U.S.N.										

B.M.S. College of Engineering, Bengaluru-560019

Autonomous Institute Affiliated to VTU

January / February 2025 Semester End Main Examinations

Programme: B.E. **Semester: V**
Branch: Electronics and Instrumentation Engineering **Duration: 3 hrs.**
Course Code: 23EI5PCDSA **Max Marks: 100**
Course: Digital Signal Processing and Its Applications

Instructions: 1. Answer any FIVE full questions, choosing one full question from each unit.
2. Missing data, if any, may be suitably assumed.

Important Note: Completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages. Revealing of identification, appeal to evaluator will be treated as malpractice.

			CO	PO	Marks
		MODULE- I			
1	a)	Describe the process of frequency domain sampling and reconstruction of discrete time signal.	CO1	PO1	**10**
	b)	Find the 4-pt DFT of the sequence x(n)={1,-2,3,2} directly.	CO1	PO1	**04**
	c)	Compute the circular convolution using DFT and IDFT for the following sequences : x1[n] = [2, 3, 1, 1] and x2[n] = [1, 3, 5, 3]	CO1	PO2	**06**
		OR			
2	a)	If x(n)=[1,2,0,3,-2, 4,7,5] evaluate the following i) X(0) ii) X(4) iii) $\sum_{k=0}^{7} \lvert X(k)\rvert^2$	CO1	PO2	**05**
	b)	Find Inverse Discrete Fourier Transform of the following signal X(k)= [6, − 1 − j1, 0, − 1 + j1]	CO1	PO1	**05**
	c)	Determine the circular convolution of the sequences using DFT-IDFT. $X_1(n)$ = {4,5,6,7} and $X_2(n)$= {1,2,3,4}.	CO1	PO2	**10**
		MODULE - II			
3	a)	Derive the DIT-FFT algorithm with necessary equation and also draw the signal flow graph for N=8	CO1	PO2	**10**
	b)	Using overlap add method, compute the output of an FIR filter with impulse response $h(n)$ = {1, 1, 1} and input $x(n)$ = {3, −1, 0, 1,3, 2, 0, 1, 2, 1}	CO1	PO2	**10**
		OR			
4	a)	Determine the DFT of the sequence x(n)={1,2,3,4,4,3,2,1} using DIF-FFT	CO1	PO2	**08**
	b)	Find the 4-point IDFT of the sequence X(k)= [6,-2+2j, -2,-2-2j] using DIT-FFT algorithm	CO1	PO2	**06**

		c)	Obtain the relationship of DFT with the Z-transform	*CO1*	*PO1*	**06**
			MODULE - III			
	5	a)	Convert the analog filter with system function $$H_a(S) = \frac{1}{(s+0.1)^2 + 3^2}$$ into a digital IIR filter by means of the impulse invariance method.	*CO3*	*PO2*	**05**
		b)	Design an analog chebyshev filter with the following specifications: Passband ripple: $1dB\ for\ 0 \leq \Omega \leq 10rad/sec.$ Stopband attenuation:$-60dB\ for\ \Omega \geq\ 50rad/sec.$	*CO3*	*PO3*	**10**
		c)	Compare analog and digital filters	*CO3*	*PO1*	**05**
			OR			
	6	a)	Derive BLT Equation $S = \frac{2}{T}\left(\frac{1-Z^{-1}}{1+Z^{-1}}\right)$	*CO3*	*PO1*	**08**
		b)	Let $H(s) = \frac{1}{s^2+\sqrt{2s}+1}$ represent the transfer function of a low pass filter with a pssband of 1rad/sec.Use frequency transformation to find the transfer function of the following analog filters i)A lowpass filter with passband of 10rad/sec ii) A highpass filter with cutoff freq of 10rad/sec	*CO3*	*PO2*	**04**
		c)	Realize the given system in cascade and parallel form $$H(z) = \frac{1+0.25Z^{-1}}{(1-2Z^{-1}+0.25Z^{-2})(1-3Z^{-1}+0.25Z^{-2})}$$	*CO3*	*PO2*	**08**
			MODULE - IV			
	7	a)	Illustrates the following with magnitude frequency response and side lobe attenuation i) Rectangular window ii) Hamming window	*CO3*	*PO2*	**08**
		b)	Determine the co-efficients h(n) for a liner phase FIR filter of length M=15,which has a symmetric unit impulse response and a frequency response that satisfies $$H\left(\frac{2\pi K}{15}\right) = \begin{cases}1 & ; K = 0,1,2,3 \\ 0 & ; K = 4,5,6,7\end{cases}$$	*CO3*	*PO2*	**08**
		c)	Realize the linear phase FIR filter having the impulse response $h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4)$	*CO3*	*PO2*	**04**
			OR			

	8	a)	The desired frequency response of the low pass filter is given by $H(e^{jw}) = e^{-j3w} \; ; \; \lvert w\rvert < \frac{3\pi}{4}$ $0 \; ; \; \frac{3\pi}{4} < \lvert w\rvert < \pi$ Determine the frequency response of FIR filter using hamming window for N=7.	*CO3*	*PO2*	**10**
		b)	Differentiate between FIR and IIR filter	*CO3*	*PO1*	**05**
		c)	Realize the system function given by $H(z) = 1 - 2z^{-1} + \frac{1}{2}z^{-2} + \frac{1}{2}z^{-3} + \frac{1}{2}z^{-4}$ Using Direct form	*CO3*	*PO2*	**05**
			MODULE - V			
	9	a)	With a neat block diagram explain the working of adaptive noise canceller	*CO4*	*PO2*	**10**
		b)	Explain sampling rate conversion by a rational factor I/D	*CO4*	*PO2*	**10**
			OR			
	10	a)	Illustrate the need of multirate signal processing	*CO4*	*PO2*	**05**
		b)	Discuss about the LMS algorithm.	*CO4*	*PO2*	**05**
		c)	With a neat block diagram describe two stage interpolator and decimator, representing multistage implementation of sampling rate conversion	*CO4*	*PO2*	**10**

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