

CBCS SCHEME

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18EE63

Sixth Semester B.E. Degree Examination, June/July 2025

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. Prove the following properties of DFT:
 - i) Frequency shift
 - ii) Linear property. (05 Marks)
- b. Find IDFT of the sequence. (07 Marks)

$$X(k) = \{ 5, 0, 1-j, 0, 1, 0, 1+j, 0 \}$$
- c. Let $X(k)$ be a 14 point DFT of length real sequence of $x(n)$. The first 8 samples of $X(k)$ are given by $X(0) = 12$, $X(1) = -1+j3$, $X(2) = 3+j4$, $X(3) = 1-j5$, $X(4) = -2+j2$, $X(5) = 6+3j$, $X(6) = -2-3j$, $X(7) = 10$. Find the remaining sample of $X(k)$. Also evaluate i) $X(0)$ ii) $X(7)$ (08 Marks)

OR

- 2 a. Obtain the linear convolved output $y(n) = x(n) * h(n)$ using circular convolution. Given that $x(n) = [1, 1, 0, -1, -1]$ and $h(n) = [1, 2, 3, 2, 1]$ (08 Marks)
- b. Find the Output of LT1 system whose impulse response $h(n) = [1, 1, 1]$ and i/p signal $x(n) = [3, -1, 0, 1, 3, 2, 0, 1, 2, \dots]$ using overlap add method. Use block length $N = 5$. (12 Marks)

Module-2

- 3 a. Find the DFT of the sequence $x(n) = 2^n$ using DIT – FFT algorithm. Where $x(n) = 2^n$ for $0 \leq n \leq 7$. (10 Marks)
- b. What are FFT algorithms? Find the number of computations required to find the DFT of an
 - i) Direct method
 - ii) DITFFT Algorithm. If $N = 16$, $N = 256$, $N = 1024$ (06 Marks)
- c. Explain the difference between DIT and DIF algorithm. (04 Marks)

OR

- 4 a. The DFT $x(k)$ of sequence is given by $x(k) = [0, 2\sqrt{2}(1-j), 0, 0, 0, 0, 0, 2\sqrt{2}(1+j)]$. Determine the corresponding time sequence $x(n)$ and write the SFG using ID1F – FCT algorithm. (10 Marks)
- b. Find the 4 point circular convolution of $x(n) = \{4, 3, 2, 1\}$ with $h(n) = \{1, 2, 3, 4\}$ using Radix – 2 decimation in time FFT Algorithm. (10 Marks)

Module-3

- 5 a. Given that $|H_a(j\Omega)|^2 = \frac{1}{1+64\Omega^6}$. Determine the analog filter system function $H_a(s)$. (10 Marks)

- b. Determine $H(z)$ using impulse invariance technique for the analog s/m

$$\text{function } H_a(s) = \frac{1}{(s+0.5)(s^2+0.55s+2)}$$

(10 Marks)

OR

- 6 a. Design a second order band pass Chebyshev filter with the passband of 200Hz to 300Hz and $\Delta p = 0.5\text{dB}$. (10 Marks)
- b. Determine $H(z)$ of lowest order Butterworth filter that will meet the following specifications.

i) 1dB ripple in passband; $0 \leq \omega \leq 0.15\pi$ rad

ii) Atleast 20db attenuation in } : $0.45\pi \leq \omega \leq \pi$ rad
stop band

Use bilinear Transformation for $T = 1$ sec.

(10 Marks)

Module-4

- 7 a. Design a digital Chebyshev – I filter that satisfies the following constraints.
 $0.800 \leq |H(j\omega)| \leq 1$ $0 \leq \omega \leq 0.27\pi$

$$|H(j\omega)| \leq 0.2 \quad 0.6\pi \leq \omega \leq \pi$$

Use impulse invariant transformation.

(12 Marks)

- b. Write the comparison between Butterworth and Chebyshev filter.

(08 Marks)

OR

- 8 a. Design a digital Chebyshev – I filter using bilinear transformation to meet the following specifications.

i) 3db ripple in passband ; $0 \leq |\omega| \leq 0.3\pi$

ii) 20db attenuation in the stop band ; $0.6\pi \leq |\omega| \leq \pi$

(10 Marks)

- b. Obtain cascade and parallel realizations for the system function given by

$$H(z) = \frac{1 + \frac{1}{4}z^{-1}}{(1 + \frac{1}{2}z^{-1})(1 + \frac{1}{2}z^{-1} + \frac{1}{4}z^{-2})}$$

(10 Marks)

Module-5

- 9 a. Design the symmetrical FIR Lowpass filters whose desired freq. response is given as

$$H_d(\omega) = \begin{cases} e^{-j\omega\tau} & \text{for } |\omega| \leq \omega_c \\ 0 & \text{Otherwise} \end{cases}$$

The length of the filter should be 7 and $\omega_c = 1\text{rad/sample}$. Use Hanning window.

(12 Marks)

- b. Realize a linear phase FIR having

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

(08 Marks)

OR

- 10 a. Design a low pass filter (FIR) using frequency sampling technique having cut – off freq. of $\frac{\pi}{2}$ rad/sample. The filter should have linear phase and length 17. (12 Marks)

- b. Realize a FIR filter with impulse response $h(n)$ is given by

$$H(n) = \left(\frac{1}{2}\right)^n [u(n) - u(n-4)]$$

(08 Marks)
