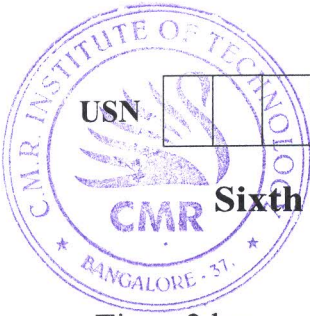


CBCS SCHEME



18EE63

Sixth Semester B.E. Degree Examination, Dec.2023/Jan.2024 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 linearity property of DFT for the given signal $x(n) = x_1(n) + x_2(n)$, where $x_1(n) = \sin \frac{n\pi}{4}$ and $x_2(n) = \cos \frac{n\pi}{4}$ for $N = 4$. (12 Marks)
- b. Given two signals $x_1(n) = n + 1, 0 \leq n \leq 3$ and $x_2(n) = \{1, 2, 1, 3\}$. Find the circular convolution of these signals using Stockham's method. (08 Marks)

OR

- 2 a. Compute the output of a LTI system whose impulse response $h(n) = \{1, 2, 1\}$, and the input signal is $x(n) = \{5, 6, 7, 1, 2, 3, 4, 5, 6, 7\}$ using overlap-add method, with circular array length $N = 6$. (10 Marks)
- b. Prove the circular time shift property for DFT. If a given signal $x(n) = \{1, -1, 2, 3\}$. Find the DFT of $x(n)$. If a new signal $y(n) = x(n - 2)$, find $Y(K)$ using time shift property. (10 Marks)

Module-2

- 3 a. Find out 8 point DFT of $x(n) = n^2, 0 \leq n \leq 7$ using DIT-FFT radix-2 algorithm. Draw the full flow diagram. (12 Marks)
- b. Explain the following terms in FFT algorithms:
 - (i) Computational complexity
 - (ii) In place computation(08 Marks)

OR

- 4 a. The first five points of a real signal is given by $X(K) = \{0, 2 + j2, -j4, 2 - j2, 0\}$. Determine the remaining DFT points and find the sequence $x(n)$ using inverse DIF-FFT radix-2 algorithm. (10 Marks)
- b. Determine the circular convolution of $x_1(n) = \{1, 1, 2, 2\}$ and $x_2(n) = \{1, 2, 3, 4\}$ using DFT-IDFT method by radix-2 DIF-FFT algorithm. (10 Marks)

Module-3

- 5 a. Explain how a frequency response is transformed from analog domain to digital domain using bilinear transformation. (10 Marks)
- b. Design a digital low pass filter using bilinear transformation to follow the following characteristics:
 - (i) Monotonic stop band and pass band
 - (ii) -3.01 dB cut off frequency of 0.5π rad
 - (iii) Stop band attenuation atleast 15 dB at 0.75π rad(10 Marks)

OR

- 6 a. Use Impulse Invariant transformation to obtain $H(z)$ for the analog transformations with $T = 1$ sec.

$$(i) H_a(s) = \frac{2}{(s+1)(s+2)} \quad (ii) H_a(s) = \frac{1}{s^2 + \sqrt{2}s + 1} \quad (10 \text{ Marks})$$

- b. Design an analog Chebyshev type 1 filter having following specifications:

- i) Pass band gain 2.5 dB at $\Omega_p = 20$ rad/sec
 ii) Stop band attenuation 30 dB at $\Omega_s = 50$ rad/sec

Show the pole positions and obtain analog filter transformation. (10 Marks)

Module-4

- 7 a. Design a digital low pass Chebyshev filter that meets the following specifications:

- i) $K_p = -1$ dB $0 \leq \omega_p \leq 0.2\pi$ rad
 ii) $K_s = -15$ dB $0.3\pi \leq \omega \leq \pi$ rad
 Use bilinear transformation.

(12 Marks)

- b. What is frequency transformation? Explain how it is used in designing filters. (08 Marks)

OR

- 8 a. A digital filter is given by

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

Obtain direct form – I and form – II structure. (08 Marks)

- b. For the equation $H(z)$ given in Q8(a), obtain (i) Cascaded realization (ii) Parallel realization (12 Marks)

Module-5

- 9 a. The desired frequency response of a lowpass filter is given by

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

Determine the filter coefficients and frequency response of the FIR filter if Hamming window is used. (10 Marks)

- b. Explain how windowing techniques are used in designing FIR filters. (10 Marks)

OR

- 10 a. Design a low pass FIR filter using frequency sampling technique, with cut off frequency

$$\omega_c = \frac{\pi}{2}. \text{ The filter must have linear phase and length } N = 17 \quad (12 \text{ Marks})$$

- b. For a FIR filter $H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$. Obtain :

- (i) Direct form structure (ii) Linear phase structure

Obtain the impulse response of the given FIR filter. (08 Marks)
