CBCS SCHEME

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22LDC/LDE/LDN12

First Semester M.Tech. Degree Examination, Dec.2023/Jan.2024
Advanced Digital Signal Processing

Time: 3 hrs. Max. Marks: 100

Note: 1. Answer any FIVE full questions, choosing ONE full question from each module.
2. M : Marks , L: Bloom's level , C: Course outcomes.

		Module – 1	M	L	C
Q.1	a.	Explain the implementation of sampling rate conversion by using polyphase structure.	10	L2	CO1
	b.	Derive an expression for decimation by a factor D.	10	L3	CO1
		OR			
Q.2	a.	Explain sampling rate conversion for subband speech signal coding and explain its advantages.	10	L2	CO1
	b.	Determine the cross correlation of sequence $\gamma_{xy}(\ell)$ of the sequences $x(n) = \{\ldots\ldots, 0, 0, 2, -1, 3, 7, \underset{\uparrow}{1}, 2, -3, 0, 0\ldots\}$ $y(n) = \{\ldots\ldots, 0, 0, 1, -1, 2, -2, \underset{\uparrow}{4}, 1, -2, 5, 0, 0\ldots\}$	10	L3	CO1
		Module – 2			
Q.3	a.	Explain the design of digital filter banks with necessary diagrams.	10	L2	CO2
	b.	With neat diagram, explain 2 – channel QMF bank.	10	L2	CO2
		OR			
Q.4	a.	Explain How Mutlirate DSP can be used as phase-shift application with diagram.	10	L2	CO2
	b.	Convert the single – pole lowpass Butterworth filter with system function $H(z) = \dfrac{0.245(1+z^{-1})}{1-0.509z^{-1}}$ into a bandpass filter with upper and lower off frequencies W_u and W_ℓ respectively. The lowpass filter has 3-DB bandwidth, $W_p = 0.2\pi$.	10	L3	CO2
		Module – 3			
Q.5	a.	Explain the method of predicting future value with block diagram of forward prediction and necessary equations.	10	L2	CO3
	b.	Explain the steps involved in Levinson Durbin algorithm for deriving the expression for Normal Equations.	10	L2	CO3
		OR			
Q.6	a.	Write a short note on : i) Random Process ii) Power density spectrum iii) Mean Ergodic process iv) Statistical Average for joint random process.	10	L1	CO3

	b.	Explain the properties of the linear prediction error filters.	10	L2	CO4
		Module – 4			
Q.7	a.	Write a note on Linear Predictive coding of speech signals.	10	L1	CO4
	b.	Explain principles of adaptive channel equalization with a neat block diagram.	10	L2	CO4
		OR			
Q.8	a.	Explain LMS algorithm with necessary equations.	10	L2	CO4
	b.	Explain adaptive Noise cancellation with an example.	10	L2	CO4
		Module – 5			
Q.9	a.	How the non-parametric methods used for power spectrum estimation. Explain Welch method for Averaging modified Periodograms.	10	L3	CO5
	b.	Write a note on ARMA model spectrum estimation.	10	L1	CO5
		OR			
Q.10	a.	Explain the relationship between Auto correlation and the model parameter with necessary equations.	10	L2	CO5
	b.	Explain Burg Method for computing the AR model parameters.	10	L2	CO5

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