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Total number of printed pages - 2

B. Tech

CPEC 5302

Fifth Semester (Special) Examination – 2013 DIGITAL SIGNAL PROCESSING

BRANCH CODE: EC

QUESTION CODE: D 274

Full Marks - 70

Time: 3 Hours

Answer Question No. 1 which is compulsory and any five from the rest.

The figures in the right-hand margin indicate marks.

1. Answer the following questions:

2×10

- (a) What is aliasing effect? How it can be avoided?
- (b) Express x (n) in terms of impulse function δ (n) when X(n) = $\{1, 3, 2, 1\}$.
- (c) State the conditions of stability for causal LTI system
- (d) Draw the basic structure of 1st order digital FIR fifter
- (e) What do you mean by region of convergence (FQC)? What is mortance in discrete time system?
- (f) Why FIR filters are inherently stable?
- (g) State time shifting property of DFT.
- (h) How many real multiplication and addition is required for computation of 8-point DFT using decimation in time algorithm?
- (i) Find the impulse response the LTI system shown below :

$$Y(n) = 0.5x(n-1) + 2x(n)$$

- (j) What are different bands in a digital filter? At which band an ideal filter is distortion less?
- (a) Determine the transient and steady state response of the system described by

$$x(n) = y(n-1) + 2y(n-2)$$

When a unit step function is applied to the system.

(b) Find out auto correlation of the following signal $x(n) = a^n u(n)$ 0 < a < 1

Where u(n) the unit step is applied to the discrete time system.

- 3. (a) Find inverse Z-transform of $X(z) = \log (1 2z) \qquad |z| > |a|$
 - (b) Determine the range values of parameter a for which the LTI system with impulse response 5

Is stable

4. (a) Convert the analog filter with the system function

 $H(s) = \frac{s+0.1}{(s+0.1)^2+9}$ Into a digital IIR filter using impulse invariance method. The digital filter is to have resonant frequency of 0.2.

- (b) Establish the relation between ω and then, bring out a mapping between them. 4
- (a) Consider the casual system
 Y(n) = 0.75y(n 1) 0.125y (n 2) + x(n) + 0.3x (n 1)
 Obtain direct form I and form II structure of the system.
 - (b) Explain how the IIR filter is designed from analog filter using bilinear transformation method.
- 6. (a) Establish the inverse Z-transform of X (Z) using Contour integration. 5
 - (b) State and prove scaling property of Z-transform.
- 7. (a) Explain Decimation in Time FFT algorithm.5(b) Prove that convolution of two signals in discrete time domain is equal to
 - multiplication in discrete frequency domain.
- 8. Write short notes on any **two** of the following: 5×2
 - (a) System Modeling
 - (b) Linear phase FIR filter by frequency sampling method
 - (c) Adaptive Line Enhancer
 - (d) Minimum Mean Square Error Criterion.

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