Duration: 3hrs **[Max Marks: 80]**

N.B. : (1) Question No 1 is Compulsory.
(2) Attempt any three questions out of the remaining five.
(3) All questions carry equal marks.
(4) Assume suitable data, if required and state it clearly.

1 Attempt **any FOUR** **[20]**

a Compute the DFT of the four point sequence x(n)={0,1,2,3}
b Explain Quantization and effects of truncation and rounding.
c Compare IIR and FIR filters.
d Explain the term linear phase and state its importance in digital filters.
e Convert an analog filter with system function H(s) into digital IIR filter using Impulse Invariance method. $H(s) = 10 / (S^2 + 7S + 10)$

2 **a** Explain the application of DSP in speech processing **[10]**
b Obtain the linear convolution of the following sequences using overlap add method. x(n) = {1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1} and h(n) = {1, 2, 3} **[10]**

3 **a** Draw architectural block diagram of DSP processor and explain functions of each block. **[10]**
b Design a Butterworth digital IIR low pass filter using BLT by assuming T=1 sec., to satisfy the following specifications **[10]**

$$0.707 \leq |H(e^{jw})| \leq 1.0; \quad 0 \leq w \leq 0.2\pi$$
$$|H(e^{jw})| \leq 0.08; \quad 0.4\pi \leq w \leq \pi$$

4 **a** Determine impulse response h(n) of a filter having desired frequency response **[10]**

$$H_d(e^{-j3w}) = e^{-j(M-1)(w/2)} \quad \text{for } 0 \leq w \leq \pi/2$$
$$= 0 \quad \text{for } \pi/2 \leq w \leq \pi$$

M=7, use frequency sampling approach.
b Design Chebyshev analog filter with maximum pass band attenuation of 2.5dB at Ω_P = 20 rad/sec and stop band attenuation of 30 dB at Ω_S = 50 rad/sec. **[10]**

5 **a** Write a note on polyphase filters. **[10]**
b Design a FIR filter with hamming window for the following specifications: **[10]**

$$H_d(w) = e^{-j3w} \quad -\pi/4 \leq w \leq \pi/4$$
$$= 0 \quad \pi/4 \leq w \leq \pi$$

6 **a** Obtain the DFT of the sequence x(n)= {1,1,1,1,1,1,1,0}using DIT FFT algorithm. **[10]**
b Write short note on Gibbs phenomenon. **[10]**

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