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. State sampling theorem.
2. What is quantization error ?
3. Write the formula for Discrete Time Cosine transform. (DCT) pair.
4. What is FFT ?
5. Define Bilinear Transformation with expressions.
6. Mention the properties of Butterworth 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 decimation ?
10. List various special audio effects that can be implemented digitally.

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

OR

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

12. (a) (i) Let $X(K) = DFT\{x(n)\}$ with n; $K = 0, 1, \ldots, (N-1)$. Determine the relationship between X(K) and the following DFTs.
(1) $DFT\ \{Re\, x(n)\}$
(2) $DFT\{x(n-1)\}$. **(9)**

(ii) State and prove any two properties of DFT. **(7)**

OR

(b) (i) Compute the DFT of the following sequence :
(1) $x(n) = [1, 0, -1, 0]$
(2) $x(n) = Cos(0.25\, \pi\, n\}, n = 0, 1, 2, \ldots, 7$. **(8)**

(ii) Write short notes on filtering methods using DFT. **(8)**

13. (a) (i) Design an analog Butterworth filter that has a –2dB passband attenuation at a frequency of 20 rad/sec and atleast lOdB 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 a 350 Hz. The sampling frequency is 5000 Hz. **(10)**

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

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) 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 a block diagram explain adaptive filtering based adaptive channel equalization. **(8)**

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