Reg. No. :

**Question Paper Code : 91363**

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2014.

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. A discrete time signal $x(n) = \{0, 0, 1, 1, 2, 0, 0, \ldots\}$. Sketch the $x(n)$ and $x(-n+2)$ signals.

2. Determine whether the following sinusoids are periodic; if periodic then compute their fundamental period.

   (a) $\cos(0.01\pi n)$

   (b) $\sin\left(\frac{\Pi 62n}{10}\right)$.

3. Using the definition $W = e^{-i(2\pi/N)}$, and the Euler identity $e^{\pm i\theta} = \cos(\theta) \pm i\sin(\theta)$, the value of $W^{(N/3)}$ is ——————.

4. In the direct computation of N-point DFT of a sequence, how many multiplications and additions are required?

5. Compare analog and digital filters.

6. Sketch the mapping of s-plane and z-plane in approximation of derivatives.

7. What are the characteristic features of FIR filters?

8. Define finite word length effects.

9. What do you mean by image enhancement?

10. State the advantages and disadvantages of digital signal processing compared to analog signal processing.

PART B — (5 × 16 = 80 marks)

11. (a) Check whether the systems described by the following equations are

(i) $y(n) = x(n)\cos(\omega_0 n)$

(ii) $y(n) = |x(n)|$

(iii) $y(n) = \mathrm{sgn}[x(n)]$

Static or dynamic

Linear or non-linear

Shift invariant or shift variant

Causal or non-causal

stable or unstable. (16)

Or

(b) Compute the linear convolution of the following sequence using Mathematical Equation, Multiplication and Tabulation methods.

$x(n) = \{0, 2, 2, 3\}$ and $h(n) = \sin\left(\frac{3\Pi n}{8}\right)$, $0 \le n \le 4$. (16)

12. (a) (i) State and prove the periodicity and time reversal properties of DFT. (8)

(ii) Obtain the 4-point DFT of the following sequences.

(1) $x(n) = 2^n$

(2) $x(n) = \{0, 1, 0, -1\}$ (8)

Or

(b) Compute the 8-point DFT of the equation $x(n) = n + 1$ using Radix-2 DIF-FFT algorithm. (16)

13. (a) Determine the system function of the IIR digital filter for the analog transfer function

$$H_a(S) = \frac{10}{(S^2 + 7s + 10)} \text{ with } T = 0.2 \text{ second}$$

using impulse invariance method. (16)

Or

(b) A digital filter with a 3 db bandwidth of $0.25\pi$ is to be designed from the analog filter whose system response is

$$H_a(s) = \frac{\Omega_c}{s + \Omega_c}$$

using bilinear transformation and obtain $H(Z)$. (16)

14. (a) Design the symmetric FIR low pass filter whose desired frequency response is given as

$$H_d(w) = \begin{cases} e^{-jw\tau} & for\ |w| \le w_c \\ 0 & otherwise \end{cases}$$

The length of the filter should be 5 and $w_c = 1$ radians/sample using Rectangular window. (16)

Or

(b) Realize a direct form and linear phase FIR filter structures with the following impulse response. Which is the best realization? Why? (16)

$$h(n) = \delta(n) + \frac{1}{3}\delta(n-1) - \frac{1}{4}\delta(n-2) + \frac{1}{3}\delta(n-3) + \delta(n-4).$$

15. (a) (i) Describe how various sound effects can be generated with the help of DSP. (8)

(ii) Explain subband coding of speech and audio signals using DSP. (8)

Or

(b) What is an adaptive filter? With neat block diagram discuss any four applications of adaptive filters. (16)

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