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Question Paper Code : 23451

B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2018.

Fifth Semester

Electronics and Communication Engineering

EC 2302 – DIGITAL SIGNAL PROCESSING

(Regulations 2008)

(Common to PTEC 2302 — Digital Signal Processing for B.E. (Part-Time) Fourth Semester — ECE — Regulations 2009)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Compare the number of multiplications required to compute the DFT of a 64 point sequence using direct computation and that using FFT.

2. What is meant by 'in place' in DIT and DIF algorithms?

3. Sketch the frequency response of an odd and an even order Chebyshev low pass filters.

4. What is bilinear transformation? What is the main advantages and disadvantages of this technique?

5. List out the advantages and disadvantages of FIR filters.

6. Write the equation of Hamming window function.

7. What do you understand by input quantization error?

8. State the methods used to prevent overflow.

9. State the various applications of DSP.

10. What is echo cancellation?

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) Design a digital Chebyshev filter to satisfy the constraints

$$0.707 \le \left|H(e^{j\omega})\right| \le 1, \quad 0 \le \omega \le 0.2\pi$$

$$\left|H(e^{j\omega})\right| \le 0.1, \quad 0.5\pi \le \omega \le \pi$$

Using bilinear transformation and assuming T = 1 sec. (16)

Or

(b) (i) For the analog transfer function

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

Determine $H(z)$ using impulse invariant method. Assume T = 1 sec. (10)

(ii) Obtain the cascade and parallel realizations for the system function given by

$$H(z) = \frac{1 + \frac{1}{4}z^{-1}}{\left(1 + \frac{1}{2}z^{-1}\right)\left(1 + \frac{1}{2}z^{-1} + \frac{1}{4}z^{-2}\right)}. \quad (6)$$

13. (a) Realize the system function $H(z) = \left(\frac{2}{3}\right)z + 1 + \left(\frac{2}{3}\right)z^{-1}$ by linear phase FIR Structure. (16)

Or

(b) Explain the designing of FIR filters using windows. (16)

14. (a) (i) Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation:

$y(n) = 0.95y(n-1) + x(n)$;

$x(n) = 0$ and $y(-1) = 13$.

Determine the dead band range of the system. (10)

(ii) Explain the effects of coefficient quantization in FIR filters. (6)

Or

(b) (i) Derive the signal to quantization noise ratio of A/D converter. (6)

(ii) Compare the truncation and rounding errors using fixed point and floating point representation. (10)

15. (a) (i) Explain the multistage implementation of sampling rate conversion with a block diagram. (8)

(ii) A signal $x(n)$ is given by $x(n) = \{0, 1, 2, 3, 4, 5, 6, 0, 1, 2, 3....\}$. (8)

(1) Obtain the decimated signal with a factor of 2.

(2) Obtain the interpolated signal with a factor of 2.

Or

(b) Explain sampling rate increase by an integer factor I and derive the input-output relationship in both time and frequency domains.