Lab 4 – filter design

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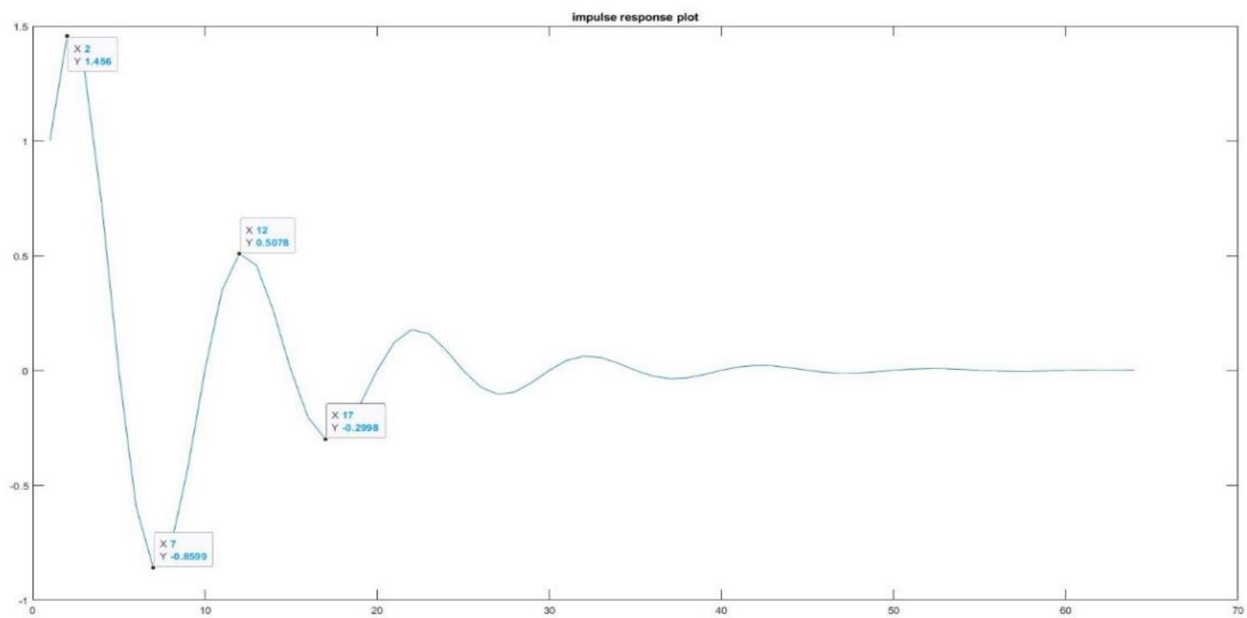
Experiments:

Task 1

- **Impulse response**

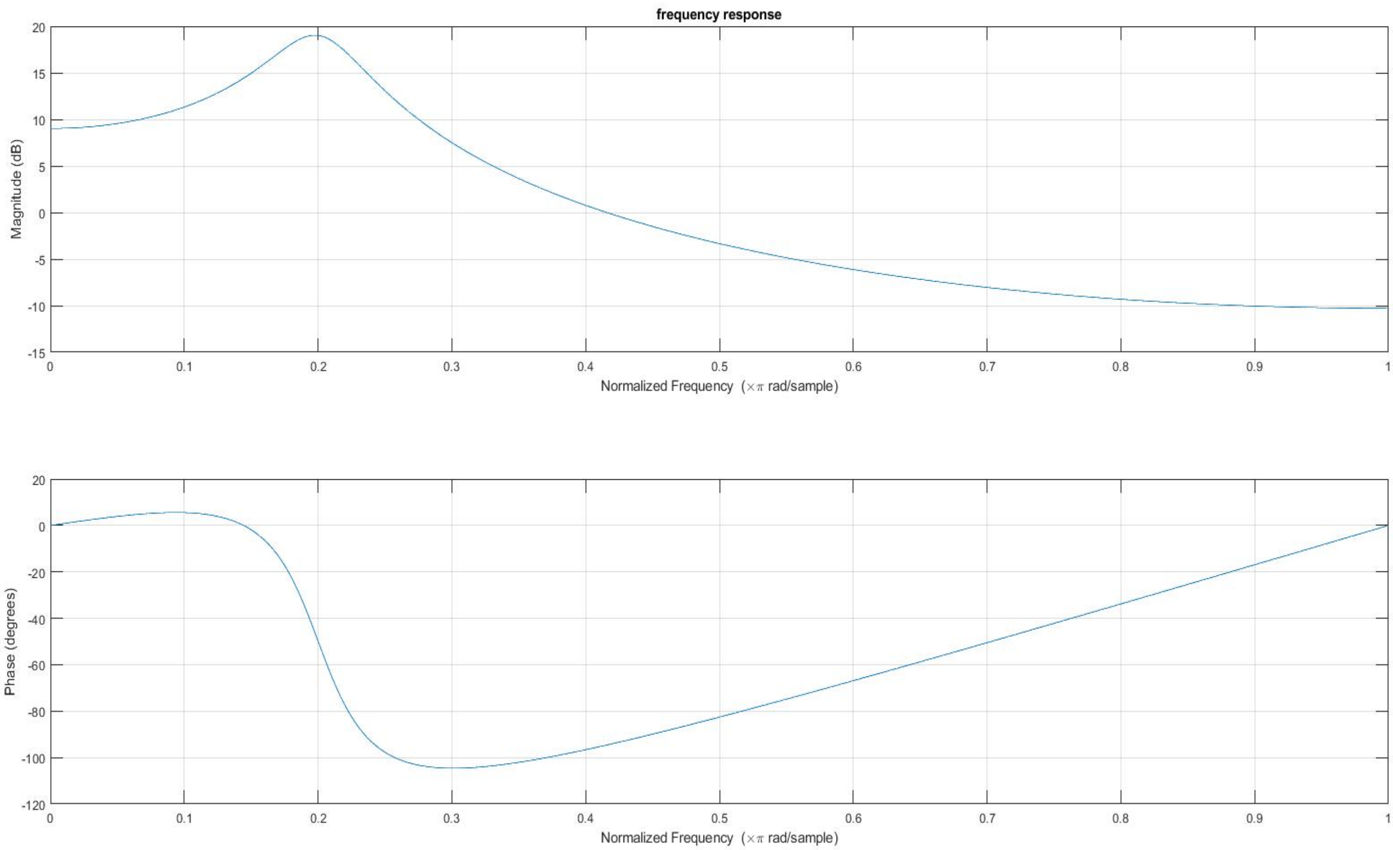


Period of oscillations h(n)=10.

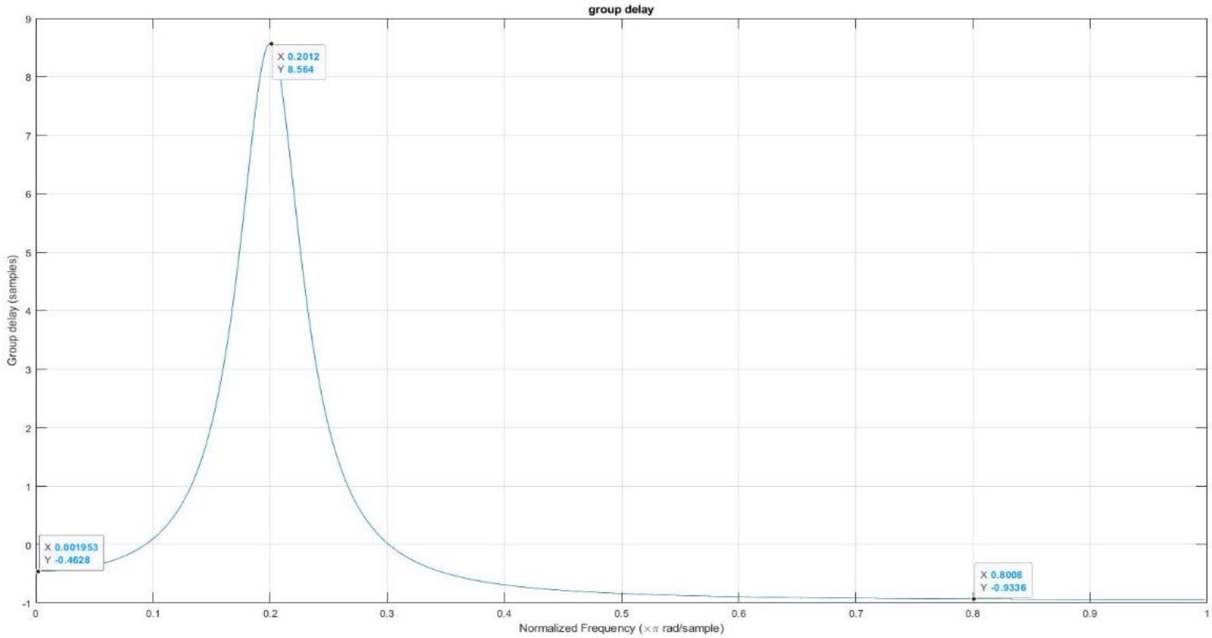
The difference between two consecutive maximums or two minimums is 10.

Thus, the Decay rate is equal to 12 seconds.

- **Frequency response**



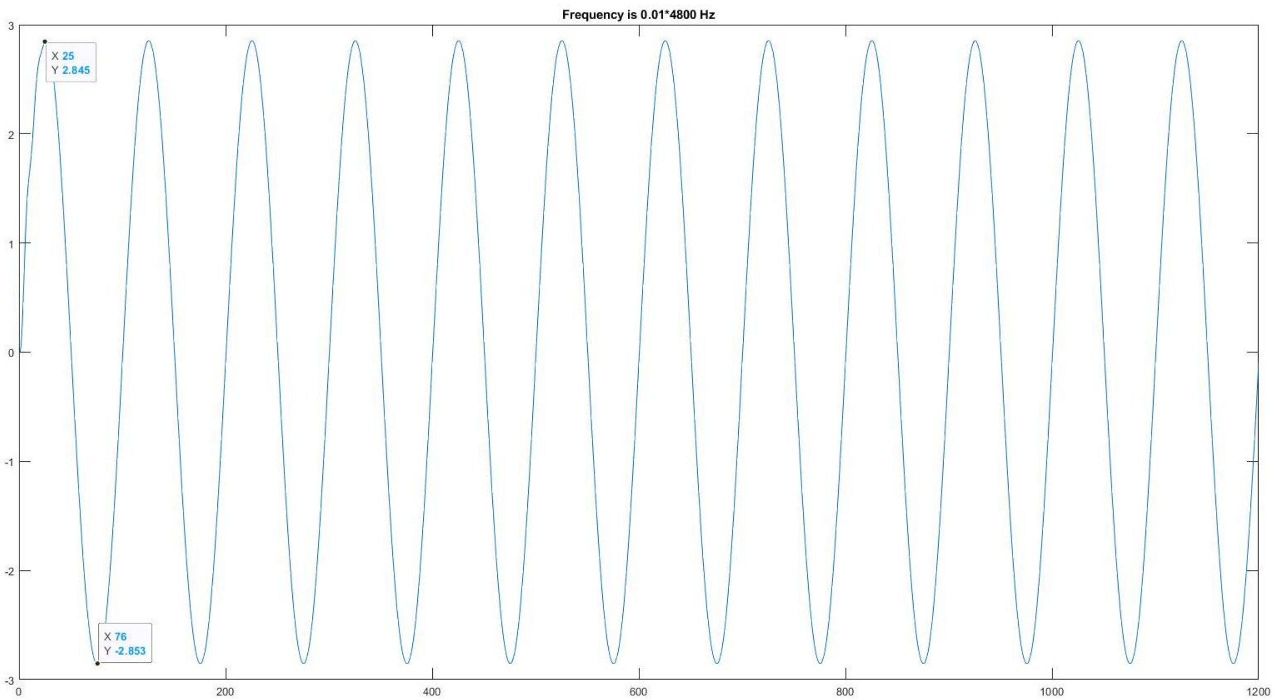
- **Group delay**



The minimum value of the group delay is -0.9, close to -1 and the maximum value of the group delay is 8.6.

Group delay peak (equal to 8.6) is located at the mark which corresponds to the 0.2 of Normalized Frequency, which is equal to the value used to specify the period of the poles 0.9e^(±j**0.2**π).

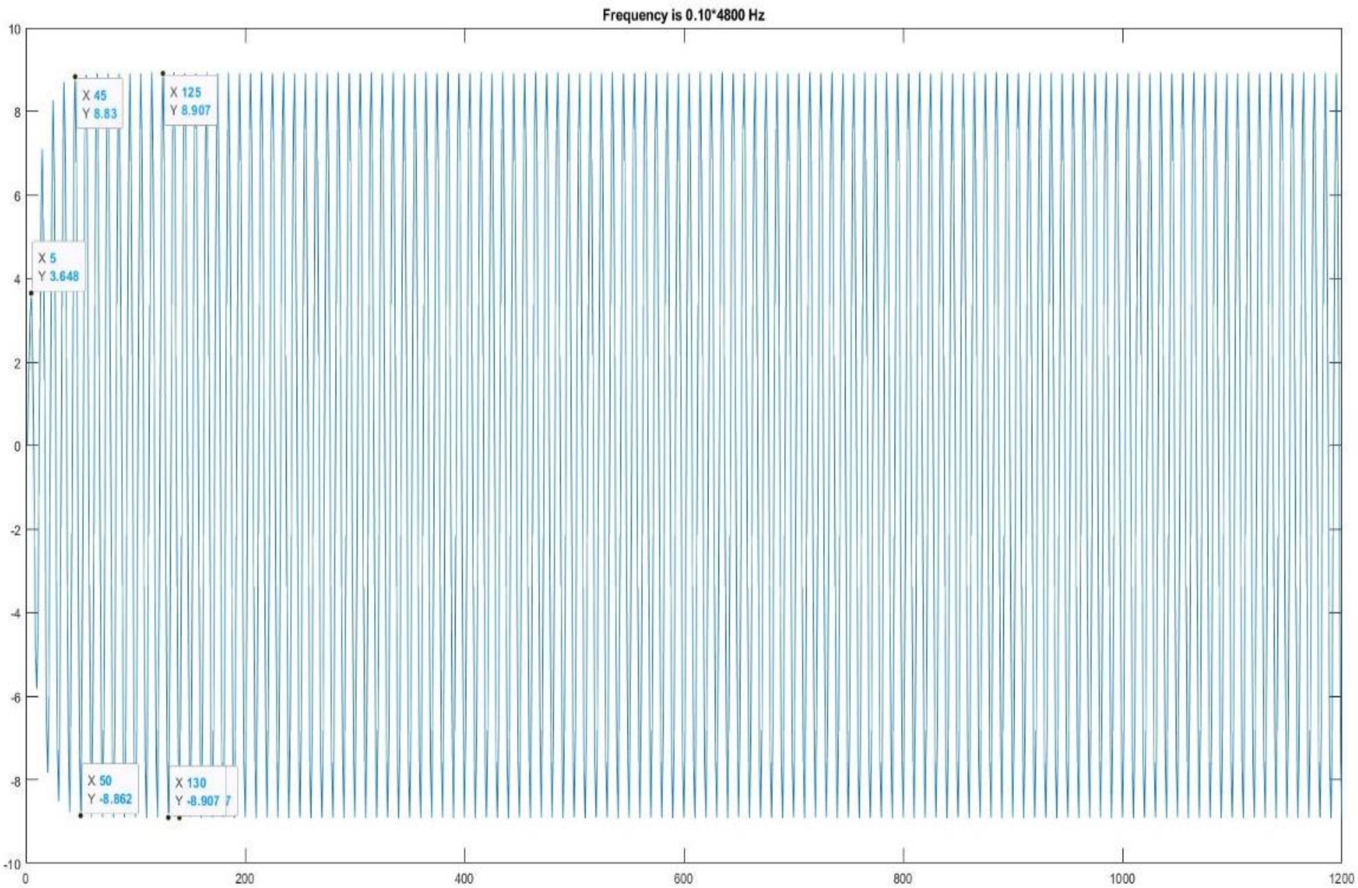
- **Responses for sine waves of different frequencies**



Chosen sample frequency was 4800Hz.

The frequency of the sine wave was set to 48 to ensure a clear and visible plot. This was calculated by multiplying 0.01 and 4800.

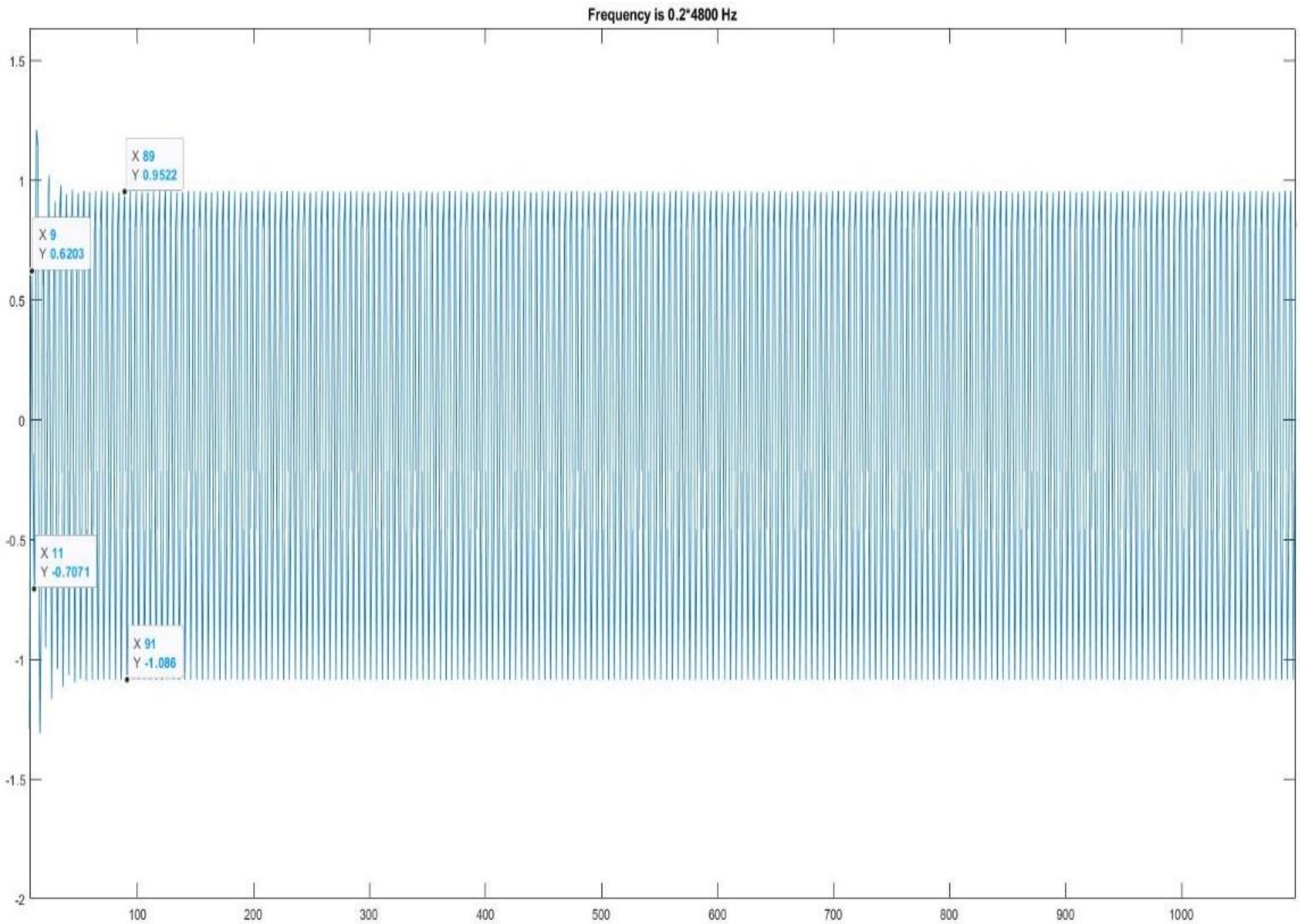
And the magnitude response is 19.



Chosen sample frequency was 4800Hz.

Frequency of the sine signal 0.10*4800 = 480 Hz, selected to have a visible and clear plot.

Magnitude response is 19.

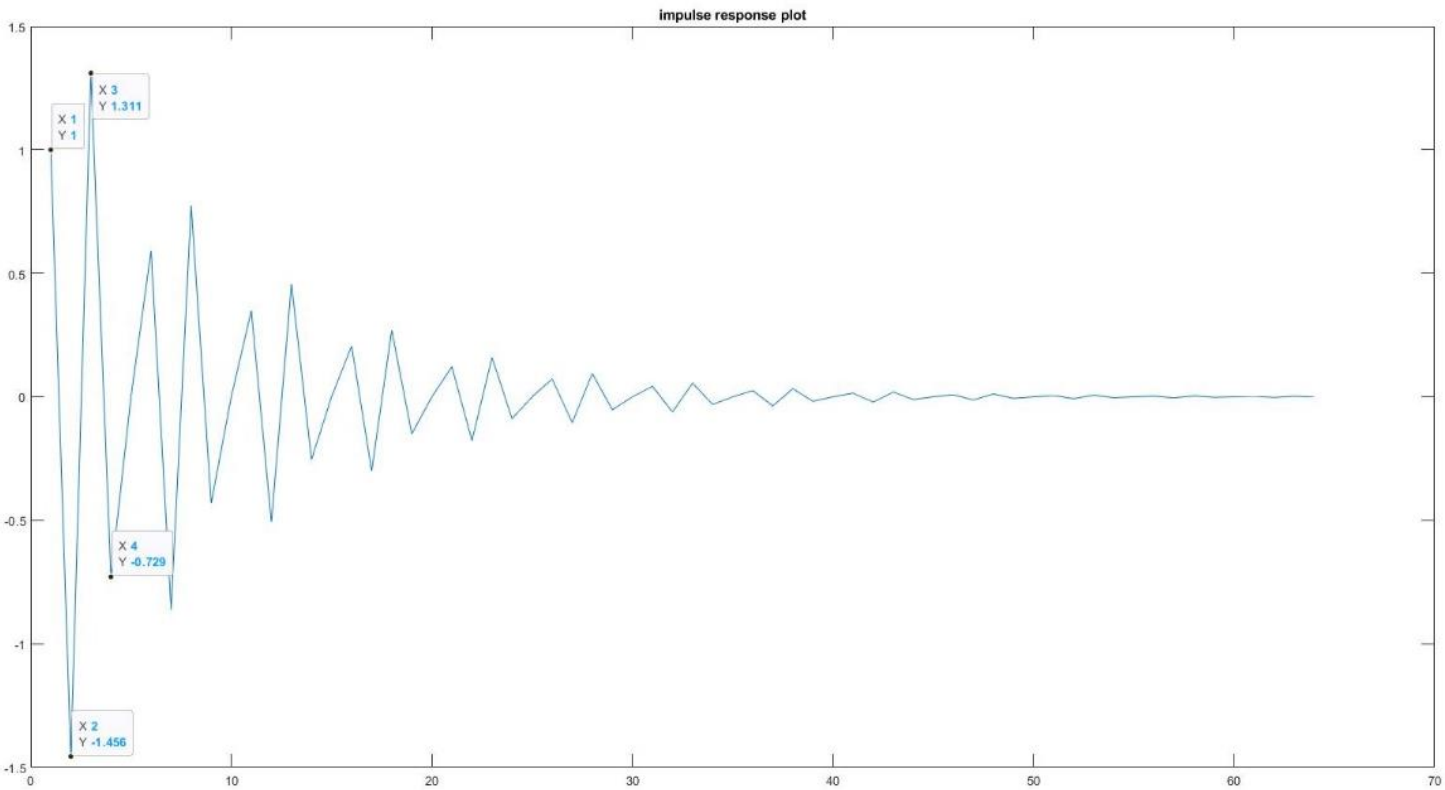


Chosen sample frequency was 4800Hz.

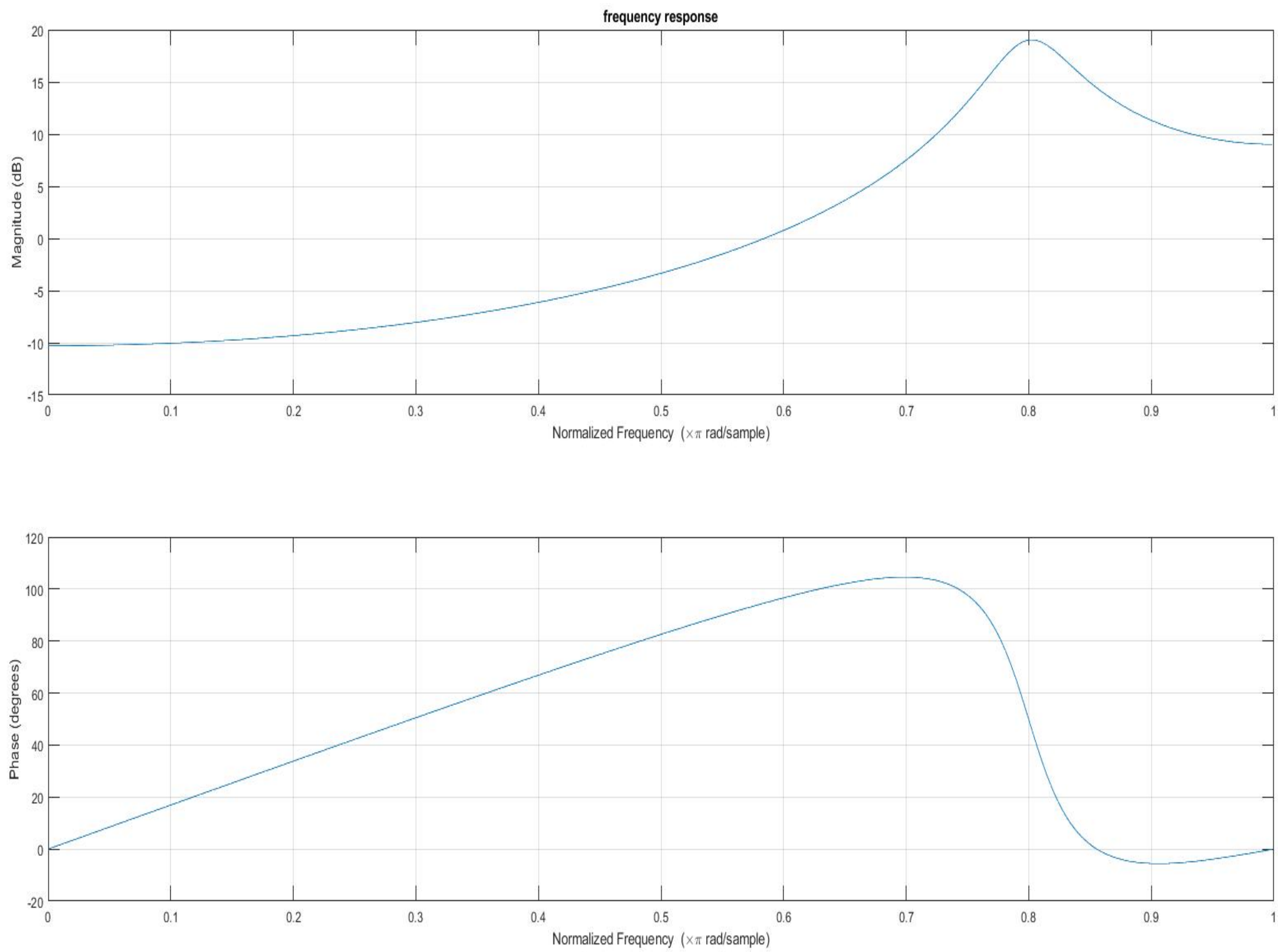
Frequency of the sine signal 0.2*4800 = 960 Hz, selected so in order to have a visible and clear plot.

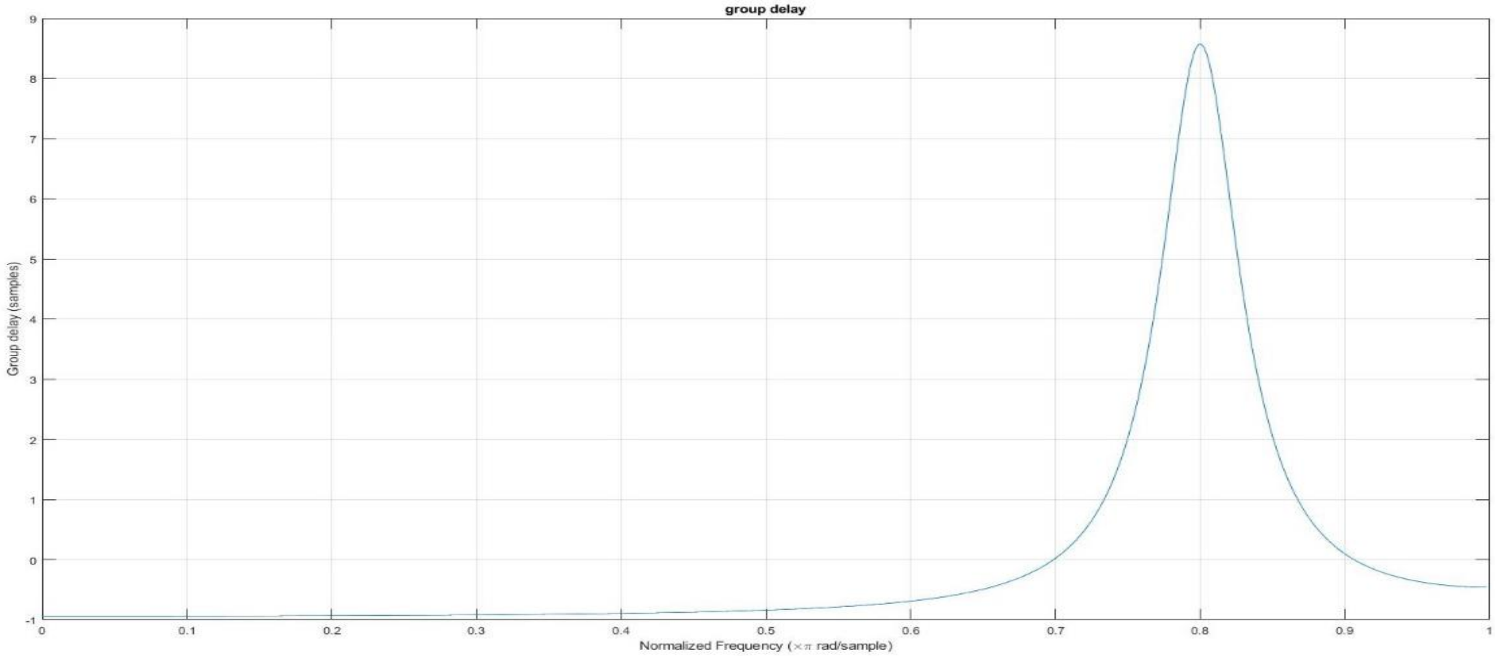
Magnitude response is 19.

Task 2



The period of oscillation for h(n) is 2, which means that the difference between two consecutive maximum or minimum values is 2. The decay rate has a value of 13 seconds.





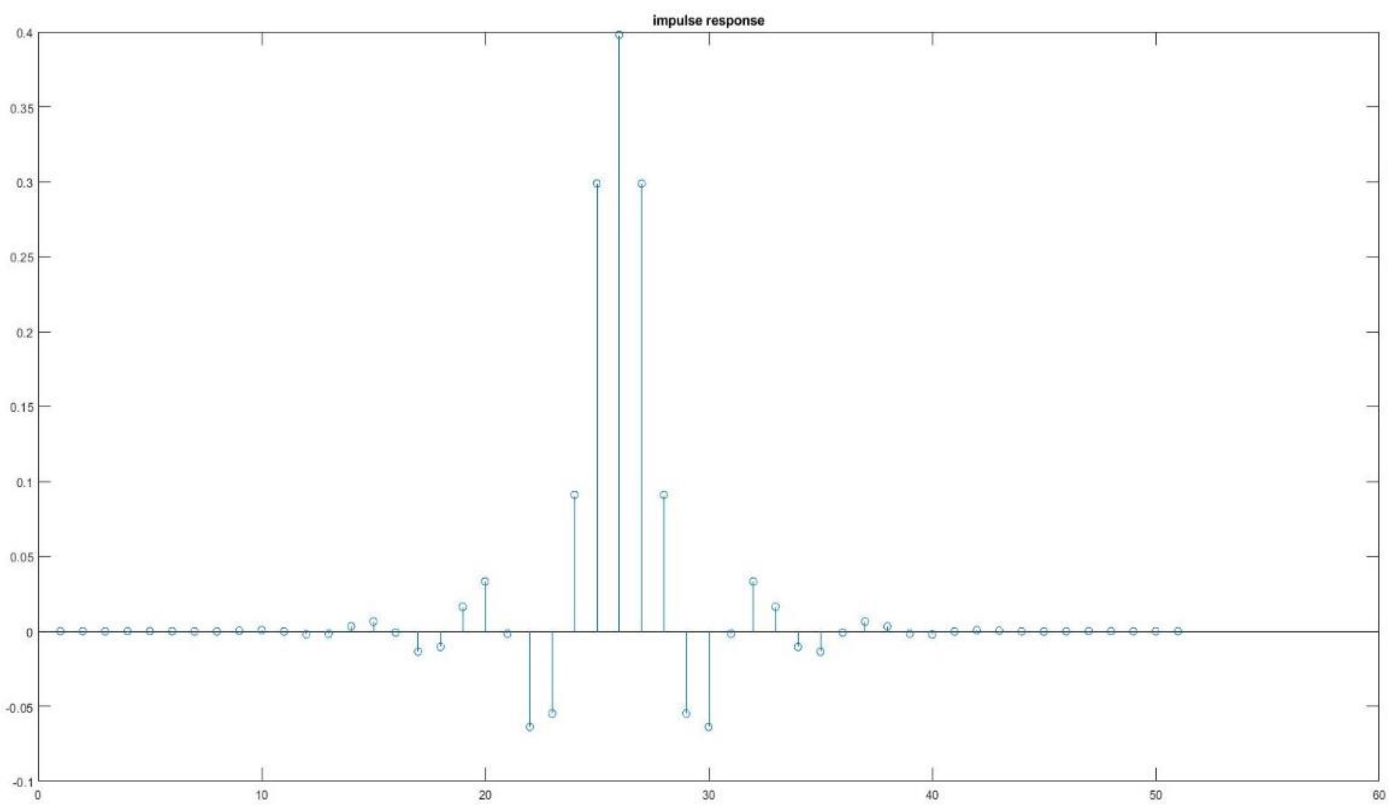
It is evident that the signal with poles at $0.9e^{\pm j0.8\pi}$ has a period of oscillation that is 5 times smaller than the signal in Task 1, which had a period of oscillation at $0.9e^{\pm j0.2\pi}$. As a result, the impulse response graph for the former signal appears sharper in comparison to the graph from Task 1.

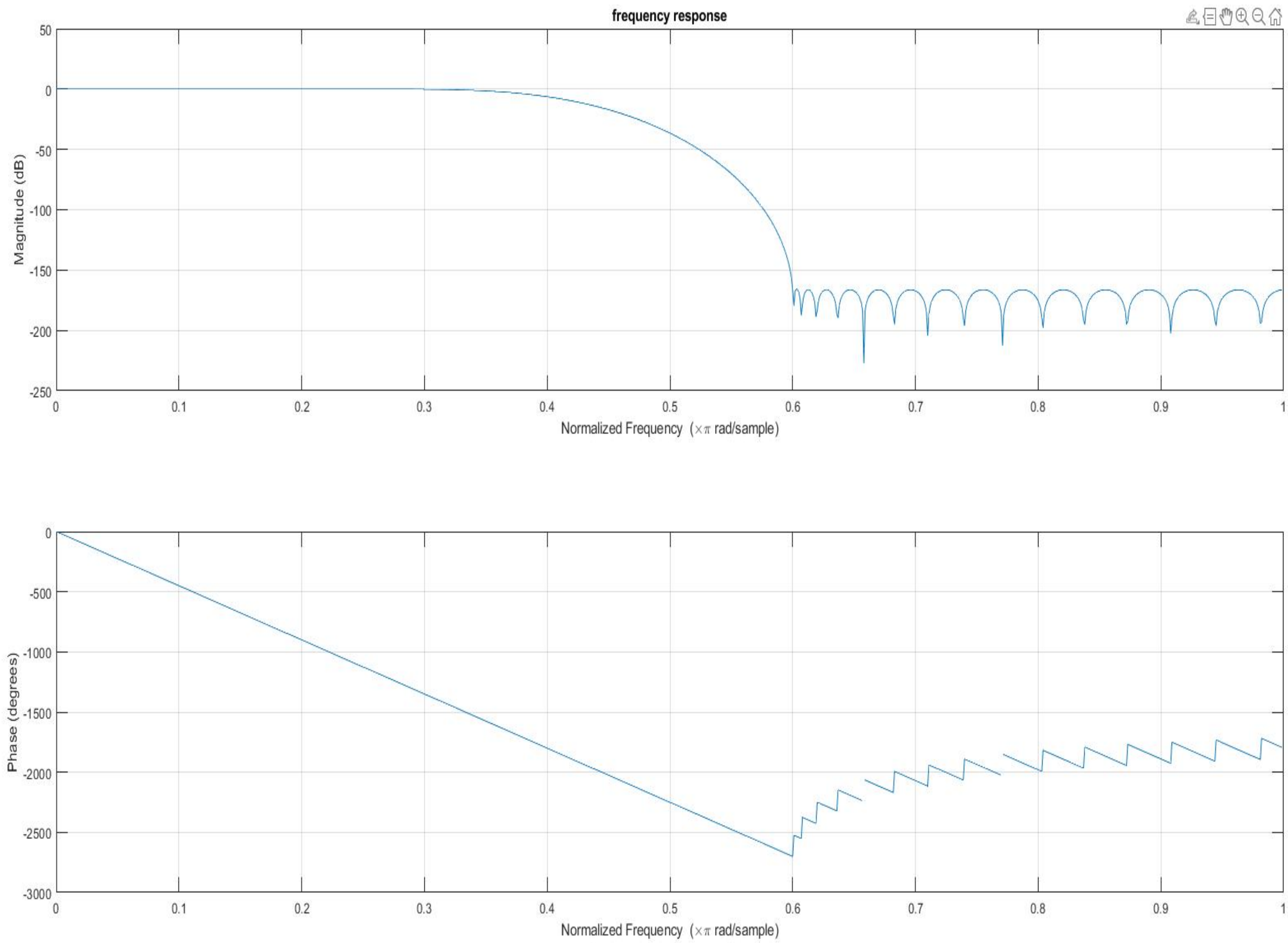
In Task 1, the frequency response graph showed a decreasing trend after reaching a frequency of 0.2. However, in Task 2, the graph can be described as increasing because it has a rising characteristic before reaching a frequency of 0.8.

The peak of the group delay graph now is also located at the different frequency mark (0.8) and in case of the task 1 it was situated at the mark of 0.2.

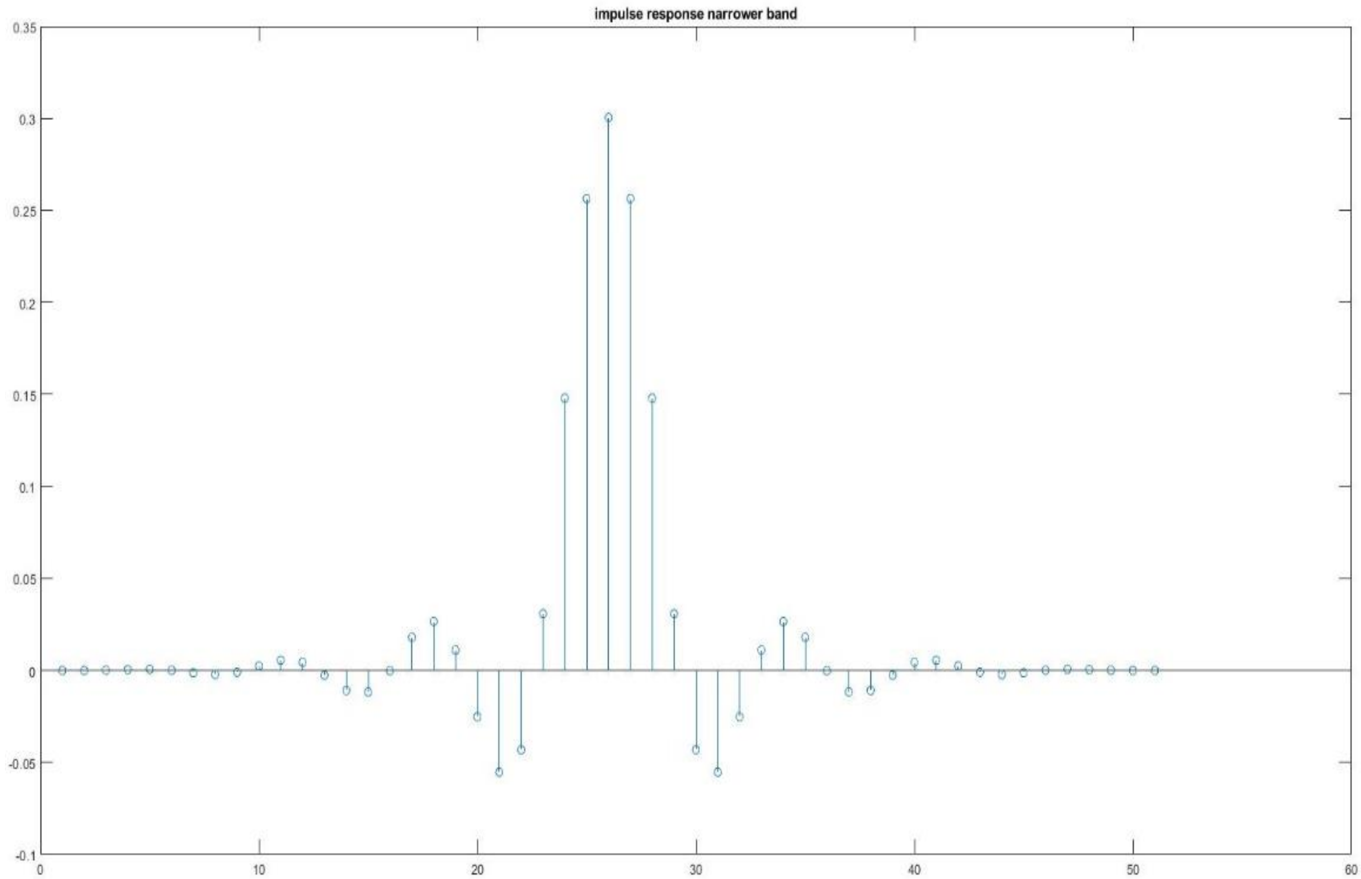
Generally, the frequency response and group delay characteristics are similar, but they are shifted with respect to the normalized frequency values. However, the impulse response characteristics differ as described earlier.

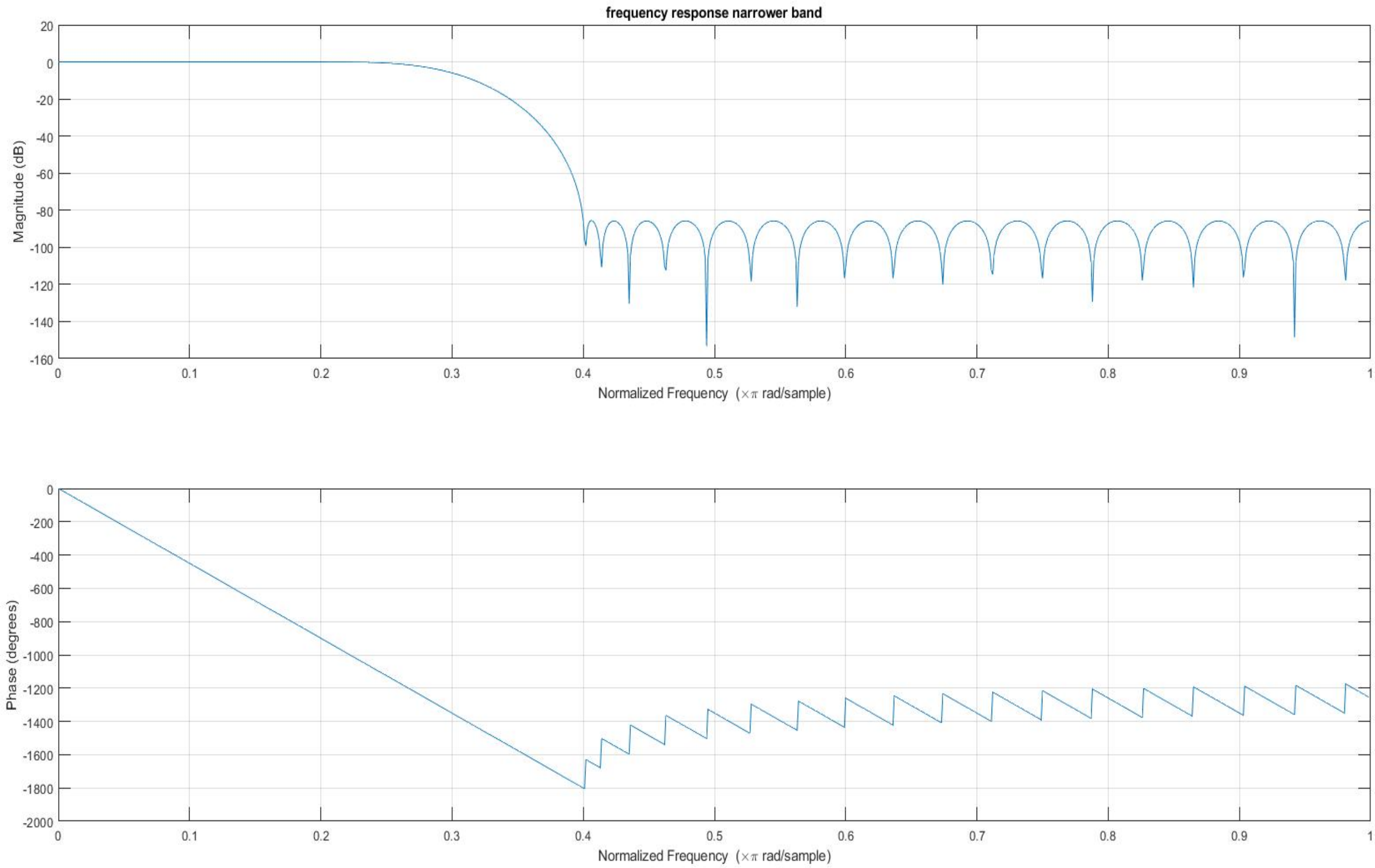
Task 3


impulse response
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
-0.1
0
10
20
30
40
50
60



frequency response
Magnitude (dB)
50
0
-50
-100
-150
-200
-250
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)
Phase (degrees)
0
-500
-1000
-1500
-2000
-2500
-3000
Normalized Frequency (×π rad/sample)


impulse response narrower band
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
-0.1
0
10
20
30
40
50
60

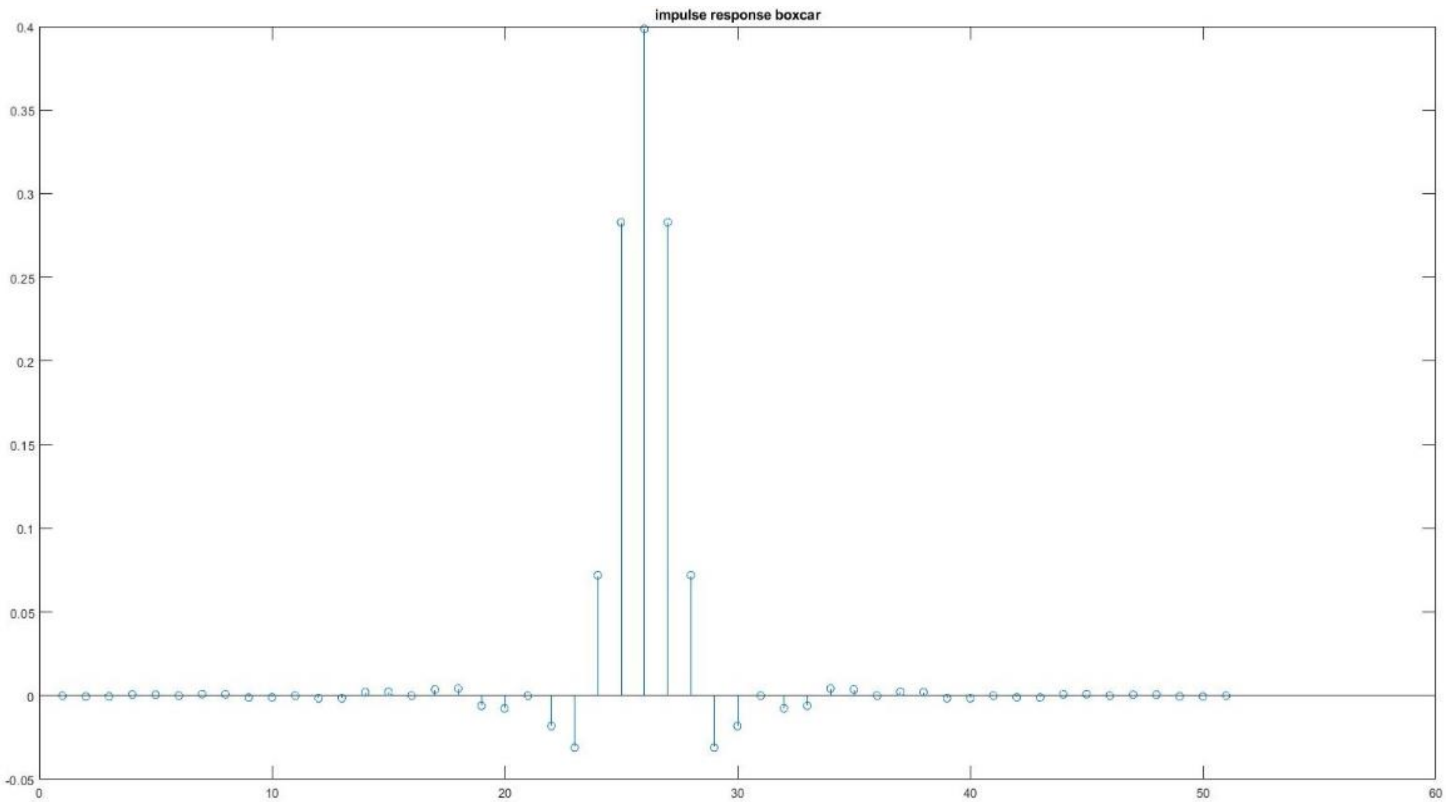


The requirements were fulfilled but a higher order value (50) was necessary to achieve a precise plot. The magnitude responses do not demonstrate an ideal rectangular characteristic, but they are of the expected design in real-world applications, appearing as sinusoidal characteristics.

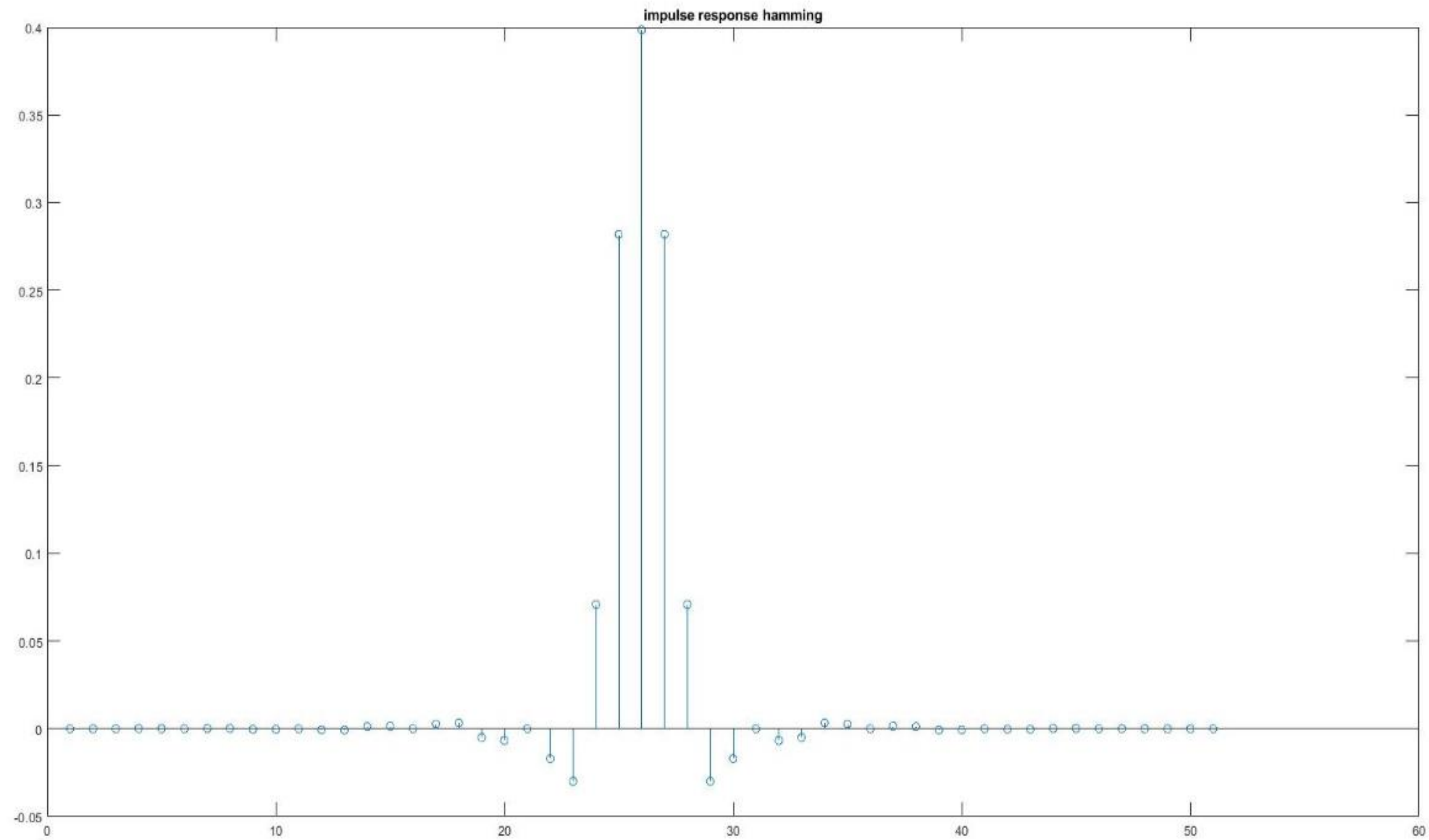
Frequency response illustrates the passband and stopband gains for both cases. Sidelobes in the stopband have sinusoidal shape, however with every part of the sinusoid being positive, they represent the noise signals which were stopped by our filter. Narrower band gives us greater amount of the sidelobes as its stopband starts at the mark of 0.4 Normalized Frequency, while in case for initial one it was 0.6.

The frequency response displays the passband and stopband gains for both cases. The stopband sidelobes have a sinusoidal shape, with each part of the sinusoid being positive. These sidelobes represent the noise signals that were filtered out by our filter. In the case of a narrower band, the stopband starts at a normalized frequency 0.4, which leads to a greater number of sidelobes in comparison to the initial filter, where the stopband started at 0.6 normalized frequency.

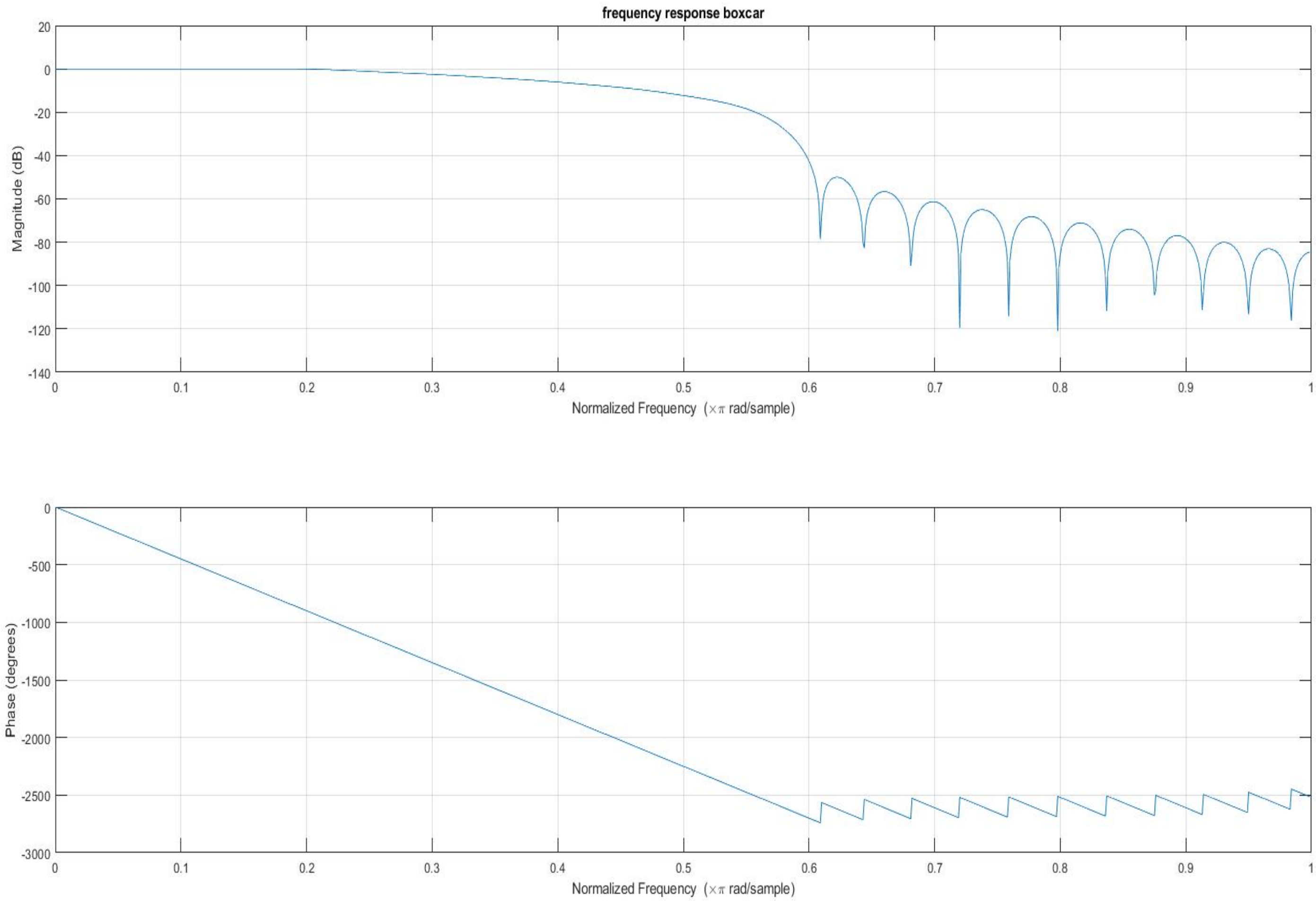
Task 4

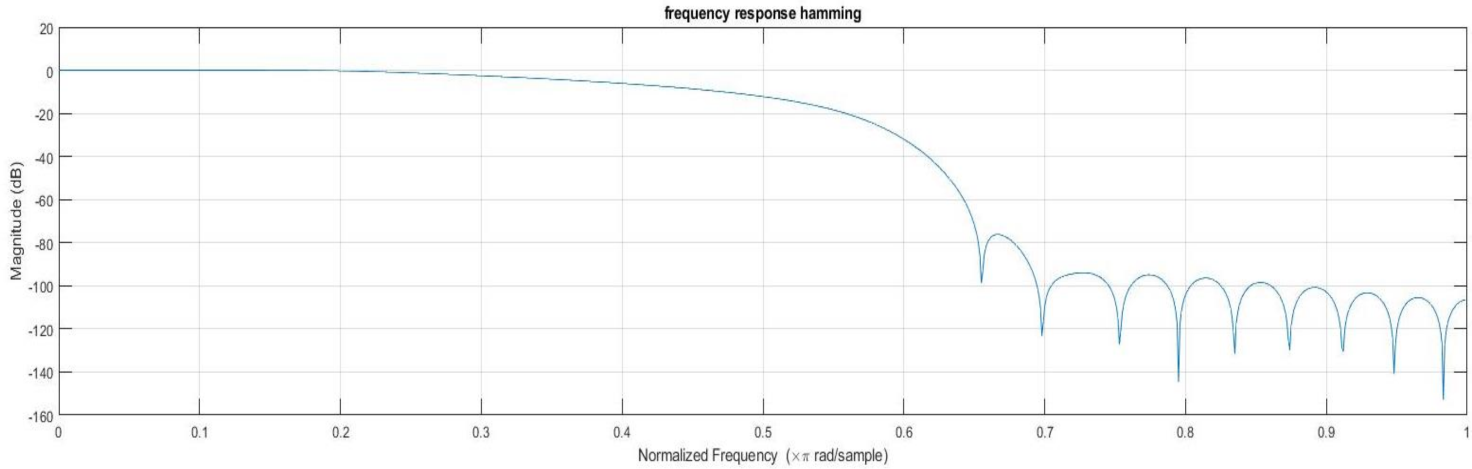


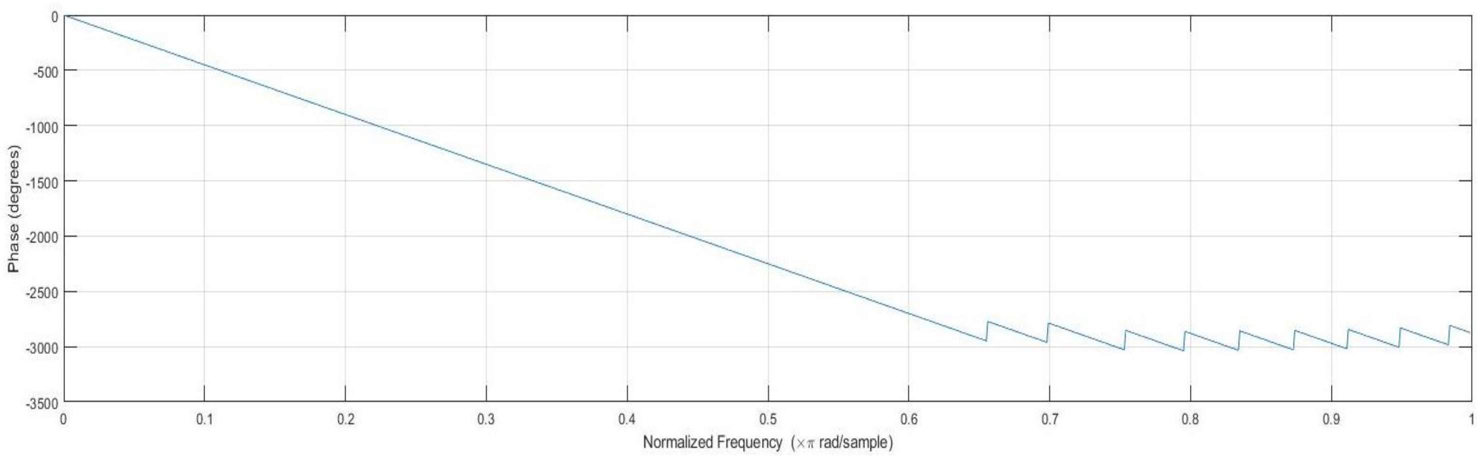
impulse response boxcar
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
0
10
20
30
40
50
60



impulse response hamming
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
0
10
20
30
40
50
60


frequency response boxcar
Magnitude (dB)
20
0
-20
-40
-60
-80
-100
-120
-140
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)
Phase (degrees)
0
-500
-1000
-1500
-2000
-2500
-3000
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)

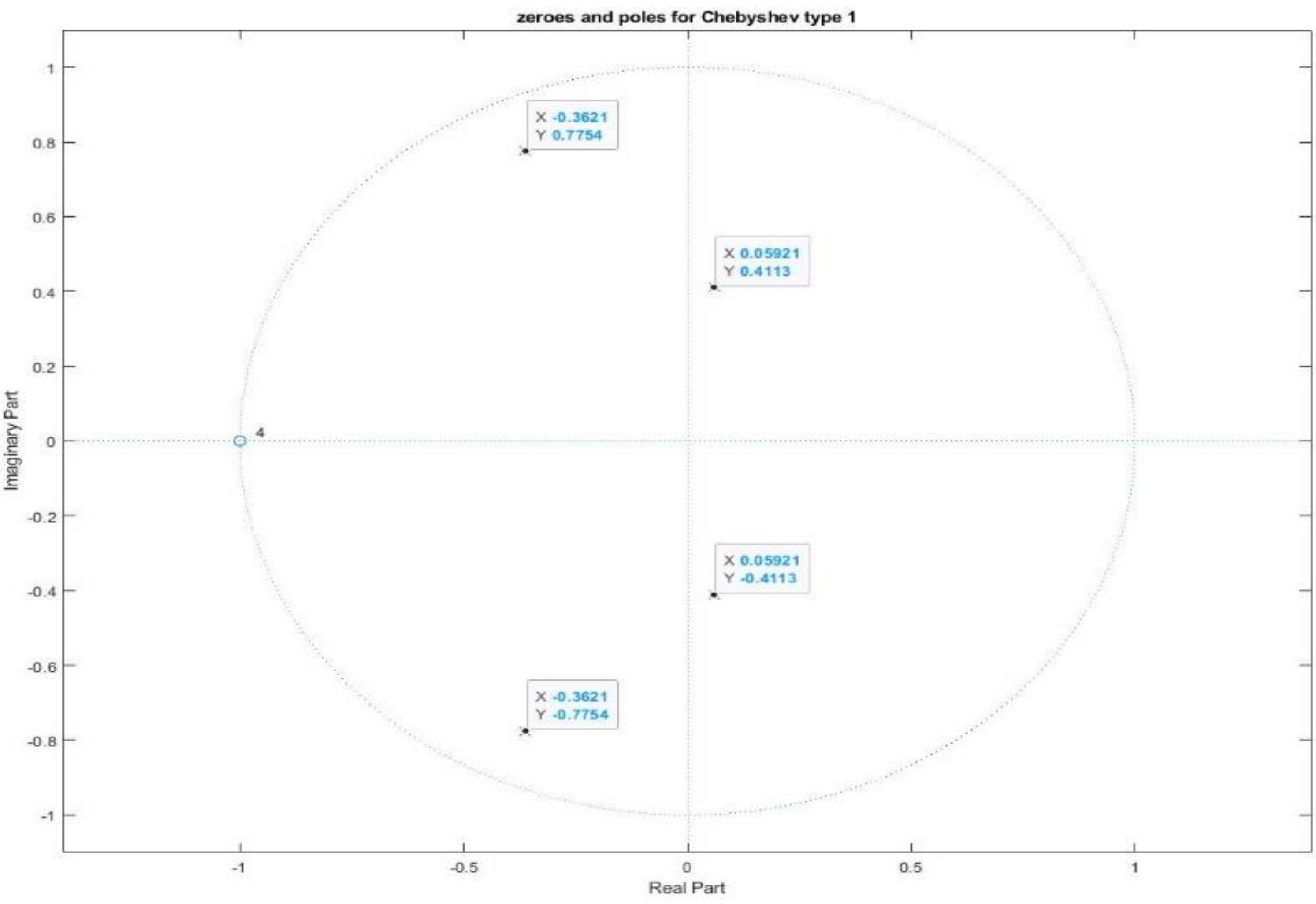


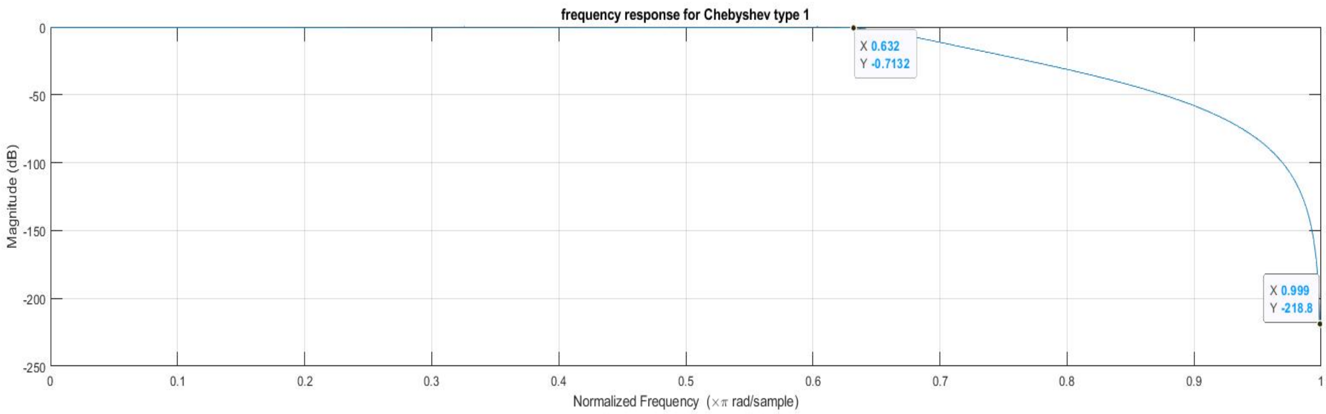


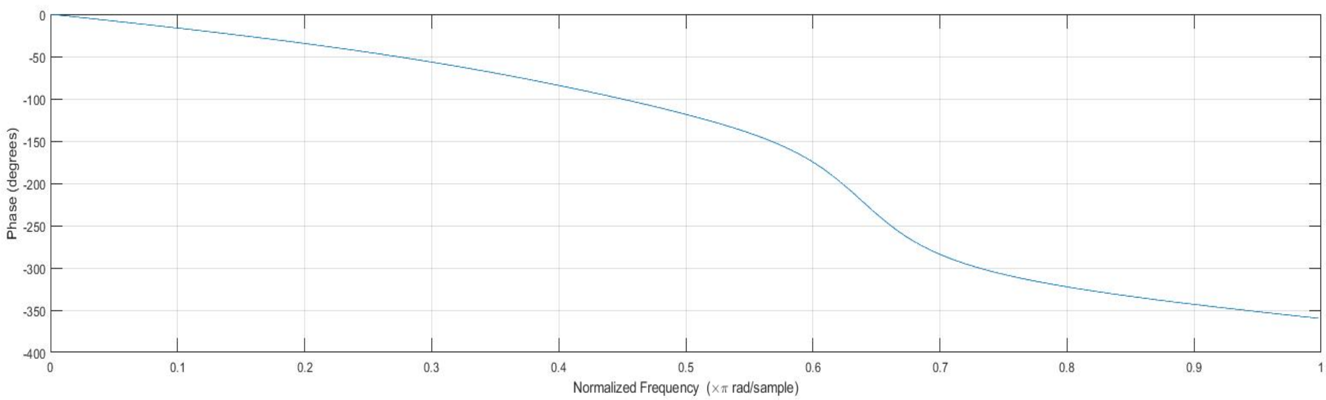
It can be observed, obtained graphs of the impulse response for two FIR filters designed with the usage of boxcar and hamming windows, look quite similar. The notable difference may be observed in case of the frequency response, where for the boxcar stopband starts right after the 0.6 Normalized Frequency mark (0.61 precisely) and for the case with hamming window we observe that stopband starts later, at the mark of 0.66 of Normalized Frequency. Beside this, the magnitude values also differ for the described points – about -80 dB in case of boxcar and a -100 in case of hamming. Due to this we can say that application of the boxcar window results in a slightly faster transition, while the usage of the hamming window gives us a smoother transition.

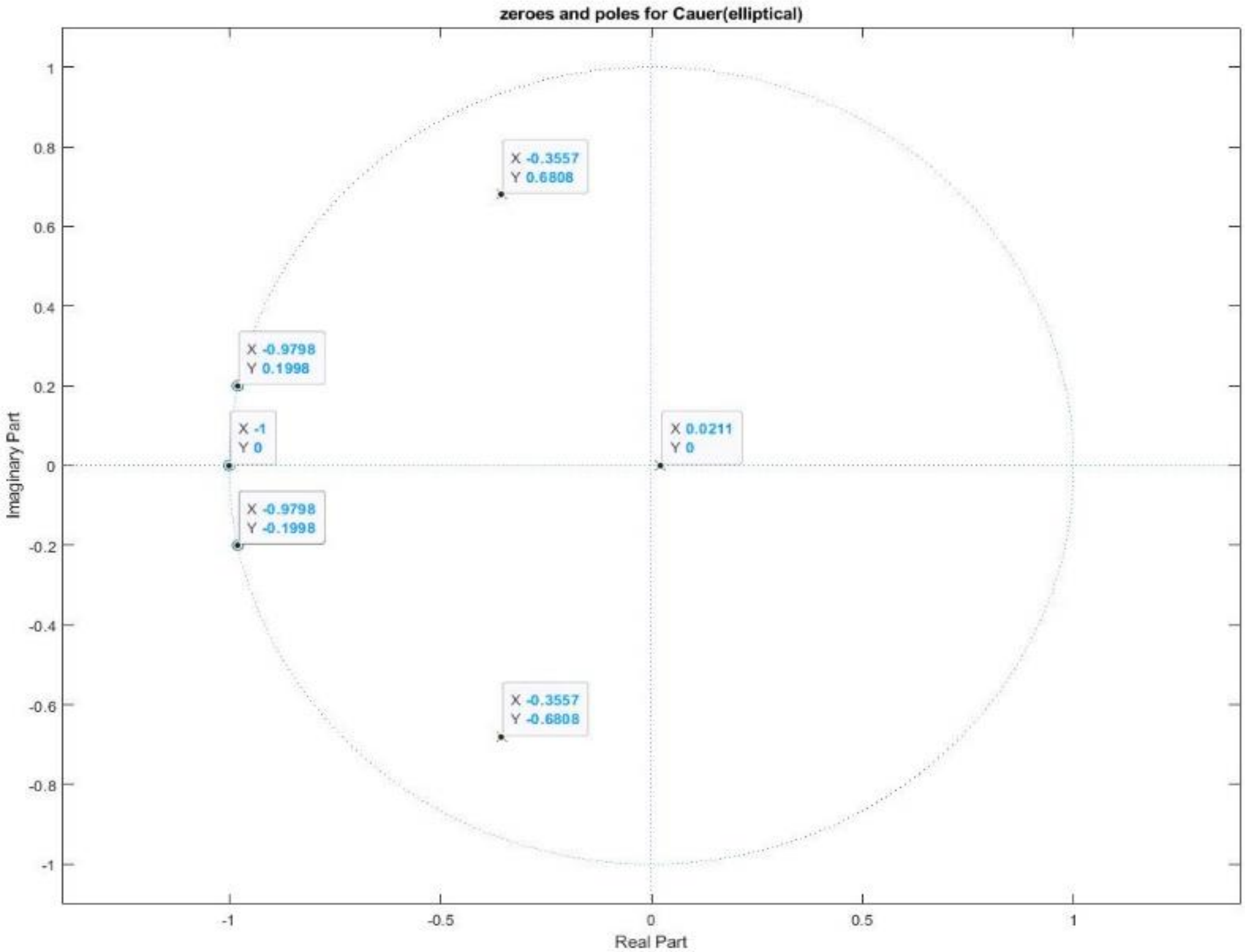
Comparing those plots with those obtained in case of testing the Parks-McClellan filter, which was done in task 3. We may see that there we had more control on the beginning of the stopband (since we specified parameters for the wide transition band). So, the transition was performed exactly from 0,2π to 0,6π with the magnitude change from 0 to -175. Afterwards, sidelobes of the stopband were in the same line, while in case of FIR filters with the given windows, the stopband sidelobes formed a declining characteristic. In addition, the number of sidelobes in stopband in case of the window methods filters is lower by half, than in the Parks-McClellan filter.

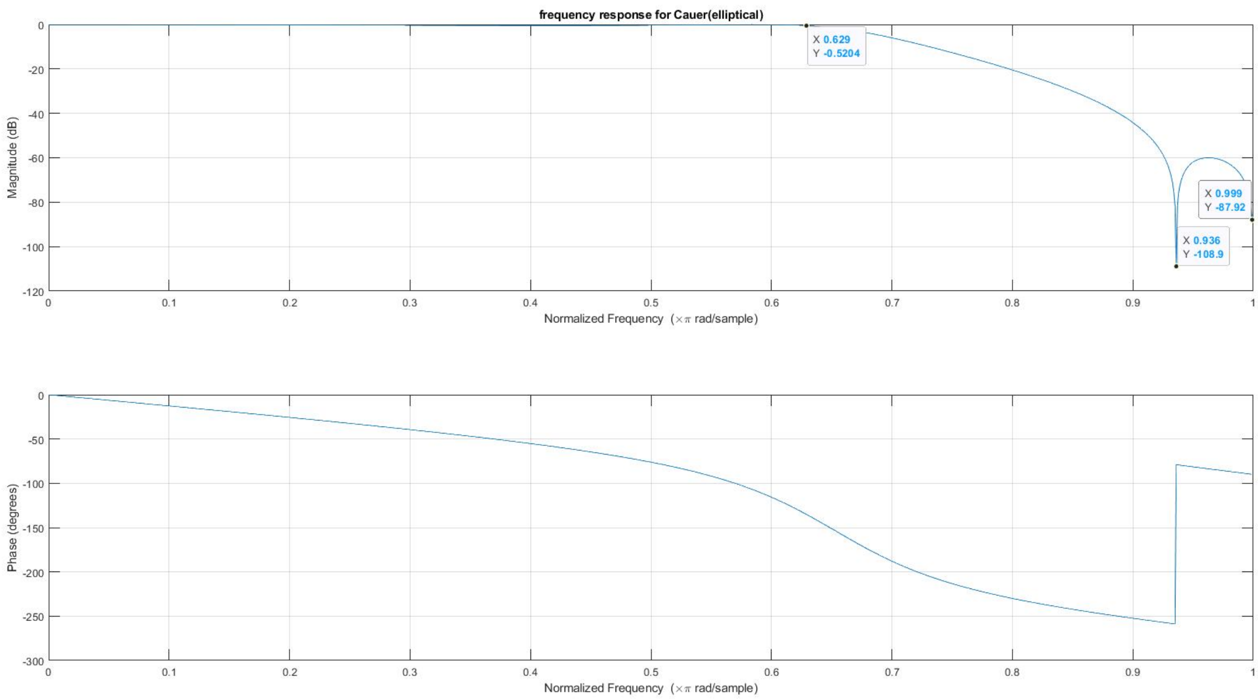
Task 5




frequency response for Chebyshev type 1
Magnitude (dB)
X 0.632
Y -0.7132
X 0.999
Y -218.8
0
-50
-100
-150
-200
-250
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)


Phase (degrees)
0
-50
-100
-150
-200
-250
-300
-350
-400
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Normalized Frequency (×π rad/sample)


zeroes and poles for Cauer(elliptical)
Imaginary Part
Real Part
X -0.3557
Y 0.6808
X -0.9798
Y 0.1998
X -1
Y 0
X 0.0211
Y 0
X -0.9798
Y -0.1998
X -0.3557
Y -0.6808



As shown in the graphs, I checked the design of an IIR LP filter using two different functions. When using Chebyshev Type 1, I obtained 4 poles distributed across the circle and 4 zeroes situated in one point. The frequency response graph shows that the transition started at the 0.63 normalized frequency mark and finished at about 1 with a magnitude response reaching -218 dB.

When using the Cauer (elliptical) method, I obtained three poles distributed inside the circle and three poles situated at the border of the circle. The transition was performed between the marks of 0.63 and 0.94 with the magnitude response decreasing to -109 dB. The stopband was clearly visible, which was not the case with the Chebyshev Type 1 method.

Overall, transition in case of Elliptical method was smoother and the magnitude response was lower.

As the above experiments show, there are differences between the IIR and FIR filters.

IIR filters are often considered more efficient than FIR filters because they require fewer coefficients to perform the same operations. As a result, we can often use a lower filter order for IIR filters compared to FIR filters. This was demonstrated in task 3 where a higher order was required for the FIR filter to obtain a more visible and precise measurement.

In contrast, the design of FIR filters provides greater flexibility and accuracy in controlling both the passband and stopband. Additionally, the phase response of FIR filters is linear, unlike the non-linear phase response of IIR filters. However, IIR filters have a smaller number of sidelobes in the stopband compared to FIR filters.

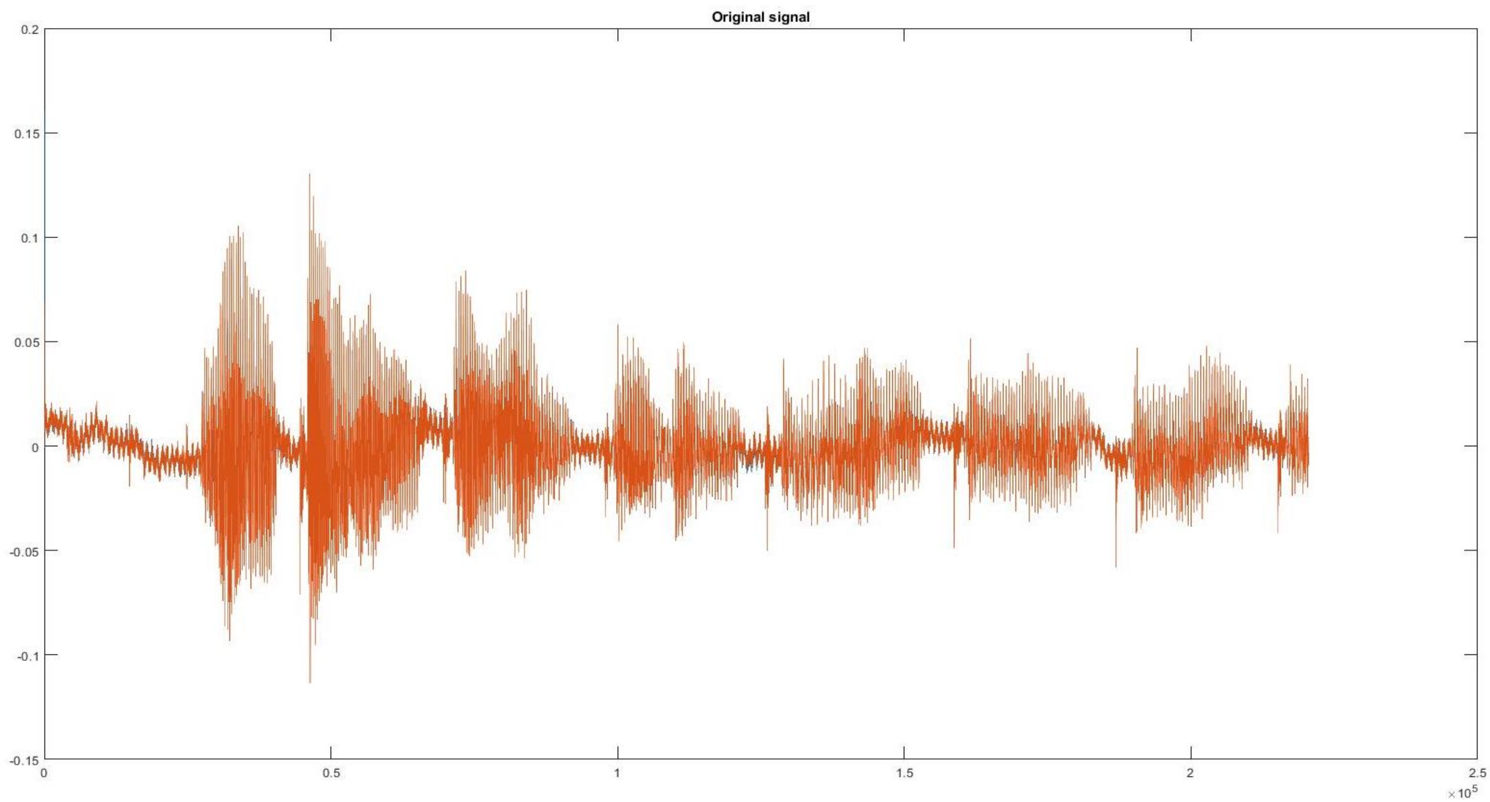
In summary, FIR filters offer better control over the stopband and passband with linear phase response but come at a higher cost and require more memory. On the other hand, IIR filters are more efficient and suitable for low-cost and fast applications, with the trade-off being a non-linear phase response and fewer stopband sidelobes.

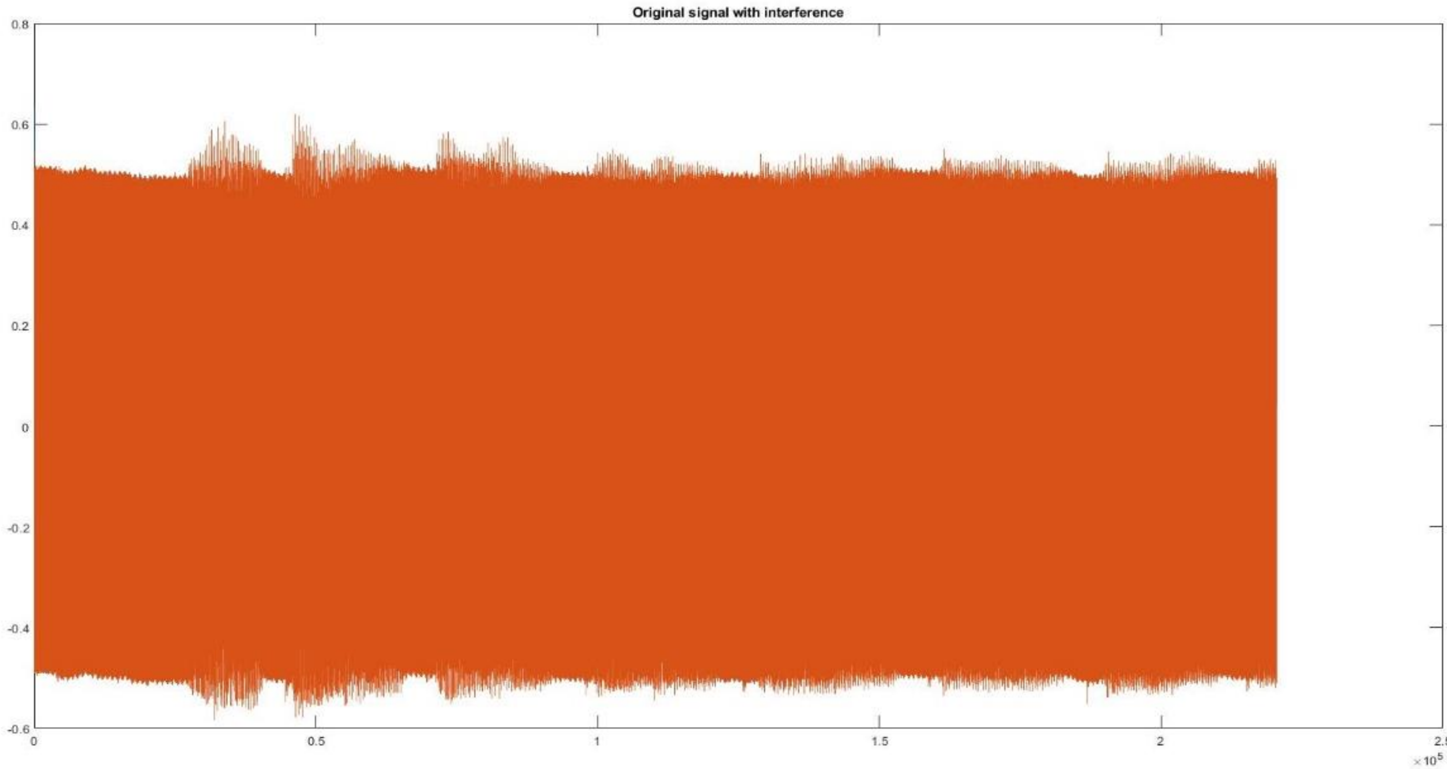
Task 6

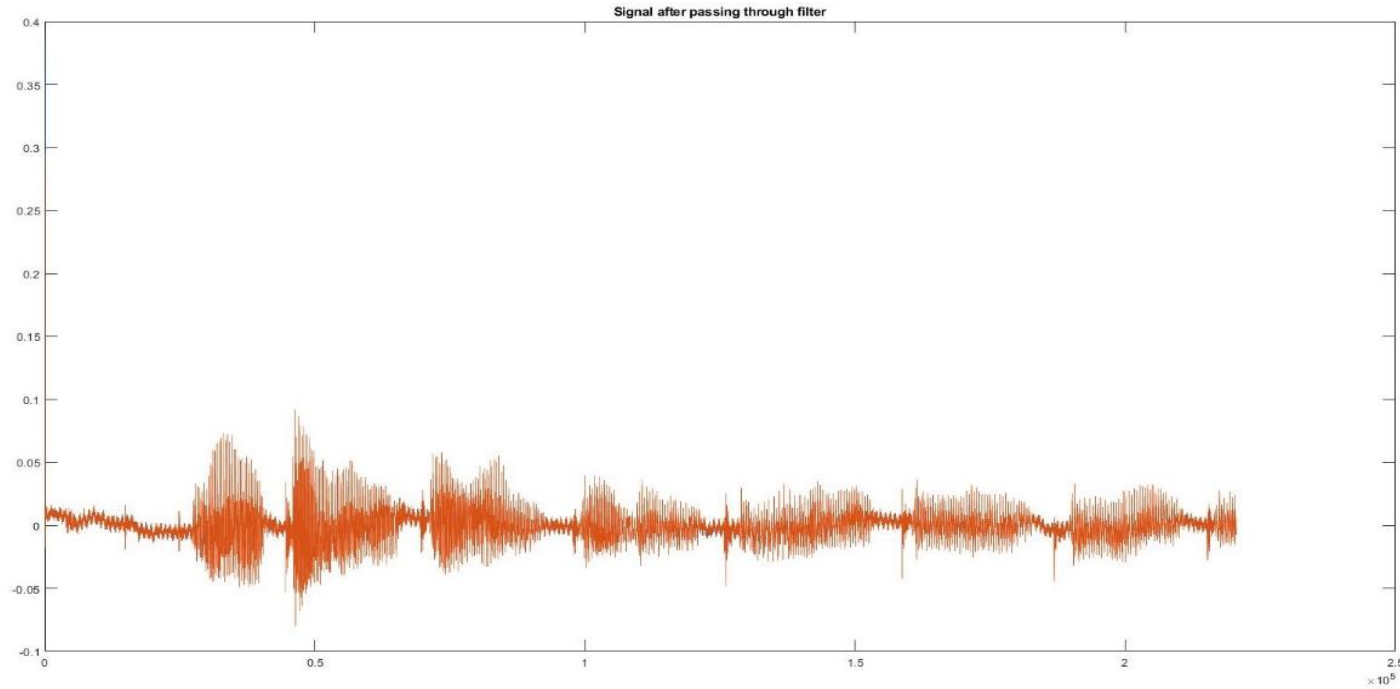
I recorded myself saying "One-two-one-two.." and applied the sinusoidal interference signal to it. I designed a filter to uncover the initial speech signal. I have chosen zeroes and ones to maximize the efficiency of the filter. Two zeroes described by $e^{(\pm 2*pi*fn)}$ and two poles given by $0.9*e^{(\pm 2*pi*fn)}$.

As a result, I managed to uncover the signal. However, some initial measurement, visible on the graph which corresponds to the signal after applying the filter, was noticed. This is due to the noises of equipment which appear when recording starts.

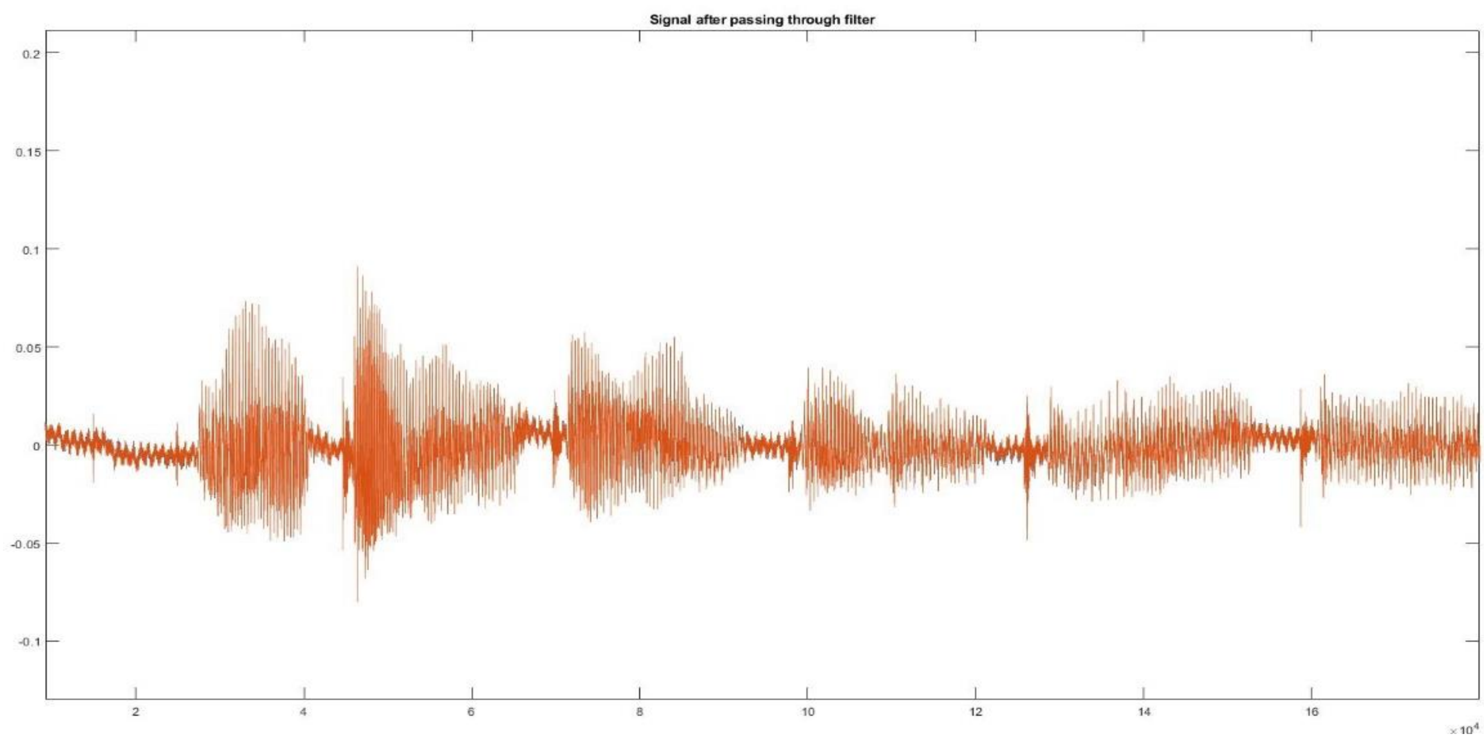
Below I attach the graphs.



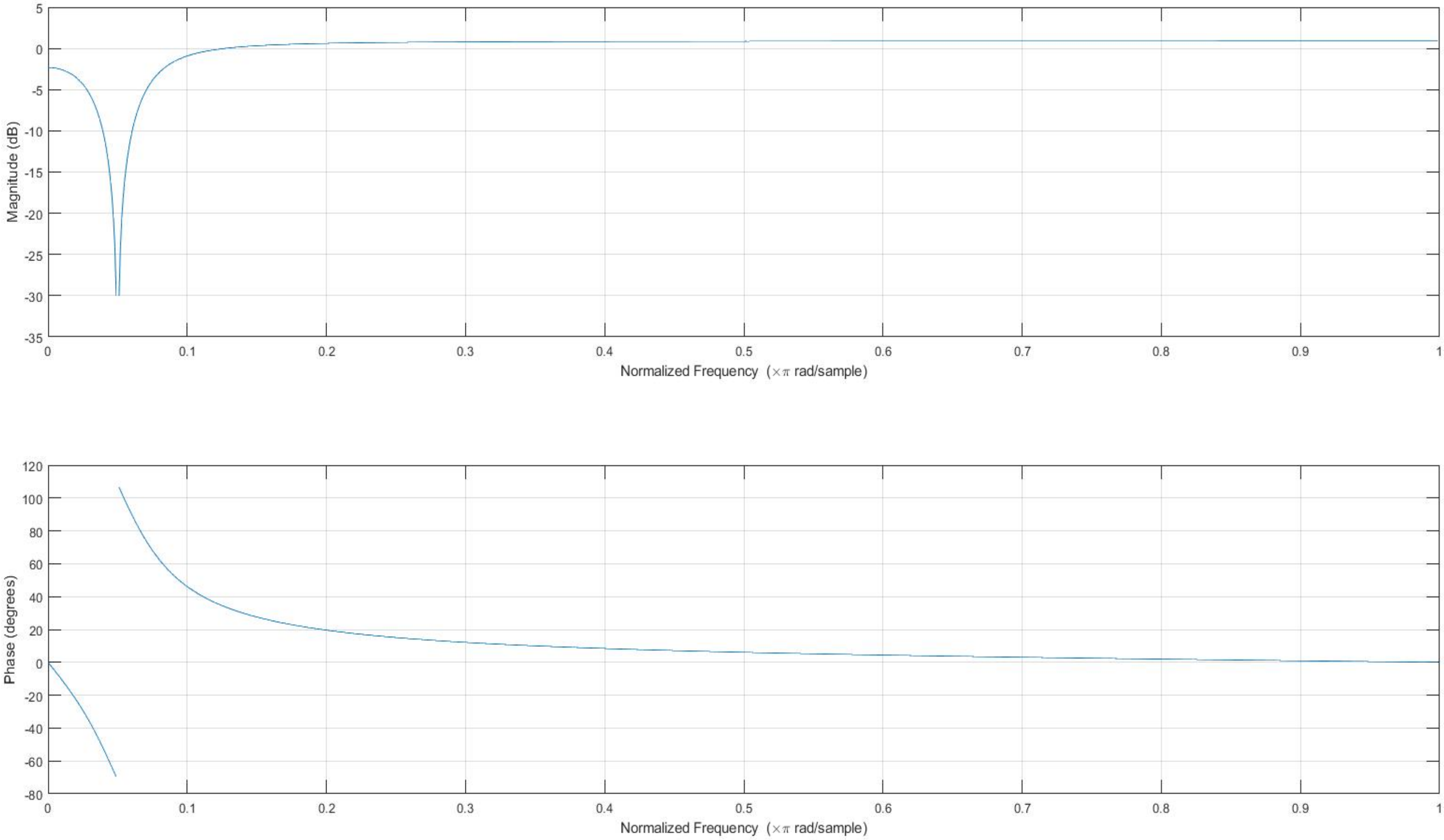

Original signal with interference
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
0
0.5
1
1.5
2
2.5
$\times 10^5$



Signal after passing through filter
0.4
0.35
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
-0.1
0
0.5
1
1.5
2
2.5
×10^5



Signal after passing through filter
0.2
0.15
0.1
0.05
0
-0.05
-0.1
2
4
6
8
10
12
14
16
×10^4



We have learned how to design IIR and FIR filters applying different methods and examined their characteristics and properties.

Appendix:

Code:

Task 1:

```
x1=0.9*exp(j*0.2*pi);
x2=0.9*exp(-j*0.2*pi);
Roots = [x1,x2];
A=poly(Roots);
B=1;

dlt=zeros(1,64);
dlt(1)=1;
X=dlt;
Y=filter(B, A, X);

%impulse response plot
figure(1)
plot(Y);
title('impulse response plot');

%frequency response
N=1000;
figure(2)
freqz(B,A,N);
title('frequency response');

%group delay
figure(3)
grpdelay(B,A);
title('group delay');

%responses for sine waves of different frequencies
 Fs = 4800;                     % samples per second
   dt = 1/Fs;                     % seconds per sample
   StopTime = 0.25;               % seconds
   t = (0:dt:StopTime-dt)';
```

```
   %Fc/Fs may be equal to 0.01 to start with
   Fc=0.01*Fs;
X=sin(2*pi*Fc*t);
Y=filter(B, A, X);

figure(4)
plot(Y);
title('Frequency is 0.01*4800 Hz');

Fc=0.10*Fs;
X=sin(2*pi*Fc*t);
Y=filter(B, A, X);
figure(5);
plot(Y);
title('Frequency is 0.10*4800 Hz');

Fc=0.2*Fs;
X=sin(2*pi*Fc*t);
Y=filter(B, A, X);
figure(6);
plot(Y);
title('Frequency is 0.2*4800 Hz');
```

Task 2:

```
x1=0.9*exp(j*0.8*pi);
x2=0.9*exp(-j*0.8*pi);
Roots = [x1,x2];
A=poly(Roots);
B=1;

dlt=zeros(1,64);
dlt(1)=1;
X=dlt;
Y=filter(B, A, X);

%impulse response plot
figure(1)
plot(Y);
title('impulse response plot');

%frequency response
```

```
N=1000;
figure(2)
freqz(B,A,N);
title('frequency response');

%group delay
figure(3)
grpdelay(B,A);
title('group delay');
```

Task 3:

```
F=[0 0.2 0.6 1];
M=[1 1 0 0];
N=50;

B = firpm(N,F,M);
%impulse response
figure(1)
stem(B);
title('impulse response');

%frequency response
figure(2)
A=1;
N=1000;
freqz(B,A,N);
title('frequency response');


%narrower transition
F=[0 0.2 0.4 1];
M=[1 1 0 0];
N=50;


B = firpm(N,F,M);
%impulse response
figure(3)
stem(B);
```

```
title('impulse response narrower band');

%frequency response
figure(4)
A=1;
N=1000;
freqz(B,A,N);
title('frequency response narrower band');
```

TASK 4:

```
%B = fir2(N,F,M,win); - N, F, M as in remez; win - chosen window (e.g
bartlett(N+1))
%Implemented windows: blackman boxcar butter chebwin hamming hanning
kaiser
%BOXCAR
F=[0 0.2 0.6 1];
M=[1 1 0 0];
N=50;
win=boxcar(N+1);

B = fir2(N,F,M,win);
%impulse response
figure(1)
stem(B);
title('impulse response boxcar');

%frequency response
figure(2)
A=1;
N=1000;
freqz(B,A,N);
title('frequency response boxcar');

%HAMMING
F=[0 0.2 0.6 1];
M=[1 1 0 0];
N=50;
win1=hamming(N+1);

B = fir2(N,F,M,win1);
%impulse response
figure(3)
stem(B);
title('impulse response hamming');
```

```
%frequency response
figure(4)
A=1;
N=1000;
freqz(B,A,N);
title('frequency response hamming');
```

TASK; 5

```
Wp=0.2*pi;
Ws=0.3*pi;
Rp=0.5;
Rs=60;
%Chebyshev type 1
[n, Wc] =cheb1ord(Wp, Ws, Rp, Rs);
[B,A]=cheby1(n, Rp, Wc);
figure(1);
zplane(B,A);
title ('zeroes and poles for Chebyshev type 1');
figure (2);
N=1000;
freqz(B,A,N);
title ('frequency response for Chebyshev type 1');

%Cauer(eleptical)
[n, Wc]=ellipord(Wp, Ws, Rp, Rs);
[B,A]=ellip(n, Rp, Rs, Wc);
figure (3);
zplane(B,A);
title ('zeroes and poles for Cauer(elliptical)');
figure (4);
N=1000;
freqz(B,A,N);
title ('frequency response for Cauer(elliptical)');
```