Subject Code: **AEC0601**

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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 question paper with 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 value of W_4^{-1} is (CO1, K1) 1

(a) 0

(b) -j

(c) j

(d) 1

1-b. DFT performs filtering operation in (CO1, K1) 1

(a) Time domain

(b) Frequency domain

(c) both time and frequency domains

(d) none of these

1-c. Non-linearity in the relationship between Ω and ω is known as (CO2, K1) 1

(a) aliasing

(b) frequency warping

(c) unwarping

(d) frequency mixing

1-d. Butterworth filters Poles lies on (CO2, K2) 1

(a) ellipse

(b) circle

(c) both (a) and (b)

(d) none of the above

1-e. For Rectangular window, the width of the main lobe is (CO3, K1) 1

(a) $4\pi/N$

(b) $8\pi/N$

(c) $12\pi/N$

(d) Adjustable

1-f. For Blackman window, the peak side lobe in dB is (CO3, K1) 1

(a) -13

(b) -31

(c) -41

(d) -58

1-g. The number of multipliers required for the realization of FIR systems is reduced if we choose (CO4), K2) 1

(a) direct form

(b) cascade form

(c) parallel form

(d) linear phase realization

1-h. The structure which in not used to realize to FIR filters (CO4), K1) 1

(a) direct form-I

(b) Linear phase

(c) cascade form

(d) Ladder form

1-i. Down sampling by a factor of D skips (CO5), K1) 1

(a) D samples

(b) $D – 1$ samples

(c) no samples

(d) $D/2$ samples

1-j. If $x(n) = \{1, 2, 3, 4, 5, 6, 7, ...\}$, then $x(2n) =$ (CO5, K2) 1

(a) $\{2, 4, 6, 8, 10, ...\}$

(b) {1, 0, 2, 0, 3, 0, 4, 0, 5, 0, 6, 0, ...}

(c) {1, 3, 5, 7, ...}

(d) {1, 0, 0, 2, 0, 0, 3, 0, 0, 4, 0, 0, 5, 0, 0 ...}

2. Attempt all parts:-

2.a.	Write the advantages of DSP over ASP.	(CO1,K2)	2
2.b.	Compare Butterworth and Chebyshev filters.	(CO2,K2	2
2.c.	Differentiate symmetric and antisymmetric FIR filters.	(CO3,K2)	2
2.d.	Explain the concept of recursive and non-recursive systems.	(CO4,K2)	2
2.e.	Explain the concept of multirate signal processing.	(CO5,K2)	2

SECTION – B 30

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

3-a. Find the Circular convolution of the sequences 6

$x_1(n)$= {1, 1, 2, 2} and $x_2(n)$= {1, 2, 3, 4} (CO1, K3)

3-b. Find the 4-point IDFT of X(k) = {3, (2+j), 1, (2-j)}. (CO1, K3) 6

3-c. Explain the Bilinear Transformation technique to convert analog low pass filter into digital low pass filter in detail. (CO2,K2) 6

3-d. For the analog transfer function 6

$$H(s) = \frac{1}{(s+1)(s+3)}$$

determine H(z) using impulse invariant technique for T= 1sec. (CO2, K3)

3-e. Compare FIR filter and IIR Filter. Also, state various disadvantages of FIR filter in detail. (CO3, K2) 6

3-f. Explain FIR Linear phase realization in detail. (CO4, K2) 6

3-g. Explain the Decimation and Interpolation processes in multirate signal processing. (CO5, K2) 6

SECTION – C 50

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

4-a. Find the 4 point DFT of the sequence x(n) = {0, 1, 2, 3} using 10

(a) DIT-FFT algorithm and (b) DIF-FFT algorithm. (CO1, K3)

4-b. Explain the followings: (CO1), K3) 10

a) Twiddle factor and its properties

b) Circular Convolution of Two Sequences in time and frequency domains.

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

5-a. Convert the analog filter with system function (CO2, K4) 10

$$Ha(s) = \frac{s + 0.1}{(s + 0.1)^2 \ + 16}$$

into a digital IIR filter by means of the bilinear transformation. The digital filter is to have a resonant frequency of $\omega_r = \pi /2$.

5-b. Determine the order and the poles of a low-pass Butterworth filter that has a 1-dB attenuation in the passband, a cutoff frequency $\Omega_P = 1000\pi$, a stopband frequency of 2000π, and an attenuation of 40 dB or more for $\Omega > \Omega_S$. (CO2, K4) 10

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

6-a. Explain the frequency response of Linear phase FIR filter for Symmetrical impulse response when N is odd. (CO3), K3) 10

6-b. The desired response of a low pass filter is 10

$H_d(e^{j\omega}) = e^{-j2\omega}$, $-\pi/4 \leq \omega \leq \pi/4$

0 , $\pi/4 \leq |\omega| \leq \pi$

Determine $H(e^{j\omega})$ for M = 5 using a rectangular window. (CO3,K3)

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

7-a. Realize the IIR filter 10

$$H(z) = \frac{3z^2 + 5z + 4}{z^2 + 6z + 8}$$

In Direct-I and ladder structures. (CO4,K3)

7-b. An LTI system is described by the equation 10

$$y(n) + 2y(n - 1) - y(n - 2) = x(n)$$

Determine the cascade and parallel realization structures of the system. (CO4,K3)

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

8-a. Explain the folloings: 10

(a) Two- channel Quadrature Mirror Filter bank,

(b) Subband coding of speech signal. (CO5,K3)

8-b. What is adaptive filter? Given the system modelling described and using the single-weight adaptive filter y(n) = wx(n) to perform the system modelling task, 10

a) Set up the LMS algorithm to implement the adaptive filter, assuming that the initial w = 0 and μ = 0:5.

b) Perform adaptive filtering to obtain y (0), y (1), y (2), and y (3) given d (0) =1, d (1) = 2, d (2) = -2, d (3) = 2,

x (0) = 0.5, x (1) = 1, x (2) = -1, x (3) = 1 (CO5,K4)