ISLAMIC UNIVERSITY OF TECHNOLOGY (IUT)

ORGANISATION OF ISLAMIC COOPERATION (OIC)

Department of Computer Science and Engineering (CSE)

SEMESTER FINAL EXAMINATION **SUMMER SEMESTER, 2021-2022**

DURATION: 3 HOURS **FULL MARKS: 150**

CSE 4631: Digital Signal Processing

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

Answer **all 6 (six)** questions. Figures in the right margin indicate full marks of questions whereas corresponding CO and PO are written within parentheses.

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1. a) i. Describe the problem of aliasing during sampling analog signals. 2+3 (CO1) (PO1)
   ii. How can this problem of aliasing be avoided?

   b) Describe how the following DSP concepts can be utilized for echo location technologies such as Radar or Sonar : 5+5 (CO1) (PO1)
   i. Sampling and Quantization
   ii. Correlation

   c) Suppose you take the following analog signal, $x_a(t) = 4\cos 2000\pi t + 7\cos 6000\pi t$, and sample it by taking ***5000*** readings per second. After that, you quantized the signal in such a way that each sample can be stored in ***5*** bits. 4+3+3 (CO2) (PO1)
   i. Determine the Nyquist rate and the folding frequency.
   ii. Can we faithfully reconstruct the original signal by interpolating from the samples obtained after sampling?
   iii. Determine the quality of the output of the A/D converter in terms of signal-to-quantization noise ratio (SQNR).

2. a) Resolve the signal $x[n] = \{-1, 3, \hat{4}, -2, 5, 7, -9, 1\}$ into its constituent parts using the following decompositions : 5+5 (CO3) (PO2)
   i. Even-odd decomposition
   ii. Interlaced decomposition

   b) Suppose a signal $x[n] = \{3, 4, -2, 5, -1, \hat{2}, 1, 8\}$ is passed through the linear systems shown in Figure 1. The impulse response of ***S1*** is $h_{s1}[n] = \{1, \hat{2}, -1\}$ and the impulse response of ***S2*** is $h_{s2}[n] = \{\hat{3}, 2\}$. The output of ***S1*** passes through a constant multiplier of **3**. 8+7 (CO2) (PO)

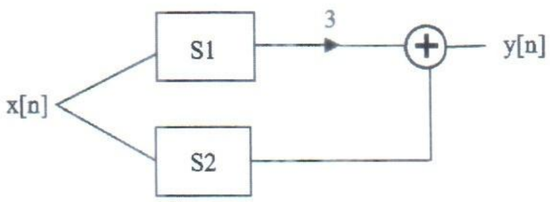

Figure 1: The system described in Question 2.b)

   i. Simplify the block diagram into one linear system and no constant multiplier. What would be the impulse response of the resultant system?
   ii. Evaluate $y[n]$.

3. a) i. Why does the problem of circular convolution occur?
   ii. How can you avoid this problem?

   b) i. Real Discrete Fourier Transform (DFT) converts an N-point time domain signal into two frequency domain signals with $N+2$ samples in total. What is the reason for these 2 additional samples? 4 (CO1) (PO1)
   ii. Why is the duality between the time domain and the frequency domain stronger when complex DFT is used instead of real DFT?

   c) Evaluate the computational complexity of forward DFT using correlation and analysis equations by calculating the required number of multiplication and addition operations. 7 (CO3) (PO2)

4. a) Analyze the trade-offs of using the following four windows in spectral analysis: 8 (CO3) (PO2)
   i. Hamming window
   ii. Blackman window
   iii. Rectangular window
   iv. Flat-top window

   b) Suppose the following signal $x[n] = \{-1, 3, \hat{4}, -2, 5, 7, -9, 1\}$ is a filter kernel of a high pass filter with cutoff frequency at **0.2** of sampling rate. Determine the filter kernel for a low pass filter with cutoff frequency at **0.3** of sampling rate. What is this process called? 8 (CO4) (PO3)

   c) Use the FFT synthesis flow diagram shown in Figure 2 to answer the following questions: 6+3 (CO1) (PO1)
   i. Explain the operations that are performed to combine two four-point frequency spectrum into an eight-point spectrum in forward FFT.
   ii. What operation does the **xS** symbol represent? Why is it necessary for calculating FFT?

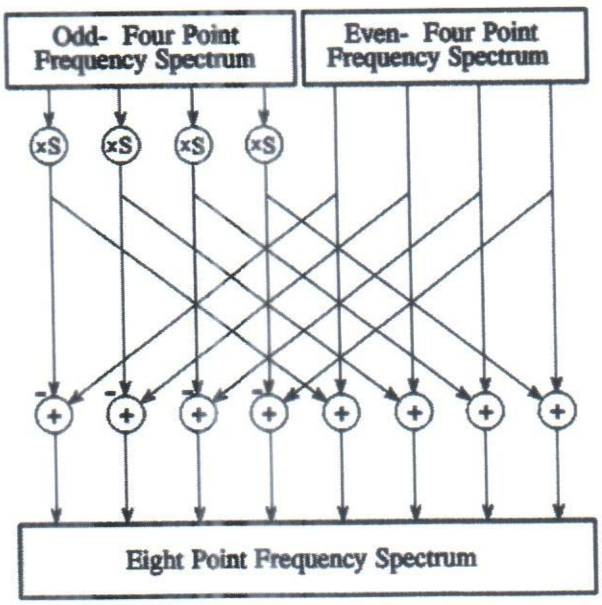

Figure 2: FFT synthesis flow diagram for Question 4.c)

a) Evaluate the performance of moving average filter as a low-pass filter by analyzing the characteristics of its frequency response. 6 (CO3) (PO2)

b) Suppose an EEG for a medical diagnosis reveals three distinct waves (alpha, beta, and gamma) with their own frequency range as shown in Figure 3. Draw the frequency response of a filter that can extract the beta wave and mark the cutoff frequencies of the filter. 9 (CO2) (PO1)

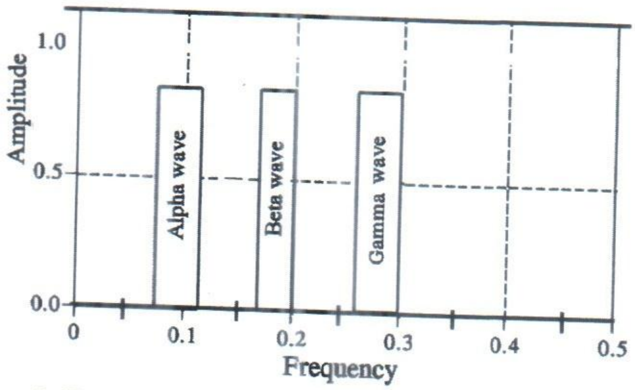

Figure 3: Frequency vs Amplitude Diagram for Question 5.b)

c) i. Suppose you are given a time domain signal x having a constant value of **2.5** in all the samples. What values will the corresponding frequency domain signals ***ReX*** and ***ImX*** contain? 5+5 (CO1) (PO1)

ii. When a time domain signal is shifted, why is the phase change for the higher frequency component signals larger than that of the lower frequency component signals?

6. a) The equation describing the general form of a sinc filter is as follows: 10 (CO4) (PO3)

$$h[i] = \frac{\sin(2\pi f_c i)}{\pi i}$$

Find the equation that describes the kernel of a windowed-sinc filter using the hamming window, where the kernel length is M, and the cutoff frequency is f_c. Explain the different components of the equation.

b) Suppose you are using a windowed-sinc filter as a low-pass filter. However, you are not satisfied with the performance in terms of its stopband attenuation and roll-off. 5+2 (CO2) (PO1)

i. What can you do to improve the performance of the filter?

ii. Will there be any downsides/limitations of this process?

c) Suppose a system's output signal x is corrupted with an undesired convolution due to a natural process. The characteristic of the undesired convolution is known in the form of an impulse response, h. 2+6 (CO2) (PO1)

i. What procedure can undo/remove the undesired convolution and restore an uncorrupted output signal?

ii. Explain the steps required to carry out the procedure with necessary equations.