Code: 15A04603 **R15**

B.Tech III Year II Semester (R15) Regular & Supplementary Examinations September/October 2021

DIGITAL SIGNAL PROCESSING

(Common to ECE & EIE)

Time: 3 hours Max. Marks: 70

PART – A

(Compulsory Question)

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1 Answer the following: (10 X 02 = 20 Marks)

- (a) Define linear and stable systems.
- (b) Give any two applications of DFT.
- (c) What do you mean by radix-2?
- (d) How many multiplications and additions are required to compute 4-point DFT using radix-2 FFT?
- (e) What are the factors that influence the choice of structure for realization of an LTI system?
- (f) What is the procedure to realize the lattice structure of FIR system?
- (g) List the three well known design techniques for linear FIR filters.
- (h) What is prewarping?
- (i) What is decimation?
- (j) Name the areas in which multirate signal processing is used.

PART – B

(Answer all five units, 5 X 10 = 50 Marks)

UNIT – I

2 (a) Verify the linearity, causality, shift-variant or invariant for the following difference equation:

$$y(n) = 3y\,2(n-1) - nx(n) - 4x(n-1) - x(n+1), \quad n \geq 0.$$

(b) A discrete-time system has a unit sample response $h(n)$ given by:

$$h(n) = \frac{1}{2}\partial(n) + \partial(n-1) + \frac{1}{2}\partial(n-2)$$

Find the system frequency response $H(e^{jw})$ and also plot magnitude response.

OR

3 Compute the DFT of the 3-point sequence $x(n) = \{2, 1, 2\}$. Using the same sequence, compute the 6-point DFT and compare the two DFTs.

UNIT – II

4 Compute 4-point DFT of a sequence $x(n) = \{0, 1, 2, 3\}$ using: (i) DIT. (ii) DIF algorithms.

OR

5 (a) Explain in detail about Goertzel algorithm.

(b) Explain the Chirp-z transform in the computation of DFT.

UNIT – III

6 (a) Discuss in detail about lattice structure of an FIR filter.

(b) Discuss about cascade form structure of FIR system.

OR

7 (a) With neat sketch, explain the Lattice-Ladder structure for IIR system.

(b) Realize the system with difference equation:

$y(n) = \frac{3}{4}\ y(n-1) - \frac{1}{8}\ y(n-2) - x(n) + \frac{1}{3}\ x(n-1)$ in cascade form.

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UNIT – IV

8 Design a filter with:

$$H_d = \left(e^{jw}\right) = e^{-jw} \qquad for \quad -\frac{\pi}{4} \leq w \leq \frac{\pi}{4}$$
$$= 0 \qquad for \quad \frac{\pi}{4} \leq w \leq \Pi$$

Using a Hamming window with N = 7.

OR

9 (a) Write the steps to design digital filter using bilinear transformation method.

(b) Discuss about frequency transformation in analog and digital domains.

UNIT – V

10 (a) Are up sampler and down sampler time variant or time invariant? Explain.

(b) Discuss about aliasing effect.

OR

11 (a) Explain multistage implementation of sampling rate conversion.

(b) Write the advantages of multirate signal processing.

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Code: 13A04602 **R13**

B.Tech III Year II Semester (R13) Supplementary Examinations September/October 2021

DIGITAL SIGNAL PROCESSING

(Common to ECE & EIE)

(For 2013, 2014 regular & 2015 lateral entry admitted batches only)

Time: 3 hours Max. Marks: 70

PART – A

(Compulsory Question)

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1 Answer the following: (10 X 02 = 20 Marks)

(a) Find the z-transform of the signal x(n) = [1, 3, 6, 8, 0, 1].

(b) When aliasing is present in a signal? Illustrate.

(c) Why direct computation of DFT is inefficient?

(d) What is the computational complexity of chirp-z transform?

(e) Compare direct form-I and direct form-II realization of IIR filter?

(f) What is the necessary and sufficient condition for linear phase characteristics of FIR filter?

(g) For the FIR filter if the unit sample response is anti symmetrical, find the filter coefficients when M is odd and even.

(h) Distinguish between type-I and type-II Chebyshev filter.

(i) With an example, show the output of a signal is decimated by factor of 3.

(j) List any four applications of multirate signal processing.

PART – B

(Answer all five units, 5 X 10 = 50 Marks)

UNIT – I

2 (a) Find x(n) for: $x(z) = \frac{1+\frac{1}{2}z^{-1}}{1-\frac{1}{2}z^{-1}}$.

(b) How frequency analysis of a signal is done using DFT?

OR

3 (a) An FIR digital filter has unit impulse response h(n) = {2, 2, 1}. Determine the output sequence in response to the input sequence x(n) = {3, 0, -2, 0, 2, 0, -2, -1, 0} using overlap add method.

(b) Explain any three properties of DFT.

UNIT – II

4 Compute the eight point DFT of the sequence:

$$x(n) = \begin{cases} 1, & 0 \le n \le 7 \\ 0, & \text{otherwise} \end{cases}$$

By using decimation in frequency FFT algorithm.

OR

5 (a) Draw the flow graph for the DIT FFT algorithm using radix-2.

(b) Explain how DFT is computed using Goertzel algorithm.

UNIT – III

6 Obtain the direct form-I, direct form-II, cascade and parallel structure for the system:

$$y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1).$$

OR

7 (a) With an example, show how lattice structure is converted to direct form in IIR system.

(b) Explain about frequency sampling FIR structures.

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UNIT – IV

8 (a) Design a FIR linear phase digital filter having ideal frequency response:

$$H_d(\omega) = 1, \text{ for } |\omega| \leq \frac{\pi}{6}$$
$$= 0, \text{ for } \frac{\pi}{6} < |\omega| \leq \pi$$

Determine the coefficients for M = 7 using rectangular window.

(b) For the analog transfer function:

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

Determine H(z) using impulse invariant technique.

OR

9 (a) A digital filter with a 3dB bandwidth of 0.25π is to be derived from the analog filter whose system response is $H(s) = \frac{\Omega c}{s+\Omega c}$.

(b) Compare all the methods used for designing FIR filter.

UNIT – V

10 Discuss about sampling rate conversion: (i) By a rational factor I/D. (ii) Of band pass signals.

OR

11 Discuss about subband coding of speech and implementation of digital filter banks.

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