Seat No.: _____ Enrolment No._____

GUJARAT TECHNOLOGICAL UNIVERSITY

BE SEMESTER– 7th EXAMINATION – SUMMER 2015

Subject Code:171003 **Date:12/05/2015**

Subject Name:Digital Signal Processing

Time:02.30PM-05.00PM **Total Marks: 70**

Instructions:

1. **Attempt all questions.**
2. **Make suitable assumptions wherever necessary.**
3. **Figures to the right indicate full marks.**

Q.1 **(a)** Attempt the following questions. **08**

1) Sketch the basic block diagram of Digital Signal Processor and explain each blocks in brief.
2) Prove convolution of two sequences in time domain is equivalent to multiplication of two spectra in frequency domain.
3) Explain associative property of Z – transform.
4) Differentiate between DTFT and DFT.

(b) Classify the following systems. **06**

i) Linear Vs non linear systems
ii) Causal Vs non causal systems

Q.2 **(a)** Determine the causality and stability of given discrete time sequences. **07**

1) $h(n) = 2^n u(n)$
2) $h(n) = n\,u(-n+2)$

(b) Find out cross correlation between following discrete time sequences. **07**

x(n)={1,2,2,1} & h(n)= {1,-1,2}

↑ ↑

OR

(b) An FIR filter has an impulse response given by h(n) ={1,2,2,3}. Find out the response of the filter y(n) if given discrete input signal is x(n)={2,-1,3}. Verify the answer using z-transform method. **07**

Q.3 **(a)** Determine the unit sample response of the system characterized by the given difference equation. **07**

y(n) = 2.5y(n-1) – y(n-2) + x(n) – 5x(n-1) + 6x(n-2)

(b) Discuss various properties of Z- transform. **07**

OR

Q.3 **(a)** For a given X(z) compute Inverse Z-transform using partial fraction expansion method such that resulting signal is causal. **07**

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

(b) Discuss architecture of Digital Signal Processor and its key features in detail. **07**

Q.4 **(a)** In practical applications while dealing with long input data sequence, how linear filtering is done using DFT? Explain any one approach in detail. **07**

(b) Compute Circular Convolution of given discrete time sequences and verify the result with DFT method (use linear transform to compute DFT). **07**

$x(n) = \{1,2,2,1\}$ & $h(n) = \{1,2,3,4\}$

OR

Q.4 **(a)** What do you mean by frequency domain sampling and how is it applicable on DTFT spectrum? Derive the equation for Discrete Fourier Transform and also describe recovery of discrete time signal from DFT spectrum. **07**

(b) For the following difference equation representing filter, obtain transfer function of the filter H(z). Realize the above filter using Direct form I, Direct form II and Cascaded form. **07**

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

Q.5 **(a)** 1) Prove and validate the following statement: Ideal filters are non causal and hence practically non-realizable. (3) **07**

2) Convert $H(S) = \frac{4s+7}{s^2+5s+4}$ to H(z) using Impulse Invariance IIR filter at S = 2 Hz. (4)

(b) Derive the equation and discuss 8 point Radix 2 DIT FFT algorithm to compute DFT. Also state the importance of bit reversal table in it. **07**

OR

Q.5 **(a)** Specify the limitations of Impulse Invariance method while implementing digital IIR filter. Discuss Binliner Transform (BLT) and explain how such limitations can be rectified using BLT method. Also mention limitations of BLT. **07**

(b) In context with overall computational complexity and speed improvement factor compare and contrast DFT, Divide and Conquer approach and Radix 2 FFT algorithms. **07**

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