E 282

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2003.

Fifth Semester

Information Technology

IF 351 — DIGITAL SIGNAL PROCESSING

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. Define a static and a stable system.

2. Define commutative and associative law of convolution.

3. Write the analysis and synthesis equations of DFT.

4. What is the need for FFT?

5. List the various forms of realizations of IIR system.

6. Write the 2's complement of the following :

   (a) + 7 (b) – 7.

7. Write the transfer function of

   (a) Hamming Window

   (b) Hanning Window.

8. Define Bilinear transformation.

9. Define Sampling theorem.

10. Write any two applications of multirate signal processing.

PART B — ($5 \times 16 = 80$ marks)

11. (i) Realise the following second order system using direct form I of realization.

$$y[n] = y(n-1) - 0.5y(n-3) + .5x(n-1).$$

(ii) Enumerate with suitable examples the truncation and the rounding off errors.

12. (a) (i) Derive the relationship between i/p and o/p of a LTI system in z transform.

(ii) Find the convolution sum of

$$x[n] = 1 \quad n = -2, 0, 1$$
$$= 2 \quad n = -1$$
$$= 0 \quad \text{elsewhere.}$$

and

$$h[n] = \delta(n) - \delta(n-1) + \delta(n-2) - \delta(n-3).$$

Or

(b) (i) Find the z–transform and ROC of

(1) $x[n] = \{5^n - 2^n\} u[n]$.

(2) $x[n] = \cos n\theta \, u[n]$

(3) $x[n] = \{1, 0, 3, -1, 2\}$.

(ii) Clearly define the following with suitable example :

(1) Linear system

(2) Time Invariant system

(3) Causal System.

13. (a) (i) Derive the relationship between DFT and z–transform

(ii) Stating the relevant equations obtain the Eight point DIT FFT algorithm.

Or

(b) (i) State and prove the following properties of DFT :

(1) Circular convolution

(2) Parseval's theorem.

(ii) (1) Find the circular convolution of

$$x_1(n) = \{1, 2, 3, 4\}$$

$$x_2(n) = \{4, 3, 2, 1\}$$

(2) Find the IDFT of $X(k) = \{5, 0, 1, j, 0, 1, 0, 1, j, 0\}$

14. (a) (i) Write the issues with windowing technique of designing FIR filter.

(ii) Using bilinear transformation convert to z domain at $T = 1$ sec

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

Or

(b) (i) State the merits and demerits of FIR filter.

(ii) Using Impulse Invariant method find $H(z)$ at $T = 1$ sec

$$H(s) = \frac{2}{s^2 + 8s + 15}.$$

15. (a) Write a detailed note

(i) Need for multirate signal processing

(ii) Identities used in multirate processing.

Or

(b) (i) Explain clearly the down sampling and up sampling in multirate signal processing.

(ii) Explain subband coding of speech signals.
