CBCS SCHEME

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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 $x_1(n) = n + 1,\ 0 \le n \le 3$ and $x_2(n) = \{\underset{\uparrow}{1}, 2, 1, 3\}$. 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 $h(n) = \{\underset{\uparrow}{1}, 2, 1\}$, 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)4, find Y(K) using time shift property. **(10 Marks)**

Module-2

3 a. Find out 8 point DFT of $x(n) = n^2,\ 0 \le n \le 7$ 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 0.5 π rad
(iii) Stop band attenuation atleast 15 dB at 0.75 π rad **(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

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

(i) $H_a(s) = \frac{2}{(s+1)(s+2)}$ (ii) $H_a(s) = \frac{1}{s^2 + \sqrt{2}s + 1}$ **(10 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) $K_p = -1\text{dB} \quad 0 \le \omega_p \le 0.2\pi$ rad

ii) $K_s = -15$ dB $\quad 0.3\pi \le \omega \le \pi$ 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 $\omega_c = \frac{\pi}{2}$. The filter must have linear phase and length N = 17 **(12 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)**

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