B.E. (Computer Technology) Seventh Semester (C.B.S.)

Elective - I : Digital Signal Processing

P. Pages : 2 **TKN/KS/16/7569**

Time : Three Hours Max. Marks : 80

Notes :
1. All questions carry marks as indicated.
2. Solve Question 1 OR Questions No. 2.
3. Solve Question 3 OR Questions No. 4.
4. Solve Question 5 OR Questions No. 6.
5. Solve Question 7 OR Questions No. 8.
6. Solve Question 9 OR Questions No. 10.
7. Solve Question 11 OR Questions No. 12.
8. Due credit will be given to neatness and adequate dimensions.
9. Assume suitable data whenever necessary.
10. Illustrate your answers whenever necessary with the help of neat sketches.

1. a) Explain advantages and imitations of DSP over ASP. **5**

b) Consider the analog signal $x(t) = 3\cos 2000\pi t + 5\sin 6000\pi t + 10\cos 12{,}000\pi t$ **9**

i) What is the nyquist rate for this signal?

ii) Assume that we sample this signal using a sampling rate $F_S = 5000$ samples/sec. What is the discrete-time signal obtained after sampling.

iii) What is the analog signal y(t) that we can reconstruct from the samples if we use ideal interpolation.

OR

2. a) What are different types of DT systems. Explain each with example. **6**

b) The impulse response of linear time-invariant system is $h(n) = \{1, \underset{\uparrow}{2}, 1, -1\}$. **8**

Determine the response of system to the input signal $x(n) = \{\underset{\uparrow}{1}, 2, 3, 1\}$ by using graphical or analytical convolution method.

3. a) State and prove any two properties of z-transform. **6**

b) Determine the z-transform of the signal $x(n) = \left(\frac{1}{2}\right)^n \mu(n)$ Also sketch the region of convergence (ROC). **7**

OR

4. a) Find the inverse z-transform of following using power series expansion method when x(n) is causal and when x(n) is anticausal $x(z) = \dfrac{1+z^{-1}}{1-2z^{-1}+z^{-2}}$ **6**

b) Find the impulse response of the system described by diff. equation $y(n) - 3y(n-1) - 4y(n-2) = x(n) + 2x(n-1)$ using z-transform. **7**

5. Find 8 point DFT of the following sequence using DIT-FFT algorithm. **14**

$x(n) = (-1)^n \cdot 0 \le n \le 7$.

Also compute the number of complex additions and multiplications required.

OR

6. Compute the circular convolution of the following sequences using DFT and IDFT **14**

$h(n) = \{\underset{\uparrow}{1}, 2, 3, 4\}, \ x(n) = \{\underset{\uparrow}{1}, 2, 2, 1\}$

7. Design a digital Buttorworth filter that satisfies the following constraints using Bilinear transformation **13**

Assume T = 1 sec.

$$0.9 \le |H(w)| \le 1, \qquad 0 \le w \le \pi/2$$

$$|H(w)| \le 0.2, \qquad \frac{3\pi}{4} \le w \le \pi$$

OR

8. A filter (LTI) is described by the following difference equation **13**

$y(n) = \frac{1}{4} y(n-1) - \frac{5}{24} y(n-2) - \frac{1}{12} y(n-3) + x(n) + \frac{1}{3} x(n-1)$ implement the system using DF-1, DF-II, cascade and parallel form of structures.

9. A low pass FIR filter is to be designed with following desired frequency response: **13**

$$Hd(w) = \begin{cases} e^{-j2w} & ; \quad \frac{-\pi}{4} \le w \le \pi/4 \\ 0 & ; \quad \pi/4 \le w \le \pi \end{cases}$$

Determine the filter coefficients h(n) using rectangular window. Also determine the frequency response H(w).

OR

10. Design FIR filter using Hamming window for a derived response: **13**

$$Hd(w) = \begin{cases} e^{-j3w} & , \quad \frac{-3\pi}{4} \le w \le 3\pi/4 \\ 0 & , \quad \frac{3\pi}{4} \le w \le \pi \end{cases}$$

Also draw the structure of the filter.

11. a) What is Multirate signal processing? Explain the applications of multirate signal processing. **6**

b) Obtain the O/P signal $y_1(n)$ from the input signal x(n) as shown below **7**

$x(n) = \{1, 2, 3, 4, 5, 6, 7, 8, 9\}$

$$x(n) \longrightarrow \boxed{\downarrow 3} \longrightarrow \boxed{\uparrow 2} \longrightarrow y_1(n)$$

OR

12. a) Explain the sampling rate conversion by rational factor with the help of block diagram. **5**

b) Explain the sub band coding of speech signals with the help of block diagram. **8**

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