Seat No.: _____ Enrolment No.______

# GUJARAT TECHNOLOGICAL UNIVERSITY

**M.E Sem-I Examination January 2010**

## Subject code: 710422

## Subject Name: Digital Signal Processing and Application

**Date:27/ 01 / 2010** **Time: 12:00 - 2:30pm**

**Total Marks: 60**

# Instructions:

1. **Attempt all questions.**
2. **Make suitable assumptions wherever necessary.**
3. **Figures to the right indicate full marks.**

**Q.1** **(a)** For each of the following systems, determine whether system is stable, causal, linear and time-invariant or not. **06**

(i) $y[n]=g[n]x[n]$ (ii) $y[n]=ax[n]+b$

**(b)** (i) Find the linear convolution of two sequences given by **02**

$x(n)=b^n u(n)$ for all n & $h(n)=a^n u(n)$ for all n

(ii) How can you find step response of a system if impulse response is known? Determine the step response of a linear-time-invariant system whose unit-impulse response is given by $h(n) = a^{-n}u(-n)$, $0<a<1$ **04**

**Q.2** **(a)** State sampling theorem. What is aliasing effect? **06**

A digital communication link carries binary coded words representing samples of an input signal $X_a(t)=3\cos 600\pi t+2\cos 800\pi t$.The link is operated at 10,000bits/sec and each input sample is quantized into 1024 different voltage levels.

(i) What is sampling frequency and folding frequency?

(ii) What is Nyquist rate for the signal $X_a(t)$?

(iii) What are the frequencies in resulting discrete time signal?

**(b)** What is multirate signal processing? Explain application of Decimation and Interpolation to A/D and D/A conversion. **06**

**OR**

**(b)** (i)For difference equation $y(n)-3y(n-1)-4y(n-2)=x(n)+2x(n-1)$ find frequency response $H(e^{jw})$ and impulse response $h(n)$ **03**

(ii) State properties of Region Of Convergence for z-transform **03**

**Q.3** **(a)** Explain Decimation-in-time algorithm to calculate 8-point DFT **06**

**(b)** A causal LTI system has impulse response h(n),for which the z-transform is $H(z)=(1+z^{-1})/[(1-0.5z^{-1})(1+0.25z^{-1})]$ **06**

(i)What is the region of convergence of H(z)?

(ii) Is the system stable?

(iii)Find the z-transform X(z) of an input x[n] that will produce the output $y[n]= -(1/3)(-1/4)^n u[n]-(4/3)(2)^n u[-n-1]$

(iv) Find the impulse response h[n] of the system.

**OR**

**Q.3** **(a)** How to obtain linear convolution from circular convolution? Perform the circular convolution of $x[n]=\{1,1,2,1\}$ and $h[n]=\{1,2,3,4\}$ using DFT and IDFT method. **06**

**(b)** Distinguish between DFT and DTFT. List out the properties of DFT and explain circular shift of a sequence property of DFT. **06**

**Q.4 (a)** Design a low pass discrete time filter by applying bilinear transformation method to an appropriate Butterworth continuous time filter for the following specification. **08**
(i) Passband magnitude is constant within 3 db for frequencies below $0.2\pi$ i.e $0 \le w \le 0.2\pi$
(ii) Stopband attenuation is greater than 25 db for frequencies between $0.45\pi$ and $\pi$ i.e $0.45\pi \le w \le \pi$. Consider T is equal to unity.

**(b)** Compare FIR and IIR digital filters **04**

**OR**

**Q.4 (a)** Give direct form I and direct form II structure of second order system realization. **04**

**(b)** What is Gibbs phenomenon? Design an FIR lowpass filter using Kaiser window filter design method satisfying the specifications $w_p=0.4\pi, w_s=0.6\pi, \delta_p=0.01$ $\delta_s=0.001$. **08**

**Q.5 (a)** Compare the fixed point and floating point arithmetic for DSP Processors. **04**

**(b)** Explain pipeline concepts in DSP. **04**

**(c)** List various application areas of DSP processors and describe application of DSP to speech processing. **04**

**OR**

**Q.5 (a)** Explain finite word length effects in FIR digital filters. **06**

**(b)** Describe Harvard Architecture in detail. **06**

*************