EE 338 Digital Signal Processing

Filter Design Assignment: Spring Semester January-April 2020

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 discrete time filters, with the specifications as per the Filter Number M assigned to the student, by the Instructor or Teaching Associates for this course, on the course Moodle webpage. This assignment of numbers M, will be put up on the course website by 8 February 2020.

This assignment is in two parts. The first part (Part I) is mandatory for all students. This part shall be scored as follows: Up to 30 marks for carrying out the filter design assignment in a satisfactory manner eventually and a further 15 marks or more, as an 'early bird and teaching assistance incentive', which will be separately awarded for deserving students.

The second part (Part II) is optional. A student may attempt it and display his/ her achievement in that part to the Teaching Associates, to earn additional credit.

The scores for both these parts will count towards the class participation score for the student.

Part I (Mandatory Component)

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:

A. Write, on top, your name, roll number and filter number M.

B. 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 an 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/ 'Early Bird Peer Teaching Assistant' for this course. The evaluation scheme will be displayed on the course web-page in due course. The report is due to be submitted and evaluation completed, 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 an analog signal, or to suppress specific parts of the analog signal.

(i) For all filters, the passband AND stopband tolerances are 0.15 in magnitude. That is, the filter magnitude response (note: NOT magnitude squared) must lie between 1.15 and 0.85 in the passband; and between 0 and 0.15 in the stopband. For the IIR Filter, the passband magnitude response must lie between 1 and 0.85.

(ii) For bandpass filters, the transition band is 20 kHz on either side of the passband. For bandstop filters, the transition band is 20 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 band-limited to 500 kHz. It is ideally sampled, with a sampling rate of 1200 kHz (1.2 MHz). 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

$B_L(m) = 150 + 17\, q(m) + 13\, r(m)$.

For example, $B_L(30) = 150 + 17\,(2) + 13\,(10) = 150 + 34 + 130 = 314$

$B_H(m) = B_L(m) + 45$.

Second Filter Specification: An analog signal is band-limited to 500 kHz. It is ideally sampled, with a sampling rate of 1200 kHz (1.2 MHz). 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 thus determined from m:

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, $r(5) = 5$, $r(30) = 10$

$B_L(m) = 157 + 17\, q(m) + 13\, r(m)$.

For example, $B_L(30) = 157 + 17\,(2) + 13\,(10) = 157 + 34 + 130 = 321$

$B_H(m) = B_L(m) + 55$.

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 (Part I): Up to 30 marks for carrying out the filter design assignment in a satisfactory manner eventually and 15 marks or more, as an 'early bird and teaching assistance incentive', which will be separately awarded for deserving students.

Part II: (Optional, for additional credit in the Class Participation Score):

Read up, from text books and reference books, about the design of Infinite Impulse Response (IIR) filters, using the Elliptic Approximation. Put down the steps involved.

Now modify the specifications of the first filter and second filter that you have been asked to design, by making BOTH the passband and the stopband equiripple and leaving all the other specifications unchanged.

Design the two filters with filter number M assigned to you, using the Elliptic Approximation. Demonstrate the design,with verification of frequency response, to the Teaching Associates, in the same manner as you did for the mandatory Part I, presenting a report in the same format, except that there will be no section on FIR filter design this time.

Compare your designs for Filter Number M in the mandatory Part I and optional Part II and explain what you observe. Particularly, compare the order, the magnitude response and the phase response and comment.

Maximum credit that can be earned for the class participation score from Part II: 30 marks.

(End of filter design assignment)