B.E / B.Tech (Full Time) DEGREE END SEMESTER EXAMINATIONS, APRIL / MAY 2013

ELECTRONICS AND COMMUNICATION ENGINEERING BRANCH

FIFTH SEMESTER

32

EC 9304 DIGITAL SIGNAL PROCESSING

(REGULATIONS 2008)

Time : 3 hr   Max. Mark : 100

Answer **ALL** Questions

Part – A (10 x 2 = 20 Marks)

1. Define energy and power of discrete time signals.
2. The first five DFT values for N = 8 is as follows
   X(k) = [28, -4+j9.656, -4+4j, -4+j1.656, -4,.........]. compute the rest of the three DFT values.
3. Obtain the Direct form I structure for the following system
   y[n] = -0.41y[n-1] + 0.32y[n-2] +3x[n] +5.6x[n-1] +0.6x[n-2]
4. Determine the order of a digital Chebyshev filter to meet the following constraints

   $0.99 \le |H(e^{j\omega})| \le 1.01 \qquad 0 \le |\omega| \le 0.2\pi$

   $|H(e^{j\omega})| \le 0.01 \qquad 0.5\pi \le |\omega| \le \pi$

5. Write the frequency response of linear phase FIR filters when impulse response is anti symmetric and N is odd
6. Define Gibbs phenomenon
7. What is an overflow? Why do they occur?
8. Compare fixed and floating point representation.
9. Why anti aliasing filter is used in decimation process.
10. Is upsampling linear operation? Is it time invariant?

Part – B (5 x 16 = 80 Marks)

11. (i). Define zero input limit cycle oscillation and explain (6)

    (ii). A digital system is described by the following equation y(n) = 0.95y(n-1) + x(n). Determine the deadband of the filter. Assume 5bit sign magnitude representation (including sign bit) (4)

    (iii). Explain Harvard architecture and describe the significance of pipelining (6)

12. a. (i). Consider the following discrete time system y[n] = 10 x[n] cos(0.25πn +θ). Where θ is a constant. Check if the system is (i) linear (ii) time invariant (iii) casual (iv) stable (4)

    (ii). Compute DFT for the following sequence x[n] = [1 -3 5 -6] (4)

    (iii). Prove the following DFT property with example. 'Multiplication of two DFTs will result in circular convolution in time domain' (8)

(OR)

12. b. Using Decimation in time FFT algorithm compute DFT for x[n] = [1 0 0 0 0 0 0 0]

13. a. (i). How mapping is achieved in bilinear transformation. Explain (8)

(ii). Using impulse invariant method find H(z) at T = 1sec (8)

$$H(z) = \frac{2}{s^2+8s+15}$$

(OR)

13. b. Design digital IIR low pass Butterworth filter to meet the following specifications.

Passband gain: 0.89, passband edge frequency: 30 Hz,stop band attenuation: 0.20, stopband edge frequency: 75Hz. Use Bilinear transformation Assume sampling frequency 200Hz

14. a. Design FIR high pass 11 tap filter for the frequency response given below

$$H(e^{j\omega}) = \begin{cases} 1, & \text{for } -\pi/4 \le |\omega| \le \pi \\ 0, & \text{for } |\omega| \le \pi \end{cases}$$

Assume Hamming window

(OR)

14. b. A LPF has the desired response as given below

$$H(e^{j\omega}) = \begin{cases} e^{-j3\omega}, & \text{for } 0 \le \omega \le \pi/2 \\ 0, & \pi/2 \le \omega \le \pi \end{cases}$$

Determine filter coefficients for N = 7 using type 1 frequency sampling method

15. a. Explain sampling rate reduction by an integer factor D and derive input – output relation in both time and frequency domain

(OR)

15. b. (i). Describe the polyphase decimation structure using D = 4 and explain why it is efficient (8)

(ii). With neat block diagram explain subband coding (8)