Seat No.: ________ Enrolment No.____________

GUJARAT TECHNOLOGICAL UNIVERSITY

M. E. - SEMESTER – I • EXAMINATION – WINTER 2012

Subject code: 714402N **Date: 09-01-2013**

Subject Name: Advanced Digital Signal Processing

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)** State the properties of the ROC for the Z-Transform. Find Inverse Z-Transform by partial-fraction method. **07**

$H(z) = [z(z + 2.0)] / [(z – 0.2)(z + 0.6)]$

(b) State and prove the convolution properties of Z- Transform and find $y[n] = x_1[n]*x_2[n]$, $x_1[n] = a^n u[n]$ and $x_2[n] = u[n]$. **07**

Q.2 **(a)** Compute the Inverse – DFT from the DFT components [2, 1 + j, 0, 1 – j]. Explain Discrete Cosine Transform shortly. **07**

(b) Explain "Decimation-In-Frequency Fast Fourier Transform (FFT)" algorithm fundamentally. **07**

OR

(b) State properties of Discrete Fourier Transform. **07**

Q.3 **(a)** Use pole-zero method of calculating filter coefficient. A bandpass digital filter is required to meet the following specifications: **07**

(1) Complete signal rejection at dc and 250 Hz;
(2) A narrow passband centred at 125 Hz;
(3) A 3 dB bandwidth of 10 Hz

Assuming a sampling frequency of 500 Hz, obtain the transfer function of the filter by suitably placing z-plane poles and zeros, and its difference equations.

(b) Explain Bilinear Z-Transform Method of coefficient calculation. **07**

OR

Q.3 **(a)** Obtain the coefficients (first three & last) of an FIR low pass filter to meet the specifications given below using the Hamming window method. **07**

Pass band edge frequency: 1.5 kHz
Transition width: 0.5 kHz
Stopband attenuation: > 50 dB
Sampling frequency: 8 kHz

(b) Discuss with all necessary equations of design of practical sampling rate converters. **07**

Q.4 **(a)** The normalized transfer function of an analog filter is given by $H(s) = 1 / (s^2 + 1.414s + 1)$, Obtain the transfer function, H(z), of an equivalent digital filter using the matched Z- Transform method. Assume a 3 dB cutoff frequency of 150 Hz and a sampling frequency of 1.28 kHz. **07**

(b) Discuss the basic LMS adaptive algorithm with flowchart. **07**

OR

Q.4	**(a)**	Explain basic "Wiener Filter" theory with neat sketch and necessary equation. Mention its Limitations	**07**
Q.4	**(b)**	Explain recursive least squares adaptive algorithm, and discuss its limitations.	**07**
Q.5	**(a)**	Compare Harvard architecture with non-Harvard architecture with timing diagram.	**07**
	(b)	Discuss simplified architecture of a first generation and second generation fixed-point DSP processor (TMS320CXX).	**07**
		OR	
Q.5	**(a)**	With Neat Sketch and Explain Architecture of TMS320C67XX DSP Processor.	**07**
	(b)	List the Various applications of DSP and explain any one in detail.	**07**

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