Seat No.: ________ Enrolment No.___________

GUJARAT TECHNOLOGICAL UNIVERSITY

ME - SEMESTER– I (New course)• REMEDIAL EXAMINATION – SUMMER 2015

Subject Code: 2710501 **Date:12/05/2015**

Subject Name: Digital Signal Processing Algorithms

Time: 10:30 am to 1:00 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, Time-variant system and Stable system. Also determine system $y(n) = Ax(n) + B$ is causal or non-causal, linear or non-linear and time-variant or time-invariant. **07**

(b) Explain discrete wavelet transform and its application in DSP. **07**

Q.2 **(a)** Obtain the direct form I, direct form II, cascade, and parallel form structures for the following system **07**

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

(b) Justify with suitable example: A linear time-invariant system is stable if its impulse response is absolutely stable. **07**

OR

(b) The impulse response of a linear time-invariant system is $h(n) = \{1, \underset{\uparrow}{2}, 1, -1\}$. Determine the response of the system to the input signal $x(n) = u(n) - u(n-3)$. **07**

Q.3 **(a)** By means of the DFT and IDFT, determine the response of the FIR filter with impulse response $h(n) = \{1 \quad 2 \quad 3\}$ to the input sequence. **07**

(b) State and explain properties of DFT. **07**

OR

Q.3 **(a)** Determine the circular convolution of the sequences: $x_1(n) = \{1 \quad 2 \quad 3 \quad 1\}$ and $x_2(n) = \{4 \quad 3 \quad 2 \quad 2\}$. **07**

(b) Develop radix-2 FFT algorithm using decimation in time approach. **07**

Q.4 **(a)** Explain Chirp-z transform algorithm. **07**

(b) Explain the Bartlett: a nonparametric method for power spectrum estimation. **07**

OR

Q.4 **(a)** Explain ARMA model for power spectrum estimation. **07**

(b) Compute the 8-point DFT of the sequence $x(n) = \{\frac{1}{2} \quad \frac{1}{2} \quad \frac{1}{2} \quad \frac{1}{2} \quad 0 \quad 0 \quad 0 \quad 0\}$ using radix-2 DIF algorithm. **07**

Q.5 **(a)** Explain frequency sampling method for FIR filter design. **07**

(b) Convert the analog filter with system function $H_a(s) = (s+0.1)/\left[(s+0.1)^2 + 16\right]$ into a digital IIR filter having a resonant frequency of $\omega_r = \pi/2$ by means of the bilinear transformation. **07**

OR

Q.5 **(a)** List the various methods of designing IIR filter. Explain the bilinear transformation method of designing of IIR filter. How does this method overcome the limitation of other methods? **07**

(b) Determine the order and the poles of a type I lowpass Chebyshev filter that has a 1-dB ripple in the passband, a cutoff frequency $\Omega_p = 1000\pi$, a stopband frequency of 2000π, and an attenuation of 40 dB or more for $\Omega \geq \Omega_s$. **07**

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