

**B.E. ELECTRICAL ENGINEERING SECOND YEAR SECOND SEMESTER
EXAMINATION 2024**

DIGITAL SIGNAL PROCESSING

Full Marks 100

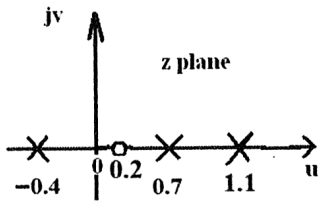
Time: Three hours

(50 marks for each part)

Use a separate Answer-Script for each part

Question No.	PART- I	Marks
	<p align="center">Answer any THREE questions Two marks reserved for well organized answers and neatness</p>	
1. (a)	<p>Determine the Nyquist rate for the continuous-time signal $x(t) = 2 \cos^2(2\pi t) \cdot \sin(6\pi t) + 3 \sin(16\pi t)$, where t is in s. If $x(t)$ is sampled at a rate of 4 Hz, and the sampled version is passed through an ideal low-pass filter with cutoff frequency 2 Hz, then determine the frequencies of the spectral components of the output of the filter. Derive the expressions used. [CO1]</p>	10
(b)	<p>Explain in brief, the necessity of using an analog lowpass filter prior to an analog-to-digital-converter in a digital signal processing system. How can the cutoff frequency of this filter be selected? Give a practical example. [CO1]</p>	6
2.(a)	<p>Cite a sequence whose Z-transform has a complex-conjugate pair of poles. Hence examine with suitable sketches the effect of location of the poles on the circumference of the unit-circle (centred at origin) in the Z-plane, outside the unit circle and inside the unit circle, and the time-domain behavior of the system. [CO1]</p>	6
(b)	<p>Obtain the closed form expression for the Z-transform of a <u>unit step sequence</u>. Also use the properties of Z-transform to obtain the transform of</p> $h[n] = n \left(\frac{1}{2} \right)^n u[n-2]$ <p>State clearly the location of the poles, their multiplicity and the region of convergence (ROC) of Z-transform in both the cases. [CO1]</p>	4

[Turn over

Question No.	PART I	Marks
(c)	Determine the closed-form expression for the inverse z-transform of $F(z) = \ln\left(1 - \frac{1}{2}z^{-1}\right); z > \frac{1}{2}$ [CO1]	6
3. (a)	<p>The pole-zero configuration of the Z-transfer function $H(z)$ of a discrete-time linear time-invariant (DTLTI) system is shown in Fig. [A].</p>  <p>Fig. [A]</p> <p>Identify all possible ROCs of $H(z)$. Comment on the stability and the causality of the system for each possible ROC, citing specific reason.</p> <p>If the DC gain of the system is 9.5, obtain the expression for $H(z)$. Determine the closed form expression for the impulse response of the system if the system is causal. [CO2]</p>	8
(b)	<p>The difference equation relating the output $y[n]$ and the input $x[n]$ of a DTLTI system is given by</p> $y[n] = bx[n] + ay[n-1], \text{ where } a \text{ and } b \text{ are real coefficients.}$ <p>Show that the system is an IIR system. Examine with the help of relevant sketches, the effect of a and b on the nature of the impulse response. [CO2]</p>	8
4.	<p>Design a low pass digital Butterworth filter with a dc gain of 0 dB, an attenuation of 3 dB at 675 Hz and an attenuation of at least 15 dB at 1350 Hz. Use bilinear transformation preceded by frequency pre-warping. Consider a sampling frequency of 10 kHz. Obtain the difference equation relating the output and the input sequences of the filter.</p> <p>Prove the warping formula used.</p> <p>Explain why the technique is free from 'Characteristic Aliasing'. [CO4]</p>	10+3+3

Question No.	PART I	Marks
5.	Write short notes on any two of the following.	8+8
(a)	Designing digital filters by impulse invariant technique. [CO4]	
(b)	Digital integrators. [CO2]	
(c)	Mapping of left half of s-plane on to z-plane. [CO1]	

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No. of Questions	PART II	Marks
	<p style="text-align: center;"><i>Answer all the questions.</i></p>	
1.	<p>Prove that computation of an N-point DFT, in its basic form, requires N^2 complex additions and N^2 complex multiplications. Then show how the number of additions and multiplications gets reduced in implementing an N-point radix-2 decimation-in-frequency in-place FFT algorithm.</p> <p style="text-align: right;">(CO1-K1)</p> <p style="text-align: center;">OR</p> <p>Explain in detail, how in computing a 4-point FFT, the concept of splitting an N-point DFT into two $(N/2)$-point DFTs is implemented.</p> <p style="text-align: right;">(CO1-K1)</p>	10
2. (a)	<p>Prove whether distortionless filtering is possible or not if the phase characteristic of a filter is linear phase with offset. How will the frequency response of an ideal digital filter get modified when the summation is carried out using finite number of terms? State any assumption made in this derivation.</p> <p style="text-align: right;">(CO2-K2)</p> <p style="text-align: center;">OR</p> <p>How is two-dimensional convolution operation employed in FIR image filters? Explain in detail how can low-pass FIR image filters be designed using concepts of simple averaging and weighted averaging.</p> <p style="text-align: right;">(CO2-K2)</p>	09
(b)	<p>A fifth order, offline, non-causal, FIR digital filter is designed to filter an input data sequence of length 10. Establish in detail which signal instants will be present and which signal instants will be absent in the filtered data sequence.</p> <p style="text-align: right;">(CO2-K2)</p> <p style="text-align: center;">OR</p> <p>Derive the frequency response of a raised-cosine window used in designing FIR filters. Plot the corresponding amplitude response and hence comment on its approximate width of main lobe.</p> <p style="text-align: right;">(CO2-K2)</p> <p style="text-align: center;">/</p>	07

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No. of Questions	PART II	Marks
3.	Write a short note on <i>any one</i> of the following: (CO3-K3)	08
(i)	Circular shift and circular symmetries of a sequence.	
(ii)	Application of FFT for digital filtering of a real data sequence.	
4.	An M -tap causal high-pass digital filter is designed with its frequency response given as: For $0 \leq \omega \leq \omega_s$, $H(\omega) = ae^{-j\omega\tau(M-1)/2}, \quad \text{for } \omega_c \leq \omega \leq (\omega_s - \omega_c)$ $= 0, \quad \text{otherwise}$ where all symbols have their usual meaning. Here, $M=5$, $a=1.1$, cut-off frequency=200 Hz, and sampling frequency=1000 Hz. The filter coefficients are smoothened using Hann window. Realize the filter and draw its schematic form. (CO4-K3)	08
5.	Justify or correct <i>any two</i> of the following statements with suitable reasons/derivations, in brief. (CO5-K4)	04×02 =08
i)	Both group delay and phase delay can always be employed to determine whether a filter designed is distortionless or not.	
ii)	The imaginary part of frequency response in an FIR digital filter always becomes zero at sampling frequency and folding frequency.	
iii)	In histogram equalization method for image contrast enhancement, the output gray intensity level for a pixel is inversely proportional to the cumulative histogram for the original pixel intensity and directly proportional to the area of the image.	