Reg. No. :

Question Paper Code : 31362

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2013.

Fifth Semester

Electronics and Communication Engineering

EC 2302/EC 52 — DIGITAL SIGNAL PROCESSING

(Regulation 2008)

(Common to PTEC 2302 – Digital Signal Processing for B.E. (Part-Time) Fourth Semester, Electronics and Communication Engineering – Regulation 2009)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — $(10 \times 2 = 20$ marks)

1. What is zero padding? What is the purpose of it?
2. How many multiplications and additions are required to compute N-point DFT using radix-2 FFT?
3. Give the steps in the design of a digital filter from analog filters.
4. What are the disadvantages of direct-form realisation?
5. State the properties of FIR filter.
6. Give the desirable characteristics of the window.
7. What do you understand by input quantization error?
8. State the methods used to prevent overflow.
9. Give the steps in multistage sampling rate converter design.
10. Write any four applications of multi-rate signal processing.

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(3)

PART B — (5 × 16 = 80 marks)

11. (a) (i) Compute the DFT of the sequence whose values for one period is given by $\tilde{x}(n) = \{1, 1, -2, -2\}$. (8)

(ii) Compute the eight-point DFT of the sequence

$$x(n) = \begin{cases} 1 & 0 \le n \le 7 \\ 0 & \text{otherwise} \end{cases}$$ by using DIT and DIF algorithms. (8)

Or

(b) (i) Summarize the Difference between overlap-save method and overlap-add method. (8)

(ii) Evaluate the 8-point DFT for the following sequence using DIT-FFT algorithm $x(n) = \begin{cases} 1 & \text{for } -3 \le n \le 3 \\ 0 & \text{otherwise} \end{cases}$. (8)

12. (a) Discuss the steps in the design of IIR filter using Bilinear transformation for any one type of filter. (16)

Or

(b) Convert the following pole-zero IIR filter into a lattice ladder structure.

$$H(z) = \frac{\left[1 + 2z^{-1} + 2z^{-2} + z^{-3}\right]}{\left[1 + \left(\frac{13}{24}\right)z^{-1} + \left(\frac{5}{8}\right)z^{-2} + \left(\frac{1}{3}\right)z^{-3}\right]}$$ (16)

13. (a) (i) Explain briefly how the zeros in FIR filter is located. (7)

(ii) Using a rectangular window technique, design a low pass filter with pass band gain of unity, cut-off frequency of 1000 Hz and working at a sampling frequency of 5 kHz. The length of the impulse response should be 7. (9)

Or

(b) Consider an FIR lattice filter with coefficients $k_1 = 1/2; k_2 = 1/3; k_3 = 1/4$ Determine the FIR filter coefficients for the direct form structure. (16)

14. (a) (i) Discuss the various common methods of quantization. (8)

(ii) Explain the finite word length effects in FIR digital filters. (8)

Or

(b) Describe the quantization in floating point realization of IIR digital filters. (16)

15. (a) (i) Explain the implementation steps in speech coding using transform coding. (8)

(ii) Discuss the design steps involved in the implementation of multistage sampling rate converter. (8)

Or

(b) Explain the efficient implementation of polyphase decimator and interpolator. (16)

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