

DIGITAL SIGNAL PROCESSING

(Common to ECE and EIE)

Time: 3 hours

Max. Marks: 70

PART – A

(Compulsory Question)

- 1 Answer the following: (10 X 02 = 20 Marks)
- What is meant by signal processing?
 - If $X(k)$ is the DFT of a sequence $x(n)$, then what are DFT of only real part of $x(n)$ and DFT of only imaginary part of $x(n)$?
 - What is the main advantage of FFT?
 - How many multiplications and additions are required to compute N-point DFT using radix-2 FFT?
 - What is the non-recursive system?
 - When does a cascade form realization is preferred in FIR filter?
 - Give the bilinear transformation equation between s-plane and z-plane.
 - What is the need for employing window technique for FIR filter design?
 - What is meant by multi rate signal processing?
 - What is the effect of upsampling and down sampling?

PART – B

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

UNIT – I

- 2 Find the response of the system described by the difference equation $y(n] + 2y(n-1) + y(n-2) = x(n) + x(n-1)$ for the input $x(n) = (\frac{1}{2})^n u(n)$, with initial conditions $y(-1) = y(-2) = 1$.

OR

- 3 Determine and sketch the magnitude and phase response of $y(n) = \frac{1}{2} [x(n) + x(n-2)]$.

UNIT – II

- 4 Find the DFT of the sequence $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ using radix-2 DIT-FFT algorithm.

OR

- Explain in detail about Goertzel algorithm.
- Explain about Quantization errors in the computation of DFT.

UNIT – III

- Discuss about signal flow graph and transposed structure.
- Determine the transposed direct form II for the given system: $y(n) = \frac{1}{2} y(n-1) - \frac{1}{4} y(n-2) + x(n) + x(n-1)$.

OR

- Obtain the cascade realization of system function: $H(z) = (1 + 2z^{-1} - z^{-2})(1 + 2z^{-1} - z^{-2})$.
- Discuss about Lattice structures of an FIR filter.

UNIT – IV

- 8 Design a filter with:

$$H_d = (e^{jw}) = e^{-j3w} \quad \text{for} \quad -\frac{\pi}{4} \leq w \leq \frac{\pi}{4}$$

$$= 0 \quad \text{for} \quad \frac{\pi}{4} \leq w \leq \pi$$

Using a Hamming window with $N = 7$.**OR**

- 9 Design a third order Butterworth digital filter using impulse invariant technique. Assume sampling period $T = 1$ sec.

UNIT – V

- Upsampler and Down sampler are time variant or invariant systems. Explain.
- Discuss about aliasing effect.

OR

- Explain multistage implementation of sampling rate conversion.
- Write the advantages of multirate signal processing.
