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

Question Paper Code : 73397

B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2017.

Seventh Semester

Computer Science and Engineering

CS 2403/CS 73 — 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 — ($10 \times 2 = 20$ marks)

1. Define energy signals and power signals.

2. What is correlation? What are its types?

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. Define Bilinear Transformation with expressions.

6. Mention the properties of Butterworth filter.

7. What is Gibb's phenomenon?

8. What are limit cycles?

9. Write the main application areas of speech coding.

10. What is adaptive filter?

PART B — ($5 \times 16 = 80$ marks)

11. (a) (i) Compute the convolution of the signals $x(n) = \{1, 2, 3, 4, 5, 3, -1, -2\}$ and $h(n) = \{3, 2, 1, 4\}$ using tabulation method. (6)

(ii) Check whether the following systems are, static or dynamic, linear or non-linear, time variant or invariant, causal or non-causal, stable or unstable. (10)

(1) $y(n) = \cos[x(n)]$

(2) $y(n) = x(-n+2)$

(3) $y(n) = x(2n)$

(4) $y(n) = x(n) \cdot \cos \omega_0(n)$.

Or

(b) (i) Describe the different types of Digital signal representation. (8)

(ii) What is Nyquist rate? Explain its significance while sampling the analog signals. (8)

12. (a) (i) Find eight point DFT of the following sequence using direct method : $\{1, 1, 1, 1, 1, 1, 0, 0\}$. (10)

(ii) State any six properties of DFT. (6)

Or

(b) (i) Compute eight point DFT of the following sequence using radix 2 decimation in time FFT algorithm : $x(n) = \{1, -1, -1, -1, 1, 1, 1, -1\}$. (10)

(ii) Discuss the use of FFT in linear filtering. (6)

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

$$0.89 \le |H(\omega)| \le 1.0; \quad 0 \le \omega \le 0.2\pi$$

$$|H(\omega)| \le 0.18; \quad 0.3\pi \le \omega \le \pi.$$

Or

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

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

$$|H(\omega)| \le 0.24; \quad 0.5\pi \le \omega \le \pi.$$

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

14. (a) (i) Using a rectangular window technique design a low pass filter with passband gain of unity, cut off frequency of 1000 Hz and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7. (10)

(ii) Compare FIR and IIR filters. (6)

Or

(b) (i) Obtain the cascade realization of system function $H(z) = (1 + 2z^{-1} - z^{-2})(1 + z^{-1} - z^{-2})$. (10)

(ii) Explain the quantization errors due to finite word length registers in digital filters. (6)

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 application with an suitable example. (16)