Seat No.: _____ Enrolment No._____

GUJARAT TECHNOLOGICAL UNIVERSITY

ME – SEMESTER I (NEW) – • EXAMINATION – SUMMER 2016

Subject Code: 2710501 **Date:16/05/2016**

Subject Name: Digital Signal Processing Algorithms

Time:02:30 pm to 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)**	Define: Causal system, Linear system, and Time-variant system. Also determine system $y(n)=x(n^2)$ is static or dynamic, causal or non-causal, linear or non-linear and time-variant or time-invariant.	**07**
	(b)	State and explain the time shifting property of z-transform. Also determine the z-transform and ROC of the signal $x_1(n)=x(n-2)$, where $x(n)=\{1 \quad \underset{\uparrow}{-1} \quad -2 \quad 1 \quad 2\}$ using time shifting property.	**07**
Q.2	**(a)**	Obtain the direct form I, direct form II, cascade and parallel form structures for the system $y(n)=\frac{3}{4}y(n-1)-\frac{1}{8}y(n-2)+x(n)+\frac{1}{3}x(n-1)$.	**07**
	(b)	State and explain properties of DFT.	**07**
		OR	
	(b)	Determine the 4-point DFT of a sequence $x(n)=\delta(n)+\delta(n-1)+\delta(n-2)$. Also find magnitude and phase spectrum of DFT.	**07**
Q.3	**(a)**	Develop radix-2 FFT algorithm using decimation in time approach.	**07**
	(b)	Determine the sequence $x_3(n)$, a circular convolution of $x_1(n)=\{2,1,2,1\}$ and $x_2(n)=\{1,2,3,4\}$ using 4-point DFT and IDFT.	**07**
		OR	
Q.3	**(a)**	Explain Chirp-z transform algorithm.	**07**
	(b)	Compute 8-point DFT of a sequence $x(n)=\{1 \quad -4 \quad 2 \quad -3 \quad 3 \quad -2 \quad 4 \quad -1\}$ using DIT FFT algorithm and draw the flow diagram.	**07**
Q.4	**(a)**	List and Explain any two windowing methods for FIR filter design.	**07**
	(b)	Determine the unit sample response $\{h(n)\}$ of a linear-phase FIR filter of length $M=4$ for which the frequency response at $\omega=0$ and $\omega=\pi/2$ is specified as $H_r(0)=1$ and $H_r(\frac{\pi}{2})=1/2$.	**07**
		OR	
Q.4	**(a)**	Explain Impulse Invariance method for IIR filter design.	**07**
	(b)	Convert the analog filter with system function $H(s)=\frac{s+0.1}{(s+0.1)^2+9}$ into a digital IIR filter using the bilinear transformation. Compare the location of the zeros in $H(z)$ with the locations of the zeros obtained by applying the impulse invariance method in the conversion of $H(s)$. (Select $T=0.1$)	**07**
Q.5	**(a)**	Explain ARMA model for power spectrum estimation.	**07**
	(b)	Write short note on discrete wavelet transform.	**07**
		OR	

Q.5	**(a)**	Write short note on the Bartlett: a nonparametric method for power spectrum estimation.	**07**
	(b)	List and briefly explain applications of DSP.	**07**

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