Reg. No. :

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B.E./B.Tech. DEGREE EXAMINATION, APRIL/MAY 2017.

Fifth Semester

Electronics and Communication Engineering

EC 2302/EC 52 — DIGITAL SIGNAL PROCESSING

(Regulations 2008)

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

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

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

2. What is need for decimation and interpolation?

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 filters.

6. What is meant by Gibbs Phenomenon?

7. What is scaling?

8. What is dead band of a filter?

9. What is effect of quantisation noise on frequency spectrum?

10. What is meant by multirate signal processing?

PART B — (5 × 16 = 80 marks)

11. (a) (i) Find the DFT of a sequence $x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ using DIT algorithm. (10)

(ii) State any six properties of DFT. (6)

Or

(b) (i) Using linear convolution find $y(n) = x(n) * h(n)$ for the sequences $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 1, 2, -1\}$ and $h(n) = \{1, 2\}$. Compare the result by solving the problem using overlap add method and overlap save method. (12)

(ii) Find the IDFT of the sequence $X(k) = \{6, -2+2j, -2, -2-2j\}$ using DIF algorithm. (4)

12. (a) Design a low pass Butterworth digital filter with the following specifications:

$W_S = 4000,\ W_P = 3000$

$A_P = 3\,dB,\ A_S = 20\,dB,\ T = 0.0001$ sec. (16)

Or

(b) A system is represented by a transfer function $H(z)$ is given by

$$H(z) = 3 + \frac{4z}{z - 1/2} - \frac{z}{z - 1/4}$$

(i) Does this $H(z)$ represent a FIR or IIR filter why? (4)

(ii) Give a difference equation realization of this system using direct form –I. (6)

(iii) Draw the block diagram for the direct form II canonic realization, and give the governing equations for implementation. (6)

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) Represent the following numbers in floating point format, with five bits for mantissa and three bits for exponent. (8)

(1) 7_{10}

(2) 0.25_{10}

(3) -7_{10}

(4) -0.25_{10}

(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 $y(n) = 0.2\,y(n-1) + 0.5\,y(n-2) + x(n)$. Assume 8 bits are used for signal representation. (8)

15. (a) (i) Explain how various sound effects can be generated with the help of DSP? (10)

(ii) State the applications of multirate signal processing. (6)

Or

(b) (i) Explain how DSP can be used for speech processing? (8)

(ii) Explain in detail about decimation and interpolation. (8)