Ref No: **Ex/EE/5/T/414/2024**

BACHELOR OF ENGINEERING (ELECTRICAL ENGINEERING) FOURTH YEAR FIRST SEMESTER EXAM 2024

SUBJECT: - DIGITAL SIGNAL PROCESSING

Time: Three hours

Full Marks 100
(50 marks for each part)

Use a separate Answer-Script for each part

PART I

No. of Questions		Marks

Answer ***any three*** *questions.* ***TWO*** *marks are reserved for neat and well organized answers.*

1.a)	"The computation of an N-point DFT, in its basic form, requires N complex multiplications and N complex additions." - Justify or correct this statement, citing suitable reasons.	05
b)	"The real parts of DFT coefficients are anti-symmetric and the imaginary parts of DFT coefficients are symmetric with respect to the line of symmetry at sampling frequency". - Justify or correct this statement, citing suitable reasons.	05
c)	Determine the DFT of the sequence $x_k = \begin{cases} \frac{1}{4}, & \text{for } 0 \le k \le 3 \\ 0, & \text{otherwise} \end{cases}$	06
2. a)	Design an M-tap causal digital FIR filter with stepped characteristic in its frequency response given as: For $\frac{-\omega_s}{2} \le \omega \le \frac{\omega_s}{2}$, $H(\omega) = ae^{-j\omega\tau(M-1)/2}$, for $\lvert\omega\rvert \le \omega_1$; $= be^{-j\omega\tau(M-1)/2}$, for $\omega_1 < \lvert\omega\rvert \le \omega_2$; $= 0$, otherwise. where $a > b$, and all other symbols have usual meaning. The filter coefficients are smoothened using Hann window. Draw the schematic realization of the filter.	09
b)	How is circular complex convolution integral employed in designing FIR filters? What is the importance of rectangular window sequence in this context?	07
3. a)	Describe in detail how can two-dimensional FIR digital filters be employed for offline analysis of two-dimensional data.	08

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3\. b) Determine the frequency response of the following non-causal window function: 08

$$w_n = 0.54 + 0.46\cos\left(\frac{2\pi n}{M-1}\right), \qquad \text{for } |n| \le \left(\frac{M-1}{2}\right).$$

$$= 0, \text{ otherwise}$$

4\. a) Prove that, for distortion-less transmission of signal through a digital filter, the phase shift must be lagging and proportional to frequency within the pass band region. 08

b) Describe in detail how can FFT be utilized to perform digital filtering of a finite real data sequence. 08

5\. Write short notes on ***any two*** of the following: 08+08

a) Computation of 8-point FFT.

b) Realization problems in FIR filters.

c) Complex Fourier series for a periodic signal.

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BACHELOR OF ENGINEERING (ELECTRICAL ENGINEERING)
FOURTH YEAR
FIRST SEMESTER EXAM 2024

DIGITAL SIGNAL PROCESSING

Time: ~~Two hours~~/Three hours/ ~~Four hours/ Six hours~~

Full Marks 100
(50 marks for each part)

Use a separate Answer-Script for each part

Question No.	PART II	Marks
	Answer any *three* questions **Two marks reserved for neat and well-organized answers**	
1.(a)	The Z-transform of a sequence x_n is $$X(z)=\frac{z^2+2z-3}{(z-0.2)(z-0.3)(z-0.4)}$$ Derive the expression for the sequence x_n.	8
(b)	Starting from the definition of Z-transform, determine the expression for the Z-transform of the sequence $f_n=\cos(\omega_0 n\tau)u_n$, where the symbols have their usual meaning. What will be the Z-transform of the sequence $$h_n=\left(\frac{1}{2}\right)^n\cos(\omega_0 n\tau)u_n ?$$	8
2.	The transfer function of a discrete-time linear time-invariant (DTLTI) system is $$H(z)=\frac{2z^3-(5/6)z^2}{[z^2-(5/6)z+(1/6)](z-1/4)}$$ Derive and draw the following structures for realizing the system. (i) Transposed Direct Form II. (ii) Cascade structure using first order subsystems. (iii)Parallel structure using first order subsystems. Write down the sets of difference equations for all the structures.	16

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Question No.	PART II	Marks
3. (a)	Show that a uniformly sampled signal can be mathematically represented by a train of scaled impulses, the strengths being the corresponding sample values.	8
(b)	Derive a relation between the frequency spectra of a continuous-time signal and those of its uniformly sampled version. Hence explain the phenomenon of *aliasing*.	8
4. (a)	Using impulse-invariant transformation, design a digital filter corresponding to the analog filter with transfer function $G(s) = 5 / (s^2 - 3s + 2)$ Consider a sampling frequency of 10 Hz. Write down the difference equation relating the output and the input sequences.	10
(b)	Derive the frequency-warping relations in connection with the bilinear transformation method of designing digital IIR filters. Explain the significance of this warping phenomenon with the help of illustrations.	6
5.	Write short notes on any *two* of the following.	
(a)	Mapping of left-half of s-plane on to z-plane.	8 +8
(b)	Digital integrators.	
(c)	Recursive and non-recursive DTLTI systems.	
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