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 (2024- 2025)
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, K2) 1

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

1-b. If x(n) and X(k) are an N-point DFT pair, then X(k+N)=? (CO1, K1) 1

(a) X(-k)
(b) -X(k)
(c) X(k)
(d) None of above

1-c. What is the relation between s and z in impulse invariant method? (CO2, K2) 1

(a) z=e^(sT)
(b) s=e^(zT)
(c) z=e^(-sT)
(d) s= (2+zT)/(2-zT)

1-d. The poles of butterworth filter lie on a… (CO2, K1) 1

(a) circle
(b) parabola

(c) ellipse
(d) helix

1-e. The main lobe width of length M bartlett window is…(CO3, K1) 1

(a) 4π/M
(b) 8π/M
(c) 12π/M
(d) 16π/M

1-f. If a linear phase filter has a phase response of 40 degree at 200 Hz, what will be its phase response at a frequency of 400 Hz. (CO3, K2) 1

(a) 35 degree
(b) 45 degree
(c) 40 degree
(d) 80 degree

1-g. If M and N are the orders of numerator and denominator of rational system function respectively, then how many multiplications are required in direct form-I realization of that IIR filter? (CO4, K2) 1

(a) M+N-1
(b) M+N
(c) M+N+1
(d) M+N-2

1-h. Which of the following is used in the realization of a system? (CO4, K1) 1

(a) Delay elements
(b) Multipliers
(c) Adders
(d) All of the mentioned

1-i. In multirate signal processing, interpolation method is used to….(CO5, K1) 1

(a) decrease the sampling rate
(b) Increase the sampling rate
(c) no change
(d) None of thses

1-j. What are the applications of adaptive filters in real time? (CO5, K1) 1

(a) system identification
(b) speech coding
(c) channel equalization
(d) all of these

2. Attempt all parts:-

2.a. Enlist the advantages of DSP over ASP. (CO1, K2) 2

2.b. Write down various approaches for designing of IIR filters from an analog filter. 2

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(CO2, K2)

2.c. Write down advantages and disadvantages of FIR filters. (CO3, K2) 2

2.d. Define canonic and non canonic structure. (CO4, K2) 2

2.e. What do you mean by multirate signal processing? (CO5, K1) 2

SECTION-B 30

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

3-a. What are the basic differences between linear and circular convolutional? (CO1, K2) 6

3-b. Derive the expression for the relationship between DFT and Z-transform.(CO1, K3) 6

3-c. Draw the frequency mapping in bilinear transformation method. Write down the advantages and disadvantages of bilinear transformation method. (CO2, K3) 6

3-d. Derive the expression for the relationship between analog and digital poles in impulse invariant method. (CO2, K3) 6

3.e. Derive the expression for linear phase and symmetric impulse response for an FIR filter. (CO3, K3) 6

3.f. Define FIR filter structure with the help of mathematical function. (CO4, K2) 6

3.g. If x(n)= {2,4,6,8,10,12,14,16} & down sampling factor=3, then what will be the value of up sampler output. Also draw the time domain representation of signal. (CO5, K3) 6

SECTION-C 50

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

4-a. State and prove circular convolution property of DFT. What is zero padding? What are its uses? (CO1, K3) 10

4-b. Find the 4-point circular convolution of x(n) and h(n) given by x(n)={1,1,1,1} & h(n)={1,0,1,0} using FFT algorithm. (CO1, K4) 10

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

5-a. Differentiate between filter and frequency transformation. Also draw the table showing digital frequency transformation for LPF to HPF, BPF and BSF. (CO2, K2) 10

5-b. 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, K4)

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, K3) 10

6-b. Define Quantization noise with the help of truncation and rounding process. Also 10 differentiate between Fixed point and floating-point representation. (CO3, K2)

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

7-a.	Briefly explain the steps used in drawing the ladder structure for IIR system. Also explain RH table. (CO4, K2)	10
7-b.	Obtain the direct form-I , direct form-II, cascade, and parallel form realization structures for the following system.(CO4, K4) $y(n) = - 0.1\, y(n-1) + 0.72\, y(n-2) + 0.7x(n) - 0.25\, x(n-2)$	10

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, K2)	10
8-b.	Derive the mathematical expression for recursive least square algorithm in detail. (CO5, K3)	10

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