B.E / B.Tech (Full Time) DEGREE END SEMESTER EXAMINATIONS, NOV / DEC 2012

ELECTRONICS AND COMMUNICATION ENGINEERING BRANCH

FIFTH SEMESTER

39

EC 9304 DIGITAL SIGNAL PROCESSING

(REGULATIONS 2008)

Time : 3 hr Max. Mark : 100

Answer **ALL** Questions

Part – A (10 x 2 = 20 Marks)

1. Find whether the following signal is power signal or energy signal. $x(n) = \cos\left[\frac{\pi}{3}n + \frac{\pi}{6}\right]$
2. Compute 4 point IDFT for X(k) = [2, 3+j, - 4, 3-j]
3. 'IIR filter does not have linear phase', Comment on the statement.
4. Write the necessary condition for a linear phase FIR filter
5. How mapping is achieved in bilinear transformation?
6. Determine the length of the filter to be designed using Blackman window given the transition width as 0.314 rad. Sampling frequency 1 KHz
7. Why rounding is preferred than truncation in realizing digital filter?
8. Consider the truncation of negative numbers represented in (b_u+1) bit, fixed point binary form including sign bit. Let (b_u – b) bits be truncated. Obtain the range of truncation error for signed magnitude, 1's complement and 2's complement representation of the negative numbers.
9. State whether downsampling is a linear operation and time invariant.
10. In sampling rate conversion by rational factor, which approach gives effective result: decimator followed by interpolator or interpolator followed by decimator. Give reason

Part – A (10 x 2 = 20 Marks)

11. (i). Give the IEEE. 754 standard format for 32 bit floating point numbers, and compare fixed point and floating point arithmetic (4)

    (ii). An IIR causal filter has the system function

    $$y[n] = x[n] - 0.75\, y[n-1]; \qquad x[n] = 0.875\delta[n]$$

    Assume 3 fractional bits in quantizer plus a sign bit. Show that the filter output enters into zero input limit cycle oscillation and obtain the dead band range. (6)

(iii). The output of an A/D converter is applied to a digital filter whose system function

$$H(z) = \frac{z(0.5)}{z-0.5}$$

Find the output noise power from the digital filter, when the input signal is quantized to eight bits. (6)

12. a.(i). Compute 8 point DFT for the following sequence x(n) = [0, 0, 1, 1, 1, 1, 0, 0] using DIT – FFT algorithm (10)

(ii). Explain over-lap add method. (6)

(or)

12. b. Verify the DFT property, 'Multiplication of the DFTs of two sequences is equivalent to the circular convolution of the two sequences in the time domain'.
Assume $x_1[n]$ =[1,2,3,4] and $x_2[n]$=[2,-1,1,-1]

13. a.(i). Design a Chebyshev filter with a maximum pass band attenuation of 2.5dB at 20rad/sec and a stop band attenuation of 30dB at 50rad/sec. (10)

(ii). Realize the following filter transfer function using cascade structure (6)

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

(or)

13.b. Design a digital high pass filter to meet the following specification.
Pass band: 2 – 4KHz, stop band: 0 – 500Hz, passband ripple 3dB,
stopband attenuation 20dB, sampling frequency 8Khz. Assume bilinear transformation and butterworth approximation

14. a. Design a FIR filter with the following desired frequency response

$$H(e^{j\omega}) = \begin{cases} 0, & -\pi/4 \le \omega \le \pi/4 \\ e^{-j2\omega}, & \pi/4 \le |\omega| \le \pi \end{cases}$$

Assume Hanning window. Realize the designed filter using linear phase structure

(or)

14.b. Determine the coefficients of a linear phase FIR filter of length 15 which has a symmetric unit sample response and a frequency response that satisfies the conditions

$$Hr\left(\frac{2\pi k}{15}\right) = \begin{cases} 1, & k = 0,1,2,3 \\ 0.4, & k = 4 \\ 0, & k = 5,6,7 \end{cases}$$

15. a.(i). Explain sampling rate reduction by an integer factor D and derive input – output relation in both time and frequency domain (12)

(ii). Why anti imaging filter is used in interpolator. Brief the significance. (4)

(or)

15. b.(i). Explain polyphase structure for interpolator (8)

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