



Fifth Semester B.E. Degree Examination, Feb./Mar. 2022
Digital Signal Processing

Max. Marks:100

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

PART – A

- 1 a. Define sampling and Aliasing. Consider the analog signal $x_1(t) = 2 \cos 2\pi (10t)$ and $x_2(t) = 2 \cos 2\pi (50t)$. Find a sampling frequency so that 50Hz signal is an alias of the 10Hz signal. (08 Marks)
- b. If $x(n) = 1 \quad 0 \leq n \leq 5$
 $= 0$ elsewhere and $x(z)$ its z transform. If $x(z)$ is sampled at $z = e^{j\frac{2\pi}{4}k} \quad 0 \leq k \leq 3$.
 Find $y(n)$ obtained as IDFT of $x(k)$. (06 Marks)
- c. Prove the following identities:
 i) DFT $(\delta(n)) = 1$ ii) DFT $[x^*(n)] = X^*(N - K)$ (06 Marks)
- 2 a. If $x(n) = \{1, 2, 3, 4\}$. Find $x_1(n) = x((n-1))_4$ and $x_2(n) = x((n+1))_4$. (04 Marks)
- b. The first five point of 8-point DFT is $x(k) = \{0.5, 1 - j, 0, 1 - j1.72, 0\}$. Find the remaining three points. (04 Marks)
- c. If the DFT of $x(n) = x(k)$, What are the DFT's of $x(-n)_{\text{mod}N}$, $X^*(N - K)$. (12 Marks)
- 3 a. Perform the circular convolution of the signals $x_1(n) = \{2, 1, 2, 1\}$, $x_2(n) = \{1, 2, 3, 4\}$. Using DFT and IDFT method. (08 Marks)
- b. With Butterfly diagram, explain radix – 2 DIT FFT algorithms. (08 Marks)
- c. State and prove Parseval's theorem pertaining to discrete Fourier transformation. (04 Marks)
- 4 a. If $x(n) = \{1, 2, 2, 3, 1, 1, 4, 2\}$. Find $X(k)$ using Radix – 2 Decimation in Frequency FFT algorithm. (10 Marks)
- b. If $X(k) = \{17, -1.12 - j7.12, j3, 3.121 + j2.87, 3, 3.12 - j2.87, -j3, -1.12 + j7.12\}$. Find $x(n)$ using radix-2 inverse DIT – FFT algorithm. (10 Marks)

PART – B

- 5 a. What is impulse invariant transformation? For the analog filter $H_a(s) = \frac{2}{(s+1)(s+2)}$.
 Find $H(z)$ using impulse invariant transformation for $T = 0.1\text{Sec}$. (08 Marks)
- b. Design Low Pass Butterworth filter using Bilinear Transformation for the following :
 Pass band = 0 – 400Hz, Stop band = 2.1 to 4Khz, Pass band ripple = 2dB, Stop band attenuation = 20dB, Sampling frequency = 10KHz (12 Marks)
- 6 a. Obtain Direct – Form II and cascade, realization of the system

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

- b. Realize the following system with minimum numbers of multipliers
$$H(z) = \left(1 + \frac{1}{2}z^{-1} + z^2\right) \left(1 + \frac{1}{4}z^{-1} + z^{-2}\right)$$
 (04 Marks)
- c. Explain procedure for designing FIR Filter using Kaiser window. (06 Marks)
- 7 a. Design Band Pass FIR Filter for $N = 8$, $w_{c1} = 0.3$ radians, $w_{c2} = 0.8$ radian, using hamming window. (10 Marks)
- b. Design a high pass FIR filter for $N = 4$, $w_{c1} = 1.5$ radians, using Hamming window. (10 Marks)
- 8 a. Design a high pass digital Chebyshev filter with cutoff frequency = 50Hz, $N = 2$, Pass band ripple = 1dB. Sampling frequency = 500Hz. (12 Marks)
- b. Design a lowpass FIR Filter using frequency sampling method having cutoff frequency $w_c = \pi/3$ radians and $N = 6$. (08 Marks)

CMRIT LIBRARY
BANGALORE - 560 037