UNIVERSITY OF MORATUWA, SRI LANKA
Faculty of Engineering
Department of Electronic and Telecommunication Engineering
Semester 3 (Intake 2020)

EN2063—SIGNALS AND SYSTEMS

Project

The objective of this project is to provide experience in the design of FIR and IIR digital filters for prescribed specifications. For FIR filters, the *windowing* method (in conjunction with the Kaiser window) is used whereas, for IIR filters, the *bilinear transformation* method is used.

The specifications of the digital filter are different from student to student, and are derived using the index numbers of the students. Let us denote the index number as $200ABC\cdot$, where A, B and C are integers in the range 0 to 9, and $\cdot$ is a letter from the English alphabet.

Table 1: Filter specifications.

Parameter	Value
Maximum passband ripple, $\tilde{A}_p$	$0.1 + (0.01 \times A)$ dB
Minimum stopband attenuation, $\tilde{A}_a$	$50 + B$ dB
Lower passband edge, Ω_{p1}	$(C \times 100) + 400$ rad/s
Upper passband edge, Ω_{p2}	$(C \times 100) + 900$ rad/s
Lower stopband edge, Ω_{s1}	$(C \times 100) + 100$ rad/s
Upper stopband edge, Ω_{s2}	$(C \times 100) + 1100$ rad/s
Sampling frequency, Ω_{sm}	$2((C \times 100) + 1500)$ rad/s

1) Using the windowing method in conjunction with the Kaiser window, design an FIR bandpass digital filter that will satisfy the specifications given in Table 1.

   a) Plot the impulse response.

   b) Plot the magnitude response of the digital filter for $\pi \leq \omega < \pi$ rad/sample.

   c) Plot the magnitude response for $\omega_{p1} \leq \omega \leq \omega_{p2}$ (in the passband), where ω_{p1} and ω_{p2} are the passband edges in the discrete-time angular frequency domain.

2) Using the bilinear transformation method, design an IIR bandpass digital filter that will satisfy the specifications given in Table 1. Here, you need to first design an appropriate analog filter, and the required digital filter should be obtained by applying the bilinear transform to the transfer function of the analog filter. Note that, *prewarping* of frequencies is essential in order to obtain the required digital filter. The approximation method (or the type) of the IIR filter is determined according to your index number as follows. Let D be the remainder after dividing the digit C of your index number by 4. Then the approximation method should be selected as presented in Table 2.

Table 2: IIR filter approximation method.

D	Approximation method
0	Butterworth
1	Chebyshev
2	Inverse-Chebyshev
3	Elliptic

a) Tabulate the coefficients of the transfer function of the IIR filter.

b) Plot the magnitude response of the digital filter for $\pi \leq \omega < \pi$ rad/sample.

c) Plot the magnitude response for $\omega_{p1} \leq \omega \leq \omega_{p2}$ (in the passband), where ω_{p1} and ω_{p2} are the passband edges in the discrete-time angular frequency domain.

3) Compare the order and the number of multiplications and additions required to process a sample by the two designed filters. Assume that the two filters are implemented using the difference equations, and the symmetry of coefficients can be exploited to reduce the number of multiplications.

- Write a report describing your results. Also, include your MATLAB or Python programs in an appendix. Submit a soft copy of the report by **11:59 pm, Sunday January 15, 2023**.

Useful MATLAB functions:

- plot: plots a function
- stem: plots a discrete-time function
- kaisrerord: computes the order of an FIR filter and the independent parameter to specify a Kaiser window
- kaiser: computes a Kaiser window
- fir1: calculates the coefficients of FIR filters designed using the windowing method
- buttord: calculates the minimum order of an analog or a digital Butterworth filter required to satisfy given specifications
- butter: calculates the coefficients of the transfer functions of analog or IIR digital Butterworth filters
- cheb1ord: calculates the minimum order of an analog or a digital Chebyshev filter required to satisfy given specifications
- cheby1: calculates the coefficients of the transfer functions of analog or IIR digital Chebyshev filters
- cheb2ord: calculates the minimum order of an analog or a digital inverse-Chebyshev filter required to satisfy given specifications
- cheby2: calculates the coefficients of the transfer functions of analog or IIR digital inverse-Chebyshev filters
- ellipord: calculates the minimum order of an analog or a digital elliptic filter required to satisfy given specifications
- ellip: calculates the coefficients of the transfer functions of analog or IIR digital elliptic filters
- bilinear: calculates the coefficients of the transfer function of an IIR digital filter for a given transfer function of an analog filter
- impz: computes the impulse response of a digital filter
- freqs: computes the frequency, magnitude, and phase response of an analog filter
- freqz: computes the frequency, magnitude, and phase response of a digital filter
- unwrap: unwraps the phase response of a digital filter
- phasedelay: computes the phase delay of a digital filter
- grpdelay: computes the group delay of a digital filter
- fft: calculates the DFT
- fftshift: shifts zero-frequency component to the center of the spectrum
- fvtool: this is a very useful and sophisticated tool, which opens a GUI and plots impulse, step, amplitude, and phase responses, delay characteristics, and can also plot many other quantities pertaining to a digital filter.