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

Question Paper Code : 30893

B.E./B.Tech. DEGREE EXAMINATIONS, APRIL/MAY 2019.

Seventh Semester

Computer Science and Engineering

CS 2403 — 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 — CSE — Regulation 2009)

Time : Three hours — Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. State low pass sampling theorem.

2. What is meant by energy and power signals?

3. What is meant by radix 2 FFT?

4. Give transform pair equation of DFT.

5. What are the limitations of Impulse invariant technique of designing filters?

6. Given the low pass transfer function $H_a(s) = \frac{1}{s+1}$. Find the high pass transfer function having a cutoff frequency to rad/sec.

7. What is linear phase response of a filter?

8. State any two important properties of FIR filter.

9. List out the application of Adaptive filtering.

10. What do you mean by speech compression?

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 $h(n) = \{1, 2, 3, 4\}$ and $x(n) = \{1, 2, 3\}$. (8)

Or

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

12. (a) (i) Discuss the properties of DFT. (8)

(ii) Discuss the use of FFT algorithm in linear filtering and correlation. (8)

Or

(b) Find DFT for {1, 1, 2, 0, 1, 2, 0, 1} using FFT DIT butterfly algorithm and plot the spectrum. (16)

13. (a) (i) Obtain the direct form I, direct form II, cascade and parallel form realization for the system

$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$. (8)

(ii) For the analog transfer function $H(s) = \dfrac{2}{(s+1)(s+2)}$. Determine $H(z)$ using impulse invariance method. Assume T = 1 sec. (8)

Or

(b) A low pass filter meeting the following specifications is required :

Passband	–	0 – 500 Hz
Stopband	–	2 – 4 kHz
Passband ripple	–	3 dB
Stopband attenuation	–	20 dB
Sampling frequency	–	8 kHz

Determine the following:

(i) Pass and stopband edge frequencies for a suitable analog prototype low pass filter.

(ii) Order N of the prototype low pass filter.

(iii) Coefficients and hence the transfer function of the discrete time filter using the bilinear z-transform.

Assume Butterworth characteristics of the filter. (16)

14. (a) Design a FIR bandstop filter to reject frequencies in the range 1.2 to 1.8 rad/sec using Hamming window, with length N = 6. Also, realize the linear phase structure of the bandstop FIR filter. (16)

Or

(b) Explain the characteristics of a limit cycle oscillation with respect to the system described by the equation $y(n) = 0.85y(n-2) + 0.72y(n-1) + x(n)$.

Determine the dead band of the filter $x(n) = \left(\frac{3}{4}\right)\delta(n)$. (16)

15. (a) (i) Explain aliasing effect in down sampling. (8)

(ii) Explain subband coding technique used in speech coding. (8)

Or

(b) (i) Explain digital processing of audio signals. (8)

(ii) Explain digital signal processing in image enhancement. (8)