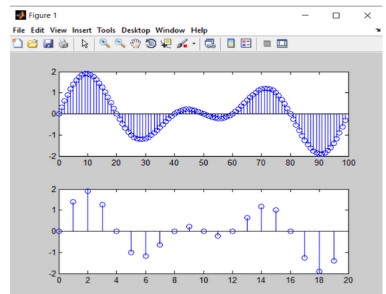
Digital Signal Processing Lab Excersie

Implement the decimation process by factor of D=5 in the following equation:

x=sin(2*pi*f1*n)+sin(2*pi*f2*n);

Program:




```
D=input('enter the downsampling factor');
L=input('enter the length of the input signal');
f1=input('enter the frequency of first sinusodal');
f2=input('enter the frequency of second
sinusodal');
n=0:L-1;
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);
y=decimate(x,D,'fir');
subplot(2,1,1);
stem(n,x(1:L));
subplot(2,1,2)
m=0:(L/D)-1;
stem(m,y(1:L/D));
```

Input:

F1=0.02, f2=0.03, n=100, D=5

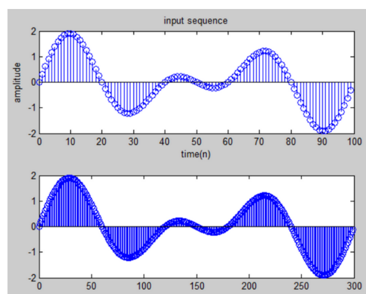
```
D=input('enter the down sampling factor');

L=input('enter the length of the input signal');

f1=input('enter the frequency of first sinusodal');

f2=input('enter the frequency of second sinusodal');

n=0:L-1;

x=cos(2*pi*30*n);

y=decimate(x,D,'fir');

subplot(2,1,1);

stem(n,x(1:L));
```

```
subplot(2,1,2)

m=0:(L/D)-1;

stem(m,y(1:L/D));
```

Q2. Implement the upsampling process by factor of L=3 in the following equation:

x=sin(2*pi*f1*n)+sin(2*pi*f2*n);

```
L=input('enter the upsampling factor');
N=input('enter the length of the input signal'); % Length should be greater than 8
f1=input('enter the frequency of first sinusodal');
f2=input('enter the frequency of second sinusodal');
n=0:N-1;
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);
y=interp(x,L);
subplot(2,1,1)
stem(n,x(1:N))
title('input sequence');
xlabel('time(n)');
ylabel('amplitude');
subplot(2,1,2)
m=0:N*L-1;
stem(m,y(1:N*L));
```



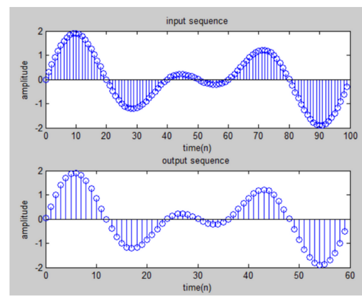
Input:

F1=0.02, f2=0.03, n=100, L=3

Q3. Implement of I/D sampling rate conversion in the following equation:

x=sin(2*pi*f1*n)+sin(2*pi*f2*n);

```
L=input('enter the upsampling factor');
D=input('enter the downsampling factor');
N=input('enter the length of the input signal');
f1=input('enter the frequency of first sinusodal');
f2=input('enter the frequency of second sinusodal');
n=0:N-1;
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);
y=resample(x,L,D);
subplot(2,1,1)
stem(n,x(1:N))
title('input sequence');
xlabel('time(n)');
ylabel('amplitude');
subplot(2,1,2)
m=0:N*L/D-1;
stem(m,y(1:N*L/D));
title('output sequence ');
xlabel('time(n)');
ylabel('amplitude');
```

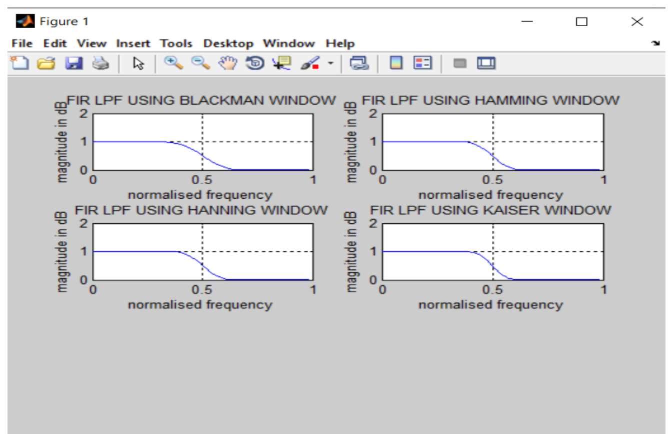


Input:

F1=0.02, f2=0.03, n=100, L=3, D=5

Q4. To Write a Matlab program of FIR Low pass filter using Hanning, Hamming, Blackman and Kaiser window. The signal cutoff frequency is 0.5 Hz and the number of samples are 25.

```
wc=.5*pi;
N=25;
w=0:0.1:pi;
b=fir1(N,wc/pi,blackman(N+1));
h=freqz(b,1,w);
subplot(3,2,1)
plot(w/pi,abs(h))
grid;
xlabel('normalised frequency');
ylabel('magnitude in dB')
title('FIR LPF USING BLACKMAN WINDOW')
b=fir1(N,wc/pi,hamming(N+1));
h=freqz(b,1,w);
subplot(3,2,2)
plot(w/pi,abs(h));
grid;
xlabel('normalised frequency');
ylabel('magnitude in dB')
title('FIR LPF USING HAMMING WINDOW')
b=fir1(N,wc/pi,hanning(N+1));
h=freqz(b,1,w);
subplot(3,2,3)
plot(w/pi,abs(h));
grid;
xlabel('normalised frequency');
ylabel('magnitude in dB')
title('FIR LPF USING HANNING WINDOW')
b=fir1(N,wc/pi,kaiser(N+1,3.5));
h=freqz(b,1,w);
subplot(3,2,4)
plot(w/pi,abs(h));
grid;
xlabel('normalised frequency');
ylabel('magnitude in dB')
title('FIR LPF USING KAISER WINDOW');
```



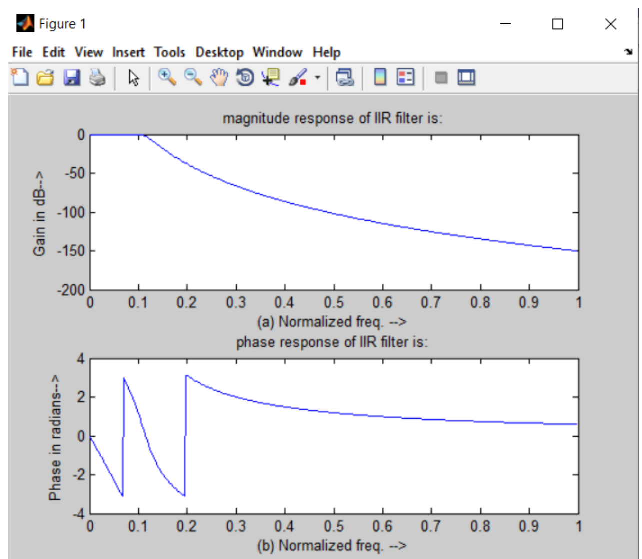
5. Write a Matlab program to implement LP IIR filter for the following given parameters.

The passband ripple rp:1
The stopband ripple rs:50
The passband freq wp:0.5
The stopband freq ws:0.7

Show the magnitude response and phase response of IIR filter.

PROGRAM:

```
clc;
clear all;
close all;
disp('enter the IIR filter design specifications');
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]=buttord(w1,w2,rp,rs,'s');
disp('Frequency response of IIR LPF is:');
[b,a]=butter(n,wn,'low','s');
w=0:.01:pi;
[h,om]=freqs(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure,subplot(2,1,1);plot(om/pi,m);
title('magnitude response of IIR filter is:');
xlabel('(a) Normalized freq. -->');
ylabel('Gain in dB-->');
subplot(2,1,2);
plot(om/pi,an);
title('phase response of IIR filter is:');
xlabel('(b) Normalized freq. -->');
ylabel('Phase in radians-->');
```



INPUT:
enter the IIR filter design specifications
enter the passband ripple rp:1

enter the stopband ripple rs:50
enter the passband freq wp:0.5
enter the stopband freq ws:0.7

01.12.2021

Question for lab exam

Set 1:

1. Write a matlab program to implement the decimation process by factor of D=5 in the following equation:

x=sin(2*pi*f1*n)+sin(2*pi*f2*n);

taking the input from keyboard of f1=0.02, f2=0.03 and n=100.

2. To Write a Matlab program of FIR Low pass filter using Hamming, and Kaiser window.

Tanking the input from keyboard of wc=.5*pi; N=25;

Set 2:

1. Write a matlab program to implement the upsampling by factor of L=3 in the following equation:

x=sin(2*pi*f1*n)+sin(2*pi*f2*n);

taking the input from keyboard of f1=0.02, f2=0.03 and n=100.

2. To Write a Matlab program of FIR Low pass filter using Blackman, Hanning window.

Tanking the input from keyboard of wc=.5*pi; N=25;

Set 3:

1. Write a matlab program to implement the I/D sampling rate conversion process by factor of L=3 and D=5 in the following equation:

x=sin(2*pi*f1*n)+sin(2*pi*f2*n);

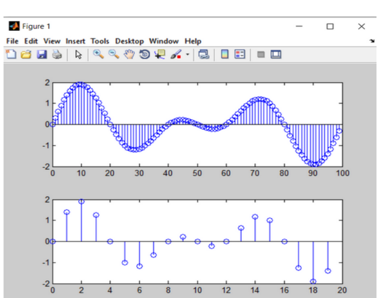
taking the input from keyboard of f1=0.02, f2=0.03 and n=100.

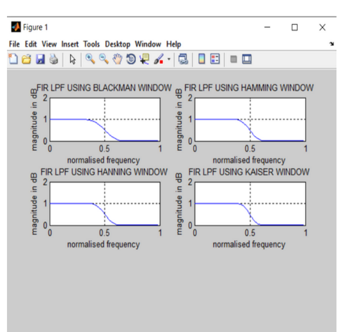
2. To Write a Matlab program of FIR Low pass filter using Hanning, and Kaiser window.

Tanking the input from keyboard of wc=.5*pi; N=25;

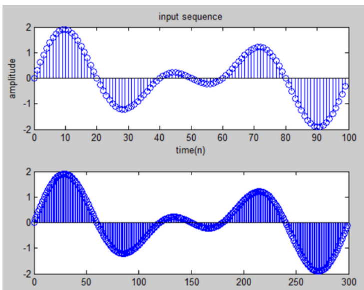
Answer:

Set1:





Set2:



input sequence
amplitude
time(n)

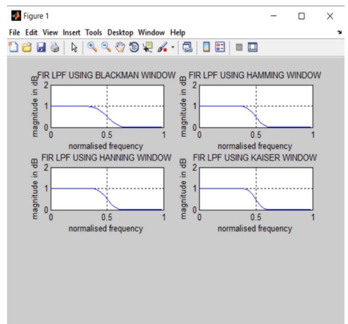
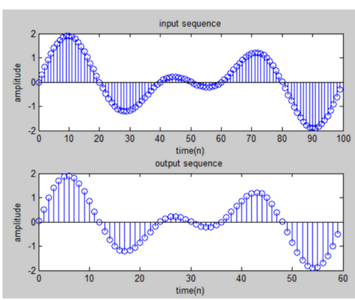


Figure 1
File Edit View Insert Tools Desktop Window Help
FIR LPF USING BLACKMAN WINDOW
FIR LPF USING HAMMING WINDOW
FIR LPF USING HANNING WINDOW
FIR LPF USING KAISER WINDOW
magnitude in dB
normalised frequency

Set3:



input sequence
amplitude
time(n)
output sequence

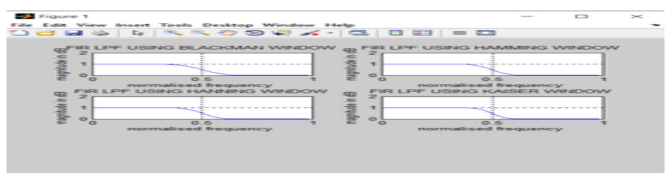


Figure 1
FIR LPF USING BLACKMAN WINDOW
FIR LPF USING HAMMING WINDOW
FIR LPF USING HANNING WINDOW
FIR LPF USING KAISER WINDOW
normalised frequency