Printed Page:- 04 **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 (2023 - 2024)
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. Which of the following conditions made digital signal processing more advantageous over analog signal processing?(CO1) 1

- (a) Flexibility
- (b) Accuracy
- (c) Storage
- (d) All of the mentioned

1-b. The total number of complex additions required computing N point DFT by radix-2 FFT is? (CO1) 1

- (a) (N/2)log2N
- (b) N/log2N
- (c) N/log2(N/2)
- (d) None of above

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 chebyshev filter lie on a....(CO2) 1

(a) circle
(b) parabola
(c) ellipse
(d) helix

1-e. FIR filter is always stable because all of its poles are…(CO3) 1

(a) at origin
(b) at ROC
(c) At infinity
(d) None of these

1-f. The main lobe width of length M bartlett window is…(CO3) 1

(a) $4\pi/M$
(b) $8\pi/M$
(c) $12\pi/M$
(d) $16\pi/M$

1-g. If we reverse the directions of all branch transmittances and interchange the input and output in the flow graph, then the resulting structure is called as _______________ (CO4) 1

(a) Direct form-I
(b) Transposed form
(c) Direct form-II
(d) None of the mentioned

1-h. The factors influence the choice of realization of structure is... (CO4) 1

(a) Memory requirement
(b) Computational complexity
(c) parallel processing & pipelining
(d) All of the mentioned

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. The choice of a particular adaptive algorithm depends on …... (CO5) 1

(a) rate of convergence
(b) steady state error
(c) computaional complexity
(d) all of these

2. Attempt all parts:-

2.a. Write down the various application of DSP in real world. (CO1) 2

2.b. What are the basic differences between impulse invariant and bilinear transformation method? (CO2) 2

2.c. What are the basic differences between infinite impulse and finite impulse response system? (CO3) 2

2.d. Define canonic and non canonic structure. (CO4) 2

2.e. What is the need of multirate signal processing? (CO5) 2

SECTION-B 30

3. Answer any five of the following:-

3-a. Derive the expression for the relationship between DFT and Z-transform.(CO1) 6

3-b. Determine the 4-point DFT of a given sequence $x(n)=\cos(\Pi n)$ using linear transformation matrix. (CO1) 6

3-c. Draw the relationship between analog and digital frequency in impulses invariant method. Also write down the advantages and disadvantages of this method. (CO2) 6

3-d. What are the basic differences between analog and digital frequency transformation. (CO2) 6

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

3.f. What is the physical significance of difference equation? Derive the expression for transfer function of All zero 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 one of the following:-

4-a. Given two sequences $x1(n) = \{1, 2, 2, 1\}$ and $x2(n) = \{2, 1, 1, 2\}$. Determine the circular convolution of x1(n) and x2(n) using: (a) Graphical Method (b) DFT/IDFT method. (CO1) 10

4-b. Write down the basic difference between DFT and FFT? Derive the expression for the DIT-FFT algorithm for N = 8 and draw the signal flow graph. (CO1) 10

5. Answer any one of the following:-

5-a. Design a digital low pass Butterworth filter that satisfies the following: 10
(a) Passband cutoff frequency: $\Omega p = 0.2\pi$
(b) Passband attenuation: Ap =7 dB
(c) Stopband cutoff frequency: $\Omega s = 0.3\pi$
(d) Stopband attenuation: As =16 dB
(e) Use the Bilinear transformation method, assume T=1 sec (CO2)

5-b. Design a Chebyshev digital low pass IIR filter using bilinear transformation to satisfy the following specifications. (Assume T=1 sec). (CO2) 10
Passband: $0.8 \le |H(e^{j\omega})| \le 1 \qquad |\omega| \le 0.2\pi$
Stopband: $|H(e^{j\omega})| \le 0.2 \qquad 0.32\pi \le |\omega| \le \pi$

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

6-a. Derive the expression for designing of FIR filter using Fourier series method. Also Explain Gibbs phenomenon with suitable diagram. (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. Explain the concept of multistage sampling rate conversion. In the system of given figure, find Y(z) in terms of X(z). Also find y(n) in terms of x(n). (CO5) 10