# CBCS SCHEME

USN

15EE63

## Sixth Semester B.E. Degree Examination, June/July 2019
## Digital Signal Processing

Time: 3 hrs. Max. Marks: 80

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



### Module-1

1 a. Determine DFT of sequence $x(n) = \frac{1}{3}$ for $0 \le n \le 2$ for N = 4. Plot magnitude and phase spectrum. **(08 Marks)**

b. Two length – 4 sequence are defined below :

$$x(n) = \cos\left(\frac{\pi n}{2}\right) \quad n = 0, 1, 2, 3$$

$$h(n) = 2^n \quad n = 0, 1, 2, 3$$

i) Calculate $x(n) \circledast_4 h(n)$ using circular convolution directly.

ii) Calculate $x(n) \circledast_4 h(n)$ using Linear convolution. **(08 Marks)**

**OR**

2 a. Compute circular convolution using DFT + IDFT for following sequence :

$$x_1(n) = \left\{\underset{\uparrow}{2}, 3, 1, 1\right\}, \quad x_2(n) = \left\{\underset{\uparrow}{1}, 3, 5, 3\right\}.$$ **(08 Marks)**

b. Find the output of the LTI system whose impulse h(n) = {1, 1, 1} and the input signal is x(n) = {3, –1, 0, 1, 3, 2, 0, 1, 2, 1}. Using the overlap save method. Use 6-pt circular convolution. **(08 Marks)**

### Module-2

3 a. What are FFT algorithms? Explain the advantages of FFT algorithms over the direct computations of DFT for a sequence x(n). **(04 Marks)**

b. What are the differences and similarities between DIT and DIF –FFT algorithms? **(04 Marks)**

c. Find the 8-pt DFT of the sequence x(n) = {1, 2, 3, 4, 4, 3, 2, 1}. Using DIT – FFT radix – 2 algorithm. **(08 Marks)**

**OR**

4 a. Find the 4-pt circular convolution of x(n) and h(n) given. Using radix-2 DIF – FFT algorithm. **(08 Marks)**

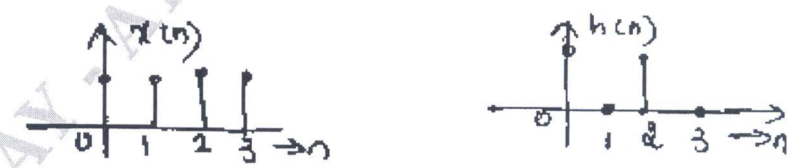


Fig.Q4(a)

b. Given x(n) = (n + 1) and N = 8. Determine X(K). Using DIF – FFT algorithm. **(08 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.

## Module-3

5 a. Convert the analog filter with system transfer function :

$$H(s) = \frac{(s+0.1)}{(s+0.1)^2 + 3^2}$$

into a digital IIR filter by mean of the impulse invariant method. **(06 Marks)**

b. Design a butter worth digital IIR lowpass filter using bilinear transformation by taking $T = 0.1$sec, to satisfy the following specification :

$$0.6 \le \left|H(e^{j\omega})\right| \le 1.0; \quad \text{for } 0 \le \omega \le 0.35\pi$$
$$\left|H(e^{j\omega})\right| \le 0.1; \quad \text{for } 0.7\pi \le \omega \le \pi$$

**(10 Marks)**

**OR**

6 a. Compare analog and digital filters. **(04 marks)**

b. Determine the poles of lowpass Butterworth filter for $N = 2$. Sketch the location of poles on s-plane and hence determine the normalized transfer function of lowpass filter. **(08 Marks)**

c. Write difference between IIR and FIR filter. **(04 Marks)**

## Module-4

7 a. Design a Chebyshev digital IIR lowpass filter using impulse invariant transformation by taking $T = 1$ sec to satisfy the following specifications;

$$0.9 \le \left|H(e^{j\omega})\right| \le 1.0; \quad \text{for } 0 \le \omega \le 0.25\pi$$
$$\left|H(e^{j\omega})\right| \le 0.24; \quad \text{for } 0.5\pi \le \omega \le \pi$$

Draw direct form – I and II structure of the filter. **(12 Marks)**

b. Write the relation between analog and digital frequency in Billnear transformation. **(04 Marks)**

**OR**

8 a. Obtain the direct form – I, direct form II realization of the LTI system governed by the relation.

$$y(n) = -\frac{3}{8}y(n-1) + \frac{3}{32}y(n-2) + \frac{1}{64y}y(n-3) + x(n) + 3x(n-1) + 2x(n-2).$$

**(08 Marks)**

b. Realize the given system in cascade and parallel form :

$$H(z) = \frac{1+0.25z^{-1}}{(1-2z^{-1}+0.25z^{-2})(1-3z^{-1}+0.25z^{-2})}.$$

**(08 Marks)**

## Module-5

9 a. The frequency response of a filter is described by : $H(\omega) = j\omega, -\pi \le \omega \le \pi$. Design the filter using a rectangular window. Take $N = 7$. **(08 Marks)**

b. Design a lowpass digital filter to be used in A/D – H(z) – D/A structure that will have – 3dB cutoff at $30\pi$ rad/sec and attenuation factor of 5dB at $45\pi$ rad/sec. The filter is required to have a linear phase and the system will use sampling frequency of 100 samples/sec. **(08 Marks)**

**OR**

10 a. Deduce the equation for the following frequency spectrum for rectangular window sequence defined by :

$$w_f(n) = \begin{cases} 1, & \frac{-(N-1)}{2} \le n \le \frac{N-1}{2} \\ 0, & \text{otherwise} \end{cases}.$$ **(06 Marks)**

b. A lowpass filter has the desired frequency response :

$$H_d(\omega) = \begin{cases} e^{-j\omega 3}, & 0 < \omega < \pi/2 \\ 0, & \text{otherwise} \end{cases}.$$

Determine h(n) based on frequency sampling method. Take K = 7. **(06 Marks)**

c. Realize the linear phase FIR filter having the following impulse response :

$$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4).$$ **(04 Marks)**

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