Printed Page:- **Subject Code:- AEC0601**

Roll. No:

NOIDA INSTITUTE OF ENGINEERING AND TECHNOLOGY, GREATER NOIDA

(An Autonomous Institute Affiliated to AKTU, Lucknow)

B.Tech

SEM: VI - THEORY EXAMINATION (2022-2023)

Subject: Digital Signal Processing

Time: 3 Hours **Max. Marks: 100**

General Instructions:

IMP: *Verify that you have received the question paper with the correct course, code, branch etc.*

1. *This Question paper comprises of* **three Sections -A, B, & C.** *It consists of Multiple Choice Questions (MCQ's) & Subjective type questions.*

2. *Maximum marks for each question are indicated on right -hand side of each question.*

3. *Illustrate your answers with neat sketches wherever necessary.*

4. *Assume suitable data if necessary.*

5. *Preferably, write the answers in sequential order.*

6. *No sheet should be left blank. Any written material after a blank sheet will not be evaluated/checked.*

SECTION A **20**

1. Attempt all parts:-

1-a. The process of converting discrete-time continuous valued signal into discrete-time discrete valued (digital) signal is known as ____________. (CO1) 1

(a) Sampling

(b) Differentiating

(c) Encoding

(d) Quantization

1-b. In DIF-FFT……doamin sequence is decimated. (CO1) 1

(a) Time

(b) Frequency

(c) Both time & frequency

(d) None of them

1-c. The transformation technique in which there is one to one mapping from s-domain to z-domain is… (CO2) 1

(a) Backward difference for the derivative

(b) Bilinear transformation method

(c) Impulse invariance method

(d) Approximation of derivatives

1-d. The poles of butterworth filter lie on a... (CO2) 1

(a) circle

(b) parabola

(c) ellipse

(d) helix

1-e. What is the peak side lobe (in dB) for a Hanning window? (CO3) 1

(a) -13

(b) -27

(c) -32

(d) -58

1-f. The finite word length effects are due to... (CO3) 1

(a) Quantization of input signal

(b) Quantization of system coefficients

(c) Quantization of product

(d) All of these

1-g. In high speed filtering applications...... (CO4) 1

(a) parallel realization is preferred

(b) Cascade realization is preferred

(c) Linear realization is preferred

(d) None of these

1-h. When the no.of delays is equal to order of the system, the structure is called... (CO4) 1

(a) Canonic

(b) Noncanonic

(c) Cascade

(d) None of these

1-i. Decimation process is used to.... (CO5) 1

(a) decrease the sampling rate

(b) Increase the sampling rate

(c) no change

(d) None of these

1-j. Quadrature mirror filter is a filter whose magnitude response is the .... of that of another filter. (CO5) 1

(a) mirror image around (π/2)

(b) optical image

(c) mirror image

(d) inverted image

2. Attempt all parts:-

2.a. What is bit reversal process in DIT-FFT algorithm? (CO1) 2

2.b. Compare analog frequency transformation with Digital frequency transformation. (CO2) 2

2.c. What is Gibbs phenomenon explain with diagram? (CO3) 2

2.d. Define all zero or moving average system. (CO4) 2

2.e. What is the need of adaptive filtering? (CO5) 2

SECTION B **30**

3. Answer any <u>five</u> of the following:-

3-a. Compute the 4-point DFT of a given sequence x(n)={0,1,2,1}. Also sketch the magnitude and phase spectrum. (CO1) 6

3-b. State and prove the Periodicity and Linearity property of DFT. (CO1) 6

3-c. Draw the frequency mapping in bilinear transformation method. Write down the advantages and disadvantages of bilinear transformation method. (CO2) 6

3-d. Determine H(z) using the impulse invariant technique for the analog system function. (CO2) 6

$$H(s) = \frac{1}{(s+0.5)(s^2+0.5s+2)}$$

3.e. Briefly explain the phenomenon of finite word length effect in digital system with suitable diagram. (CO3) 6

3.f. Write a short note on (a) recursive (b) non recursive system. (CO4) 6

3.g. Consider the discrete time signal x(n)={1,2,3,4,5,6,7,8,9,10,11,12} , determine the down sampled version of signals for sampling rate reduction factor (a) D=2 (b) D=3. (CO5) 6

SECTION C **50**

4. Answer any <u>one</u> of the following:-

4-a. An input sequence x(n) = {2, 1, 0, 1, 2} is applied to a DSP system having an impulse sequence h(n) = {5, 3, 2, 1}. Determine the output sequence by linear convolution, and verify the same through circular convolution. (CO1) 10

4-b. Given x(n) = (n + 1), and N = 8 find X(K) using DIF FFT algorithm. Also, calculate the computational reduction factor. (CO1) 10

5. Answer any <u>one</u> of the following:-

5-a. Design a Chebyshev digital low pass IIR filter using bilinear transformation to satisfy the following specifications. (Assume T=1 sec). (CO2) 10

Passband: $0.8 \le |H(e^{j\omega})| \leqslant 1 \qquad |\omega| \leqslant 0.2\pi$

Stopband: $|H(e^{j\omega})| \leqslant 0.2 \qquad 0.32\pi \leqslant |\omega| \leqslant \pi$

5-b. Find the system function H(z) of the digital Butterworth filter that meets the following specification: (CO2) 10

a) 1 dB ripple in passband $0 \le \omega \ge 0.3\pi$

(b) at least 40 dB attenuation in stopband $0.8\pi \le \omega \ge \pi$

Use BLT when T=1sec.

6. Answer any <u>one</u> of the following:-

6-a. Derive the frequency response of linear phase FIR filter when antisymmetric impulse response and M is even. (CO3) 10

6-b. Design a linear phase low pass digital filter if the desired frequency response is giving by 10

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

Using the bartlett window and choosing a suitable length of filter length M, find the impulse response and frequency response of designed filter. Determine the system function and difference equation. Also draw the linear phase structure of designed filter. (CO3)

7. Answer any <u>one</u> of the following:-

7-a. Obtain the direct form-I , direct form-II, cascade, and parallel form realization structures for the following system.(CO4) 10

$y(n) = -0.1\, y(n-1) + 0.72\, y(n-2) + 0.7x(n) - 0.25\, x(n-2)$

7-b. Determine the coefficients of a continued-fraction expansion of H(z); Also draw ladder realization structure of a given IIR system. (CO4) 10

$$H(z) = \frac{2 + 8z^{-1} + 6z^{-2}}{1 + 8z^{-1} + 12z^{-2}}$$

8. Answer any <u>one</u> of the following:-

8-a. Briefly explain the phenomenon of Subband coding of speech signals with neat diagram. (CO5) 10

8-b. Derive the mathematical expression for recursive least square algorithm in detail. (CO5) 10