CBCS SCHEME

USN | | | | | | | | | | | **21EC42**

Fourth Semester B.E. Degree Examination, June/July 2023
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. Define. DFT and IDFT and solve for the 4-point DFT of the sequence x(n) = [0, 1, 2, 3] and also write program to find N-point DFT. **(10 Marks)**

b. Explain the process of frequency domain sampling and reconstruction of discrete time signal. **(10 Marks)**

OR

2 a. Summarize multiplication of two DFT properties and also write a program to verify Pasval's theorem. **(08 Marks)**

b. Make use of DFT and IDFT to compute circular convolution of the sequence.
x(n) = [2, 3, 1, 1] and h(n) = [1, 3, 5, 3]. **(08 Marks)**

c. The five samples of 8-point DFT X(K) are given X(0) = 0.5, X(1) = - j2, X(4) = X(6) = 0. X(5) = + j2. Make use property to find remaining Samples and also find x(0). **(04 Marks)**

Module-2

3 a. Explain the computational arrangement of 8-point DFT using Radix – 2 DIT-FFT algorithm. **(12 Marks)**

b. Examine the o/p y(n) = x(n) ∗ h(n) if x(n) = [1, 0] and h(n) – [1, 3, 1] using Radix – 2 DIT – FFT algorithm. **(08 Marks)**

OR

4 a. Examine the output of y(n) of a filter where impulse response h(n) = [3, 2, 1] input sequence x(n) = [2, 1, +1, –2, 3, 5, 6, –7, 2, 0, 2, 1]. Use 8-point circular convolution in your approach using overlap add method. **(08 Marks)**

b. Solve for 8-point DFT of the sequence x(n) = [1, 1, 1, 1] using Radix – 2 DIT-FFT algorithm. **(08 Marks)**

c. What is the speed improvement factor in calculating 128 point DFT of sequence using direct computation and FFT algorithm? **(04 Marks)**

Module-3

5 a. What are the different design techniques are available for FIR filter? Explain the four window techniques for the designing of FIR filter. **(08 Marks)**

b. A low pass filter is to be designed with the following desired frequency response.

$$H_d(e^{f\omega}) = \begin{cases} e^{f3\omega} & \text{for} \quad |\omega| \le 3\pi/4 \\ 0 & \text{for} \quad \text{otherwise} \end{cases}$$

Determine $H(e^{f\omega})$ for M = 7 using Hamming window. **(08 Marks)**

c Determine the direct form Relaization of the following :

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

OR

6 a. Formulate the expression for symmetric FIR filter. **(08 Marks)**

b. Write a program and design for FIR Lowpass filter using humming window for M = 7 and

$$\omega_c = 3\pi/4 \quad H_d(\omega) = \begin{cases} e^{-t3\omega} & \text{for} \quad |\omega| \le \omega_c \\ 0 & \text{for} \quad \text{otherwise} \end{cases}.$$ **(08 Marks)**

c. Realize a linear phase FIR filter with following Impulse. Response

$H(z) = 1 + \frac{3}{4}z^{-1} + \frac{17}{8}z^{-2} + \frac{3}{4}z^{-3} + z^{-4}$ in cascade form. **(04 Marks)**

Module-4

7 a. Given that $|H_a(\Omega)|^2 = \dfrac{1}{1+16\Omega^4}$. Determine the Analog filter system function $H_a(S)$. **(08 Marks)**

b. Develop an analog filter with maximally flat response. In pass band with acceptable, attenuation of 2dB at 20rad/sec, the alteration in stop band more than that 10dB beyond 30rad/sec. **(08 Marks)**

c. Write program to implementation of IIR Butterworth Lowpass filter. **(04 Marks)**

OR

8 a. Realization of direct form – I and direct form – II of IIR filter is given by

$$H(z) = \frac{3+4z}{z-\frac{1}{2}} - \frac{2}{z-\frac{1}{4}}.$$ **(06 Marks)**

b. Make use of Bilinear transformation to obtain digital filter with $w_r = \pi/2$ and $\Omega = 4$ form given analog filter $H_a(s) = \dfrac{s+0.1}{(s+0.1)^2+16}$. **(08 Marks)**

c. Write a program. Design and implementation of high pass filter to meet specification. **(06 Marks)**

Module-5

9 a. Describe the IEEE single precision floating point digital signal processors. **(08 Marks)**

b. Describe the digital signal processes following units :

i) Multiplier and accumulator

ii) Address generation unit. **(08 Marks)**

c. Determine following number into Q_{15} notation.

i) 0560123 ii) –0.160123. **(04 Marks)**

OR

10 a. Explain fixed point digital signal processors of TMS320 family. **(08 Marks)**

b. Explain digital signal processor using Harvard architecture. **(06 Marks)**

c. Write a program for linear convolution of two sequences. Using DSK6713 DSP processor. **(06 Marks)**

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