Reg. No. : [ ][ ][ ][ ][ ][ ][ ][ ][ ][ ][ ][ ]

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B.E./B.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2014.

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 – ECE – Regulation 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. Distinguish between Butterworth and Chebyshev filter.

4. What is prewarping?

5. Give the equations specifying Hamming and Blackman window.

6. Realize the following causal linear phase FIR system function

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

7. What is scaling?

8. What is dead band of a filter?

9. Define decimator and interpolator.

10. List the applications of multi rate 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 digital Chebyshev filter to satisfy the constraints

$$0.707 \le \left|H\left(e^{j\omega}\right)\right| \le 1, \quad 0 \le \omega \le 0.2\pi$$
$$\left|H\left(e^{jw}\right)\right| \le 0.1, \quad 0.5\pi \le \omega \le \pi$$

Using bilinear transformation and assuming $T = 1\,\text{sec}$. (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 $T = 1\,\text{sec}$. (10)

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

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

13. (a) (i) A low pass filter has the desired response as given below

$$H_d\left(e^{j\omega}\right)=\begin{cases} e^{-j3\omega}, & 0 \le \omega < \dfrac{\pi}{2} \\ 0, & \dfrac{\pi}{2} \le \omega \le \pi \end{cases}$$

Determine the filter coefficients $h(n)$ for M = 7, using type-I frequency sampling technique. (10)

(ii) What is a linear phase filter? What are the conditions to be satisfied by the impulse response of an FIR system in order to have a linear phase. (6)

Or

(b) Design a bandpass filter which approximates the ideal filter with cut off frequencies at 0.2 rad / sec and 0.3 rad / sec. The filter order is M = 7. Use the Hanning window function. (16)

14. (a) Discuss the following:

(i) Product quantization error (8)

(ii) Limit cycle oscillations. (8)

Or

(b) (i) Derive the equation for rounding and truncation errors. (8)

(ii) Derive the equation for quantization noise power. (8)

15. (a) Explain with block diagram the general poly phase frame work for decimator and interpolator. (16)

Or

(b) Implement a two stage decimator for the following specifications:

Sampling rate of the input signal = 20,000Hz

M = 100

Passband = 0 to 40 Hz

Transition band = 40 to 50 Hz

Passband ripple = 0.01

Stopband ripple = 0.002. (16)