Time: 3 hrs   **Max. Marks: 100**

Answer ALL Questions

Part – A (10 x 2 = 20 Marks)

1.   State and prove convolution property of Z transform

2.   Find linear convolution of using Z transform
       {1, 0, 5} {2, 1, 2}
        ↑   ↑

3.   List DFT properties.

4.   Draw butterfly signature for 2 point FFT DIT method.

5.   Compare Butterworth and Chebyshev filter.

6.   Find digital filter equivalent for

$$H(s) = \frac{1}{S-7}$$

7.   Discuss about windowing technique in FIR filter design

8.   Realize $(2Z^{-2}+3Z^{-1}+1)/(Z^{-2}+Z^{-1}+1)$ using direct form I

9.   Signal (2K) is coded with 8 bits. Find bandwidth (bit rate) requirement for this signal.

10.   Design a sampling rate converter for 4 and list problems due to this rate conversion.

Part – B (5 x 16 = 80 Marks)

11(i)   Find convolution of
          {1, 0, 3, 2} and {3, 0, 0, 2} using linear convolution   (6)
(ii) find the inverse Z transform using long division method

$$H(Z) = \frac{1 + 5 Z^{-1} - 2Z^{-2}}{1-4Z^{-1}+5Z^{-2}}$$ (10)

12. a. Derive and draw 8 point FFT of DIT method using Radix 2 butterfly structure. (16)

(OR)

b. Find FFT for (1,2,2,1,2,1,1,2) using FFT- DIF method (16)

13 a. Design linear phase FIR filter with N = 9 for the following specification with hanning Window (16)

$$H(e^{jw}) = 1 \qquad \Pi \geq |w| \geq \Pi/3$$
$$= 0 \qquad \text{otherwise}$$

(OR)

b. Design a Lowpass filter of finite impulse response with hanning window and N = 7 for the following specification. (16)

$$H(e^{jw}) = e^{-j3w} \qquad -3\Pi/4 \leq w \leq 3\Pi/4$$
$$= 0 \qquad 3\Pi/4 < |w| < \Pi$$

14 a. Design a HPF with -3dB loss at 200 Hz and -20dB loss at 100Hz and convert this filter into digital filter with T = 1sec using impulse invariant technique. (16)

(OR)

b. Realize the following filter using DF-I and DF-II technique

$$H(Z) = \frac{1 + 5 Z^{-1}}{(1-4Z^{-1}+8Z^{-2})(1+8Z^{-1})} \qquad (16)$$

15.a(i) Discuss in detail about Multi rate signal processing, Interpolator, decimator and periodogram (16)

(OR)

b.(i) Discuss about Vocoder, and application of DSP on speech signal