Reg. No. :

Question Paper Code : 23397

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2018.

Seventh Semester

Computer Science and Engineering

CS 2403 — DIGITAL SIGNAL PROCESSING

(Common to Fifth Semester — Information Technology)

(Regulations 2008)

(Also common to PTCS 2403 — Digital Signal Processing for B.E. (Part-Time) — Sixth Semester — CSE — Regulations 2009)

Time : Three hours | Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. State the convolution property of Z transforms.

2. Define sampling theorem.

3. Find the circular convolution of two sequences $x_1(n) = \{1, 2, 2, 1\}$ and $x_2(n) = \{1, 2, 3, 1\}$

4. Calculate the number of multiplications needed in the calculation of DFT using FFT algorithm with 32-point sequence.

5. Compare digital and analog filters.

6. What is meant by impulse invariant method of designing IIR filter?

7. What is linear phase response of a filter?

8. State any two important properties of FIR filter.

9. What is the need for multirate signal processing?

10. What is adaptive equalization?

PART B — (5 × 16 = 80 marks)

11. (a) (i) Determine whether each of the following systems below is

(1) Causal (2) Linear

(3) Dynamic (4) Time invariant

(5) Stable

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

(B) $y(n) = x(n)\sum_{k=-\infty}^{\infty}\delta(n-2k)$. (8)

(ii) Explain sampling theorem and reconstruction of the analog signal from its samples. (8)

Or

(b) (i) Explain the properties of cross correlation and autocorrelation sequences. (8)

(ii) Find the discrete convolution of the following sequences $u(n) * u(n-3)$. (8)

12. (a) (i) The input $x(n)$ and impulse response $h(n)$ of a system are given by $x(n) = \{-\underset{\uparrow}{1}, 1, 2, -2\}$; $h(n) = \{0.\underset{\uparrow}{5}, 1, -1, 2, 0.75\}$. Determine the response of the system using DFT. (10)

(ii) State and prove convolution property of DFT. (6)

Or

(b) Compute the FFT of the sequence $x(n) = n^2 + 1$ for $0 \le n \le N-1$, where N = 8 using DIT algorithm. (16)

13. (a) Design a Butterworth digital filter using bilinear transformation that satisfy the following specifications (16)

$0.89 \le |H(w)| \le 1.0 \,;\, 0 \le w \le 0.2\pi$

$|H(w)| \le 0.18 \,;\, 0.3\pi \le w \le \pi$.

Or

(b) The specification of the desired lowpass digital 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) Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = h(N-1-n)$. Also discuss symmetric and anti symmetric cases of FIR filter when N is even. (16)

Or

(b) Explain in detail about Finite word length effects in digital filters. (16)

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.