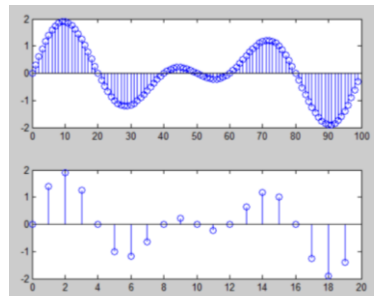
Digital Signal Processing Lab Excersie

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

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

<u>Program:</u>

```
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

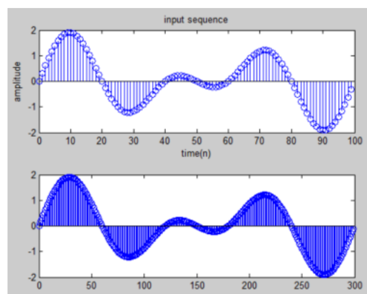
```
D=input('enter the down sampling factor');

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));
```



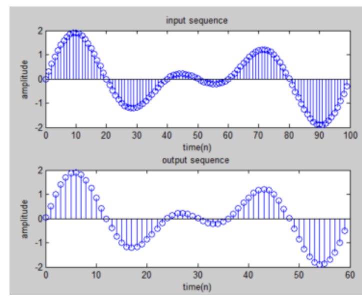
Input:

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');
```

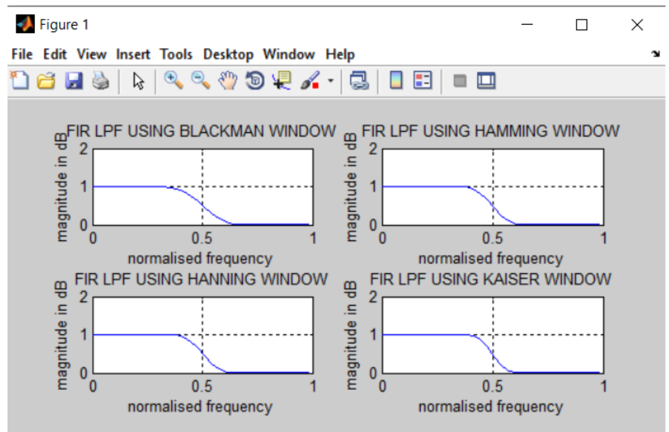


Input:

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');
```



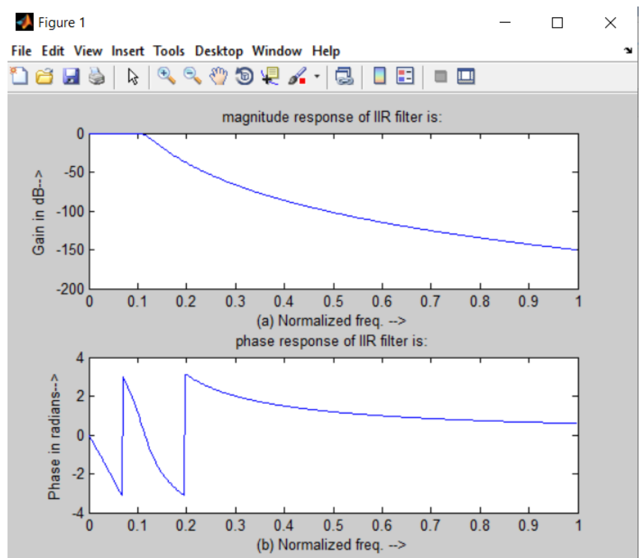
5. Write a Matlab program to implement LP IIR filter for the following given parameters.

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:

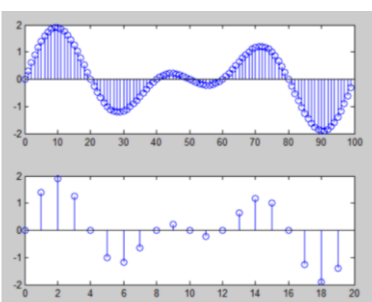
x=sin(2*pi*f1*n)+sin(2*pi*f2*n);

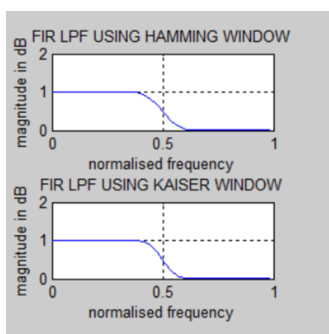
taking the input from keyboard of f1=0.02, f2=0.03 and n=100.

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

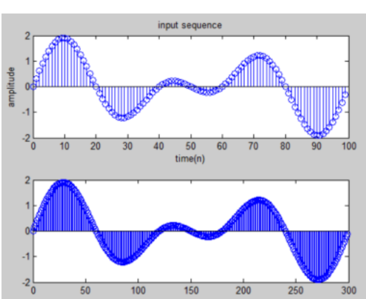
Tanking the input from keyboard of wc=.5*pi; N=25;
Answer:

Set1:

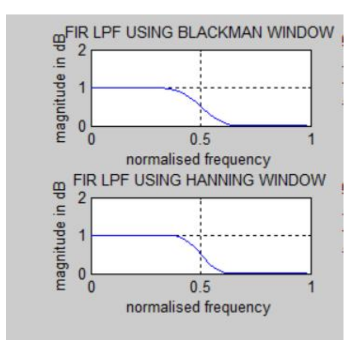


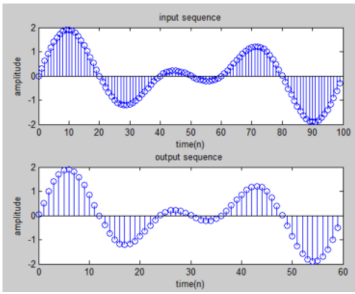


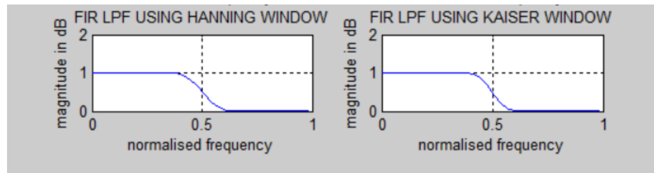
Set2:



Set3:




input sequence
amplitude
time(n)
output sequence
amplitude
time(n)


FIR LPF USING HANNING WINDOW
magnitude in dB
normalised frequency
FIR LPF USING KAISER WINDOW
magnitude in dB
normalised frequency