CBCS SCHEME

USN

21EC42

Fourth Semester B.E./B.Tech. Degree Examination, Dec.2025/Jan.2026
Digital Signal Processing

Time: 3 hrs. Max. Marks: 100

Note: ***Answer any FIVE full questions, choosing ONE full question from each module.***

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages.
2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

Module-1

1 a. Why is it necessary to perform frequency domain sampling? Illustrate the relationship between the sampled Fourier transform and the DFT. **(08 Marks)**

b. Find the four-point (N = 4) DFT of the sequence :

$$x(n) = \cos\left(\frac{\pi}{4}n\right) + \sin\left(\frac{\pi}{4}\right)n$$

use linearity property. **(08 Marks)**

c. Prove the periodicity property of DFT. **(04 Marks)**

OR

2 a. Show that the DFT can be viewed as a Linear transformation. **(04 Marks)**

b. For the sequences $x_1(n) = \cos\left(\frac{2\pi n}{N}\right)$ and $x_2(n) = \sin\left(\frac{2\pi n}{N}\right)$, $0 \le n \le N-1$, find the N – point circular convolution $x_1(n) \circledast_N x_2(n)$. **(08 Marks)**

c. Let x(n) = (1, 2, 0, 3, –2, 4, 7, 5). Evaluate :

i) X(0) ii) X(4) iii) $\sum_{k=0}^{7} X(k)$.

Also write a program to compute N-point DFT of a sequence using DSK 6713 simulator. **(08 Marks)**

Module-2

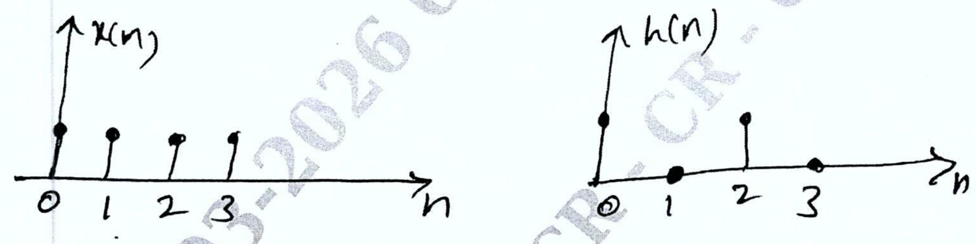
3 a. For h(n) = {3, 2, 1, 1} and x(n) = {1, 2, 3, 3, 2, 1, –1, –2, –3, 5, 6, –1, 2, 0, 2, 1}, find the output using overlap add method assuming block length as 7. **(10 Marks)**

b. i) Compute the 4-point DFT of the sequence x(n) = (1, 0, 1, 0) using DIT – FFT radix – 2 algorithm.
ii) Also, find x(n) found in part (i) using two different methods. **(10 Marks)**

OR

4 a. Perform x(n) ∗ h(n), $0 \le n \le 11$ for h(n) = (1, 1, 1) and x(n) = (1, 2, 0, –3, 4, 2, –1, 1, –2, 3, 2, 1, –3) using overlap save method. **(10 Marks)**

b. Find the 4-point circular convolution of x(n) and h(n) shown below using radix-2 DIF – FFT algorithm.



Also write a program to compute circular convolution using DFT and IDFT **(10 Marks)**

Module-3

5 a. Explain the following with relevant equations and response :
i) Rectangular window
ii) Bartlett window
iii) Hamming window
iv) Hanning window. **(10 Marks)**

b. The desired frequency response of a lowpass filter is given by :

$$H_d(e^{j\omega}) = H_d(\omega) = \begin{cases} e^{-j3\omega}, & |\omega| < \frac{3\pi}{4} \\ 0, & \frac{3\pi}{4} < |\omega| < \pi \end{cases}$$

Determine the frequency response of the FIR filter if Hamming window is used with N = 7. Also write a program to realize FIR low pass filter for particular specifications. **(10 Marks)**

OR

6 a. Realize the Linear-Phase FIR filter having the following impulse response :

$$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4).$$ **(10 Marks)**

b. Given the FIR filter with the following difference equation :

$$y(n) = x(n) + 3.1x(n-1) + 5.5x(n-2) + 4.2x(n-3) + 2.3x(n-4)$$

sketch the Lattice realization of the filter. **(10 Marks)**

Module-4

7 a. Design a lowpass Butterworth filter with the following specifications :

i) 3dB attenuation at passband frequency of 1.5 KHz
ii) 10dB stopband attenuation at frequency of 3 KHz
iii) Sampling frequency of 8 KHz. **(12 Marks)**

b. A filter is specified by the following transfer function. Obtain direct form – I and Direct form –II realization of the system :

$$H(z)=\frac{(z^2-1)(z^2-2z)}{\left(z^2+\frac{1}{16}\right)\left(z^2-z+\frac{1}{2}\right)}.$$ **(08 Marks)**

OR

8 a. Explain the concept of bilinear transformation and frequency warping with relevant equations and mapping properties. **(08 Marks)**

b. Given a lowpass prototype $H_p(s)=\frac{1}{s+1}$

i) Determine high pass filter with a cut off frequency of 40 radians per second.
ii) Determine the high pass filter with a cut off frequency 100 radians per second. **(08 Marks)**

c. Write a program to implement Butterwroth LPF to meet given specifications. **(04 Marks)**

Module-5

9 a. With neat block diagram, explain digital signal processors based on Harvard architecture and its special hardware units : MAC unit, shifters and address generators. **(12Marks)**

b. With example, explain :
i) Fixed point
ii) Floating point formats in DSP. **(08Marks)**

OR

10 a. With neat block diagram, explain fixed-point digital signal processors. **(10 Marks)**

b. The following IIR filter, y(n) = 2x(n) + 0.5y(n -1) uses the direct form-I and for a particular application, the maximum input is I_{max} = 0.010 . . . . 0_2 = 0.25. Develop the DSP implementation equations in the Q – 15 fixed point system. **(10 Marks)**

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