4/6/16 AN

Reg. No. :

Question Paper Code : 51399

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2016.

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. What is twiddle factor?

4. List the uses of FFT in linear filtering?

5. What is meant by bilinear transformation method of designing IIR filter?

6. Draw the direct form realization of IIR system.

7. What are Gibbs oscillations?

8. Distinguish between FIR and IIR filters.

9. What is decimation?

10. List various special audio effects that can be implemented digitally.

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) The input $x(n)$ and impulse response $h(n)$ of a system are given by

$x(n) = \{-1, \underset{\uparrow}{1}, 2, -2\}$; $h(n) = \{0.5, 1, \underset{\uparrow}{-1}, 2, 0.75\}$

Determine the response of the system using DFT. (10)

(ii) State and prove convolution property of DFT. (6)

Or

(b) Compute the FFT of the sequence $x(n) = n^2 + 1$ for $0 \le n \le N-1$, where $N = 8$ using DIT algorithm. (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) Discuss the design procedures of FIR filter using frequency sampling method.

Or

(b) Design an ideal differentiator with frequency response

$H(e^{jw}) = jw; \ -\pi \le w \le \pi$ using Hamming window with $N = 7$.

15. (a) (i) Explain the method for converting the sampling rate by a factor I/D with block diagram and equations. (8)

(ii) Discuss sub band coding process in detail. (8)

Or

(b) (i) With block diagram explain adaptive filtering based adaptive channel equalization. (8)

(ii) What is image enhancement? Explain various image enhancement techniques. (8)

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