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

Question Paper Code : 21362

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 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 × 2 = 20 marks)

1. Find the 4–point DFT of the sequence $x(n) = \{1, 1, -1, -1\}$.

2. What is meant by in-place Computation?

3. Mention the advantages of cascade realization.

4. Convert the given analog transfer function $H(s) = \frac{1}{s+a}$ into digital by impulse invariant method.

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

6. Write the equation of Hamming window function.

7. What are the two types of quantization employed in digital system?

8. Define zero input limit cycle oscillations.

9. What is anti-imaging filter?

10. Give the applications multi-rate DSP.

PART B — $(5 \times 16 = 80$ 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) A desired low pass filter with the following specification is

$0.8 \le |H(\omega)| \le 1.0; 0 \le \omega \le 0.2\pi$

$|H(\omega)| \le 0.2;\ 0.3\pi \le \omega \le \pi$

Design Butterworth digital filter using impulse invariant transformation.

Or

(b) (i) Obtain the cascade form realization of the digital system

$$y(n) = \frac{3}{4}y(n-1) - \left(\frac{1}{8}\right)y(n-2) + \frac{1}{3}x(n-1) + x(n). \quad (8)$$

(ii) Convert the given analog filter with a transfer function

$H(s) = \dfrac{2}{(s+1)(s+2)}$ into a digital IIR filter using bilinear transformation. Assume T=1 sec. (8)

13. (a) (i) Determine the frequency response of FIR filter defined by $y(n) = 0.25x(n) + x(n-1) + 0.25x(n-2)$.

Calculate the phase delay and group delay. (8)

(ii) Discuss the design procedure of FIR filter using frequency sampling method. (8)

Or

(b) Design a FIR filter with the following desired specification

$$H_d(e^{jw}) = \begin{cases} 0, & \dfrac{-\pi}{4} \le \omega \le \dfrac{\pi}{4} \\ e^{-j2\omega}, & \dfrac{\pi}{4} \le |\omega| \le \pi \end{cases}$$

using a Hanning window with N = 5.

14. (a) (i) Represent the following numbers in floating point format with five bits for mantissa and three bits for exponent.

(1) 7_{10}

(2) 0.25_{10}

(3) -7_{10}

(4) -0.25_{10} . (8)

(ii) Draw the product quantization noise model of second order IIR system. (8)

Or

(b) (i) Explain how signal scaling is used to prevent overflow limit cycle in the digital filter implementation with an example. (8)

(ii) Determine the dead band of the system $y(n) = 0.2y(n-1) + 0.5y(n-2) + x(n)$.

Assume 8 bits are used for signal representation. (8)

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.
