Seat No.: _____ Enrolment No._____

GUJARAT TECHNOLOGICAL UNIVERSITY

M.E Sem-I Remedial Examination April 2010

Subject code: 710422

Subject Name: Digital Signal Processing and Application

Date: 09 / 04 / 2010 **Time: 12.00 noon – 02.30 pm**

Instructions: **Total Marks: 60**

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

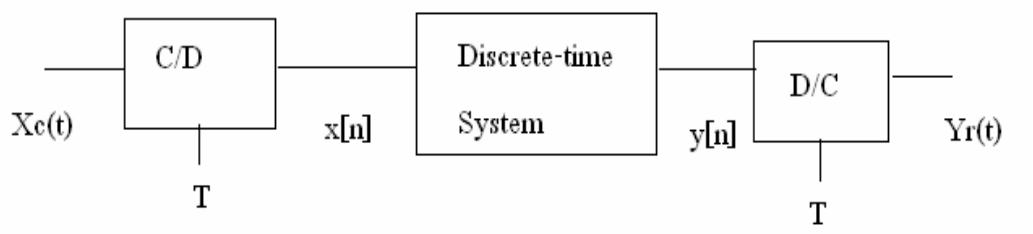
Q.1 **(a)** Answer the following questions: **06**

1) What is difference between discrete-time signal and digital signal? Explain with example.
2) State the different stages involve in digital filter designing?
3) Define inverse system. Also find inverse of the system given by function $H(z)=\frac{1-0.5z^{-1}}{1-0.9z^{-1}}$ with ROC |z| > 0.9.

(b) A causal linear time-invariant system is described by the difference equation $y[n]-5y[n-1]+6y[n-2]=2x[n-1]$ **06**

1) Determine the impulse response of the system.
2) Determine the step response of the system.

Q.2 **(a)** Consider the system of figure, with the discrete-time system an ideal low-pass filter with cutoff frequency $\pi/8$ radians/sec. **06**



1) If $x_c(t)$ is band limited to 5 KHz, what is the maximum value of T that avoid aliasing in C/D converter?
2) If 1/T = 10 KHz, what will the cutoff frequency of effective continuous time filter be?
3) Repeat part (2) for 1/T=20 KHz.

(b) Explain linear convolution using DFT and compare it with circular convolution for the system input x[n] = {1, 2, 3} and system impulse response h[n] = {2, 1}. **06**

OR

(b) What is difference between DFT and DTFT? Compute the value of DFT and DTFT for the given sequence x[n] = {0, 1, 2, 3} and compare it. **06**

Q.3 **(a)** When the input to an LTI system is $x[n]=\left(\frac{1}{3}\right)^n u[n]+(2)^n u[-n-1]$ the corresponding output is $y[n]=5\left(\frac{1}{3}\right)^n u[n]-5\left(\frac{2}{3}\right)^n u[n]$. **06**

1) Find the system function H (z) of the system. Plot the pole(s) and Zero(s) of H (z) and indicate region of convergence.
2) Is the system stable? Is it causal?

(b) Compute 8-point DFT of the sequence x[n] = u[n]-u [n-8] using decimation in time FFT algorithm. Also compare the computational complexity with direct DFT computation and using FFT algorithm. **06**

OR

Q.3 **(a)** Find inverse z-transform of **06**

$$X(z)=\frac{1}{1-1.5z^{-1}+0.5z^{-2}}$$

1) If x[n] is right side sequence.
2) If x[n] is left side sequence.
3) If x[n] is double side sequence.

(b) In what way FFT algorithm is better than the conventional DFT implementation? Explain decimation in frequency decomposition FFT algorithm with example. **06**

Q.4 **(a)** Compare IIR and FIR filter. Also justify FIR filter are stable. **06**

(b) Using Hamming window design a low-pass FIR filter for the following specifications: **06**

Cut-off frequency =500 Hz
Sampling frequency =2 KHz
Order of the filter N=10
Filter length required L=N+1=11

(Hint: $\omega_{HAM}=0.54+0.46\cos\left(\frac{\pi n}{M}\right);\quad -M\le n\le M$)

OR

Q.4 **(a)** What is concept of bilinear transformation technique of IIR filter design? An analog filter has the following transfer function $H_a(s)=\frac{1}{S+1}$.Using bilinear transformation technique, determine the transfer function of digital filter H (z) and also write the difference equation of digital filter. Assume T=1sec. **06**

(b) Draw the signal flow graph for implementation of the discrete time system $H(z)=\frac{1+2z^{-1}+z^{-2}}{1-0.75z^{-1}+0.125z^{-2}}$ in each of the following forms: **06**

1) Direct form 1
2) Cascade form
3) Parallel form

Q.5 **(a)** Explain the role of DSP in speech coding. **06**

(b) What is Multirate signal processing? What is need for Multirate signal processing? Explain the frequency domain representation of decimation process with an example. **06**

OR

Q.5 **(a)** Compare DSP processor and general purpose microprocessor. Also Differentiate floating point DSP processor and fix point DSP processor. **06**

(b) Explain the effect of coefficient quantization in digital filter with example. **06**

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