Reg. No. :

Question Paper Code : 21321

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 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 — ($10 \times 2 = 20$ marks)

1. Define energy signals and power signals.

2. What is correlation? What are its types?

3. What is meant by radix 2 FFT?

4. Give transform pair equation of DFT.

5. What are the characteristics of Chebyshev filter?

6. Write the transformation equation to convert low pass filter into band stop filter.

7. Write the equation for Blackman window.

8. What is zero input limit cycle oscillation?

9. What is decimation?

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

PART B — ($5 \times 16 = 80$ marks)

11. (a) (i) Consider the analog signal

$x_a(t) = 3 \cos 2000\pi t + 5 \sin 6000 \pi t + 10 \cos 12000 \pi t$.

(1) What is the Nyquist rate for this signal?

(2) Assume now that we sample this signal using a sampling rate $F_s = 5000$ samples/s. What is the discrete time signal obtained after sampling?

(3) What is the analog signal $y_a(t)$ that we can reconstruct from the samples if we use ideal interpolation? (8)

(ii) Derive the equation for convolution sum and summarize the steps involved in computing convolution. (8)

Or

(b) (i) Determine the z transform and ROC of the signal $x(n) = -\alpha^n u(-n-1)$. (6)

(ii) Check whether the discrete time system $y(n) = \cos[x(n)]$ is

(1) Static or dynamic

(2) Linear or nonlinear

(3) Time invariant or time varying

(4) Causal or non-causal

(5) Stable or unstable. (10)

12. (a) (i) Find eight point DFT of the following sequence using direct method:

{1, 1, 1, 1, 1, 1, 0, 0} (10)

(ii) State any six properties of DFT. (6)

Or

(b) (i) Compute eight point DFT of the following sequence using radix 2 decimation in time FFT algorithm.

$x(n) = \{1, -1, -1, -1, 1, 1, 1, -1\}$ (10)

(ii) Discuss the use of FFT in linear filtering. (6)

13. (a) (i) Obtain the direct form I, direct form II, cascade and parallel form realization for the system

$$y(n) = -0.1\,y(n-1) + 0.2\,y(n-2) + 3\,x(n) + 3.6\,x(n-1) + 0.6\,x(n-2).$$ (8)

(ii) For the analog transfer function $H(s) = \dfrac{2}{(s+1)(s+2)}$. Determine $H(z)$ using impulse invariance method. Assume T = 1 sec. (8)

Or

(b) A low pass filter meeting the following specifications is required :

Passband	–	0-500 Hz
Stopband	–	2-4 kHz
Passband ripple	–	3 dB
Stopband attenuation	–	20 dB
Sampling frequency	–	8 kHz

Determine the following :

(i) Pass and stopband edge frequencies for a suitable analog prototype low pass filter.

(ii) Order N of the prototype low pass filter.

(iii) Coefficients and hence the transfer function of the discrete time filter using the bilinear z-transform.

Assume Butterworth characteristics of the filter. (16)

14. (a) (i) Given a three stage lattice filter with coefficients $K_1 = \dfrac{1}{4}$, $K_2 = \dfrac{1}{4}$, $K_3 = \dfrac{1}{3}$, determine the FIR filter coefficients for the direct form structure. (8)

(ii) Determine the coefficients of a linear phase FIR filter of length M = 15 has a symmetric unit sample response and a frequency response that satisfies the conditions $H\left(\dfrac{2\pi k}{15}\right) = \begin{cases} 1 & k = 0, 1, 2, 3 \\ 0 & k = 4, 5, 6, 7 \end{cases}$. (8)

Or

(b) Design an ideal high pass filter with a frequency response

$$H_d(e^{j\omega}) = \begin{cases} 1 & \text{for } \frac{\pi}{4} \le |\omega| \le \pi \\ 0 & \text{for } |\omega| \le \frac{\pi}{4} \end{cases}$$

Find the value of $h(n)$ for $N = 11$ using hamming window. Find $H(z)$ and compute magnitude response. (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 block diagram explain adaptive filtering based adaptive channel equalization. (8)

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