Register No : ________________

THE KAVERY ENGINEERING COLLEGE

(An Autonomous Institution)

M.E DEGREE END SEMESTER THEORY EXAMINATIONS, NOV/DEC 2025

First Semester

Applied Electronics

PAPT2401 -Advanced Digital Signal Processing

Regulations - 2024

Duration : 3 Hrs *Maximum : 100 Marks*

*PART – A (ANSWER ALL QUESTIONS) (Marks 10*2=20)*

Q.No	Questions	BLT	CO	Marks
1.	What is the minimum sampling rate for an analog signal?	L1	1	2
2.	Find output of digital filter given input and transfer function. $$H(z) = \frac{0.86 - 1.72z^{-1} + 0.86z^{-2}}{1 - 1.71z^{-1} + 0.73z^{-2}}$$	L2	1	2
3.	Write down the system function H(z) for a causal IIR Wiener filter.	L2	2	2
4.	What are the advantages of lattice structure?	L1	2	2
5.	Outline Welch's method for Averaging modified periodograms.	L2	3	2
6.	State applications of power spectrum estimation.	L2	3	2
7.	Name two applications of Multirate sampling.	L2	4	2
8.	Draw Interpolation or anti imaging lowpass filter functional diagram.	L2	4	2
9.	Interpret Spectrum after down sampling in Multirate signal processing.	L2	5	2
10.	Summarize the effects of sub band coding of speech signals.	L2	5	2

*PART – B (ANSWER ALL QUESTIONS) (Marks 5*16 = 80)*

Q.No			Questions	BLT	CO	Marks
11.	a		Select a suitable Image coding scheme from the available methods and judge why it is best?	L3	1	16
			Or			
	b	i	(Given x(n) = n+1, and N=8, Solve for X(K) using DIF FFT algorithm.	L3	1	8
		ii	Use 4-point inverse FFT for the DFT result {6,-2+2j,-2,- 2-2j} and identify the input sequence.	L3	1	8
12.	a	i	Apply impulse invariance method to design IIR digital filter.	L3	2	8
		ii	Convert the analog filter with transfer function $H_a(s) = s + 0.1 / (s + 0.1)^2 + 9$.	L3	2	8
			Or			
	b		With suitable examples state advantages of Lattice IIR filter design techniques.	L3	2	16

13.	a	Make use of Blach man turkey method to plan a power spectrum estimation.	L3	3	16
		Or			
	b	Model a report to solve issues arised in Barlett's method in periodogram averaging.	L3	3	16
14.	a	Inference the concepts of decimation by a factor D.	L4	4	16
		Or			
	b	Inference the concepts of interpolation by a factor I.	L4	4	16
15.	a	Explain digital to analogue conversion for a CD player using x8 oversampling.	L3	5	16
		Or			
	b	Explain about over sampling A/D and D/A Conversion in DSP Integrated circuits.	L3	5	16