EN2063 - SIGNALS and SYSTEMS

Project

Thilakarathne D.L.J - 200650U

Calculation of digital filter specifications

Index = 200650U

Comparing with 200ABC·

A = 6 B = 5 C = 0 · = U

Maximum passband ripple, $\tilde{A}_p = 0.1600$ dB

Minimum stopband attenuation, $\tilde{A}_a = 55$ dB

Lower passband edge, $\Omega_{p1} = 400$ rad/s

Upper passband edge, $\Omega_{p2} = 900$ rad/s

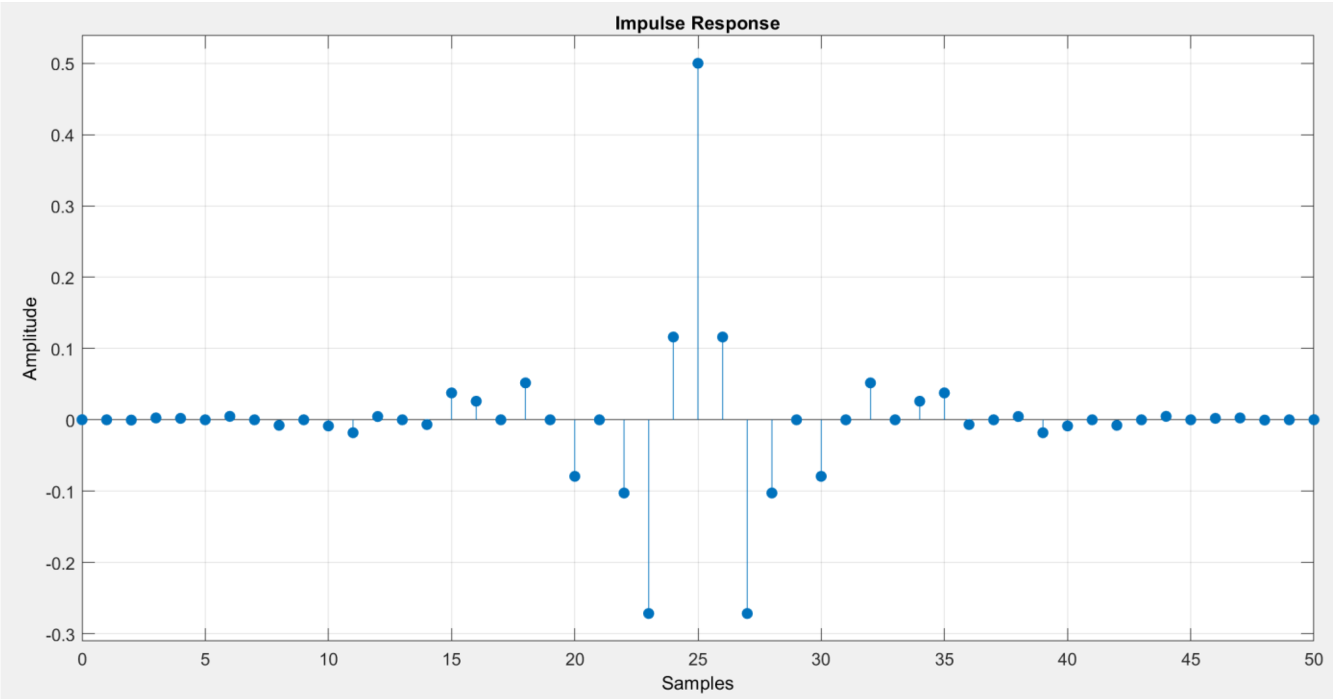
Lower stopband edge, $\Omega_{s1} = 100$ rad/s

Upper stopband edge, $\Omega_{s2} = 1100$ rad/s

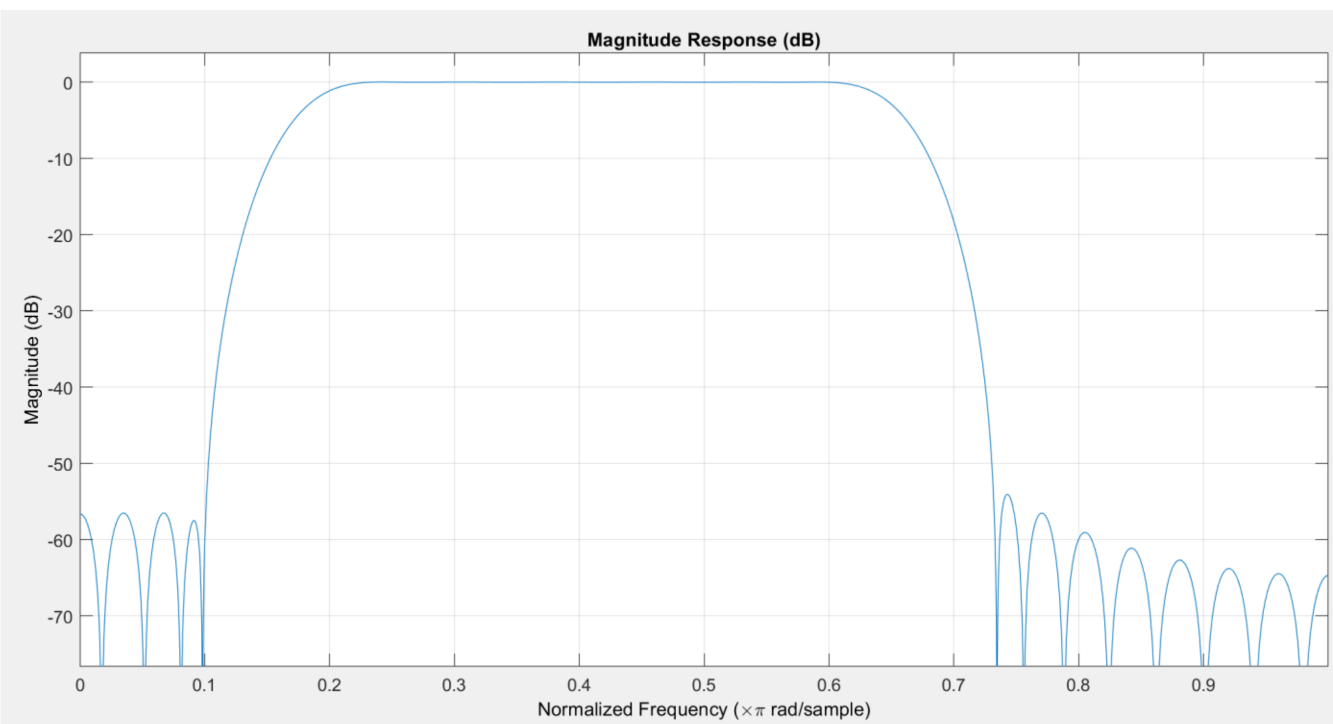
Sampling frequency, $\Omega_{sm} = 3000$ rad/s

Question 1

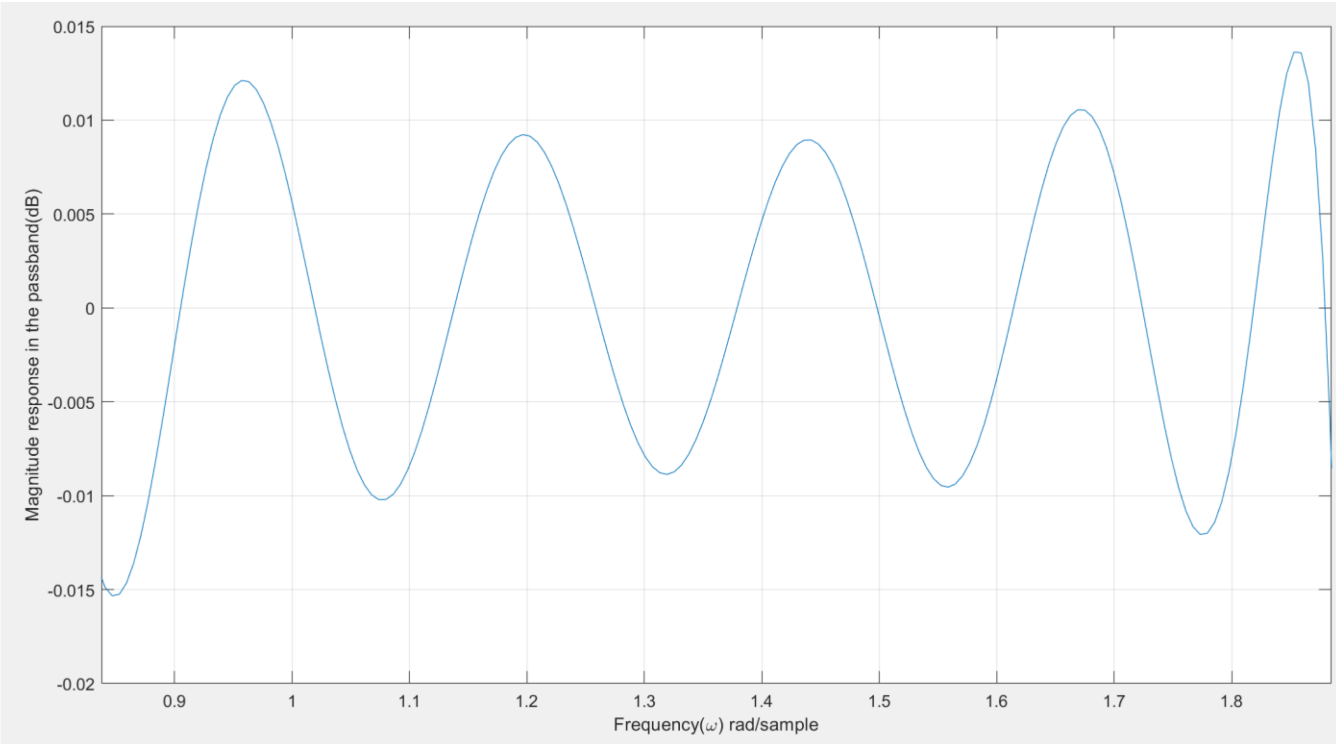
a) Impulse response



b) Magnitude response of the digital filter



c) Magnitude response in the passband



Following are the information relating to this FIR filter,

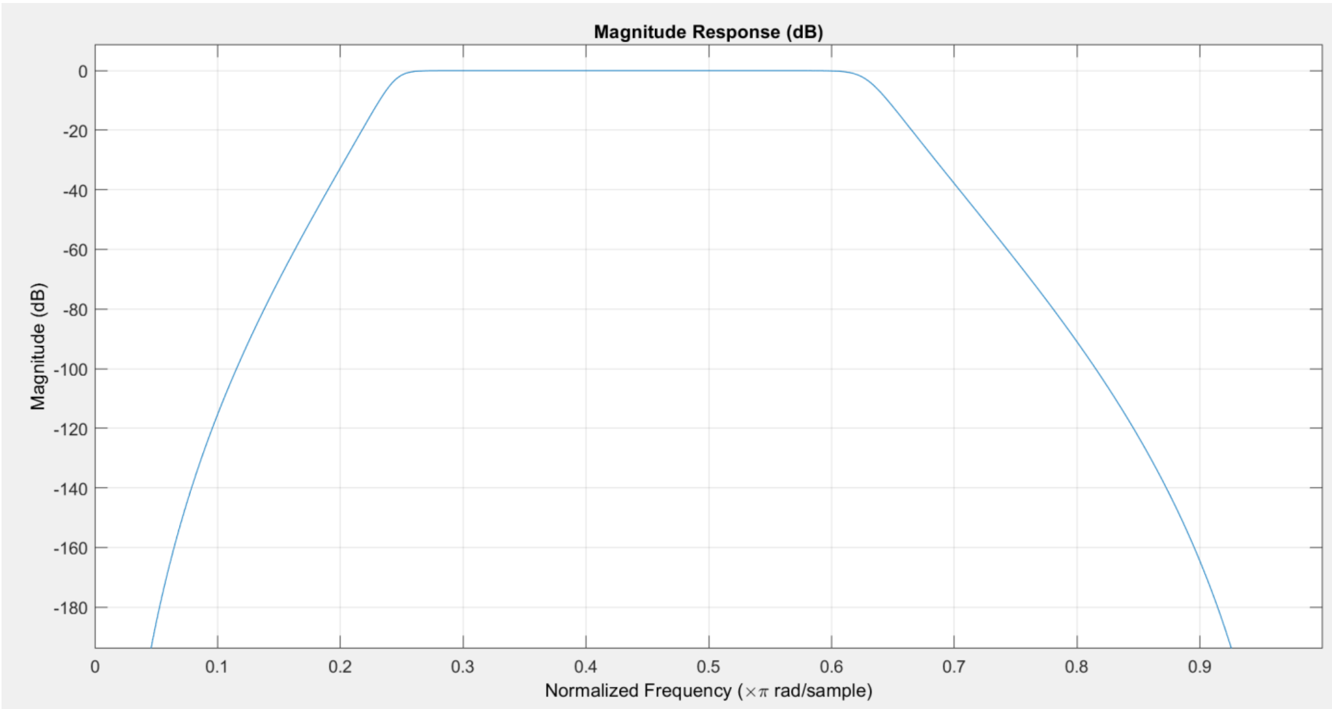
```
Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure     : Direct-Form II Transposed
Numerator Length     : 51
Denominator Length   : 1
Stable               : Yes
Linear Phase         : Yes (Type 1)

Implementation Cost
Number of Multipliers            : 51
Number of Adders                 : 50
Number of States                 : 50
Multiplications per Input Sample : 51
Additions per Input Sample       : 50
```

Question 2

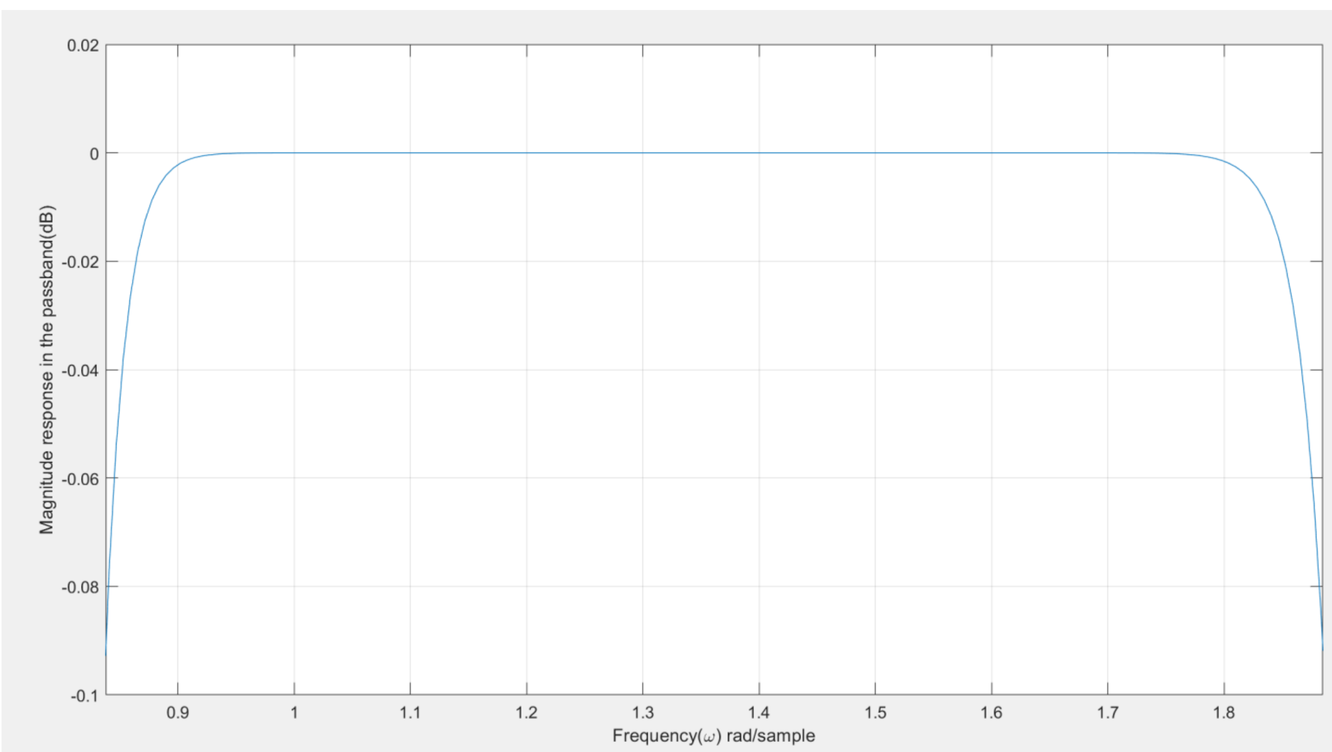
a) Coefficients of the transfer function of the IIR filter

Numerator	Denominator
0.0002	1.0000
0.0000	-3.2535
-0.0018	7.4395
0.0000	-12.6518
0.0089	18.9141
0.0000	-24.1157
-0.0268	27.6710
0.0000	-28.1470
0.0537	26.2539
0.0000	-22.0749
-0.0751	17.0864
0.0000	-11.9774
0.0751	7.7417
-0.0000	-4.5184
-0.0537	2.4203
-0.0000	-1.1574
0.0268	0.5037
-0.0000	-0.1908
-0.0089	0.0647
-0.0000	-0.0180
0.0018	0.0044
-0.0000	-0.0007
-0.0002	0.0001

b) Magnitude response of the digital filter



c) Magnitude response in the passband



Following are the information relating to this IIR filter,

```
Discrete-Time IIR Filter (real)
-------------------------------
Filter Structure     : Direct-Form II Transposed
Numerator Length     : 23
Denominator Length   : 23
Stable               : Yes
Linear Phase         : No

Implementation Cost
Number of Multipliers            : 45
Number of Adders                 : 44
Number of States                 : 22
Multiplications per Input Sample : 45
Additions per Input Sample       : 44
```

<u>Question 3</u>

Order of the designed FIR and IIR filters are respectively, 51 and 23.

According to the filter information panels for both filters,

```
Discrete-Time FIR Filter (real)
-------------------------------
Filter Structure    : Direct-Form II Transposed
Numerator Length    : 51
Denominator Length  : 1
Stable              : Yes
Linear Phase        : Yes (Type 1)

Implementation Cost
Number of Multipliers            : 51
Number of Adders                 : 50
Number of States                 : 50
Multiplications per Input Sample : 51
Additions per Input Sample       : 50
```

```
Discrete-Time IIR Filter (real)
-------------------------------
Filter Structure    : Direct-Form II Transposed
Numerator Length    : 23
Denominator Length  : 23
Stable              : Yes
Linear Phase        : No

Implementation Cost
Number of Multipliers            : 45
Number of Adders                 : 44
Number of States                 : 22
Multiplications per Input Sample : 45
Additions per Input Sample       : 44
```

Both number of additions and multiplications for the IIR filter is less than for the FIR filter.

MATLAB codes

Question 1

```
%-------------------------- index = 200650U ------------------------------
A = 6;
B = 5;
C = 0;

%----------- calculation of digital filter specifications -----------------

maxPassbandRipple = 0.1 + (0.01 * A); %dB
minStopbandAttenuation = 50 + B; %dB
lowerPassbandEdge = (C * 100) + 400; %rad/s
upperPassbandEdge = (C * 100) + 900; %rad/s
lowerStopbandEdge = (C * 100) + 100; %rad/s
upperStopbandEdge = (C * 100) + 1100; %rad/s
samplingFrequency = 2*((C * 100) + 1500); %rad/s

%----------------------------- question_1 ---------------------------------

sampFreq = samplingFrequency/(2*pi); %sampling frequency in Hz
frequencyEdges = [lowerStopbandEdge lowerPassbandEdge upperPassbandEdge
upperStopbandEdge]/sampFreq;
magnitudes = [0 1 0];
deviations = [1/(10^(minStopbandAttenuation/20)) 1/(10^(maxPassbandRipple/20))
1/(10^(minStopbandAttenuation/20))];

[n, Wn, beta, ftype] = kaiserord(frequencyEdges, magnitudes, deviations, 2*pi);
filterCoefficients = fir1(n, Wn, ftype, kaiser(n+1,beta), 'noscale');

%GUI which plots the digital filter
fvtool(filterCoefficients, 1)

%computing the frequency response vector - "magnitude" and the corresponding angular
frequency vector - "phase"
[magnitude, phase] = freqz(filterCoefficients, 1);

%------------------magnitude response in the passband----------------------
plot(phase, 20*log10(abs(magnitude)))
xlim([lowerPassbandEdge upperPassbandEdge]*(2*pi/samplingFrequency))
xlabel('Frequency(\omega) rad/sample')
ylabel('Magnitude response in the passband(dB)')
grid on
```

Question 2

```
%-------------------------- index = 200650U -------------------------------
A = 6;
B = 5;
C = 0;
D = 0;
%therefore the IIR filter approximation method is Butterworth.

%----------- calculation of digital filter specifications -----------------

maxPassbandRipple = 0.1 + (0.01 * A); %dB
minStopbandAttenuation = 50 + B; %dB
lowerPassbandEdge = (C * 100) + 400; %rad/s
upperPassbandEdge = (C * 100) + 900; %rad/s
lowerStopbandEdge = (C * 100) + 100; %rad/s
upperStopbandEdge = (C * 100) + 1100; %rad/s
samplingFrequency = 2*((C * 100) + 1500); %rad/s

sampFreq = samplingFrequency/(2*pi); %sampling frequency in Hz

passbandEdges = [lowerPassbandEdge upperPassbandEdge]/(sampFreq);
stopbandEdges = [lowerStopbandEdge upperStopbandEdge]/(sampFreq);

%prewarping the critical frequencies
passbandEdges = 2 / (1/sampFreq) * tan(passbandEdges/2);
stopbandEdges = 2 / (1/sampFreq) * tan(stopbandEdges/2);

[minimumOrder, cutoffFrequencies] = buttord(passbandEdges, stopbandEdges, maxPassbandRipple,
minStopbandAttenuation, 's');
[numerator, denominator] = butter(minimumOrder, cutoffFrequencies, 's');

%Digital filter coefficients
[numeratorDiscrete, denominatorDiscrete] = bilinear(numerator, denominator, sampFreq);

%GUI which plots the digital filter
fvtool(numeratorDiscrete, denominatorDiscrete)

%computing the frequency response vector - "magnitude" and the corresponding angular
frequency vector - "phase"
[magnitude,phase]=freqz(numeratorDiscrete,denominatorDiscrete);

%-------------------magnitude response in the passband---------------------
plot(phase, 20*log10(abs(magnitude)))
xlim([lowerPassbandEdge upperPassbandEdge]*(2*pi/samplingFrequency))
xlabel('Frequency(\omega) rad/sample')
ylabel('Magnitude response in the passband(dB)')
grid on
```