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SPRING 2020-2021
DUE: APRIL 2 - 13:40

EE433 REAL-TIME APPLICATIONS OF DIGITAL SIGNAL PROCESSING
EXPERIMENT 2 - PRELIMINARY WORK

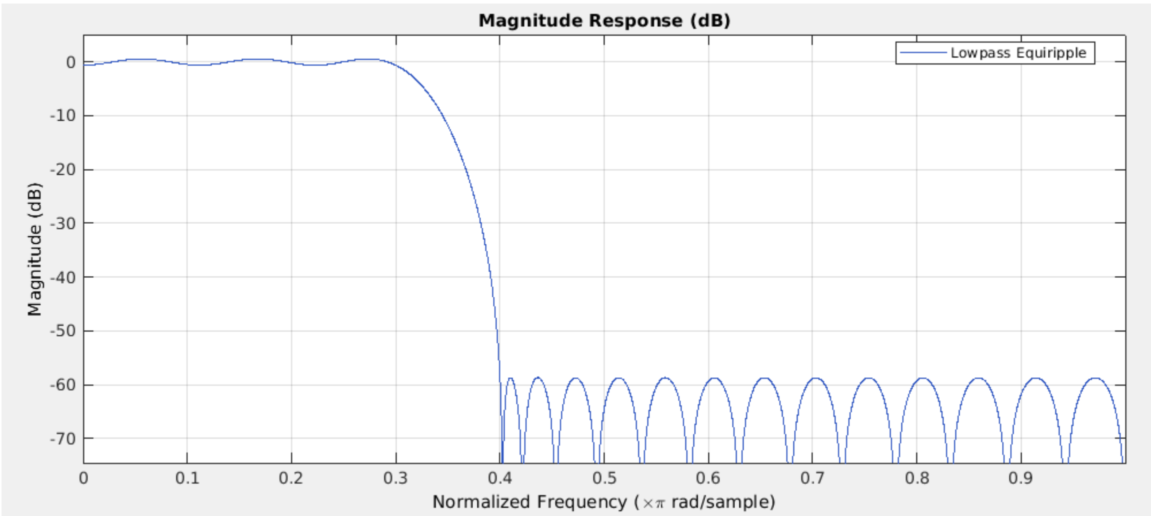
1. Figures showing magnitude response, phase response, pole-zero plot, and filter coefficients are shown respectively for both FIR lowpass equiripple and IIR Chebyshey Type II, which is also indicated by legends on the plots.

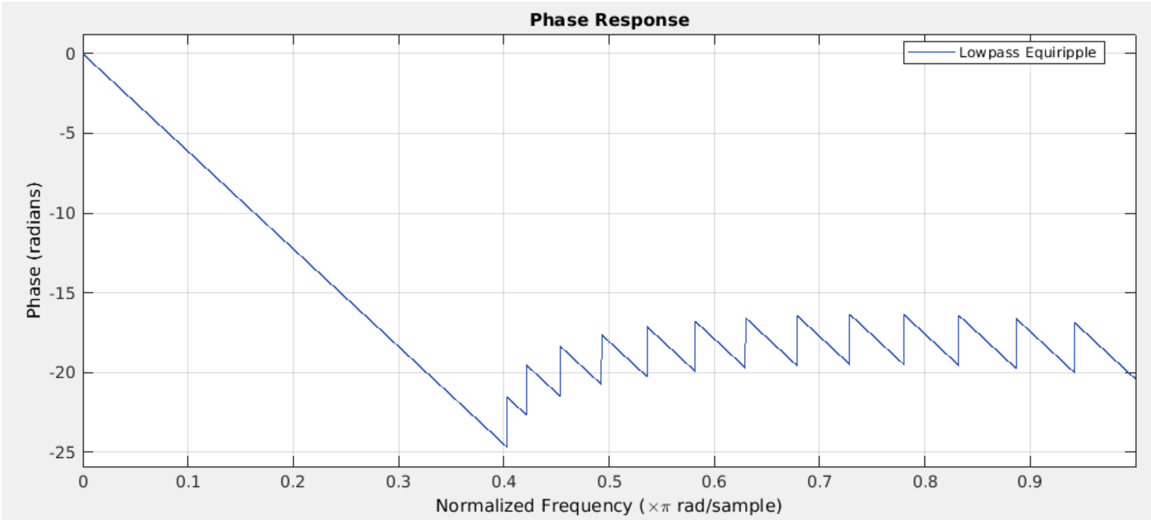
   Observations for FIR lowpass equiripple:

   - Its length is 39, indeed a finite impulse response filter.
   - Lowpass characteristics are observed from the magnitude response as expected.
   - It can be seen from impulse response that it is a symmetric filter.
   - By phase response, it can be concluded that it is a linear-phase filter.
   - From pole-zero plot, we observe one pole at zero, which was expected since it is a FIR filter. Furthermore, we see that zeros are accumulated around -1, which indicates high frequency components are eliminated.

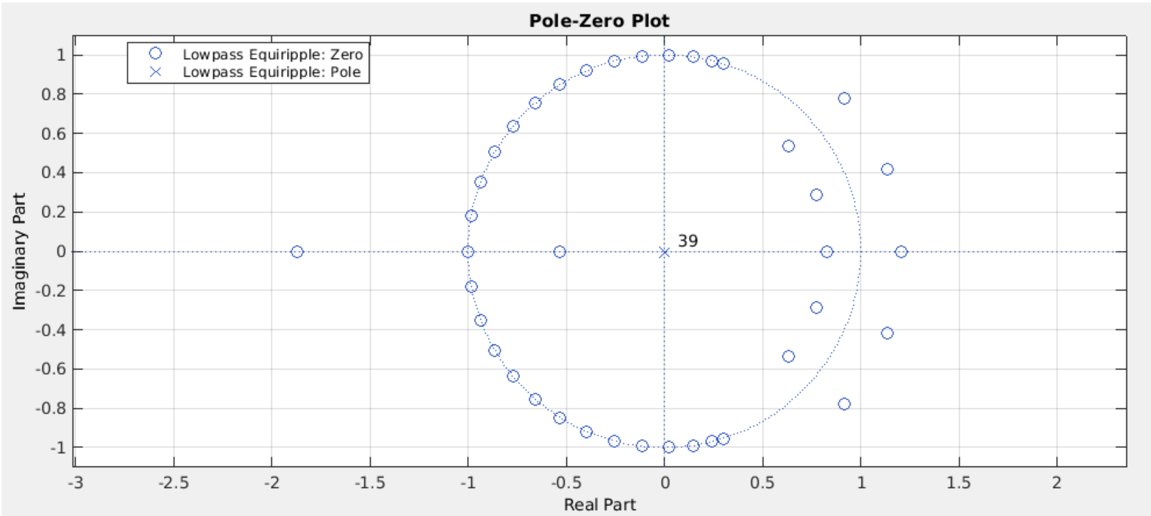
   Observations for IIR Chebyshey Type II:

   - Minimum order of IIR filter is 10, while minimum order of FIR filter is 39.
   - From the magnitude response, there can be spotted 5 frequencies where the response is zero. Since magnitude response is symmetric, 10 zeros should be observed in pole-zero plot, where we can also see 10 zeros.
   - Furthermore, for pole-zero plot, we see that all poles are different from zero, which is expected since it is an IIR filter.

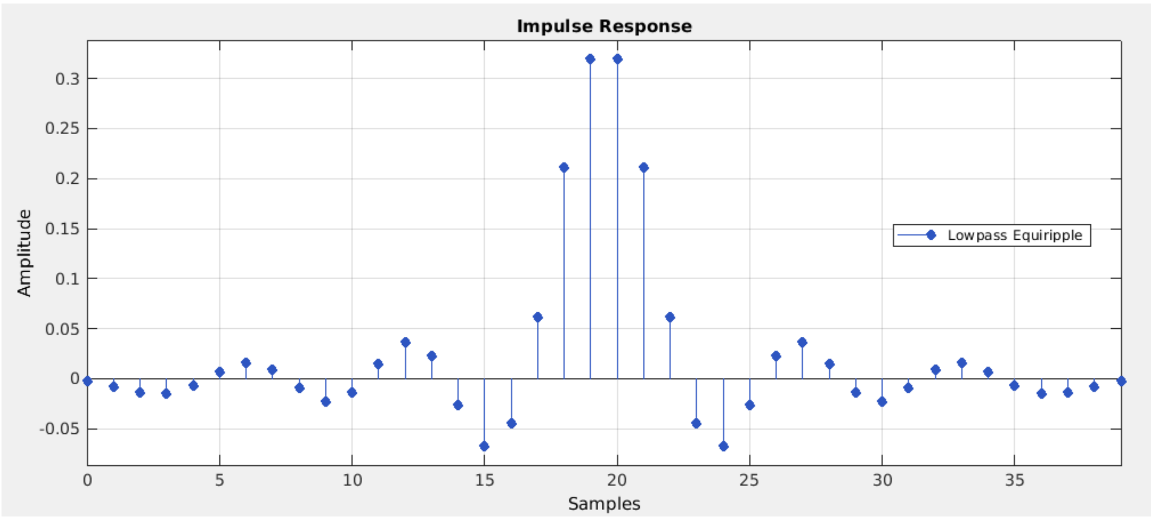




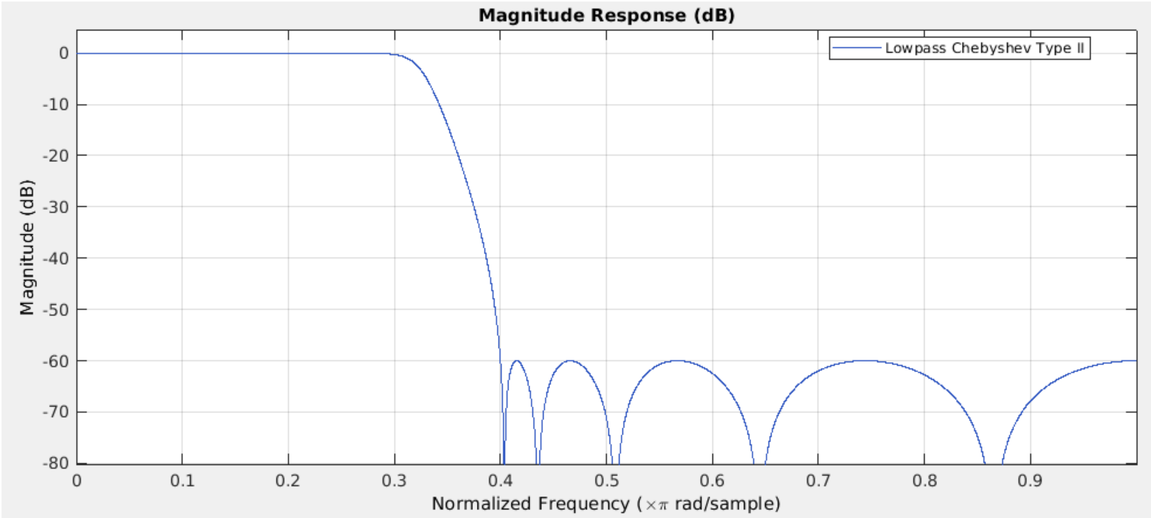
Phase Response
Lowpass Equiripple
Phase (radians)
Normalized Frequency (×π rad/sample)

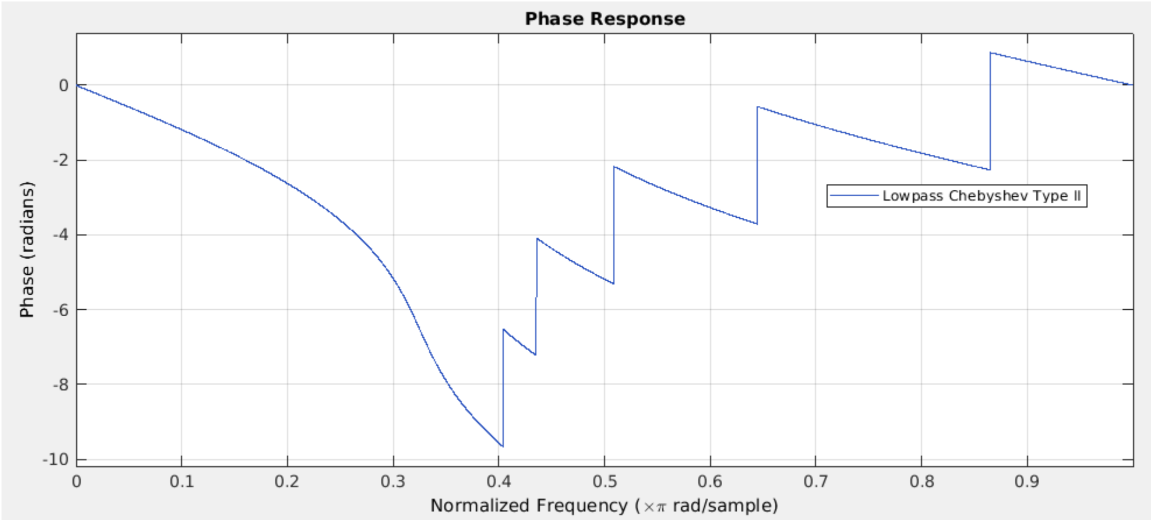


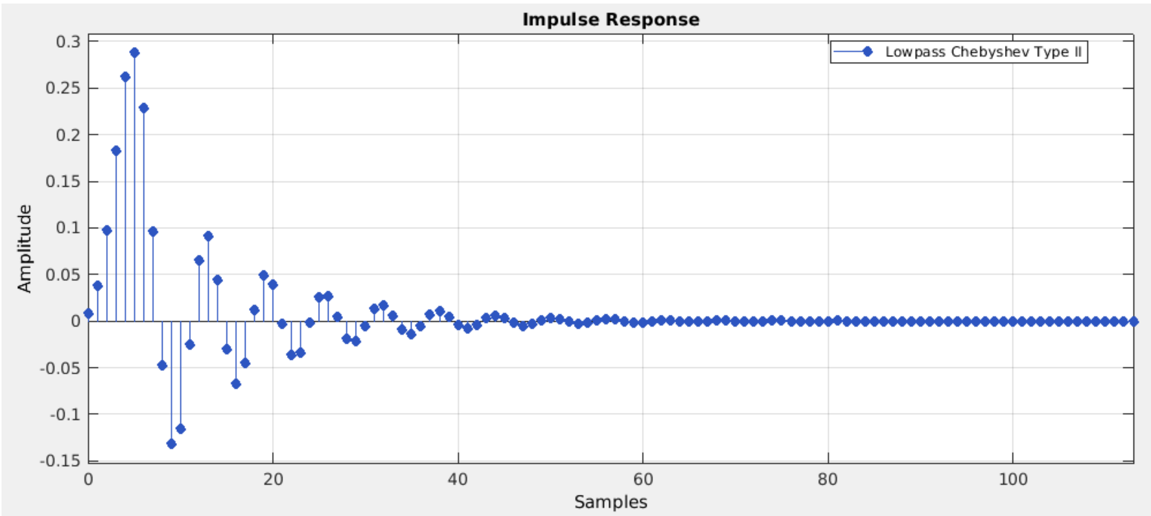
Pole-Zero Plot
Lowpass Equiripple: Zero
Lowpass Equiripple: Pole
39
Imaginary Part
Real Part

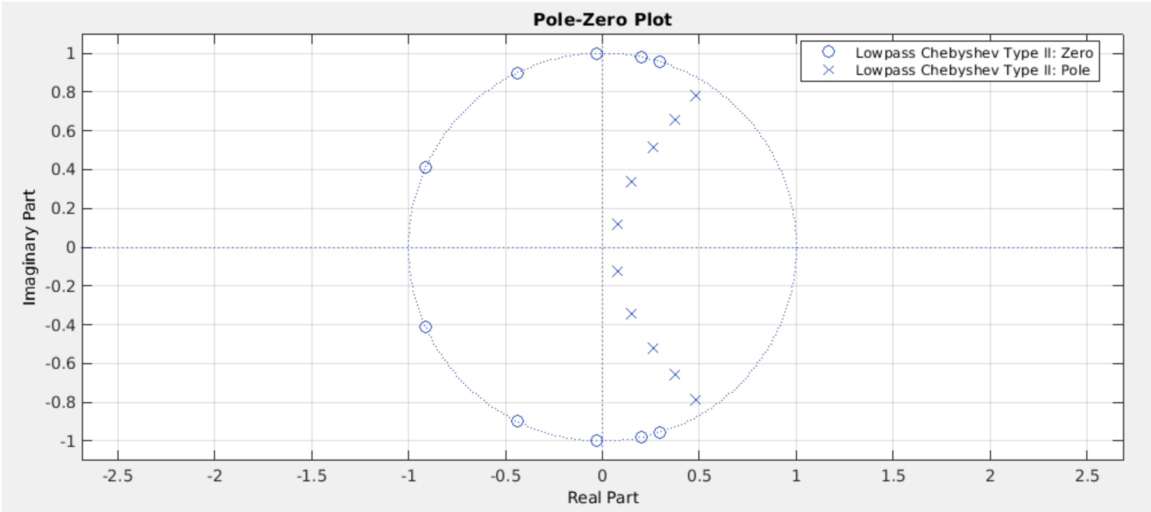


Impulse Response
Lowpass Equiripple
Amplitude
Samples


Magnitude Response (dB)
Lowpass Chebyshev Type II
Magnitude (dB)
0
-10
-20
-30
-40
-50
-60
-70
-80
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9
Normalized Frequency (×π rad/sample)


Phase Response
Lowpass Chebyshev Type II
Phase (radians)
0
-2
-4
-6
-8
-10
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9
Normalized Frequency (×π rad/sample)


Impulse Response
Lowpass Chebyshev Type II
Amplitude
0.3
0.25
0.2
0.15
0.1
0.05
0
-0.05
-0.1
-0.15
0 20 40 60 80 100
Samples


Pole-Zero Plot
Lowpass Chebyshev Type II: Zero
Lowpass Chebyshev Type II: Pole
Imaginary Part
Real Part
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
-2.5
-2
-1.5
-1
-0.5
0
0.5
1
1.5
2
2.5

2. MATLAB code is as follows:

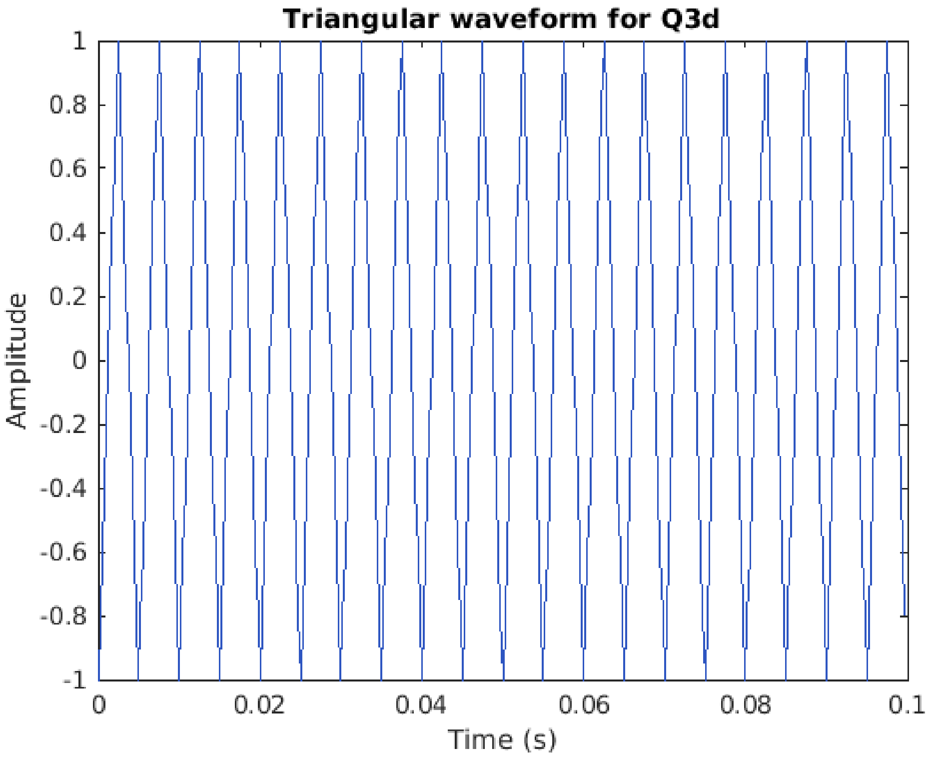
```
clear; clc; close all;

t = 0:1/4000:0.1-1/4000;
x = sawtooth(2*pi*200*t,0.5);
dftx = my_dft(x);

% FIR
fir = [1 1];
y1 = my_conv(x,fir);
dfty1 = my_dft(y1);

% IIR
y2 = 0;
for i = 1:length(x)
    y2(i+1) = y2(i) + x(i);
end
dfty2 = my_dft(y2);

figure
plot(t,x)
title('Triangular waveform for Q3d')
xlabel('Time (s)')
ylabel('Amplitude')
figure
plot(abs(dftx))
title('The magnitude response of the input')
xlabel('Frequency Bins')
ylabel('Amplitude')

t = 0:1/4000:0.1;
figure
plot(t,y1);
title('The output of the FIR filter')
xlabel('Time (s)')
ylabel('Amplitude')
figure
plot(abs(dfty1))
title('The magnitude response of the output for FIR filter')
xlabel('Frequency Bins')
ylabel('Amplitude')
figure
plot(t,y2);
title('The output of the IIR filter')
xlabel('Time (s)')
ylabel('Amplitude')
figure
plot(abs(dfty2))
title('The magnitude response of the output for IIR filter')
xlabel('Frequency Bins')
ylabel('Amplitude')
```

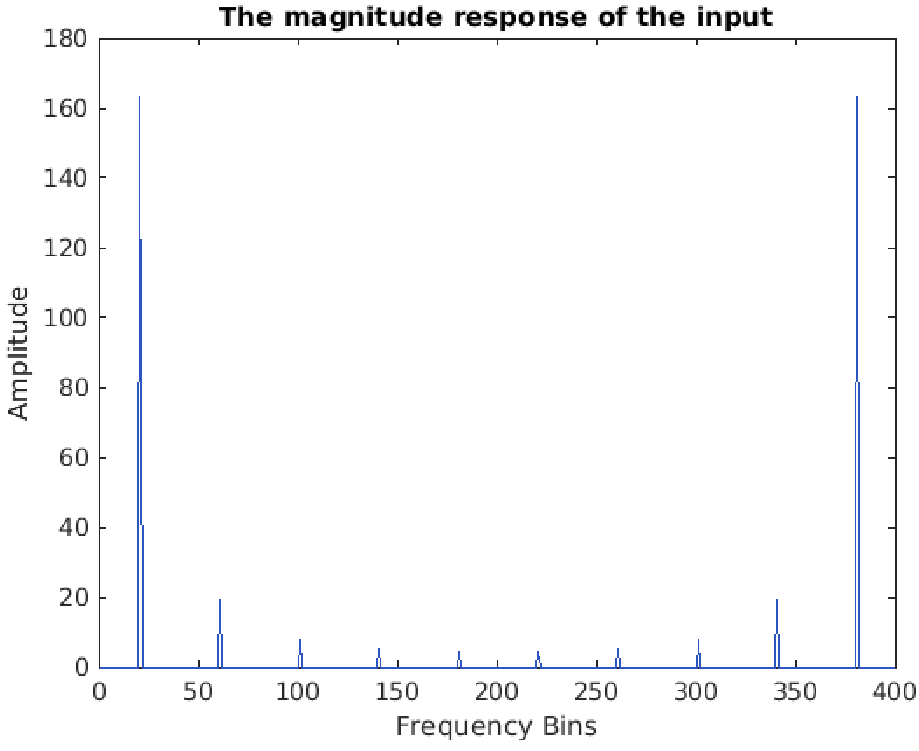
```
function y = my_dft(x)
N = length(x);
w = exp(-2*pi*1i/N);
W = zeros(N);
for n = 0:N-1
    for m = 0:N-1
        W(m+1,n+1) = w^(m*n);
    end
end
y = W*x.';
end

function y = my_conv(x,h)
m = length(x);
n = length(h);
x = [x,zeros(1,n)];
h = [h,zeros(1,m)];
y = zeros(1,m+n-1);
for i = 1:m+n-1
    y(i) = 0;
    for j = 1:m
    if((i+1)>j)
        y(i) = y(i)+x(j)*h(i-j+1);
    end
    end
end
end
```

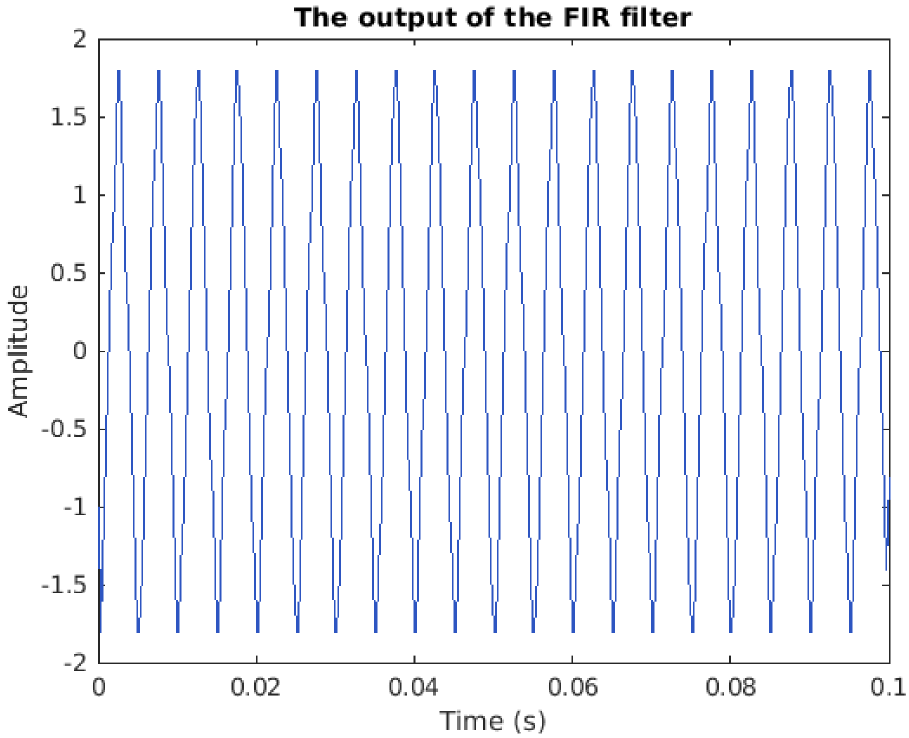
Figures showing both waveforms and their magnitude responses are given for input waveform, the output waveform of FIR filter, and the output waveform of IIR filter respecively.

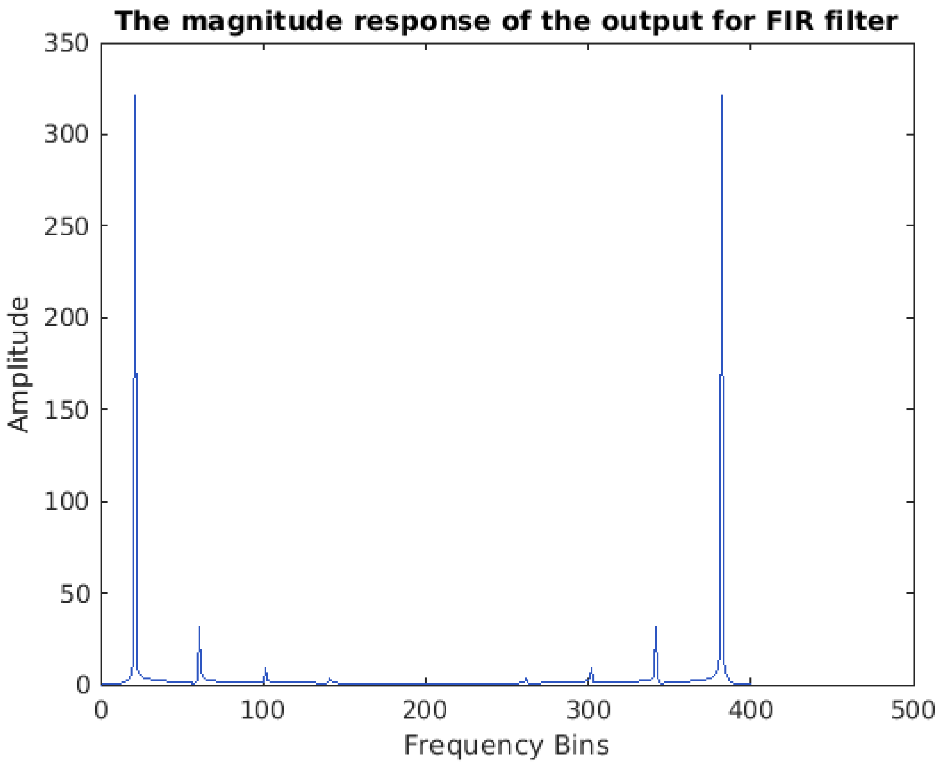
Steps for implementing first-order FIR and IIR lowpass filters:

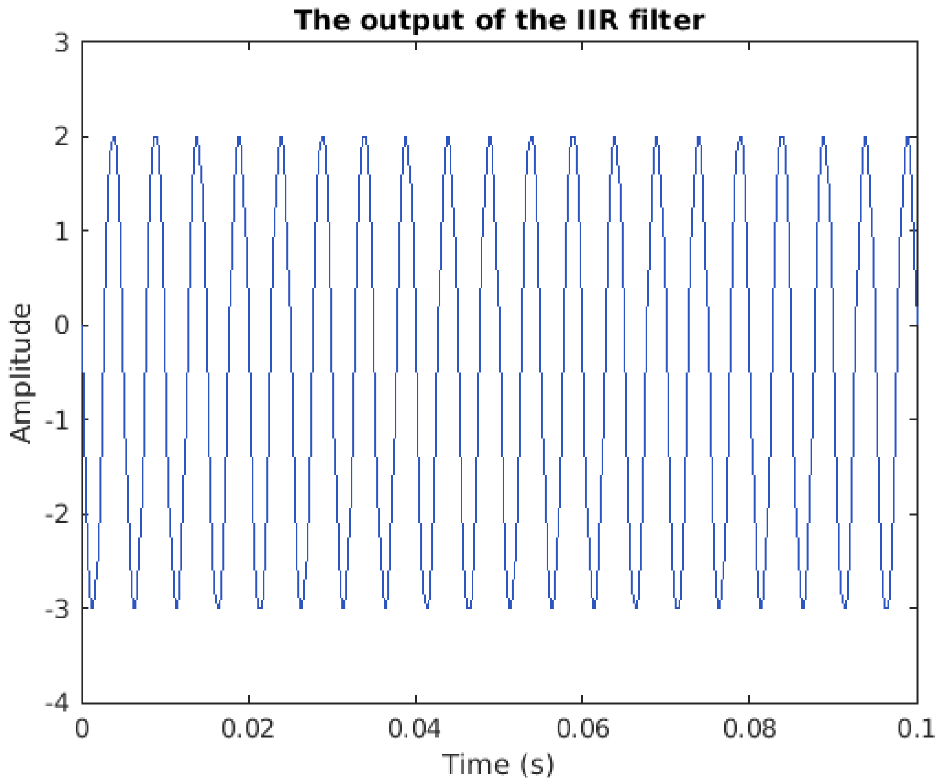
- Dicrete fourier transform in matrix form and convolution are implemented in MATLAB, since the built-in functions can not be used.
- The desired input, which was a triangular waveform where the amplitude is 1 and frequency is 200Hz with sampling frequency 4000 samples/sec, is obtained and plotted with its magnitude response. The magnitude response is obtained with the written DFT function.
- The convolution is used to compute the output of FIR filter. Multiplication in Fourier domain could be selected for this problem, however, I preferred to use convolution since I had an already written function of convolution for EE430. In this procedure, high frequency components are eliminated by lowpass FIR filter as can be seen from the plot titles as "The magnitude response of the output for FIR filter".
- After selecting the pole location, by using z-transform, the difference equation is found and implemented. Initial value is randomly selected. Again, it can be seen that high frequency components are eliminated. In its waveform, we observe more smooth transition unlike triangular waveform.

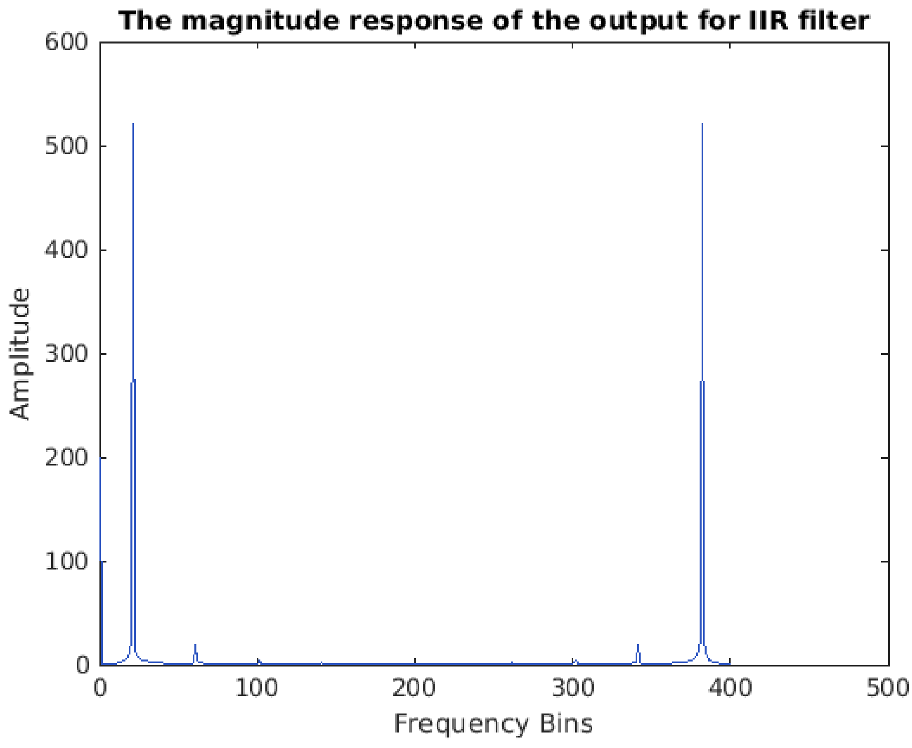

Triangular waveform for Q3d
Amplitude
Time (s)
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0
0.02
0.04
0.06
0.08
0.1


The magnitude response of the input
Amplitude
Frequency Bins
180
160
140
120
100
80
60
40
20
0
0
50
100
150
200
250
300
350
400


The output of the FIR filter
Amplitude
Time (s)
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
0.02
0.04
0.06
0.08
0.1


The magnitude response of the output for FIR filter
Amplitude
Frequency Bins
350
300
250
200
150
100
50
0
0
100
200
300
400
500


The output of the IIR filter
Amplitude
Time (s)
3
2
1
0
-1
-2
-3
-4
0
0.02
0.04
0.06
0.08
0.1


The magnitude response of the output for IIR filter
Amplitude
Frequency Bins
600
500
400
300
200
100
0
0
100
200
300
400
500