B. Sc. Engg. CSE 6th Semester

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ISLAMIC UNIVERSITY OF TECHNOLOGY (IUT)
THE ORGANIZATION OF THE ISLAMIC COOPERATION (OIC)
Department of Computer Science and Information Engineering (CSE)

SEMESTER FINAL EXAMINATION **SUMMER SEMESTER, 2016-2017**

DURATION: 3 Hours **FULL MARKS: 150**

CSE 4631: Digital Signal Processing

Programmable calculators are not allowed. Do not write anything on the question paper.

There are 8 (eight) questions. Answer any 6 (six) of them.

Figures in the right margin indicate marks.

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1. a) Why is DSP important? Evaluate the necessity of DSP in *telecommunication*. 10

   b) One of the key contribution areas of Digital Signal Processing is *Image Processing*. Why is this distinct than the other areas? Describe the importance of this subgroup. 10

   c) Define *Signal* and *System* with example. 5

2. a) Name the time domain and frequency domain parameters for evaluating filter performance. Describe an example scenario where you would need a filter which works good both in time domain and frequency domain. 6+4

   b) What is Superposition? Why is this property considered as the foundation of DSP? 3+5

   c) In DSP, Linear systems are appreciated as they are easy to work with. But not all the systems in the world are linear. What are alternatives we can use if a system is not linear? 7

3. a) A digital communication link carries binary-coded words representing samples of an input signal. 18

   $$x_a(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12000\pi t$$

   This link is operated at 500 bits/s and each input sample is quantized into 1024 different voltage level.

   i. What are the sampling frequency and folding frequency?

   ii. What is the Nyquist rate for the signal $x_a(t)$?

   iii. What is the discrete-time signal obtained after sampling?

   iv. What is the analog signal $y_a(t)$ that we can reconstruct from the samples if we use ideal interpolation?

   v. What is the resolution Δ?

   b) Describe the synthesis process of Fourier Analysis in brief. 7

4. a) Name the four types of Fourier transforms. Mention the three ways you can calculate DFT. 4+3

   b) Calculate the time domain for the following frequency values: 12

   Real part: {64,64,64,64,64,64,64,64,64,64,64,64,64,64,64,64}

   Imaginary part: {0,0,0,0, 0,0,0,0,0,0,0,0,0,0,0,0}

   c) You are told that the following signals are the frequency domain of a 32 point real DFT. Give two reasons why this is not possible. 6

   Real part: {1,2,3,4,5,6,7,8,7,6,5,4,3,2,1,0}

   Imaginary part: {8,7,6,5,4,3,2,1,0,1,2,3,4,5,6,7}

5. a) A signal containing 1000 points is to be convolved with a signal containing 128 points. Answer the following questions: 2+2+7

   i. What is the length of the resulting signal?
   ii. If frequency domain convolution is used, what length of DFT is appropriate?
   iii. If a 1024 point DFT is used to perform frequency domain convolution, how many samples are correct, and how many are corrupted? How do you explain this corruption?

b) Why would you use Polar notation instead of Rectangular notation? Describe a real time use of Spectral Analysis. 4+4

c) If x[n] has the frequency domain: Xreal[f] and Ximag[f], and y[n] has the frequency domain: Yreal[f] and Yimag[f], calculate the frequency domain of the following signals: 6

   i. x[n] + y[n]
   ii. 3.14x[n] + y[n]/3.14

6. a) Suppose, you are given an input signal of length 2000 and a filter of length 200 and told to evaluate the output. If you are not allowed to use convolution, how can you perform the task? Contrast the performance with traditional convolution. 6

b) DFT coverts a time domain signal into a frequency domain signal which has two parts. In frequency domain, a signal can have two different representation: *Rectangular Notation* or *Polar Notation.* 4+3+7

   i. Show the relation between these two notations.
   ii. Which one is better for signal representation? Why?
   iii. What are the nuisances associated with *Polar Notation*?

c) Why does aliasing occur during sampling of an analog signal? What are the ways to avoid it? 5

7. a) A sinusoid at 1.7 kHz is digitized at 10,000 samples per second. The signal is passed through a 2048 point DFT, and converted to polar form. Draw four sketches of the magnitude, one for each of the four ways that the frequency domain's independent variable can be expressed. Be sure to indicate the frequency symbol used, the range of values, the units, and at what frequency the sinusoid appears. 12

b) Two signals, x(n) and h(n), are defined by: 9

   x(n): 1, 0, 2, 3, 2, 1,-1,-2,-1, 0, 2, 3, 3, 2, 1, 1 (samples 0-15)
   h(n): 1, 2, 3,-3,-2,-1 (samples 0-5)

   If y(n) = x(n)*h(n), use the input side algorithm to determine the contribution to y(n) from the indicated sample:

   i. x(2)
   ii. x(6)
   iii. x(9)

c) Why is Hamming Window used in spectral analysis? 4

8. a) What is a filter kernel? Differentiate between Infinite Impulse Response (IIR) filters and Finite Impulse Response (FIR) filters. 2+4

b) The original filter kernel of a low-pass filter is given below and the original frequency response is shown in Figure 1. The bold sample denotes the center of the symmetry. 12

Original filter kernel = $\{1,2,3,2,1,-1,-2,-1,0,1,2,3,4,5,4,3,2,1,0,-1,-2,-1,1,2,3,2,1\}$

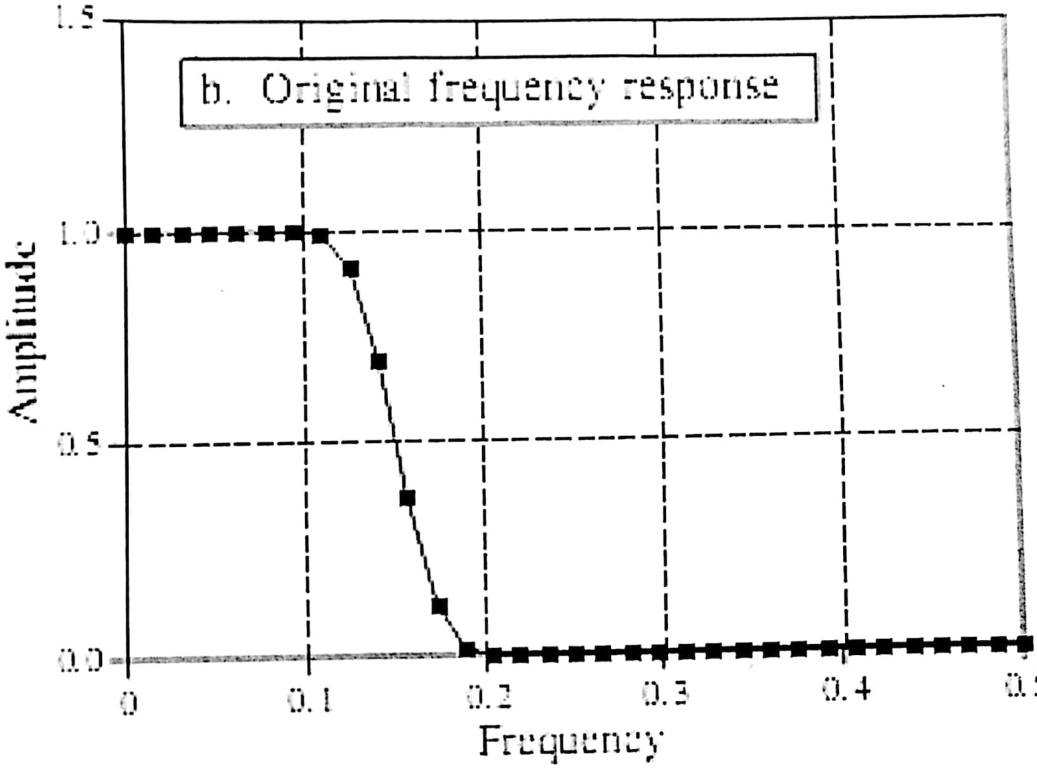

Figure 1: Filter kernel of a low-pass filter

Calculate the filter kernel and draw the frequency response for the following two cases:

i. The original filter is converted to high-pass filter using spectral inversion

ii. The original filter is converted to high-pass filter using spectral reversal

c) If you are told to design a band-stop filter using the Original filter in question (1b), how will you design it? Draw the frequency response of it. 7