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.
Branch: ES Cluster (EI /EC)
Course Code: 19ES5CCDSP
Course: Digital Signal Processing

Semester: V
Duration: 3 hrs.
Max Marks: 100

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**
		UNIT – I			
1	a)	Evaluate the following function without computing the DFT $\sum_{k=0}^{9} e^{-j\frac{4\pi k}{6}} X(k)$ for a given 10-point sequence x(n) = {8, 4, 7, -1, 2, 0, -2, -4, -5, 1}	CO 1	PO1	**06**
	b)	If X(k) is the 10-point DFT of the sequence $x[n] = \delta(n-1) + 2\delta(n-4) - \delta(n-7)$. What sequence y(n), has a 10 Point DFT $Y(k) = 2X(k)\cos\left(\frac{6\pi k}{10}\right)$	CO 1	PO1	**08**
	c)	By means of circular convolution, determine the response of the FIR filter with impulse response h(n)= {4,1,3} to the input sequence x(n) ={2,5,0,4}.	CO 1	PO1	**06**
		OR			
2	a)	For the sequence x(n)= [1,2,3,4], Compute i) 4 point DFT X(K) ii) Magnitude and phase spectrum.	CO 1	PO1	**06**
	b)	Compute the 4-point circular convolution of $x_1[n] = \left(\frac{1}{2}\right)^n$ *and* $x_2[n] = 2\cos\left(\frac{3\pi n}{4}\right)$, both signals defined over n = 0,1,2,3.	CO 1	PO1	**08**
	c)	Given x(n)= (1,2,3,4). Find the energy and hence verify Parseval's theorem.	CO 1	PO1	**06**
		UNIT - II			
3	a)	An FIR filter has the unit impulse response sequence h(n)={2,2,1}.Determine the output sequence in response to the input sequence x(n) ={3,0,-2,0,2,1,0,-2,-2,0} using the overlap add convolution method.	CO 2	PO2	**10**

		b)	We have 15 seconds of a speech waveform that is sampled at 8kHz and we wish to filter it with an FIR filter h[n] of length 55, using 512-point DFTs. How many DFTs and IDFTs will be required to perform this filtering task using the overlap-add method? Does your answer change with the overlap save method?	CO 1	PO1	**05**
		c)	Compute the 4-point DFT of the sequence x(n) = (1,0,1,0) using DIT-FFT radix -2 algorithm.	CO 1	PO1	**05**
			OR			
	4	a)	Explain the computation of the 4-point DFT of a given sequence using DIF -FFT algorithm. Suppose N – the number of DFT points has to be increased to 8, what change can be made in the input sequence? Also determine number of complex multiplications and complex additions needed if butterfly structure is used to compute 8-point DFT.	CO 1	PO1	**05**
		b)	Find the linear convolution of x(n) = {1,2,3,4,5,6,7,8,9,10} and h(n) = {1,2,1} using overlap-save method.	CO 1	PO1	**07**
		c)	Find the 8-point DFT of x(n) = {3,7,8,1} using DIT-FFT algorithm.	CO 1	PO1	**08**
			UNIT - III			
	5	a)	Obtain the cascade and parallel form realization of the given LTI system governed by the difference equation $y(n) = \frac{5}{8}\, y(n-1) - \frac{1}{16} y(n-2) + x(n) - 3x(n-1) + 3x(n-2) - x(n-3)$	CO 2	PO2	**10**
		b)	Design a Butterworth analog lowpass filter that will meet the following specifications: a. Maximum passband ripple = 2dB b. Passband edge frequency = 100 rad/sec c. Minimum stopband attenuation = 20dB d. Stopband edge frequency = 200 rad/sec	CO 2	PO2	**10**
			OR			
	6	a)	Derive the expression for the order of Butterworth, Low-Pass Filter.	CO2	PO1	**6**
		b)	Draw the direct form-I structure for the following difference Equation: y(n)-0.5y(n-1) +0.2y(n-2) = x(n)+2x(n-1) +2.5x(n-2)	CO2	PO1	**4**
		c)	Derive the expression for Digital filter conversion from analog filter using impulse invariant method. Also discuss the steps involved in digital filter design.	CO2	PO2	**10**
			UNIT – IV			
	7	a)	What is the significance of windowing in FIR Filter? Discuss.	CO 3	PO3	**04**
		b)	Draw the linear phase structure for an FIR Filter characterized by $h(n) = \delta(n) + \frac{1}{2}\delta(n-1) - \frac{1}{4}\delta(n-2) + \frac{1}{2}\delta(n-3) + \delta(n-4)$ Implement the same using direct form.	CO 3	PO2	**08**

		c)	A low pass filter is to be designed with the following desired frequency response $$H_d(e^{j\omega}) = \begin{cases} e^{-j2\omega} & -\frac{\pi}{4} \le \omega \le \frac{\pi}{4} \\ 0 & \frac{\pi}{4} \le \omega \le \pi \end{cases}$$ Determine the filter coefficients $h_d(n)$ if the window function is defined as $$\omega(n) = \begin{cases} 1 & 0 \le n \le 4 \\ 0 & otherwise \end{cases}$$ Also determine the frequency response $H(e^{j\omega})$ of the desired filter.	CO 3	PO1	**08**
			OR			
	8	a)	Draw the Direct form structure for following FIR filter y(n)=x(n)+ 2x(n-1) + 3x(n-2) + x(n-3)	CO3	PO1	**04**
		b)	Design and realize a linear phase digital lowpass filter (LPF) having 3dB cutoff frequency of 7.5KHz. and stopband attenuation of at least 45dB at 35KHz. Find the impulse response and the frequency response of this filter. Use f_s = 100KHz. [Hint: Use Hamming Window]	CO3	PO2	**10**
		c)	Discuss the steps involved in designing the FIR filters using frequency sampling technique. Also draw the frequency sampling structure of FIR Filter.	CO3	PO1	**06**
			UNIT – V			
	9	a)	Given a DSP up sampling system with the following specifications Sampling rate=6000Hz, input audio frequency range =0-800Hz, Passband ripple=0.02dB, Stopband attenuation=50dB, up sample factor L=3. Determine the FIR filter length, cut off frequency and window type if window design method is used.	CO 4	PO2	**06**
		b)	With a neat block diagram and relevant equations explain the working principle of echo cancellation in data transmission over telephone channels.	CO 4	PO2	**08**
		c)	With a neat block diagram and relevant equations explain the working principle of system identification using Adaptive filters.	CO 4	PO2	**06**
			OR			
	10	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	PO2	**05**
		b)	Illustrate the working of an Adaptive filters in implementing following Applications (i) Channel Equalizer (ii) Random Noise Cancellation	CO4	PO1	**08**

		c)	Illustrate the significance of the up-sampling and down-sampling by considering the following sequence as an example. $x(n) = \{6,5,1,6,0,6,4,2,0,-2,5,-5,-4,4\}$ (i) Consider the up-sampling factor, L to be 3. (ii) Consider down-sampling factor M to be 2. Sketch the sequence in time domain both for interpolation and decimation operation.	CO 4	PO2	**07**

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