Question 1: Mohammed Khaleel(21EE01027)

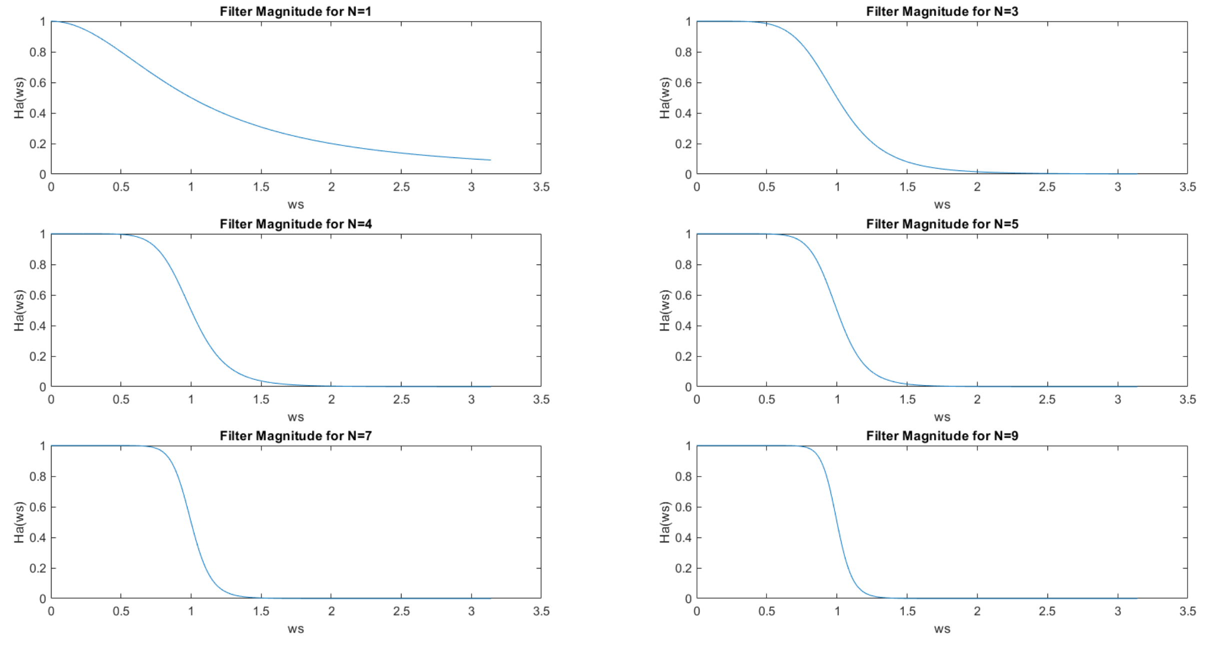
Q.1. a) Implement the equation of Analog Butterworth Low Pass Filter Approximation and observe the effect of order N on the magnitude response.

Part A

```
syms N
ws = 0:0.01:pi;
Ha(N) = 1./(1+ws.^(2*N))
```

Magnitude Plots

```
N_subs = [1,3,4,5,7,9];

for i=1:6
    Ha_subs = subs(Ha,N,N_subs(i));
    subplot(3,2,i)
    plot(ws,abs(Ha_subs))
    ylabel('Ha(ws)')
    xlabel('ws')
    title(['Filter Magnitude for N=',num2str(N_subs(i))])
end
```



Part B:

b) Plot the poles of the filter for different values of order N.

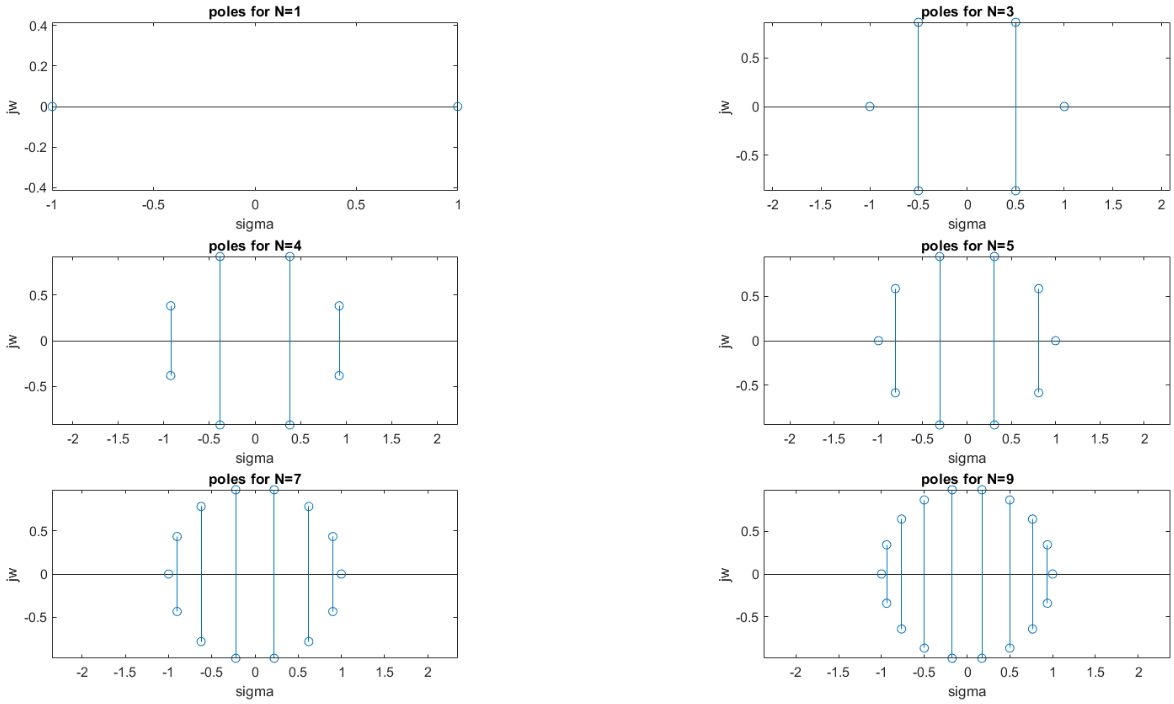
```
syms k
poles = 1j*exp(1j*(2*k+1)*pi/(2*N))
```

poles =

$$e^{\frac{\pi\,(2\,k\,\mathrm{i}+\mathrm{i})}{2\,N}}\,\mathrm{i}$$

Plotting poles

```
for i=1:6
    poles_N_subs = subs(poles,N,N_subs(i));

    k_sub = 0:2*N_subs(i)-1;
    poles_k_subs = subs(poles_N_subs,k,k_sub);

    subplot(3,2,i)
    stem(real(poles_k_subs),imag(poles_k_subs))
    axis equal;
    ylabel('jw')
    xlabel('sigma')
    title(['poles for N=',num2str(N_subs(i))])
end
```



Part c:

c) Plot the curves for the Bilinear Transformation and it's inverse so as to compensate the frequency warping effect.

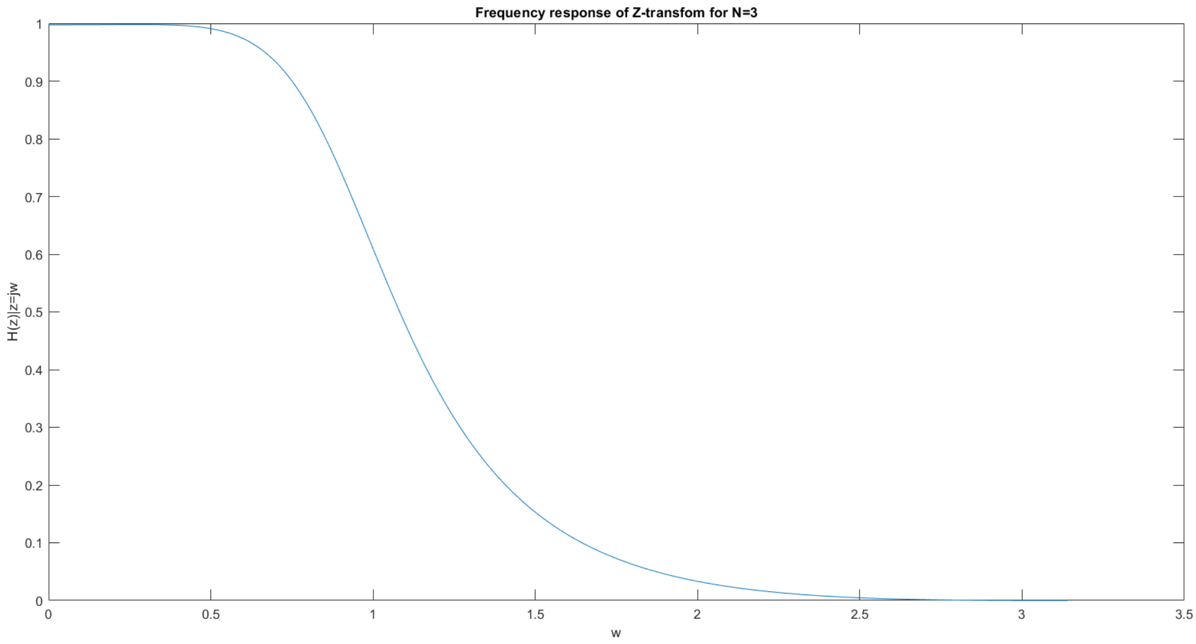
Applying bilinear transform

```
[numd,dend] = bilinear(1,[1,2,2,1],1)
```

```
numd = 1×4
    0.0476    0.1429    0.1429    0.0476
dend = 1×4
    1.0000   -1.1905    0.7143   -0.1429
```

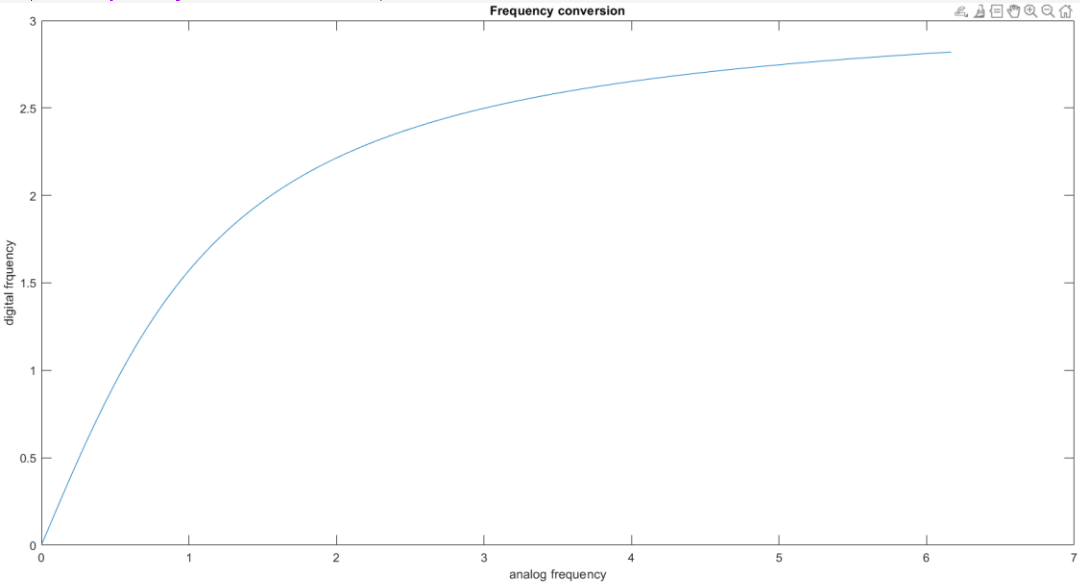
Frequency response of z-transform

```
w1 = 0:0.01:pi;
numerator = 0.0476 + 0.1429 * exp(-1i*w1) + 0.1429 * exp(-2i*w1) + 0.0476 *
exp(-3i*w1);
denominator = 1 - 1.19 * exp(-1i*w1) + 0.7143 * exp(-2i*w1) - 0.1423 * exp(-
3i*w1);
H = numerator ./ denominator;
clf
plot(w1,abs(H))
ylabel('H(z)|z=jw')
xlabel('w')
title('Frequency response of Z-transfom for N=3')
```



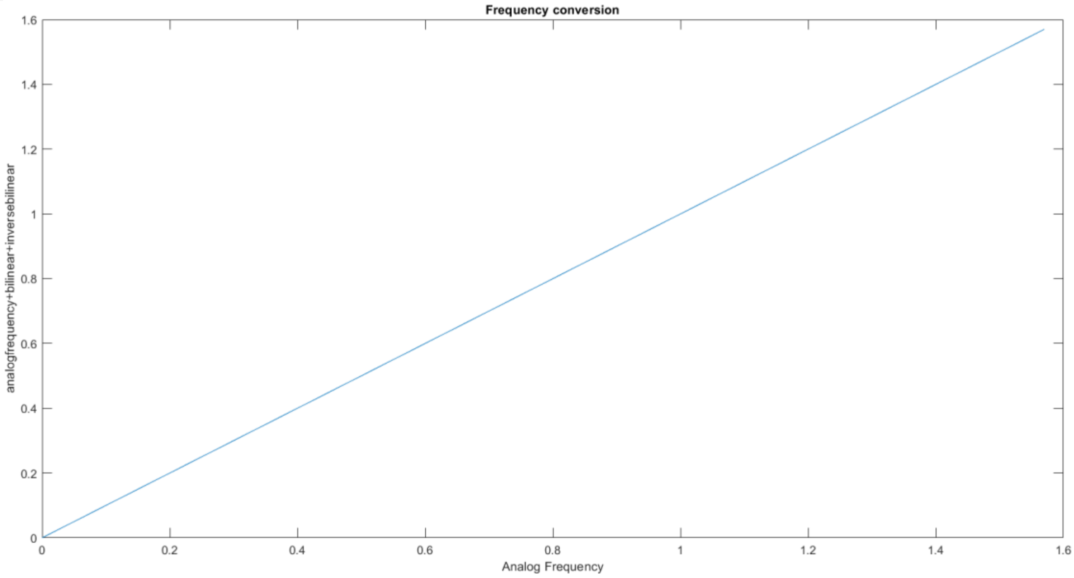
Relation between digital and analog frequency

```
digital_frquency = 0:0.01:pi*0.9;
analog_frequency = tan(digital_frquency/2);

clf
plot(analog_frequency,digital_frquency)
xlabel('analog_frequency')
ylabel('digital_frquency')
title('Frequency conversion')
```



Relationship

```
w_analog = 0:0.01:pi/2;
w_bilinear = 2*atan(w_analog);
w_bilinear_inverseBilinear = tan(w_bilinear/2);

plot(w_analog,w_bilinear_inverseBilinear)
xlabel('Analog Frequency')
ylabel('analogfrequency+bilinear+inversebilinear')
title('Frequency conversion')
```



Question 2: Mohammed Khaleel(21EE01027)

Q.2. a) Design and realize the IIR LP filter with the specifications as discussed in the class.

Defining IIR low pass

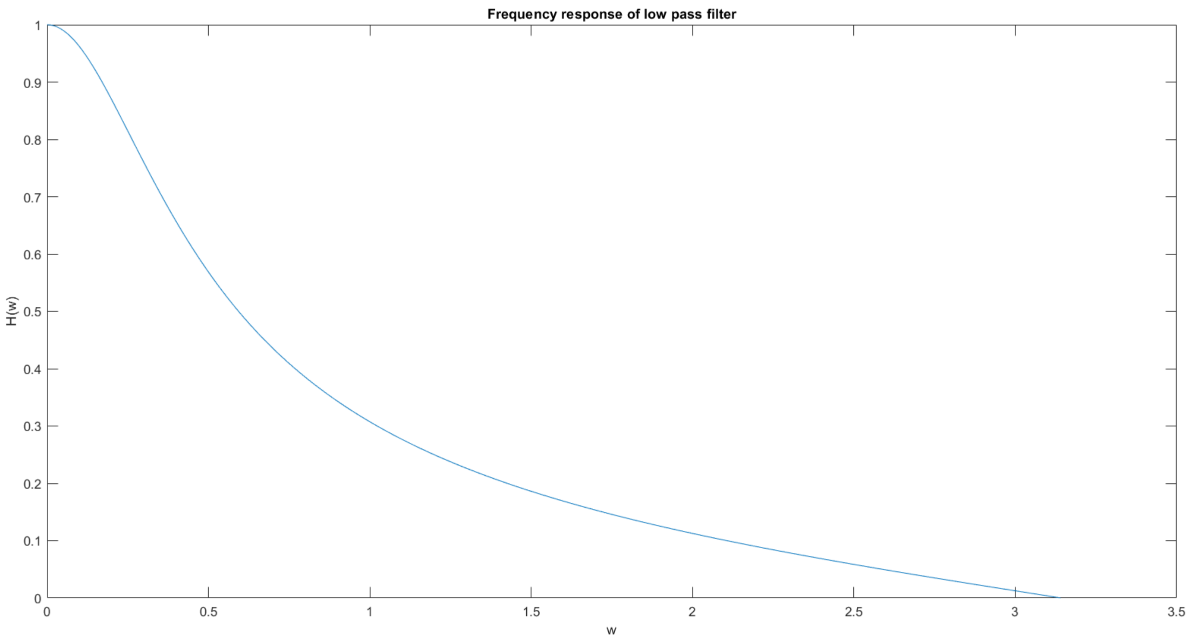
```
syms z
alpha = 0.7;
Hz_low = (1+z^-1)/(1-alpha*z^-1)*0.5*(1-alpha)
```

Hz_low =

$$-\frac{3\left(\frac{1}{z}+1\right)}{20\left(\frac{7}{10\,z}-1\right)}$$

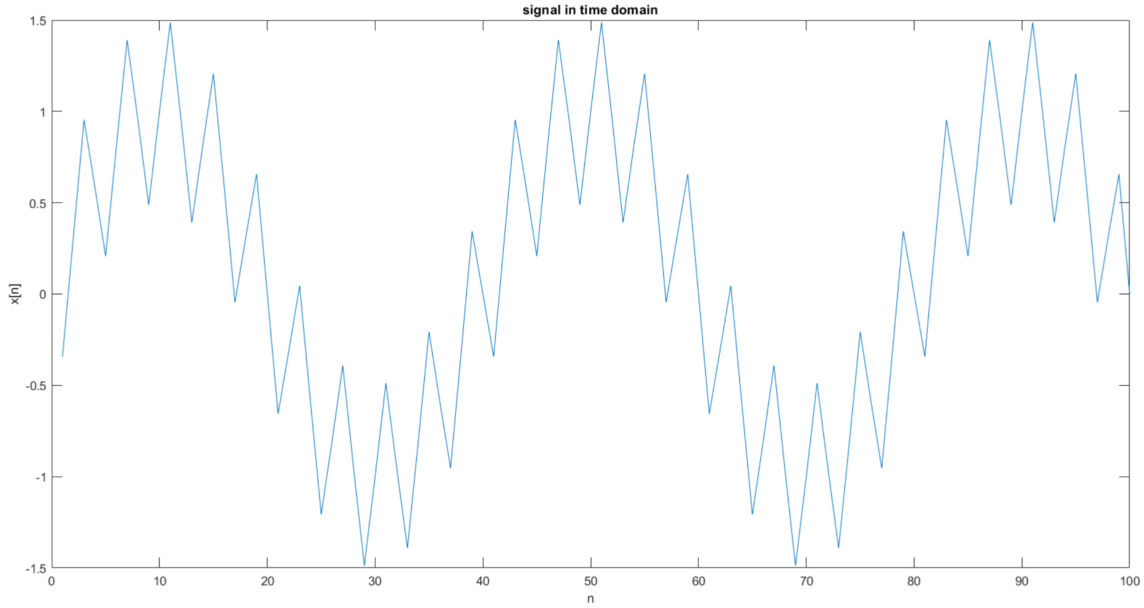
```
w_subs = 0:0.01:pi;
z_subs = exp(1j*w_subs);
```

```
Hw = subs(Hz_low,z,z_subs);
plot(w_subs,abs(Hw))
ylabel('H(w)')
xlabel('w')
title('Frequency response of low pass filter')
```

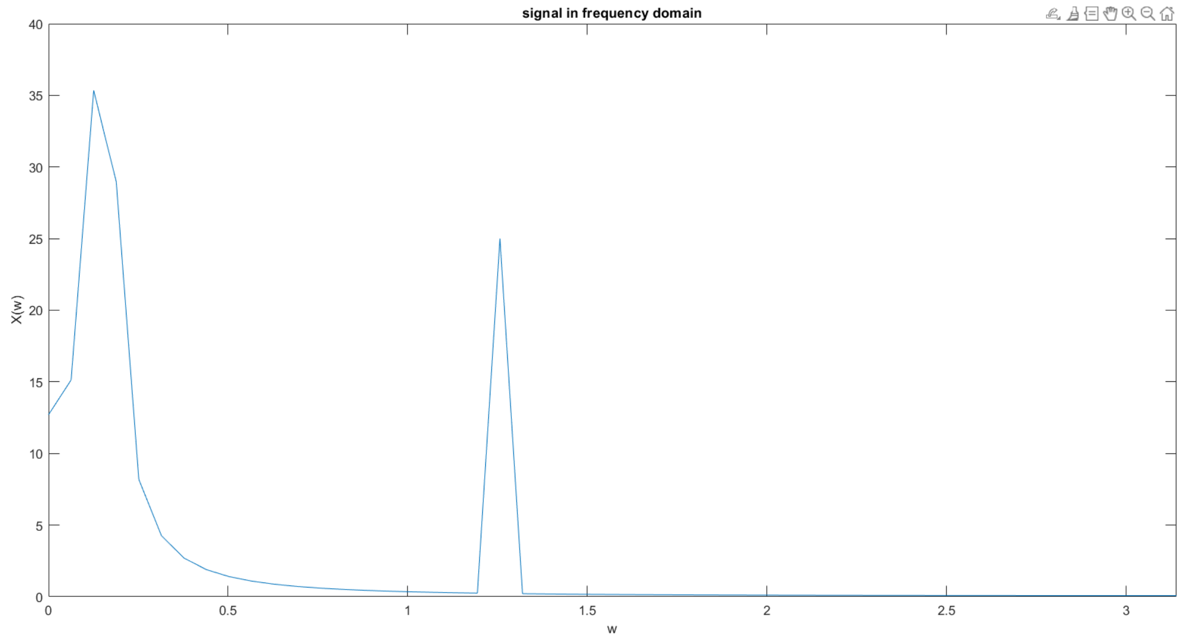


Making signal

```
n = 1:100;
test_signal = sin(pi*0.05*n) + 0.5*sin(pi*0.4*n); % Test signal (sum of
sinusoids)

plot(n,test_signal)
title('signal in time domain')
ylabel('x[n]')
xlabel('n')
```

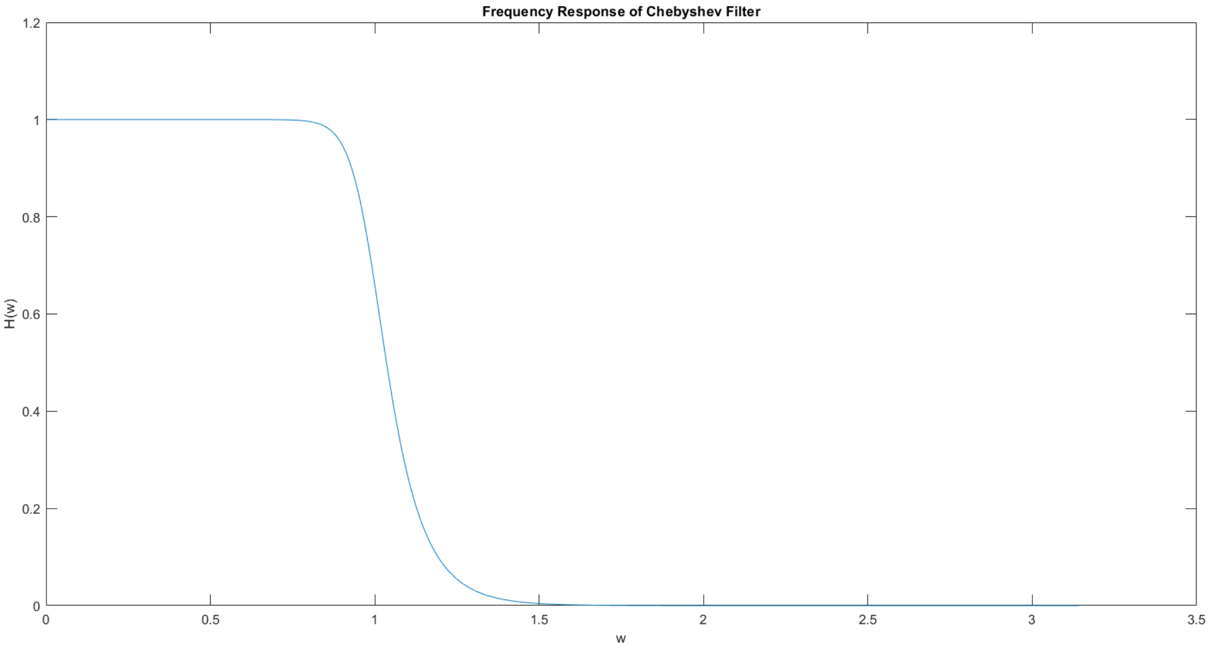



```
freq_response = fft(test_signal);
x_axis = 0:length(freq_response)-1;
plot(x_axis*2*pi/length(test_signal),abs(freq_response));
title('signal in frequency domain')
```

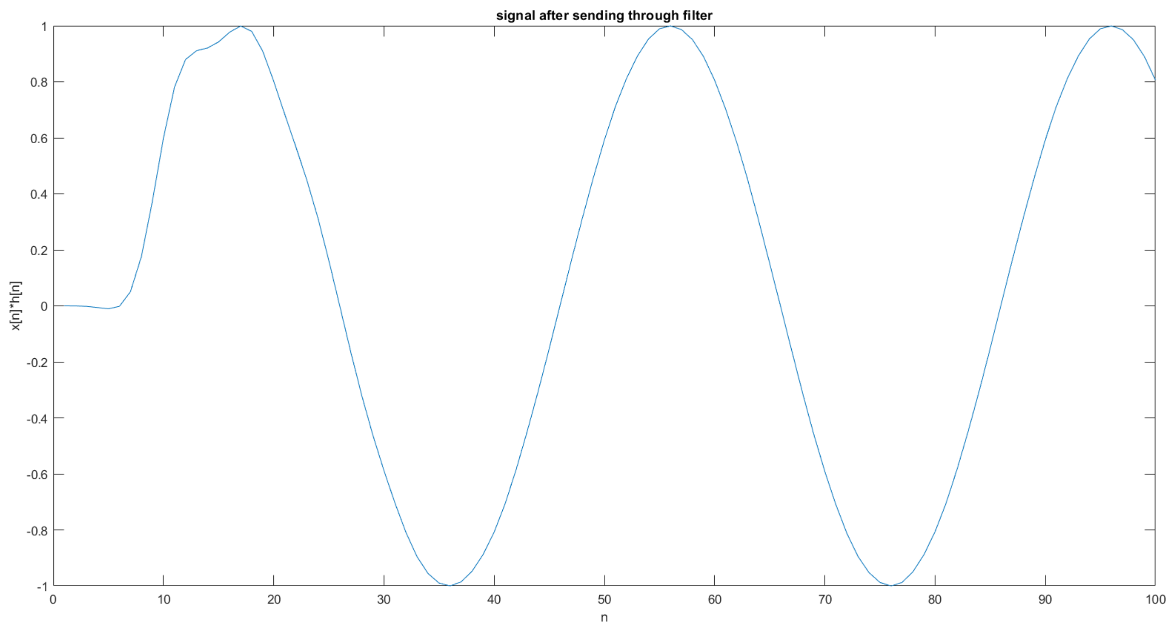


Butterworth Filter

```
[b,a] = butter(10,0.1*pi);
Hw = freqz(b,a);
plot(linspace(0,pi,512),abs(Hw))
title('Frequency Response of Chebyshev Filter')
ylabel('H(w)')
xlabel('w')
```




```
y2 = filter(b,a,test_signal);
plot(n,y2)
title('signal after sending through filter')
ylabel('x[n]*h[n]')
xlabel('n')
ylim([-1,1])
```



Discussions:

2(b) Generate two sinusoids one within passband and other out of passband, add them and pass through the filter as designed in part a). Plot the input and output signals and verify whether the desired specifications are satisfied or not.

We have defined a test signal to be sum of 2 sine signals of frequency 0.05π (low frequency) and 0.4π (High frequency), where the cutoff frequency of filter is 0.1π. By sending the signal which consisted of a low and a high frequency component through a Butterworth Low pass filter, only the low frequency term (0.05π) remains which we see as a sine wave as observed in the last figure. Hence the designed Low pass filter functions properly.