Reg. No. :

Question Paper Code : 31321

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2013.

Seventh Semester

Computer Science and Engineering

CS 2403/CS 73 — DIGITAL SIGNAL PROCESSING

(Common to Fifth Semester – Information Technology)

(Regulation 2008)

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 DTFT of $x(n) = -b^n . u(-n-1)$.

4. Compute the IDFT of $Y(k) = \{1, 0, 1, 0\}$.

5. Define Bilinear Transformation with expressions.

6. Mention the properties of Butterworth filter.

7. What are Gibbs oscillations?

8. Distinguish between FIR and IIR filters.

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) 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 noncausal, stable or unstable. (10)

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

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

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

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

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) 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) The specification of the desired low pass filter is

$$0.8 \le |H(\omega)| \le 1.0; \quad 0 \le \omega \le 0.2\pi$$
$$|H(\omega)| \le 0.2 \quad ; \quad 0.32\pi \le \pi.$$

Design butterworth digital filter using impulse invariant transformation. (16)

Or

(b) (i) Discuss the limitation of designing an IIR filter using impulse invariant method. (6)

(ii) Convert the analog filter with the system transfer function $H_a(s)=[s+0.3]/[(s+0.3)^2+16]$ using bilinear transformation. (10)

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) Explain how the speech compression is achieved. (8)

Or

(b) Explain in detail about Image Enhancement technique. (16)