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Question Paper Code : 30947

B.E./B.Tech. DEGREE EXAMINATIONS, APRIL/MAY 2019.

Fifth Semester

Electronics and Communication Engineering

EC 2302 — 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 × 2 = 20 marks)

1. State the difference between DFT and DTFT.

2. What is Bit Reversal?

3. Why do we go for analog approximation to design a digital filter?

4. Give any two properties of chebyshev filters.

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. Define decimator and interpolator.

10. List the applications of multi rate signal processing.

PART B — (5 × 16 = 80 marks)

11. (a) (i) State the following properties of DFT.

(1) Time reversal

(2) Parsavel's theorem. (8)

(ii) Perform the linear convolution of the given sequences $x(n) = (1, -1, 1-1)$, $h(n) = \{1, 2, 3, 4\}$ using DFT method. (8)

Or

(b) Derive the butterfly diagram of 8 point radix-2 DIF–FFT algorithm and fully label it.

12. (a) Design a digital Chebyshev filter to satisfy the constraints

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

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

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 :

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

$x(n) = 0$ 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 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)
