EE 338 Digital Signal Processing
Filter Design Assignment: Spring Semester 2016-2017

This is an **individual** filter design assignment for each student who has registered for the course. In the discussion below, the specification of each filter is provided according to its number. Each student who has registered for the course, for credit or audit, is required to design **two** Filters with the specifications as per the Filter Number M assigned to the student, by the Teaching Associates for this course, on the course Moodle webpage. This assignment of numbers M, will be put up on the course website by 1 March 2016.

In the filter design, you are encouraged to partly make use of SCILAB/ MATLAB or any equivalent open source software as available. You are encouraged to contact the Computer Centre/ the PC Laboratory in the Department of Electrical Engineering for more information on open source software available in the Institute. It is not mandatory to use any package, of course! You could write a small C program / high level language program as well.

For designing the IIR and FIR filters, you are NOT permitted to use “filter design” commands directly. Neither are you permitted to carry out the whole design simply by using a filter design package. You may use basic SCILAB/ MATLAB statements relating to matrix operations, window function generation, and so on. You may write small programs in MATLAB. Your final design submission must be as follows, in an electronic file to be uploaded appropriately as per instructions from Teaching Associates, in the ‘Filter Design Assignment Forum’ on the Course Moodle Page:

- Write, on top, your name, roll number and filter number M.
- The following data pertinent to each of the two filter designs assigned to you, must be submitted, in that order.

1. The un-normalized discrete time filter specifications: including whether the passband and stopband are equiripple or monotonic respectively.
2. The corresponding normalized digital filter specifications.
3. The corresponding analog filter specifications for the same type of analog filter using the bilinear transformation.
4. The frequency transformation to be employed with relevant parameters.
5. The frequency transformed lowpass analog filter specifications.
6. The analog lowpass filter transfer function $H_{analog,LPF}$ (s_L).
7. The analog transfer function for the appropriate type of filter.
8. The discrete time filter transfer function.
9. Its realization using Direct Form II..
10. An FIR Filter Transfer function for realizing the same specifications using the Kaiser Window. You may use a MATLAB statement for generating the Kaiser window coefficients directly.

As it is tedious to write out coefficients and data by hand each time, you are welcome to include a printout or electronic write-out of results/ data from a computer program/ SCILAB/ MATLAB program wherever appropriate. **Further, you must demonstrate the frequency response of the filter that you have designed in MATLAB/ any other means.** The demonstration must be made to a Teaching Associate for this course. The evaluation scheme will be displayed on the course web-page in due course. **The report is due to be submitted at least one week prior to the semester-end examination in this course**.

The Filter Specifications:

We wish to build a series of discrete time filters, as described below, to extract specific bands of this analog signal, or to suppress specific parts of the analog signal.

(i) For all filters, the passband AND stopband tolerances are 0.1 in magnitude. That is, the filter magnitude response (**note: NOT magnitude squared**) must lie between 1.1 and 0.9 in the passband; and between 0 and 0.1 in the stopband.

(ii) For bandpass filters, the transition band is 1 kHz on either side of the passband. For bandstop filters, the transition band is 1 kHz on either side of the stopband.

(iii) When realized as an IIR Filter, all these filters have a monotonic stopband. (For FIR Filters, of course, we do not have a choice of the nature of stopband / passband).

First Filter Specification: An analog signal is bandlimited to 60 kHz. It is ideally sampled, with a sampling rate of 140 kHz. The first filter to be designed by each student is a **bandpass** filter. Filter numbers 1 to 75 have a monotonic passband, whereas filter numbers 76 to 150 have an equiripple passband. For filter numbers m and 75+m; m going from 1 to 75; the passband is from $B_L(m)$ kHz to $B_H(m)$ kHz, where $B_L(m)$ and $B_H(m)$ are numbers determined from m as follows. Define:

q(m) = greatest integer strictly less than 0.1m. For example, q(5) = 0, q(30) = 2
r(m) = m – 10q(m). For example, r(5) = 5, r(30) = 10
$B_L(m) = 2 + 0.7\,q(m) + 2\,r(m)$. For example, $B_L(30) = 2 + 0.7\,(2) + 2\,(10) = 23.4$
$B_H(m) = B_L(m) + 5$.

Second Filter Specification: An analog signal is bandlimited to 40 kHz. It is ideally sampled, with a sampling rate of 90 kHz. The second filter to be designed by each student is a **bandstop** filter. Filter numbers 1 to 75 have an equiripple passband, whereas filter numbers 76 to 150 have a monotonic passband. For filter numbers m and 75+m; m going from 1 to 75; the stopband is from $B_L(m)$ kHz to $B_H(m)$ kHz, where $B_L(m)$ and $B_H(m)$ are numbers determined from m as follows. Define:

q(m) = greatest integer strictly less than 0.1m. For example, q(5) = 0, q(30) = 2
r(m) = m – 10q(m). For example, r(5) = 5, r(30) = 10
$B_L(m) = 2 + 0.6\,q(m) + 1.5\,r(m)$. For example, $B_L(30) = 2 + 0.6\,(2) + 1.5\,(10) = 18.2$
$B_H(m) = B_L(m) + 3$.

Use either **Butterworth or Chebyshev approximation** to design the **IIR Filter** as appropriate. For the **FIR Filter**, use the **Kaiser window** as stated above.

Total credit for filter design assignment: 40 marks (20 percent weight)
(End of filter design assignment)