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Question Paper Code :

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2011.

Seventh Semester

Computer Science and Engineering

CS 2403 — DIGITAL SIGNAL PROCESSING

(Common to Fifth Semester Information Technology)

(Regulation 2008)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Define and express the transfer function of N^{th} order LTI system.

2. Compare linear convolution and circular convolution.

3. What is the relation between DFT and Z-Transform?

4. What is phase factor or twiddle factor?

5. Sketch the various tolerance limits to approximate an ideal lowpass and highpass filter.

6. What is the importance of poles in filter design?

7. What is frequency warping?

8. What is Butterworth approximation?

9. List various voice compression and coding techniques.

10. What are the various enhancement techniques in image processing?

PART B — (5 × 16 = 80 marks)

11. (a) (i) Find the inverse Z-Transform of $X(z)=\frac{Z^2+Z}{(Z-1)(Z-3)}$, $ROC : |Z|>3$, using

(1) Residue method and

(2) Convolution method. (8)

(ii) State and prove circular convolution. (8)

Or

(b) LTI system is described by the difference equation $y(n)=a\,y(n-1)+b\,x(n)$. Find the impulse response, magnitude function and phase function. Solve b, if $|H(w)|=1$. Sketch the magnitude and phase response for a = 0.6. (1

12. (a) (i) Evaluate the 8-point for the following sequences using DIT-FFT algorithm (8)

$$x(n)=\begin{cases}1 & \text{for } -3\le n\le 3\\ 0 & \text{otherwise}\end{cases}$$

(ii) Calculate the percentage of saving in calculations in a 1024-point radix -2 FFT, when compared to direct DFT. (8)

Or

(b) Determine the response of LTI system when the input sequence $x(n)=\{-1, 1, 2, 1, -1\}$ by radix 2 DIT FFT. The impulse response of the system is $h(n)=\{-1, 1, -1, 1\}$. (16)

13. (a) The specification of the desired lowpass filter is

$$\frac{1}{\sqrt{2}}\le|H(w)|\le 1.0 \;;\; 0\le w\le 0.2\pi$$

$$|H(w)|\le 0.08 \;;\; 0.4\pi\le w\le\pi$$

Design a Butterworth digital filter using bilinear transformation. (16)

Or

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

$$0.9\le|H(w)|\le 1.0 \;;\; 0\le w\le 0.25\pi$$

$$|H(w)|\le 0.24 \;;\; 0.5\pi\le w\le\pi$$

Design a Chebyshev digital filter using impulse invariant transformation. (16)

14. (a) (i) Design a single tier notch filter to reject frequencies in the range 1 to 2 rad/sec using rectangular window with $N = 7$. (8)

(ii) Compare Hamming window and Kaiser window. (8)

Or

(b) (i) Explain the characteristics of a limit cycle oscillation with respect to the system described by the equation $y(n) = 0.95\, y(n-1) + x(n)$. Determine the dead band of the filter. (8)

(ii) Explain Gibb's phenomenon (or Gibb's oscillation). (8)

15. (a) Explain the methods of speech analysis and synthesis in detail. (16)

Or

(b) Explain how image enhancement restoration and coding can be done using signal processing. (16)