



Fifth 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. Compute the N point DFT of the given sequence.

$$x(n) = \frac{\cos 2\pi}{N} K_0 n \quad 0 \leq n \leq N-1. \quad (05 \text{ Marks})$$
- b. State and prove the circular frequency shift property of DFT. (05 Marks)
- c. G(k) and H(k) are the 6 point DFTs of sequence g(n) and h(n) respectively. The DFT G(k) is given as

$$G(k) = \{1 + j, -2.1 + j3.2, -1.2 - j2.4, 0, 0.9 + j3.1, -0.3 + j1.1\}$$
 The sequence g(n) and h(n) are related by the circular time shift as $h(n) = g((n-4))_6$. Determine H(k), without computing the DFT. (10 Marks)

OR

- 2 a. Let x(n) be the sequence, $x(n) = \delta(n) + 2\delta(n-2) + \delta(n-3)$.
 i) Find the 4 point DFT of x(n).
 ii) If y(n) is the 4-point circular convolution of x(n) with it self, find y(n) and four point DFT Y(K). (10 Marks)
- b. Given the sequence $x_1(n) = \{1, 2, 3, 1\}$ and $x_2(n) = \{4, 3, 2, 2\}$. Find y(n) such that $Y(K) = X_1(K) \cdot X_2(K)$. (05 Marks)
- c. The even samples of the 11-point DFT of length – 11 real sequence are given by $X(0) = 4$, $X(2) = -1 + j3$, $X(4) = 2 + j5$, $X(6) = 9 - j6$, $X(8) = -5 - j8$ and $X(10) = \sqrt{3} - j2$. Determine the missing odd samples of the DFT. (05 Marks)

Module-2

- 3 a. Compute y(n) of the filter whose impulse response is $h(n) = (3, 2, 1, 1)$ and input is given by $x(n) = \{1, 2, 3, 3, 2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1\}$. Use overlap save method. Assume the length of the block is 9. (10 Marks)
- b. Let X(K) be a 14-point DFT of a length 14 real sequence x(n). The first 8 samples of X(K) are given by
 $X(0) = 12, X(1) = -1 + j3, X(2) = 3 + j4, X(3) = 1 - j5, X(4) = -2 + j2, X(5) = 6 + j3,$
 $X(6) = -2 - j3, X(7) = 10$. Determine the remaining samples of X(K). Also evaluate the following functions without computing the IDFT.
- i) x(0) ii) x(7) iii) $\sum_{n=0}^{13} x(n)$ iv) $\sum_{n=0}^{13} |x(n)|^2$. (10 Marks)

OR

- 4 a. Compute the speed improvement factor in calculating 64 point DFT of a sequence using direct computations and FFT algorithms. (05 Marks)
- b. Find the 4 point DFT of the sequence $x(n) = \{2 + j2, -3 + j1, 2 + j1, -1 + j3\}$. Make use of this result compute the DFT of the sequence $y(n) = \{1, -1.5, 1, -0.5\}$ and $z(n) = \{j4, j2, +j2, j6\}$. (10 Marks)
- c. List and prove the 2 property which improves the efficiency of computation of DFT using FFT algorithm. (05 Marks)

Module-3

- 5 a. Given $x(n) = n + 1$ for $0 \leq n \leq 7$. Compute the $X(K)$ using DIF FFT algorithm. (12 Marks)
- b. Illustrate Chirp Z transform concept with contours for different values of r , R θ and ϕ . (08 Marks)

OR

- 6 a. If $x_1(n) = \{1, 2, 0, 1\}$ and $x_2(n) = \{1, 3, 3, 1\}$. Obtain 4-point circular convolution by using DIT-FFT algorithm. (10 Marks)
- b. Discuss the Goertzel algorithm in computation of sparse DFT and list its advantages. (10 Marks)

Module-4

- 7 a. A digital filter is given by

$$H(z) = \frac{1 - \frac{1}{2}z^{-1}}{\left(1 - \frac{1}{3}z^{-1}\right)\left(1 - \frac{1}{4}z^{-1}\right)}$$

Obtain the parallel form structure. (06 Marks)

- b. Design an IIR – digital filter that when used in the pre filter A/D – H(Z) – D/A structure will satisfy the following equivalent analog specifications: Use BLT.
- LPF with -1db cut off at 100π rad/sec.
 - Stopband attenuation of 35db or greater at 1000π rad/sec.
 - Monotonic stopband and passband.
 - Sampling rate of 2000 samples/sec.
- (14 Marks)

OR

- 8 a. Convert the following second order analog filter with system transfer function

$$H(S) = \frac{s + a}{(s + a)^2 + b^2}$$

into a digital filter with impulse response by the use of impulse invariance mapping technique. (08 Marks)

- b. Consider the first order LPF with passband edge frequency Ω_p having its transfer function

$$H_a(s) = \frac{\Omega_p}{s + \Omega_p}$$

Transfer the filter to

- LPI with passband edge frequency Ω_1 .
 - HPF with cutoff frequency Ω_2 .
- (06 Marks)
- c. Determine the order and magnitude squared function of the chebyshev filter for the following specifications:
- Maximum passband ripple is 1db.
 - Stopband attenuation is 40db for $\Omega \geq 4$ rad/sec.
- (06 Marks)

Module-5

- 9 a. Realize the linear phase FIR filter with the following impulse response. Give the necessary equation. $h(n) = \delta(n) + \frac{1}{2}\delta(n-1) - \frac{1}{4}\delta(n-2) + \delta(n-4) + \frac{1}{2}\delta(n-3)$. (06 Marks)
- b. Obtain the lattice structure corresponding to the FIR filter described by the system function $H(z) = 1 + 2z^{-1} + \frac{1}{3}z^{-2}$. (06 Marks)
- c. Design the symmetric FIR lowpass filter whose desired frequency response is $H_d(w) = \begin{cases} e^{-jw\tau} & \text{for } |w| \leq W_c \\ 0 & \text{otherwise} \end{cases}$. The length of the filter should be 7 and $W_c = \pi/4$ rad/sample. Use rectangular window. (08 Marks)

OR

- 10 a. Derive the steps involved in design of an FIR filter using window function. Also list the advantage of FIR filter. (10 Marks)
- b. The desired frequency response of a LPF is
- $$H_d(e^{jw}) = \begin{cases} e^{-j3w} & -\frac{3\pi}{4} \leq w \leq \frac{3\pi}{4} \\ 0 & \frac{3\pi}{4} < |w| \leq \pi \end{cases}$$
- Design it using Hamming window ($N = 7$). Also obtain its frequency response. (10 Marks)

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