Reg. No. : | | | | | | | | | | |

Q 2208

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2007.

Fifth Semester

Electronics and Communication Engineering

EC 333 — DIGITAL SIGNAL PROCESSING

(Common to Bio-Medical Engineering)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. What is causality? What is the necessary and sufficient condition for causality of an LTI system?

2. Draw and explain the basic butterfly diagram of DIF radix-2 FFT algorithm.

3. What is warping effect?

4. Compare the Butterworth and Chebyshev filter.

5. What are the desirable characteristics of the frequency response of window function?

6. What is meant by optimum equiripple design criterion?

7. What is meant by truncation and rounding?

8. Define dead band of the filter.

9. What are the advantages of multirate signal processing?

10. Define variance and covariance of discrete random signals.

PART B — ($5 \times 16 = 80$ marks)

11. (a) (i) Explain sampling rate conversion by a rational factor I/D and derive the output expression in both time and frequency domain. (9)

(ii) Draw the block diagram of subband coding process and explain. (7)

Or

(b) (i) Give the effects of finite word lengths in FIR digital filters. (8)

(ii) Explain the quantization noise model. (8)

12. (a) (i) Find the response of an LTI system with impulse response $h(n) = \{2, 1, 3\}$ for the input $x(n) = \{1, 2\}$ using DIT radix-2 FFT algorithm. (9)

(ii) State and prove convolution theorem of DFT. (7)

Or

(b) (i) A discrete-time system is characterized by the equation

$$y(n) = \sum_{k=-\infty}^{n} 2^{k-n} x(k+1)$$

Check whether the system is (1) LTI and (2) BIBO stable. (8)

(ii) Show that DTFT of a real valued signal is Hermitian symmetry. (8)

13. (a) (i) Derive the mapping formula for the bilinear transformation. (6)

(ii) Using impulse invariant technique convert the following analog transfer function into digital with sampling period $T = 0.2$ second. (10)

$$H(s) = \frac{1}{(s+1)(s+2)}.$$

Or

(b) Design a Chebyshev IIR digital filter to meet the following desired frequency response $H(\omega)$ by using impulse invariant technique with sampling period $T = 1$ second.

$$0.9 \le |H(\omega)| \le 1 \quad 0 \le \omega \le \frac{\pi}{4}$$

$$0 \le |H(\omega)| \le 0.24, \frac{\pi}{2} \le \omega \le \pi.$$

14. (a) (i) Design a bandpass filter which approximates the ideal filter with cut-off frequencies at 0.2 rad/sec and 0.3 rad/sec. The filter order is $N = 7$. Use Hamming window. (12)

(ii) Compare the Hamming window and Blackman window. (4)

Or

(b) (i) Explain the polyphase decomposition for FIR filter structure. (7)

(ii) Obtain linear phase structure with minimum number of multipliers for the system described by the equation : $y(n) = x(n) + 1/2\, x(n-1) - 1/4\, x(n-2) + 1/2\, x(n-3) + x(n-4)$. (5)

(iii) What is the principle of designing FIR filter using frequency sampling method? (4)

15. (a) Find the output roundoff noise power for the system having transfer function $H(z) = \dfrac{1}{(1-0.5z^{-1})(1-0.4\,z^{-1})}$ which is realized in cascade form. Assume word length is 4 bits.

Or

(b) Explain signal scaling to prevent overflow limit cycle in the second order digital filter implementation.
