OF ARTS & SCIENCE

(AUTONOMOUS)

MSc DEGREE EXAMINATION MAY 2026

(Fourth Semester)

Branch – **APPLIED ELECTRONICS**

DIGITAL SIGNAL PROCESSING

Time: Three Hours

Maximum: 75 Marks

SECTION-A (10 Marks)

Answer **ALL** questions

ALL questions carry **EQUAL** marks (10 × 1 = 10)

Module No.	Question No.	Question	K Level	CO
1	1	A signal is defined as ___________. (i) A mathematical model of a physical system (ii) A function that conveys information about a phenomenon (iii) A hardware device (iv) A processing unit	K1	CO1
	2	Identify the memoryless system from the following options and choose the correct answer. (i) $y[n]=x[n]+x[n-1]$ (ii) $y[n]=x[n+1]$ (iii) $y[n]=2x[n]$ (iv) $y[n]=\sum x[n]$	K2	CO1
2	3	The DFT of an N-point sequence produces __________. (i) N frequency samples (ii) N/2 frequency samples (iii) 2N frequency samples (iv) Infinite samples	K1	CO2
	4	Circular convolution suffers from __________. (i) Aliasing (ii) Attenuation (iii) Amplification (iv) Quantization	K2	CO2
3	5	An ideal low pass filter has _____________. (i) Linear phase (ii) Finite impulse response (iii) Constant group delay (iv) Non-causal impulse response	K1	CO3
	6	The impulse response of an ideal filter is __________. (i) Exponential (ii) Sinc function (iii) Delta function (iv) Step function	K2	CO3
4	7	Direct Form–II realization requires ___________. (i) More memory than Direct Form–I (ii) Same memory as Direct Form–I (iii) Less memory than Direct Form–I (iv) No delay elements	K1	CO4
	8	Linear phase realization is possible in FIR filters when coefficients are ___________. (i) Random (ii) Complex (iii) Negative (iv) Symmetric or anti-symmetric	K2	CO4
5	9	Short-Time Fourier Transform (STFT) is used to _______. (i) Analyze time-varying frequency content (ii) Compress speech (iii) Remove noise (iv) Increase sampling rate	K1	CO5
	10	Anti-aliasing filter is used before ________. (i) Quantization (ii) encoding (iii) decoding (iv) Sampling	K2	CO5

Cont…

SECTION - B (35 Marks)

Answer **ALL** questions

ALL questions carry **EQUAL** Marks (5 × 7 = 35)

Module No.	Question No.	Question	K Level	CO
1	11.a.	Define a signal and a system. Classify signals with suitable examples.	K2	CO1
		(OR)		
	11.b.	Explain the properties of Linear Time-Invariant (LTI) systems.		
2	12.a.	Explain Radix-2 Decimation in Time (DIT) FFT algorithm.	K2	CO2
		(OR)		
	12.b.	Derive the expression for DFT and IDFT.	K3	
3	13.a.	Given a first-order recursive system, apply the structure of a simple IIR filter to derive its difference equation and compute the output for a given input sequence.	K3	CO3
		(OR)		
	13.b.	Using the rectangular window method, apply the procedure to design an FIR low pass filter for given cutoff frequency and filter length.		
4	14.a.	Describe Lattice structure of FIR filters.	K3	CO4
		(OR)		
	14.b.	Describe Parallel form realization using partial fraction expansion.		
5	15.a.	Analyze the working of a subband coding system with block diagram and explain its advantages over PCM.	K4	CO5
		(OR)		
	15.b.	Examine Doppler processing in radar signal analysis and explain its significance.		

SECTION -C (30 Marks)

Answer **ANY THREE** questions

ALL questions carry **EQUAL** Marks (3 × 10 = 30)

Module No.	Question No.	Question	K Level	CO
1	16	Explain in detail the elementary discrete-time signals. Derive the relationship between unit step and unit impulse signals.	K2	CO1
2	17	With suitable example, compute 8-point DFT using Radix-2 DIT FFT algorithm.	K2	CO2
3	18	Explain ideal digital filters (LPF, HPF, BPF, BSF) with magnitude response and impulse response characteristics.	K3	CO3
4	19	Given a transfer function, explain how to realize it using Cascade and Parallel forms.	K4	CO4
5	20	Analyze digital audio signal processing steps including sampling, quantization, filtering, and compression.	K4	CO5

Z-Z-Z END