C 3267

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2007.

Fourth Semester

Information Technology

IT 1252 — DIGITAL SIGNAL PROCESSING

(Regulation 2004)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Find the Z transform of $\{ 1, 0, 2, 0, 3 \}$.

2. Check whether the system $y(n) = e^{x(n)}$ is linear.

3. Draw the radix-4 FFT-DIF butterfly diagram.

4. Find the values of W_N^K when $N = 8$ and $K = 2$ and also for $K = 3$.

5. Draw the response curve for butterworth, Chebyshev and Eliptic filters.

6. Write the equation for frequency transformation from lowpass to bandpass filter.

7. Find digital filter equivalent for $H(S) = \frac{1}{S+8}$.

8. Explain Gibb's phenomenon.

9. State sampling theorem.

10. Explain briefly the musical sound processing.

PART B — (5 × 16 = 80 marks)

11. (a) (i) Find the response of the system for the input signal. $x(n) = \{1, 2, 2, 3\}$ and $h(n) = \{1, 0, 3, 2\}$. (8)

(ii) Find the inverse Z–transform of

$$\frac{1}{\left(1-\frac{1}{2}z^{-1}\right)\left(1-\frac{1}{4}z^{-1}\right)} \quad (8)$$

Or

(b) Check whether the following systems are linear time invariant.

(i) $y(n)=A+B\cdot x(n)$

(ii) $y(n)=A\cdot x(n)+B\cdot[x(n-1)]^2$

(iii) $y(n)=e^{x(n)}$.

12. (a) Derive and draw the radix-2 DIT algorithm for FFT of 8 points. (16)

Or

(b) Compute the DFT for the sequence $\{1, 2, 0, 0, 0, 2, 1, 1\}$. Using radix-2 DIF FFT algorithm. (16)

13. (a) (i) Design a digital filter using $H(S) = \frac{1}{S^2 + 9S + 18}$ with $T = 0.2$ sec. (10)

(ii) Design a second order band reject filter with W_1 and W_2 as cut off frequency and sampling interval as T. (6)

Or

(b) (i) Realize the given transfer function using direct form–1 and parallel methods

$$H(Z) = \frac{4Z^2 + 11Z - 2}{(Z+1)(Z-3)} \quad (10)$$

(ii) If $H(S) = \frac{1}{(S+1)(S+2)}$ find $H(Z)$ using impulse invariance method for sampling frequency of 5 samples/sec. (6)

14. (a) Design a Linear phase FIR digital filter for the given specifications using Hamming window of length $M = 7$.

$$H_d(W) = \begin{cases} e^{-j3W}, & \text{for} \quad |W| \le \frac{\pi}{6} \\ 0, & \text{for} \quad \frac{\pi}{6} < |W| \le \pi \end{cases} \qquad (16)$$

Or

(b) Design and implement linear phase FIR filter of length $N = 15$ which has following unit sample sequence

$H(K) = 1 \; K = 0, 1, 2, 3$

$= 0 \; K = 4, 5, 6, 7$

15. (a) Explain in detail about finite word length effects in the filter design. (16)

Or

(b) Explain briefly :

(i) Multi rate signal processing. (8)

(ii) Vocoder. (8)