



Sixth Semester B.E. Degree Examination, June/July 2023
Digital Signal Processing

Time: 3 hrs

Max. Marks:100

Note: Answer any FIVE full questions, selecting atleast TWO questions from each part.

PART – A

- 1 a. State and prove that following properties of DFT
 i) Linearity ii) circular frequency shift (06 Marks)
- b. Find the 8-point DFT of the sequence
 $x(n) = 1$ for $0 \leq n \leq 3$
 $= 0$, for $4 \leq n \leq 7$ (08 Marks)
- c. Let $x(n) = (1, 2, -3, 0, 1, -1, 4, 2)$. Evaluate the following without computing DFT
 i) $x(0)$ ii) $x(4)$ iii) $\sum_{k=0}^7 X(k)$ iv) $\sum_{k=0}^7 |X(k)|^2$ (06 Marks)
- 2 a. Determine the circular convolution of the following sequence using DFT and IDFT method.
 $x_1(n) = \{1, 1, 2, 1\}$ $x_2(n) = \{1, 2, 3, 4\}$ (10 Marks)
 Find the output $y(n)$ of a filter whose impulse response is $h(n) = (1, 1, 1)$ and input
 b. $x(n) = (3, -1, 0, 1, 3, 2, 0, 1, 2, 1)$, using overlap add method. Use only 5-point circular convolution. (10 Marks)
- 3 a. Find the number of computations required to find the DFT of 32-point sequence using
 i) Direct method ii) DIT FFT algorithm. Also find the speed improvement factor. (06 Marks)
- b. Find the 8-point DFT of a sequence $x(n) = (1, 1, 1, 1, 0, 0, 0, 0)$ using DIT-FFT radix-2 algorithm. (10 Marks)
- c. What are the differences and similarities between DIT and DIF algorithms? (04 Marks)
- 4 a. Find the circular convolution of the sequences $x(n) = (1, 1, 1, 1)$ and $h(n) = (1, 0, 1, 0)$ using DIF – FFT algorithm. (10 Marks)
- b. Develop DIT – FFT algorithm for composite value of $N = 9$. Draw the corresponding signal flow graph. (10 Marks)

PART – B

- 5 a. The system function of analog filter is given as $H_a(s) = \frac{10}{s^2 + 7s + 10}$. Obtain $H(z)$ using impulse invariant transformation. Take $T = 0.2$ seconds. (12 Marks)
- b. A digital low pass Butterworth filter is required to meet the following specifications
 i) -3.01 dB cut-off frequency of 0.5π radians
 ii) Stop band attenuation of atleast 15 dB at 0.75π radians.
 Find the system function $H(z)$ using Bilinear transformation for $T = 1$ Sec. (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.

- 6 a. Distinguish between Butterworth and Chebyshev – I filter. (04 Marks)
 b. Design a Chebyshev analog lowpass filter that has – 3.01dB cutoff frequency of 100 rad/sec and a stopband attenuation of 25dB or greater for all frequencies past 250 rad/sec. (10 Marks)
 c. Derive expression for poles from squared magnitude response of Butterworth lowpass filter for cutoff frequency of 1 rad/sec. (06 Marks)

- 7 a. A lowpass filter has the desired frequency response

$$H_d(\omega) = H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} & 0 < \omega < \frac{\pi}{2} \\ 0 & \frac{\pi}{2} < \omega < \pi \end{cases}$$

Determine $h(n)$ based on frequency sampling method. Take $N = 7$.

(12 Marks)

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

$$H_d(e^{j\omega}) = \begin{cases} e^{-j2\omega} & -\frac{\pi}{4} \leq \omega \leq \frac{\pi}{4} \\ 0 & \frac{\pi}{4} < \omega \leq \pi \end{cases}$$

Determine the filter coefficient $h_d(n)$ and $h(n)$ if the window function is defined as

$$W_R(n) = \begin{cases} 1, & 0 \leq n \leq 4 \\ 0 & \text{Otherwise} \end{cases}$$

(08 Marks)

- 8 a. Determine the direct forms I and II for the IIR filter given by
 $y(n) = 2b \cos \omega_0 y(n-1) - b^2 y(n-2) + x(n) - b \cos \omega_0 x(n-1)$. (08 Marks)
 b. Determine the parallel form realization of the IIR digital filter transfer function

$$H(z) = \frac{3(2z^2 + 5z + 4)}{(2z + 1)(z + 2)}$$

(06 Marks)

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

$$h(n) = \delta(n) - \frac{1}{4} \delta(n-1) + \frac{1}{2} \delta(n-2) + \frac{1}{2} \delta(n-3) - \frac{1}{4} \delta(n-4) + \delta(n-5)$$

(06 Marks)
