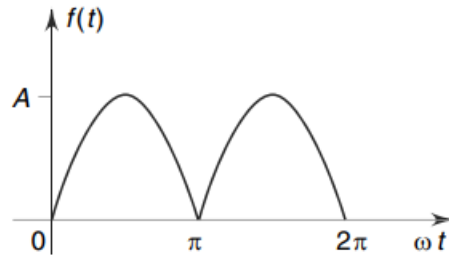


THE KAVERY ENGINEERING COLLEGE*(An Autonomous Institution)***B.E/B.TECH DEGREE END SEMESTER THEORY EXAMINATIONS, APRIL-MAY 2025***Fourth semester**Biomedical Engineering***BM3401 - Signal Processing***Regulations -2021***Duration : 3 Hrs****Maximum : 100 Marks****PART – A (ANSWER ALL QUESTIONS) (Marks 10*2=20)**

Q.No	Questions	BLT	CO	Marks
1.	Determine whether the following systems are static or dynamic $y(t) = x(2t) + 2x(t)$.	L2	1	2
2.	Differentiate between causal and non-causal signal.	L2	1	2
3.	State the necessary and sufficient conditions for the existence of the Fourier series representation for a signal.	L2	2	2
4.	Describe the relationship between DTFT and z-transform.	L2	2	2
5.	What are 'twiddle factors' of the DFT?	L1	3	2
6.	Explain the overlap-add method in linear convolution.	L2	3	2
7.	Summarize the advantages and disadvantages of FIR and IIR filters.	L2	4	2
8.	Discuss the characteristics of Chebyshev filters.	L2	4	2
9.	Explain the concept of Interpolation in multirate signal processing.	L2	5	2
10.	What is a linear phase FIR filter? Highlight its importance in digital signal processing.	L2	5	2

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

Q.No	Questions	BLT	CO	Marks
11.	a i Determine whether the signals are energy or power signal $x(t) = e^{-3t}u(t)$.	L3	1	8
	i Explain different classifications of systems based on their properties.	L2	1	8
	i			
	Or			
	b i Determine whether the system $y(n) = \sin x(n)$ is linear or nonlinear and static or dynamic.	L3	1	8
	i	L2	1	8
	i Illustrate with example, the aliasing effect in signal processing.			
12.	a Obtain the Fourier series for the full wave rectified sine waves.	L4	2	16



Or

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|---|--|----|---|----|
| b | Derive the DFT of the sample data sequence $x(n) = \{1, 1, 2, 2, 3, 3\}$ and compute the corresponding amplitude and phase spectrum. | L4 | 2 | 16 |
|---|--|----|---|----|
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|-----|---|---|--|----|---|---|
| 13. | a | i | Determine the convolution of the two sequences $x(n) = \{2, 1, 0, 0.5\}$ and $h(n) = \{2, 2, 1, 1\}$. | L3 | 3 | 8 |
| | | i | Find the linear convolution through circular convolution of | L3 | 3 | 8 |
| | | i | $x_1(n) = \delta(n) + \delta(n-1) + \delta(n-2)$ $x_2(n) = 2\delta(n) - \delta(n-1) + 2\delta(n-2)$ | | | |
- Or
- | | | | | | | |
|---|---|---|---|----|---|---|
| b | i | Determine the 8-point DFT of the sequence. | L3 | 3 | 8 | |
| | | $x(n) = \begin{cases} 1, & -3 \leq n \leq 4 \\ 0, & \text{otherwise} \end{cases}$ | | | | |
| | | i | Summarize the computation efficiency of FFT over DFT. | L3 | 3 | 8 |
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|-----|---|--|----|---|----|
| 14. | a | Use bilinear transform to design a first-order Butterworth LPF with 3 dB cut-off frequency of 0.2π . | L3 | 4 | 16 |
|-----|---|--|----|---|----|
- Or
- | | | | | |
|---|--|----|---|----|
| b | Convert the analog filter to digital filter whose system function is | L3 | 4 | 16 |
|---|--|----|---|----|
- $$H(s) = \frac{36}{(s+0.1)^2 + 36}$$
- The digital filter should have a resonant frequency of $\omega_r = 0.2\pi$. Use impulse invariant mapping.
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|-----|---|--|----|---|----|
| 15. | a | Derive the expression for the impulse response of a low-pass FIR filter using the frequency sampling method. | L2 | 5 | 16 |
|-----|---|--|----|---|----|
- Or
- | | | | | |
|---|--|----|---|----|
| b | Compare and contrast the characteristics of FIR and IIR filters in terms of stability, complexity, and computational efficiency. | L2 | 5 | 16 |
|---|--|----|---|----|