BIRLA INSTITUTE OF TECHNOLOGY, MESRA, RANCHI
(END SEMESTER EXAMINATION)

CLASS:	**BTECH**	**SEMESTER : V**	
BRANCH:	**ECE**	**SESSION : MO/2024**	

SUBJECT: EC305 SIGNAL PROCESSING TECHNIQUES

TIME: 3 Hours | **FULL MARKS: 50**

INSTRUCTIONS:
1. The question paper contains 5 questions each of 10 marks and total 50 marks.
2. Attempt all questions.
3. The missing data, if any, may be assumed suitably.
4. Before attempting the question paper, be sure that you have got the correct question paper.
5. Tables/Data handbook/Graph paper etc. to be supplied to the candidates in the examination hall.

			CO	BL
Q.1(a)	Define the stability and causality of LTI system in Z-domain. Find the impulse response of the LTI system defined as $H(z) = \frac{5z}{(z-1)^2} - \frac{2z}{(z-0.5)^2}$	[5]	CO1	1
Q.1(b)	Compare the computational complexity of DFT with FFT. Determine the 8- point DFT of the continuous time signal x(t)=sin(2πft) with f=50 Hz using decimation in frequency FFT method.	[5]	CO1	3
Q.2(a)	Discuss the finite word length effect in the realization of discrete time systems. Realize the system $H(z) = \frac{0.5(1-z^{-2})}{1+1.3z^{-1}+0.36z^{-2}}$ in parallel form structure.	[5]	CO2	4
Q.2(b)	What is the advantage of cascaded structure over direct form structure of LTI system. Given a system function $H(z) = \frac{3z^2+5z+4}{z^2+6z+8}$, Realize it using ladder structure.	[5]	CO2	4
Q.3(a)	Distinguish between Butterworth and Chebyshev filter. Design a Butterworth lowpass filter having the specification as: $f_p = 6kHz,\ f_s = 10kHz\ ,\ \delta_p = \delta_p = 0.1$	[5]	CO5	6
Q.3(b)	Explain the impulse invariance method of IIR filter design and establish the relation between analog frequency and digital frequency. Compare between the FIR and IIR filter.	[5]	CO3	4
Q.4(a)	Define linear phase characteristics of a filter. Explain the frequency sampling method of FIR filter design.	[5]	CO4	2
Q.4(b)	State the characteristics of the window in windowing method of FIR filter design. Evaluate the system function of the digital filter using impulse invariance method at 5 Hz sampling frequency from the analog filter given as $H_a(s) = \frac{2}{(s+1)(s+2)}$	[5]	CO4	5
Q.5(a)	Explain decimation and interpolation with suitable example and its spectrum. Describe the periodogram method of power spectrum estimation.	[5]	CO5	3
Q.5(b)	Define noble identity. Design an up-sampler system that interpolate the input signal by a factor 3 using polyphase structure.	[5]	CO5	6

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