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

Autonomous Institute Affiliated to VTU

April 2025 Semester End Make-Up Examinations

Programme: B.E.

Semester: V

Branch: Electronics and Communication Engineering

Duration: 3 hrs.

Course Code: 23EC5PCDSP

Max Marks: 100

Course: Digital Signal Processing

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

UNIT - I			CO	PO	Marks
1	a)	Given $x_a(t)=10\cos(2\pi \times 4 \times 10^3 t) + 6\cos(2\pi \times 8 \times 10^3 t)$. Determine following i) Nyquist rate for this signal ii) The folding frequency iii) Determine the discrete time signal $x[n]$ if $x_a(t)$ is sampled at a rate of 16 kHz.	CO 1	PO 2	6
	b)	Find the 4 point DFT of the sequence of $x[n] = 4 \cos(\pi n/2)$. Draw its magnitude spectrum.	CO 1	PO 2	4
	c)	Consider the length-10 sequence, defined for $0 \leq n \leq 11$, $x[n] = \{8, 4, 3, 1, -4, -5, 0, -2, -1, 2, 4, 7\}$ with a 12-point DFT given by $X[k]$, $0 \leq k \leq 11$. Evaluate the following functions of $X[k]$ without computing DFT: i) $X[0]$, ii) $X[6]$, iii) $\sum_{k=0}^{11} X(k)$ iv) $\sum_{k=0}^{11} X(k) e^{-j\pi k/6}$	CO 1	PO 2	10
OR					
2	a)	Let $x[n]$ be a 4-point real sequence with DFT $X(k)$. $X(k) = [0, 1-j, 0, 1+j]$. Using the properties of DFT, find the DFTs of the following sequences .i) $x_1[n] = x((n-2))_4$, ii) $x_2[n] = x^*((-n))_4$	CO 1	PO 2	6
	b)	Apply DFT-IDFT method to compute the circular convolution of two given sequences $x[n] = \{2, 0, 1, 1\}$, $h(n) = \{1, 2, 2, 1\}$.	CO 1	PO 2	8
	c)	If $X(k) = [1, 2+j, 3-j, 4]$ be a 6-point DFT of a length 6 real sequence. Find (i) $x(0)$ (ii) $x(3)$ (iii) $\sum_{n=0}^5 x(n)$ without computing IDFT.	CO 1	PO 2	6
UNIT - II					
3	a)	Compute the output $y(n)$ of a filter whose impulse response is $h(n) = \{1, 1, 1\}$ and input signal $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ using overlap and save method. Use 5-point circular convolution.	CO 2	PO 2	8

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.

	b)	Given $x(n)=\{1,2,-1,2,4,2,-1,2\}$, determine the frequency domain samples corresponding to the 8-point DFT using DIF-FFT radix-2 algorithm.	CO 2	PO 2	8
	c)	Find the relation between DFT and Z transform.	CO 2	PO 2	4
	OR				
4	a)	A long sequence $x[n]=\{1,2,0,-3,4,2,-1,1,-2,3,2,1,-3\}$ is filtered through a filter with impulse response, $h(n)=\{1,1,1\}$. Compute the output $y[n]$ of a filter using overlap-add method. Use 8-point circular convolution	CO 2	PO 2	8
	b)	Compute 8-point DFT of $x(n)=\{0,1,2,3,4,5,6,7\}$ using DIT-FFT radix-2 algorithm.	CO 2	PO 2	8
	c)	Compare the complex multiplications and additions needed for a 1024 point sequence in the calculation of DFT using (i) Direct computation (ii) radix-2 FFT algorithm	CO 2	PO 1	4
	UNIT - III				
5	a)	Design a digital IIR low pass filter using Butterworth approximation for the following analog specifications: Pass band edge: 1000 Hz, Stop band edge: 3000 Hz, Pass band gain: -2dB, Stop band attenuation: -20dB, Sampling Frequency: 8000 Hz. Use Bilinear transformation. Transform the analog filter.	CO 3	PO 2	10
	b)	Obtain the cascade form and Parallel form realization for the following system. $y[n] = 0.75y[n - 1] - 0.125y[n - 2] + 6x[n] + 7x[n - 1] + x[n - 2]$	CO 3	PO 2	10
	OR				
6	a)	Design a type-I Chebyshev analog lowpass filter to meet the following specifications. Pass band ripple ≤ 2.5 dB Pass band edge frequency= 20 rad/sec. Stop band attenuation ≥ 30 dB Stop band edge frequency=50 rad/sec	CO 3	PO 2	10
	b)	Transform the analog filter $H(s) = \frac{1}{(s^2+3s+2)}$, into $H(z)$ using impulse invariant transformation. Assume sampling interval of 0.2 sec.	CO 3	PO 2	6
	c)	Compare Butterworth and Chebyshev filters	CO 3	PO 1	4
	UNIT - IV				
7	a)	Design and realize a linear phase digital lowpass filter (LPF) having 3dB cutoff frequency of 7.5KHz. and stopband attenuation of at least 40dB at 35KHz. Find the difference equation and the frequency response of this filter. Use $f_s = 100$ KHz. Hint: Use Hamming Window to attain the necessary attenuation.	CO 3	PO 2	12

	b)	Mention the different types of window functions used in FIR filter design and also write their mathematical expressions.	CO 3	PO 2	8
		OR			
8	a)	Use frequency sampling method to design a FIR filter with cutoff frequency 0.3π rad with 7 taps.	CO 3	PO 2	10
	b)	List the advantages of FIR filter	CO 3	PO 2	4
	c)	A FIR system is described by the following input -output relation $y(n) = x(n) + \frac{5}{6}x(n-1) + \frac{13}{6}x(n-2) + \frac{5}{6}x(n-3) + x(n-4)$ Represent the system using direct form realization and Linear Phase form	CO 3	PO 2	6
		UNIT - V			
9	a)	With the help of the necessary equations for Mean Square Error (MSE) for weight adaptation of the coefficients using the LMS algorithm, illustrate the convergence of an Adaptive filter.	CO 4	PO 1	10
	b)	With a neat block diagram describe two stage interpolator and decimator, representing multistage implementation of sampling rate conversion	CO 4	PO 1	10
		OR			
10	a)	Illustrate the significance of the up-sampling and down-sampling by considering the following sequence as an example. Consider the up-sampling factor, L to be 3 and down-sampling factor M to be 2. $x(n) = \{6, 5, 1, 6, 0, 6, 4, 2, 0, -2, 5, -5, -4, 4\}$ Sketch the sequence in time domain both for interpolation and decimation operation. Given the impulse response of the filter, $h(n) = \{\frac{1}{3}, \frac{1}{2}, 1, \frac{1}{2}, \frac{1}{3}\}$ and $x(n) = \{6, 6, 1, 2, -4\}$ Compute the interpolated sequence at the output of an interpolation filter with L=2.	CO 4	PO 1	10
	b)	Illustrate the working of an Adaptive filters in implementing following Applications 1. Channel Equalizer 2. Random Noise Cancellation	CO 4	PO 1	10
