Roll No.												

PRESIDENCY UNIVERSITY

BENGALURU

Mid - Term Examinations – March 2026

Date: 12- 03- 2026 **Time:** 09:30am – 11.00am

School:SOE	**Program:**B.Tech	
Course Code: ECE2520	**Course Name:** Digital Signal Processing	
Semester:IV	**Max Marks**:50	**Weightage**:25%

CO - Levels	CO1	CO2	CO3	CO4	CO5	CO6
Marks	26	24	0		0	

Instructions:

(i) Read all questions carefully and answer accordingly.

(ii) Do not write anything on the question paper other than roll number.

Part A

Answer ALL the Questions. Each question carries 2marks. **5Q x 2M=10M**

1	An analog signal contains frequency components up to 10 kHz. What is the minimum sampling rate required to sample this signal without aliasing?	**2 Marks**	**L2**	**CO1**
2	Describe the relationship between Discrete Fourier Transform and Discrete-Time Fourier Transform.	**2 Marks**	**L2**	**CO1**
3	State and prove periodicity property of DFT.	**2 Marks**	**L2**	**CO1**
4	How many multiplications and additions are required to compute N-point DFT using radix-2 FFT algorithm?	**2 Marks**	**L2**	**CO2**
5	Draw the basic butterfly diagram of radix-2 DIT FFT.	**2 Marks**	**L2**	**CO2**

Part B

Answer the Questions. **Total Marks 40M**

6.	**a.**	Convolution is used to determine the output of a linear time-invariant (LTI) system by combining the input signal with the system's impulse response. It helps analyze system behavior and is widely used in filtering, signal analysis, and system modeling. The input x(n) and impulse response h(n) of a LTI system are given by, x(n) = {0,1, 1, 2, 2}; h(n) = {0.5, 1, 1, 2, 0.75} Determine the response of the system a) linear convolution via circular convolution and b) using circular convolution.	**10 Marks**	**L2**	**CO1**
	b.	A digital communication system processes a long data sequence x(n) using a finite impulse response (FIR) filter with impulse response h(n) = {1, 1} The input sequence x(n) is: x(n) = {1, 1, 2, 2, 3, 3, 4, 4}. Given that the system employs the Overlap-Add method for efficient convolution,	**10 Marks**	**L2**	**CO1**
		Or			
7.	**a.**	In an audio processing application, you have recorded a discrete-time signal x[n] = {2, 1, 2, 1} representing a musical note sampled at N=4. Find the DFT of the sequence and sketch the frequency response.	**10 Marks**	**L2**	**CO1**
	b.	In a digital communication system, a transmitted signal x(n) x(n) = {1, 2, 3, 1, 2, 3, 4, 5, 6} samples are received with some distortion. To mitigate this, a low-pass FIR filter with impulse response h(n) = {2, 1, 1} is applied using the Overlap-Save method.	**10 Marks**	**L2**	**CO1**

8.	**a.**	FFT efficiently computes the DFT of a discrete-time signal by reducing computational complexity from $\boldsymbol{O(N^2)}$to $\boldsymbol{O(N\log_2 N)}$. Compared to direct DFT, FFT requires significantly fewer arithmetic operations, making it suitable for real-time signal processing. Examine the 8-point DFT of the sequence x(n) = {2,2,2,2,1,1,1,1} using DIT- FFT algorithm.	**10 Marks**	**L3**	**CO2**
	b.	IDFT using the DIT algorithm efficiently reconstructs the time-domain signal from its frequency-domain representation by decomposing the computation into smaller stages. Find the IDFT of the sequence X(K)= {6, -2+2j, -2, -2-2j} using Radix 2 -DIT algorithm.	**10 Marks**	**L3**	**CO2**
		Or			

9.	**a.**	DIT FFT computes the DFT by decimating the input sequence into even and odd samples, reducing computational complexity. Find the 8-point DFT of a given sequence x(n) = {1,2,2,1,1,2,2,1} using DIT- FFT algorithm.	**10 Marks**	**L3**	**CO2**
	b.	DIT butterfly combines two input samples using addition, subtraction, and twiddle factor multiplication to efficiently compute the DFT. With a neat sketch find 4-point DFT of the sequence x(n)=[1,6,7,4] using radix2 DIT-FFT algorithm**.**	**10 Marks**	**L3**	**CO2**