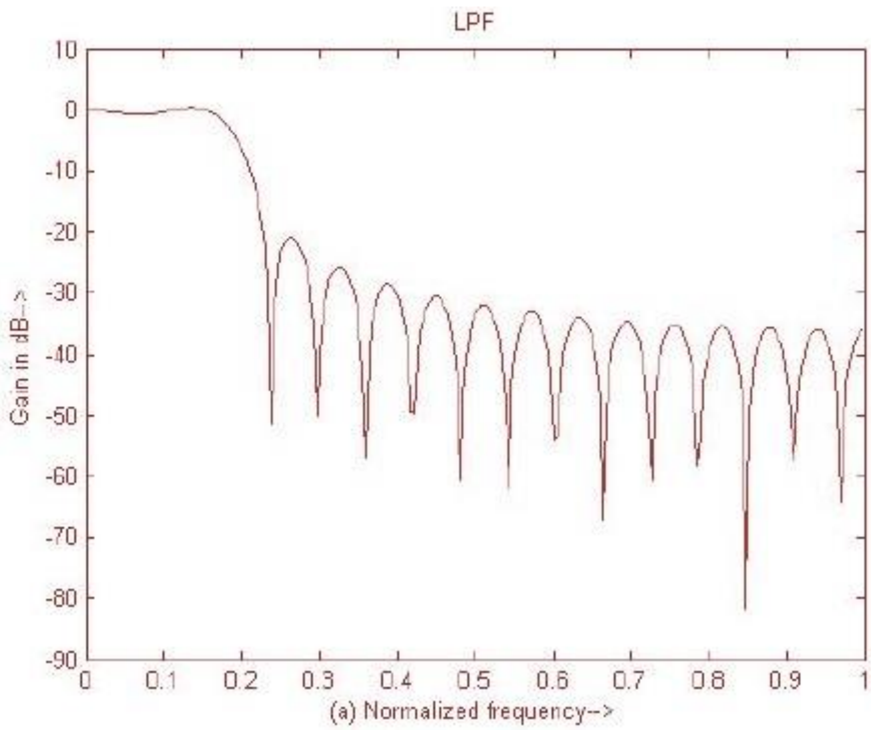
```
end
ch=input('give type of filter 1:LPF,2:HPF');
switch ch
case 1
b=fir1(n,w1,y);
[h,o]=freqz(b,1,256);
m=20*log10(abs(h));
plot(o/pi,m);
title('LPF');
xlabel('(a) Normalized frequency-->');
ylabel('Gain in dB-->');
case 2
b=fir1(n,w1,'high',y);
[h,o]=freqz(b,1,256);
m=20*log10(abs(h));
plot(o/pi,m);
title('HPF');
xlabel('(b) Normalized frequency-->');
ylabel('Gain in dB-->');
end
```

OUTPUT:

enter passband ripple 0.02
enter the stopband ripple
enter passband freq 1000
enter stopband freq 1500
enter sampling freq 10000
enter beta value 5
enter your choice of window function 1. rectangular 2.Hamming 3.kaiser:1

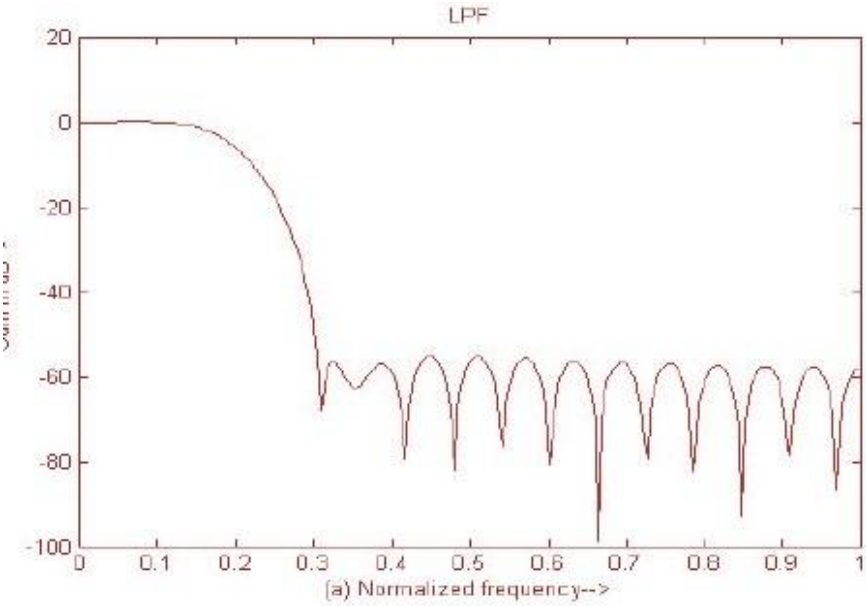
Rectangular window filter response
give type of filter 1:LPF,2:HPF
1:LPF

Low passfilter using Rectangular Window



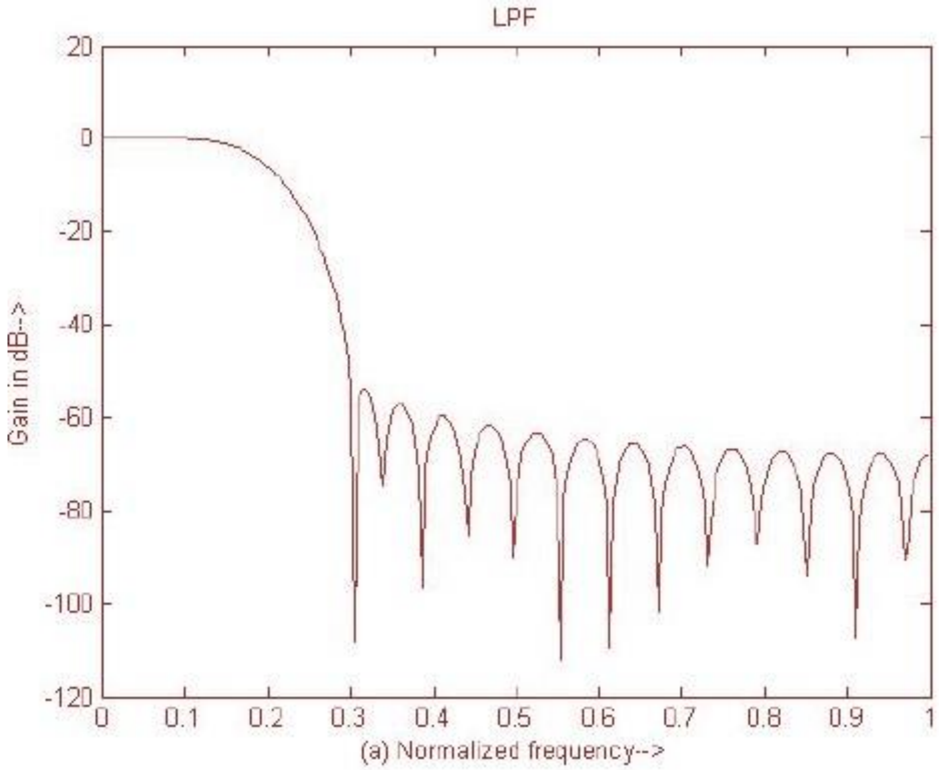
enter your choice of window function 1. rectangular 2. Hamming 3.kaiser:
2
Hamming window filter response
give type of filter 1:LPF,2:HPF
1:LPF

Low pass FIR filter using Hamming window



enter your choice of window function 1. rectangular 2. Hamming 3.kaiser:
3
kaiser window filter response
give type of filter 1:LPF,2:HPF
1:LPF

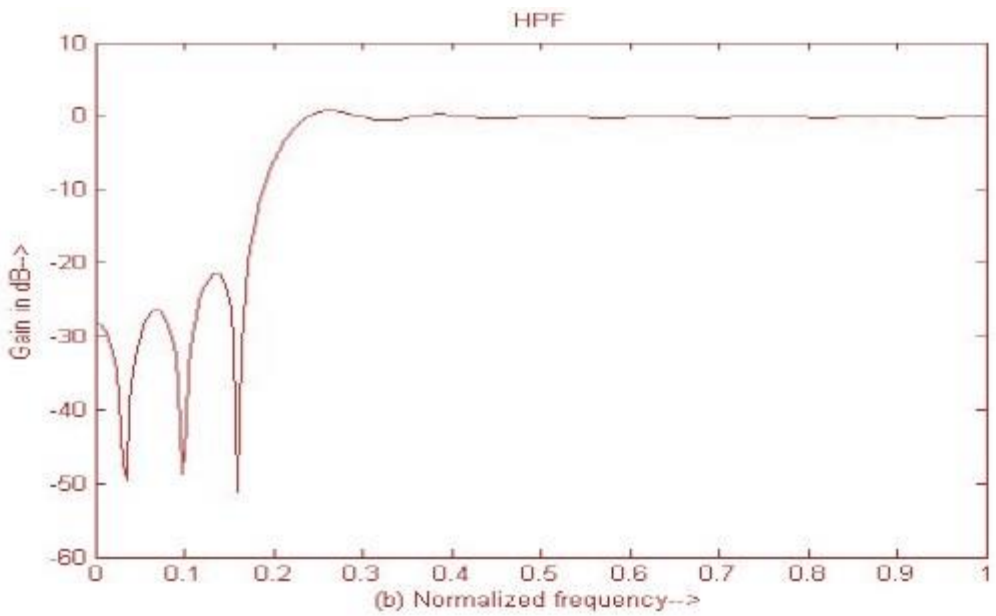
Low pass FIR filter using Kaiser Window



FIR High pass Filter design

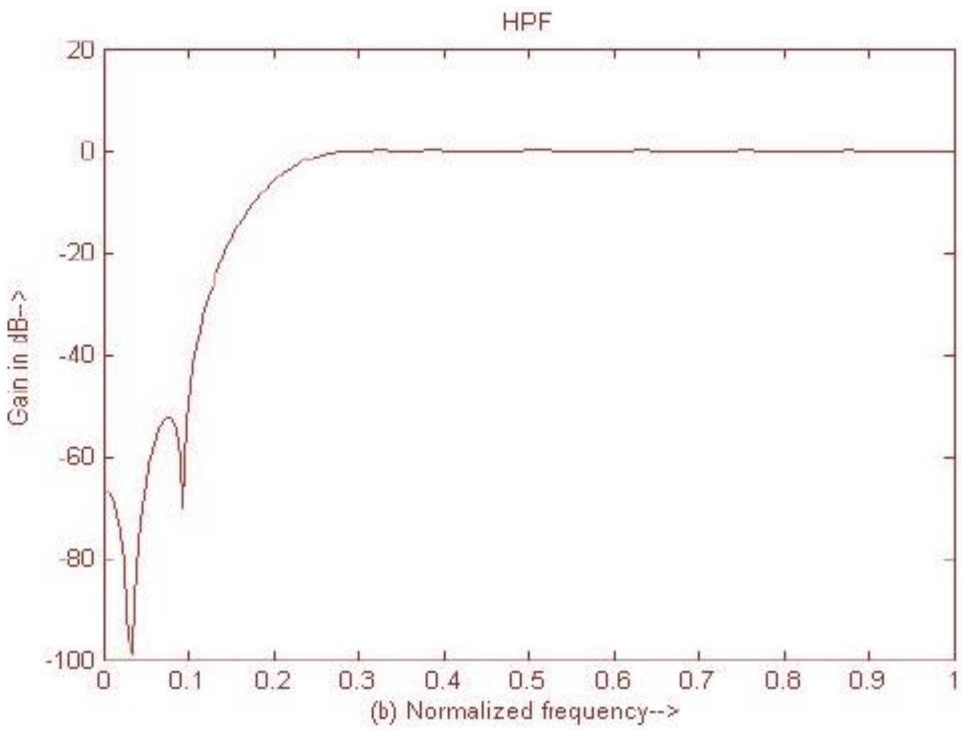
enter passband ripple
enter the stopband ripple
enter passband freq 1000
enter stopband freq 1500
enter sampling freq 10000
enter beta value 5
enter your choice of window function 1. rectangular 2.
Hamming 3.kaiser:
1
Rectangular window filter response
give type of filter 1:LPF,2:HPF
2:HPF

High pass FIR filter using Rectangular Window



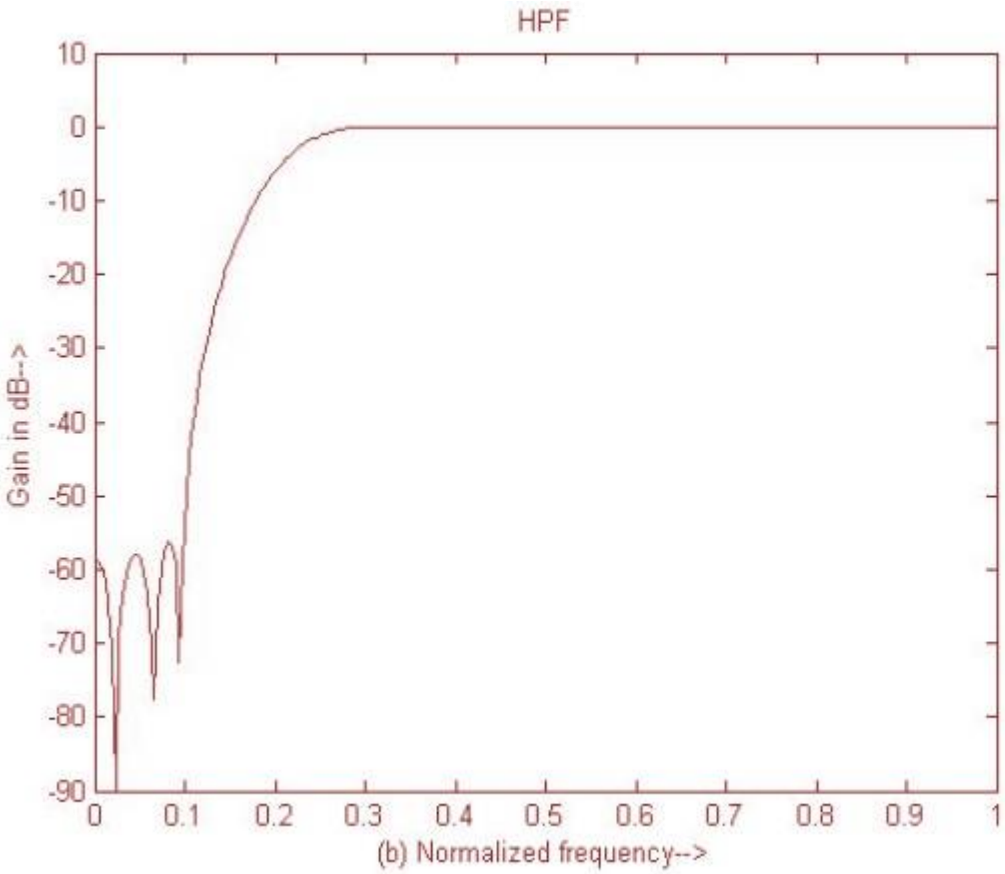
enter your choice of window function 1. rectangular 2. Hamming 3.kaiser:
2
Hamming window filter response
give type of filter 1:LPF,2:HPF
2:HPF

High pass FIR filter using Hamming Window



enter your choice of window function 1. rectangular 2. Hamming 3.kaiser:
3
kaiser window filter response
give type of filter 1:LPF,2:HPF
2: HPF

High pass FIR filter using Kaiser Window



RESULT:

CONCLUSIONS:

VIVA QUESTIONS:

1. What is filter?

2. What is FIR and IIR filter define, and distinguish between these two?

3. What is window method? How you will design an FIR filter using window method?

4. What are low-pass and band-pass filter and what is the difference between these two?

5. What is the matlab command for Hamming window? Explain.