

# ELEC 3004/7312: Signals Systems & Controls

## Assignment 2, Due: 20<sup>th</sup> of April 2012

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**Note:** This assignment is worth **10%** of the final course mark. You should spend approximately 3 hours preparing for the tutorial. The tutors will *not* assist you further unless there is real evidence you have attempted questions prior to the tutorial. Beyond the lecture and tutorial sessions, it is estimated that you will need 4 to 5 hours to complete the assignment (**7-8 hours total**).

**Total marks: 100**

1. Multiply the following polynomials in  $z$  by using the fft algorithm (MATLAB).

$$\begin{aligned} &1 + 2z^{-1} + 4z^{-2} + 5z^{-3} + 11z^{-4} + 25z^{-5} \\ &1 - 4z^{-1} - 5z^{-2} + 15z^{-3} - 13z^{-4} + 19z^{-5} \end{aligned}$$

(10 marks)

2. Multiply the following base 10 numbers together and then show how the same result can be obtained via the DFT. Explain your method.

$$537283 \text{ multiplied by } 423715$$

(10 marks)

3. Consider the polynomial

$$F(z) = 1 + 2z^{-1} + 3z^{-2} + 5z^{-3} + 12z^{-4} + 25z^{-5}$$

- a) Write down and plot the positions of all the poles and zeros (MATLAB roots/zplane).
- b) Evaluate  $|F(z)|$  at 256 uniformly spaced points around the unit circle using the Discrete Fourier Transform and plot the result.

(10 marks)

4. Consider a 10 Hz sine wave sampled at 40 Hz. If the samples are plotted and displayed using linear interpolation, the reconstructed waveform will not resemble the original continuous-time sine wave. Show how  $\sin x$  on  $x$  interpolation can be used to reconstruct the original continuous-time signal. (Hint: In this question you are being asked to resample the sine wave at a higher sampling rate)

(10 marks)

5. An  $N$  point DFT can be considered to be a set of  $N$  filters; each filter tuned to detect one particular frequency. Consider the filter corresponding to the DC component with  $N=10$ .

- a) Plot the frequency response of this filter.
- b) How many times does the magnitude of this filter go to zero around the unit circle?
- c) Plot the zeros of the filter (use roots and zplane in Matlab)

(10 marks)

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6. An FIR filter with symmetric or antisymmetric coefficients will have linear phase. Show that for such a filter the zeros will either be on the unit circle or in reciprocal conjugate pairs. Hint – For symmetric coefficients show that

$$H(z) = z^{-M} (H(z^{-1}))$$

and examine the zero locations.

(10 marks)

7. Design a LP FIR filter to meet the following specifications using the window method. Use a Blackman window.

$$F_s = 10 \text{ kHz}$$

$$F_c = 1.5 \text{ kHz (3 dB down)}$$

$$\text{Attenuation} = 60 \text{ dB at 2.5 kHz}$$

Give all the relevant plots (impulse, frequency responses) and the performance of the final filter. Compare this filter to one designed using the optimal (remez) method

(20 marks)

8. Design a HP FIR filter to meet the following specifications using the window method. Use a Kaiser window.

$$F_s = 10 \text{ kHz}$$

$$F_c = 3.5 \text{ kHz (1 db down)}$$

$$\text{Attenuation} = 60 \text{ dB at 3 kHz}$$

Give all the relevant plots (impulse, frequency responses) and the performance of the final filter. Compare this filter to one designed using the optimal (remez) method.

(20 marks)