A 1120

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2006.

Fifth Semester

Computer Science and Engineering

CS 331 — DIGITAL SIGNAL PROCESSING

Time : Three hours　　　　Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. Define Shift–Invariant system.

2. Define Causality.

3. List the advantages of FIR systems over IIR systems.

4. Define 'twiddle factor' of FFT.

5. What are the conditions to be satisfied for constant phase delay in linear phase FIR filters?

6. List the well known design techniques for linear phase FIR filter.

7. What is the main drawback of impulse–invariant mapping?

8. Why IIR filters do not have linear phase?

9. Distinguish between first order and second order filters.

10. What is the need for scaling in digital filters?

PART B — ($5 \times 16 = 80$ marks)

11. (i) Obtain and sketch the impulse response of shift invariant syst described by

$$f(n) = 0.4x(n) + x(n-1) + 0.6(n-2) + x(n-3) + 0.4x(n-4).$$

(ii) An LTI system is described by the equati $y(x) = x(n) + 0.81x(n-1) + 0.81x(n-2) - 0.45y(n-2)$. Determine t transfer function of the system. Sketch the poles and zeros on t y–plane. (

12. (a) (i) State and prove convolution property of discrete Fourier transform (

(ii) State the shifting property of DFT and give its proof. (8

Or

(b) (i) Determine the Fourier transform of the signal $x(n) = a^{|n|}; -1 < a < 1$ (8

(ii) Determine the inverse Fourier transform for the first orde recursive filter $H(w) = \left(1 - a e^{-jw}\right)^{-1}$. (8

13. (a) (i) The desired frequency response of a digital filter is

$$H_d(w) = \begin{cases} e^{-j3w} & -\dfrac{\pi}{4} \le w \le \dfrac{\pi}{4} \\ 0 & \dfrac{\pi}{4} < |w| \le \pi \end{cases}.$$

Determine the filter coefficients if the window function is defined as $w(x) = \begin{cases} 1 & 0 \le x \le 5 \\ 0 & \text{otherwise} \end{cases}$. (10)

(ii) List the various steps in designing F.I.R. filters. (6)

Or

(b) (i) Derive the frequency response of linear phase FIR filter when impulse response is symmetrical and N is odd. (8)

(ii) How is design of linear phase FIR filter done by frequency sampling method? Explain. (8)

14. (a) (i) Apply the bilinear transformation of

$$H_a(s) = \frac{2}{(s+1)(s+2)} \text{ with } T = 1 \text{ sec and find } H(z). \quad (8)$$

(ii) The normalized transfer function of an analog filter is given by

$$H_a(s_n) = \frac{1}{s_n^2 + 1.414 s_n + 1}.$$

Convert the analog filter to a digital filter with a cutoff frequency of 0.4π using bilinear transformation. (8)

Or

(b) (i) Enumerate the various steps involved in the design of low pass digital Butterworth IIR filter. (6)

(ii) The specification of the desired low pass filter is

$$0.8 \le |H(w)| \le 1.0 \; ; \; 0 \le w \le 0.2\pi$$
$$|H(w)| \le 0.2 \; ; \; 0.32\pi \le w \le \pi$$

Design a Butterworth digital filter using impulse invariant transformation. (10)

15. (a) Explain in detail the design and various implementation steps for filter using sampling rate conversion system. (16)

Or

(b) (i) Discuss on sampling rate conversion of national factor (Z/D). (5)

(ii) Explain dead band in limit cycles. (5)

(iii) Differentiate between Auto and Cross correlation of random signals. (6)
