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ISLAMIC UNIVERSITY OF TECHNOLOGY (IUT)

ORGANISATION OF ISLAMIC COOPERATION (OIC)

Department of Computer Science and Engineering (CSE)

SEMESTER FINAL EXAMINATION — SUMMER SEMESTER, 2022-2023

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 with corresponding COs and POs in parentheses.

1. a) A low-pass FIR filter is to be designed with the following specifications: passband 10 kHz, stopband 11 kHz, stopband attenuation 50 dB, and a sampling frequency of 44 kHz. Assume that having better stopband attenuation is more important than having faster rolloff. Determine the impulse response $h(n)$ for this filter. 15 (CO2) (PO1)

   b) Compute the convolution of the following input signal $x(n)$ with the impulse response $h(n)$. 10 (CO2) (PO1)

$$x(n) = u(n+1) - u(n-4) - \delta(n-5)$$
$$h(n) = [u(n+2) - u(n-3)].(3 - |n|)$$

2. a) The frequency response $H(n)$ of a system is given in the polar form as follows: 9 + 4 (CO3) (PO2)

$$MagH = \{80, 31.305, 8\}$$
$$PhaseH = \{0, -0.4636, 3.1416\}$$

   where $PhaseH(n)$ is given in radians.
   Find the impulse response and step response of this system.

   b) Explain how time domain aliasing and frequency domain aliasing occur within a signal. 6 (CO3) (PO2)

   c) Suppose you are given a 60 point time domain audio signal and you wish to generate a higher resolution version of this signal having 1000 points. Describe the frequency domain interpolation procedure needed to obtain your desired result. 6 (CO3) (PO2)

3. a) Describe the problems associated with using the polar notation to represent the frequency domain of a signal. 7 (CO1) (PO1)

   b) Identify the factors that limit the frequency resolution of a signal. 8 (CO1) (PO1)

   c) The frequency domain representation of the input, $X(n)$, and the output, $Y(n)$, of a system is given in the rectangular form as follows: 10 (CO2) (PO1)

$$X = \{80 + 0i, 28 - 14i, -8 + 0i\}$$
$$Y = \{-80 + 0i, -70 + 140i, 24 + 0i\}$$

   Find the frequency response of the system in rectangular form.

4. a) Illustrate the effect on the phase of a signal due to a right shift or left shift in the time domain with relevant examples. 8 (CO1) (PO1)

b) Show how the bit reversal sorting algorithm is used to find the time domain decomposition of a 16 point signal in the Fast Fourier Transform (FFT) algorithm. 7 (CO2) (PO1)

c) Explain how the FFT butterfly functions as a fundamental computation unit in the FFT synthesis step. Illustrate how the FFT butterfly is used to combine two 8 point frequency spectra into a single 16 point frequency spectra. 5 + 5 (CO2) (PO1)

5. a) Explain how FFT is more advantageous over DFT in terms of both time complexity and accuracy. 9 (CO1) (PO1)

b) Suppose the filter kernel of a low-pass filter having the cutoff frequency 200 Hz is given as $h_l = \{-1, 0, 1, 2, 1, 0, -1\}$. Assuming that the sampling frequency is 1 kHz, Calculate the filter kernel for the following filters:

i. A high-pass filter having a cutoff frequency at 300 Hz. 5 (CO2) (PO1)

ii. A band-pass filter that allows frequencies in the range of 200 Hz to 300 Hz. 6 (CO2) (PO1)

iii. A band-reject filter that stops frequencies in the range of 200 Hz to 300 Hz. 5 (CO2) (PO1)

6. a) Suppose you are given an arbitrary time domain input signal, $x(n)$, and your DFT function is providing the following frequency domain representation after calculating a 16 point real DFT of the input signal: 8 (CO3) (PO2)

$$X(n) = \{4 + 2i, 5 - 2i, 6 + 6i, 7 - 7i, 8 + i, 7 + i, 6 - 2i, 5 + 2i\}$$

Justify how you know that your implementation of the DFT function is not correct.

b) The following input-output pairs have been observed during the operation of a linear system: 7 (CO2) (PO1)

$$x_1(n) = \{-1, 2, 1\} \xrightarrow{\tau} y_1(n) = \{1, 2, -1, 0, 1\}$$

$$x_2(n) = \{1, -1, -1\} \xrightarrow{\tau} y_2(n) = \{-1, 1, 0, 2\}$$

$$x_3(n) = \{0, 1, 1\} \xrightarrow{\tau} y_3(n) = \{1, 2, 1\}$$

Here, the leftmost point is the origin for each of the signals and τ represents the system.

Provide justifications regarding the time invariance of the system.

c) Explain why amplitude modulation is used in DSP. Describe its effect on the corresponding frequency domain of the modulated signals. 4 + 6 (CO3) (PO2)