EN2570 Digital Signal Processing - Project
Semester 4 (November 2017)

The objective of this project is to provide experience in the design of FIR digital filters for prescribed specifications using the windowing method in conjunction with the Kaiser window.

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 $140ABC\cdot$, 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, $\tilde{A}_p$	$0.05 + (0.01 \times A)$ dB
Minimum stopband attenuation, $\tilde{A}_a$	$40+B$ dB
Lower passband edge, Ω_{p1}	$(C \times 100) + 300$ rad/s
Upper passband edge, Ω_{p2}	$(C \times 100) + 850$ rad/s
Lower stopband edge, Ω_{a1}	$(C \times 100) + 400$ rad/s
Upper stopband edge, Ω_{a2}	$(C \times 100) + 700$ rad/s
Sampling frequency, Ω_s	$2[(C \times 100) + 1200]$ rad/s

1. Using the windowing method in conjunction with the Kaiser window, design an FIR *bandstop* digital filter that will satisfy the specifications given in Table 1.
2. Plot the causal impulse response.
3. Plot the magnitude response of the digital filter obtained for the frequency range 0 to $\frac{\Omega_s}{2}$ rad/s. Note that $\frac{\Omega_s}{2}$ rad/s is equivalent to π rad/sample.
4. Plot the magnitude response of the digital filter for the frequencies in the lower and upper passbands.
5. Check the operation of the filter by plotting the time-domain response of the digital filter to an excitation

$$x(nT) = \sum_{i=1}^{3} \sin(\Omega_i nT),$$

where Ω_1 is the middle frequency of the lower passband, Ω_2 is middle frequency of the stopband, and Ω_3 is middle frequency of the upper passband. Compare the output signal of the designed digital filter with the expected output signal if an ideal bandstop filter is used, i.e., one that has a gain of 1 in the passbands and 0 in the stopband. Note that 200 to 300 samples might be required to achieve a steady-state response.

6. By using the DFTs of the input and the output signals (estimated using an FFT algorithm), demonstrate that the output signal is a filtered version of the input signal and that the correct frequencies have been passed.

7. Write a report describing your results. The report should include an abstract, introduction, basic theory in brief, results, conclusions, and references where appropriate. Also, include your MATLAB programs in an appendix.

   Submit a hard copy of the report by Nov 22nd, 2017.

Note: The required filter design for this project can be done very quickly by using functions available in MATLAB (such as *fir1*) but this is not the idea of the project. You must write the MATLAB programs you need to complete your design following the steps of the windowing method covered during the lecturers. Note that you can not use the *besseli* function available in MATLAB. However, you can check your design using MATLAB functions that can be used to plot magnitude, phase, and time-domain responses or magnitude and phase spectra of signals such as FFT functions. If you have any doubt about the functions that can or cannot be used, check with the lecturer.

Useful MATLAB functions:

- plot: plots a function
- impz: computes the impulse response of a digital filter
- freqs: computes the frequency, magnitude, and phase response of an analog filter
- freqz, angle: compute the frequency, magnitude, and phase response of a digital filter
- phasedelay, grpdelay: compute the phase and group delay of a digital filter
- unwrap: unwraps the phase response of a digital filter
- 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.
- fft: calculates the DFT
- fftshift: shift zero-frequency component to the center of the spectrum