Seat No.: ________ Enrolment No.___________

# GUJARAT TECHNOLOGICAL UNIVERSITY

**ME - SEMESTER– I EXAMINATION – WINTER 2014**

**Subject Code: 3715203** **Date: 09/01/2015**

**Subject Name: Digital Signal Processing**

**Time:02:30 to 05:00** **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 the convolution $y(n)$ between the following signals $x(n)=u(n)-u(n-4)$ and $h(n)=4\delta(n+2)+3\delta(n+1)+2\delta(n)+\delta(n-1)$ **07**

**(b)** Determine the impulse response of the following causal system: $y(n)=3y(n-1)+4y(n-2)+x(n)+2x(n-1)$ **07**

**Q.2** **(a)** Determine the causal signal $x(n)$ if its Z transform $X(z)$ is given by: **07**

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

(ii) $X(z)=\dfrac{1+2z^{-2}}{1+z^{-2}}$

**(b)** Determine the eight-point DFT of the signal $x(n)=\{1 \;\; 1 \;\; 1 \;\; 1 \;\; 1 \;\; 1 \;\; 0 \;\; 0\}$ and sketch its magnitude and phase. **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 $x(n)=\{1 \;\; -1 \;\; -1 \;\; -1 \;\; 1 \;\; 1 \;\; 1 \;\; -1\}$ using DIT FFT algorithm and draw the flow diagram. **07**

**(b)** Describe the window method for designing linear phase FIR filter. **07**

**OR**

**Q.3** **(a)** Develop radix-2 FFT algorithm using decimation in time approach. **07**

**(b)** For the given frequency response **07**

$H_r(0)=1 \quad for\ \omega=0$ and $H_r\left(\dfrac{\pi}{2}\right)=\dfrac{1}{2} \quad for\ \omega=\dfrac{\pi}{2}$

Determine the unit sample response $h(n)$ of a linear-phase FIR filter of length $M=4$.

**Q.4** **(a)** Design an analog Butterworth filter to obtain $H(s)$ with the following filter specifications: **07**

Passband frequency = 1000 Hz

Passband attenuation=1 dB

Stopband frequency=12000 Hz

Stopband attenuation=80 dB

**(b)** Write short note on (i) Analog to Digital Converter and (ii) Digital to Analog Converter with their significance. **07**

**OR**

**Q.4** **(a)** Obtain the expressions for the order and cut-off frequency of an analog Butterworth type low pass system from the specifications of the LPF system. **07**

**(b)** Explain in detail following analog system transformation methods: **07**

(i) Low pass filter to High pass filter

(ii) Low pass filter to Band pass filter

**Q.5** **(a)** Obtain the direct form I, 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)** Explain IIR filter design by Bilinear Transformation method. **07**

**OR**

**Q.5** **(a)** Draw transposed direct form II and cascade form structure for the function, **07**

$$H(z) = \frac{\left(1 + 2z^{-1} - 3z^{-2}\right)}{\left(1 + 5z^{-1} + 6z^{-2}\right)}$$

**(b)** Design the first order digital Butterworth lowpass filter having 3-dB frequency is $0.4\pi$, using Bilinear transformation. **07**

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