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

Question Paper Code : 21399

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2015.

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 – Computer Science and Engineering – Regulations 2009)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. A discrete-time signal $x(n) = \{-2, -1, 0, \underset{\uparrow}{1}, -1, 1\}$ is multiplied by $u(-n-2)$. What is the resulting signal?

2. What is a shift-invariant system? Give an example.

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 conditions on the FIR sequence $h(n)$ are to be imposed in order that this filter can be called a linear phase filter?

8. What is the effect of quantization on pole locations?

9. What is the need for anti-imaging filter after up sampling a signal?

10. What is a channel vocoder?

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) Explain any four properties of DFT. (8)

(ii) Compute the eight-point DFT of the sequence $x(n) = \{0.5, 0.5, 0.5, 0.5, 1, 2, -1, 0\}$ using the inplace-radix-2 DIT algorithm. (8)

Or

(b) (i) Draw the flowgraph of a two-point radix-2 DIF-FFT algorithm. What is the basic operation of DIF algorithm? (8)

(ii) Find the IDFT of the sequence

$X(k) = \{4, 1-j2.414, 0, 1-j0.414, 0, 1+j0.414, 0, 1+j2.414\}$ using DIF algorithm. (8)

13. (a) (i) Design an analog Butterworth filter that has a –2dB passband attenuation at a frequency of 20 rad/sec and atleast –10dB stopband attenuation at 30 rad/sec. (10)

(ii) Explain the steps of design of digital filters from analog filters. (6)

Or

(b) (i) Using the bilinear transform, design a high pass filter, monotonic in passband with cutoff frequency of 1000 Hz and down 10 dB at 350 Hz. The sampling frequency is 5000 Hz. (10)

(ii) Explain the methods of realization of digital filters. (6)

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) 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)