Reg. No. : | | | | | | | | | | | | |

Question Paper Code : 71399

B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2015.

Seventh Semester

Computer Science and Engineering

CS 2403/CS 73 — DIGITAL SIGNAL PROCESSING

(Common to Fifth Semester – Information Technology)

(Regulation 2008)

(Also Common to PTCS 2403 – Digital Signal Processing for B.E. (Part-Time) Sixth Semester – Computer Science and Engineering Regulation 2009)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. State sampling theorem.

2. What is quantization error?

3. Compute DFT of the signal $x(n) = \delta(n)$.

4. What is meant by radix - 2 FFT?

5. What is meant by bilinear transformation method of designing IIR filter?

6. Draw the direct form realization of IIR system.

7. What is linear phase response of a filter?

8. State any two important properties of FIR filter.

9. Write the main application areas of speech coding.

10. What is adaptive filter?

PART B — (5 × 16 = 80 marks)

11. (a) (i) Find the convolution of given signals.

$x(n) = 3^n\ u(-n)$ and $h(n) = [1/3]^n\ u(n-2)$. (8)

(ii) Applying concentric circle method, compute circular convolution of the sequences $h(n) = \{1, 2, 3, 4\}$ & $x(n) = \{1, 2, 3\}$. (8)

Or

(b) Explain the process of analog to digital conversion of signal in terms of sampling, quantization and coding.

12. (a) Find the 8 point - DFT of a sequence using radix-2 DIT algorithm

$x(n) = 1; \quad 0 \le n \le 2$

$= 0;$ *otherwise*

Or

(b) Compute 8-point DFT of the sequence $x(n) = \{1, -1, 1, -1, 0, 0, 0, 0\}$ using radix - 2 DIF algorithm.

13. (a) Design a digital butterworth filter satisfying the constraints

$0.707 \le |H(e^{jw})| \le 1; 0 \le w \le \pi/2$

$|H(e^{jw})| \le 0.2; \ 3\pi/4 \le w \le \pi$

- with T = 1 sec using bilinear transformation method.

Or

(b) Obtain the direct form I, direct form II and cascade form realization of the following system functions.

$y(n) = -0.1\ y(n-1) + 0.2\ y(n-2) + 3x(n) + 3.6\ x(n-1) + 0.6\ x(n-2)$

14. (a) Discuss the design procedures of FIR filter using frequency sampling method.

Or

(b) Design an ideal differentiator with frequency response

$H(e^{jw}) = jw; -\pi \le w \le \pi$ using Hamming window with N = 7.

15. (a) (i) Discuss about multi rate signal processing. (8)

(ii) Write short note on speech compression. (8)

Or

(b) Discuss the role of DSP in image enhancement with an example.
