Seat No.: ________ *Enrolment No.*___________

GUJARAT TECHNOLOGICAL UNIVERSITY

M. E. - SEMESTER – I • EXAMINATION – WINTER • 2014

Subject code: 2710501 **Date: 13-01-2015**

Subject Name: Digital Signal Processing Algorithms

Time: 02:30 pm - 05:00 pm **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)** Determine and sketch the convolution $y(n)$ of the signals $x(n)=\{1 \quad 2 \quad -1 \quad 0 \quad 2\}$ and $h(n)=\{1 \quad 0 \quad -1 \quad -1\}$ **07**

(b) **(i)** Check the following systems for time invariance and linearity, **04**
(i) $y(n)=x(n^2)$ (ii) $y(n)=x^2(n)$

(ii) List and briefly explain applications of DSP. **03**

Q.2 **(a)** Obtain the direct form II, cascade and parallel form structures for the following system **07**
$$y(n)=-0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)$$

(b) Determine the 4-point DFT of a sequence $x(n)=u(n)-u(n-3)$. Also find magnitude and phase spectrum of DFT. **07**

OR

(b) Determine the sequence $x_3(n)$, a circular convolution of $x_1(n)=\{1,2,3,1\}$ and $x_2(n)=\{4,3,2,2\}$ using 4-point DFT and IDFT. **07**

Q.3 **(a)** Compute 8-point DFT of a sequence $x(n)=\{1 \quad -1 \quad -1 \quad -1 \quad 1 \quad 1 \quad 1 \quad -1\}$ using DIT FFT algorithm and draw the flow diagram. **07**

(b) State and explain circular time shift and circular time reversal property of DFT. **07**

OR

Q.3 **(a)** Compute 8-point DFT of a sequence $x(n)=\{1 \quad -1 \quad -1 \quad -1 \quad 1 \quad 1 \quad 1 \quad -1\}$ using DIF radix-2 FFT algorithm. **07**

(b) Draw a signal flow graph representation of the DIT FFT method for finding 8-point DFT. **07**

Q.4 **(a)** Explain IIR filter design by Bilinear Transformation method. **07**

(b) List and Explain any two windowing methods for FIR filter design. **07**

OR

Q.4 **(a)** Design a single-pole lowpass filter with a 3-dB bandwidth of 0.2´, using the bilinear transformation applied to the analog filter **07**

$H(s)=\dfrac{\Omega_c}{s+\Omega_c}$ where Ω_c is the 3-dB bandwidth of the analog filter.

(b) Explain Impulse Invariance method for IIR filter design. **07**

Q.5 **(a)** Explain model based methods for power spectral density estimation. **07**

(b) Write short note on discrete wavelet transform. **07**

OR

Q.5 **(a)** Write short note on Bartlett and Welch method for spectral estimation. **07**

(b) State and prove the properties of z-transform **07**

(i) Convolution of two sequences

(ii) Differentiation in z domain

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