



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.

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 :  

$$x(n) = \cos\left(\frac{\pi}{4}n\right) + \sin\left(\frac{\pi}{4}n\right)$$
 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  $x_2(n) = \sin\left(\frac{2\pi n}{N}\right)$ ,  $0 \leq n \leq N - 1$ , find the N – point circular convolution  $x_1(n) \otimes_N x_2(n)$ . (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)

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.

OR

- 4 a. Perform  $x(n) * h(n)$ ,  $0 \leq n \leq 11$  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 :
- Rectangular window
  - Bartlett window
  - Hamming window
  - 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 :
- 3dB attenuation at passband frequency of 1.5 KHz
  - 10dB stopband attenuation at frequency of 3 KHz
  - 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 :

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

**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  $H_p(s) = \frac{1}{s+1}$
- Determine high pass filter with a cut off frequency of 40 radians per second.
  - Determine the high pass filter with a cut off frequency 100 radians per second. (08 Marks)
- c. Write a program to implement Butterworth 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 :
- Fixed point
  - 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  $I_{\max} = 0.010 \dots 0_2 = 0.25$ . Develop the DSP implementation equations in the Q – 15 fixed point system. (10 Marks)

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