Seat No.: _____ Enrolment No._____

GUJARAT TECHNOLOGICAL UNIVERSITY

M.E –Ist SEMESTER–EXAMINATION – JULY- 2012

Subject code: 710422N **Date: 11/07/2012**

Subject Name: Digital Signal Processing and Applications

Time: 2:30 pm – 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) Give answer of following questions. **07**

(i) Show that the accumulator and the moving average systems are both Linear Time-Invariant (LTI) systems. Under which condition moving average system is causal? Check whether the system represented by equation $y[n]-2y[n-1] = (2)^{x[n]}x[n]$ is LTI system?

(ii) Determine the inverse z-transform of

$X(z) = \dfrac{z}{3z^2 - 4z + 1}$, if x[n] is two-sided sequence.

(b) Give answer of following questions. **07**

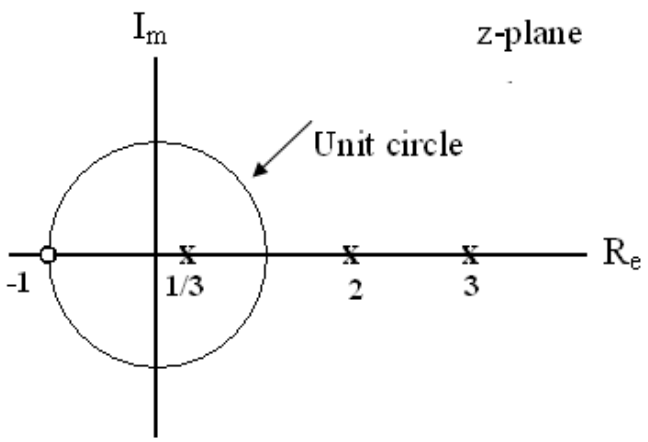
(i) What is the need for anti-aliasing filter prior to down-sampling? The continuous-time signal $x_c(t)=\cos(4000\pi t)$ is sampled with a sampling period T to obtain a discrete-time signal $x[n]=\cos\left(\dfrac{\pi n}{3}\right)$. Determine a choice for T consistent with this information. Is choice for this T unique? Justify your answer.

(ii) Find inverse of the system given by the function

$$H(z) = \frac{z^{-1} - 0.5}{\left(1 - 0.9z^{-1}\right)}, \qquad |z| > 0.9$$

(iii) Determine the input x[n] of the system with impulse response h[n] = {1,2,3,2} and output y[n] = {1,3,7,10,10,7,2}.
↑ ↑

Q.2 (a) Describe the usefulness of the z-transform including its applications in the analysis of discrete-time LTI systems. Consider the z-transform X(z) whose pole-zero plot is as shown in figure below. **07**



(i) Determine the region of convergence of X(z) if it is known that the Fourier transform exists. For this case, determine whether the corresponding sequences x[n] is right sided, left sided, or two sided.

(ii) How many possible two-sided sequences have the pole-zero plot shown in figure.

(iii) Is it possible for pole-zero plot in figure to be associated with a sequence that is both stable and causal? If so, give the appropriate region of convergence. Justify your answer.

(b) When the input to a causal LTI system is **07**

$$x[n] = -\frac{1}{3}\left(\frac{1}{2}\right)^n u[n] - \frac{4}{3}2^n u[-n-1],$$

the z-transform of the output is $Y(z) = \dfrac{1+z^{-1}}{\left(1-z^{-1}\right)\left(1+0.5\,z^{-1}\right)\left(1-2z^{-1}\right)}$,

(i) Find the z-transform of x[n].

(ii) What is the region of convergence of Y(z)?

(iii) Find the impulse response of the system.

(iv) Is the system stable?

OR

(b) Determine the transfer function H(z) of a discrete-time LTI system for which the following information is given. Also determine the value of A and deduce other properties (stability, causality) of the system. **07**

(i) If the input to the system is

$x_1[n] = \left(\frac{1}{6}\right)^n u[n]$ then output is $y_1[n] = \left[A\left(\frac{1}{2}\right)^n + 10\left(\frac{1}{3}\right)^n\right]u[n]$

Where A is real number.

(ii) If the input to the system is $x_2[n]=(-1)^n$ then the output is $y_2[n] = \frac{7}{4}(-1)^n$

Q.3 **(a)** Determine linear convolution of the given sequences x[n]={1,2} & h[n]={2,1} using DFT and IDFT methods. Compare it with circular convolution of x[n] & h[n]. **07**

(b) Give answer of following questions. **07**

(i) Explain circular shift of a sequence property of DFT.

(ii) Give the difference between overlap-save method & overlap-add method.

(iii) The analog signal has a bandwidth of 4 KHz. If we use N-point DFT with $N=2^m$ (m is an integer) to compute the spectrum of the signal with resolution less than or equal to 25 Hz. Determine the minimum number of required samples and the minimum length of the analog signal record.

OR

Q.3 **(a)** Consider a causal linear time-invariant system whose system function is **07**

$$H(z)=\frac{1-\frac{1}{5}z^{-1}}{\left(1-\frac{1}{2}z^{-1}+\frac{1}{3}z^{-2}\right)\left(1+\frac{1}{4}z^{-1}\right)}$$

Draw the signal flow graphs for implementations of the system in each of the following forms:

(i) Direct form I

(ii) Cascade form using first- and second-order direct form II sections.

(iii) Parallel form using first- and second-order direct form II sections.

(b) Compare computational complexity using direct DFT computation and using FFT. **07**

Compute the 8-point DFT of the sequence $x[n]=\begin{cases}1 & 0\le n\le 7\\ 0 & otherwise\end{cases}$

by DIT-FFT algorithm. (Show flow graph)

Q.4 **(a)** State the procedure for the design of IIR filter from analog filter specifications. The system function of a discrete-time system is **07**

$$H(z)=\frac{2}{\left(1-e^{-0.2}z^{-1}\right)}-\frac{1}{\left(1-e^{-0.4}z^{-1}\right)}$$

(i) Assume that this discrete-time filter was designed by the impulse invariance method with $T_d=2$; i.e., $h[n]=2h_c(2n)$, where $h_c(t)$ is real. Find the system function $H_c(s)$ of a continuous-time filter that could have been the basis for the design. Is your answer unique? If not, find another system function $H_c(s.)$

(ii) Assume that H(z) was obtained by the bilinear transform method with $T_d=2$.Find the system function $H_c(s)$ that could have been the basis for the design. Is your answer unique? If not, find another $H_c(s.)$

(b) Compare the commonly used windows of FIR filter design method. **07**

Use Kaiser window method to design a discrete-time filter with generalized linear phase that meets specifications of the following form:

$$\begin{aligned} |H(e^{j\omega})| &\le 0.01, && ,0\le|\omega|\le 0.25\pi,\\ 0.95\le|H(e^{j\omega})| &\le 1.05, && ,0.35\pi\le|\omega|\le 0.6\pi,\\ |H(e^{j\omega})| &\le 0.01, && ,0.65\pi\le|\omega|\le\pi. \end{aligned}$$

Determine the minimum length (M+1) of the impulse response and the value of the Kaiser window parameter β for a filter that meets the preceding specifications. What is the delay of the filter?

OR

Q.4 **(a)** What are the advantages and disadvantages of non-parametric methods for power spectrum estimation? Determine the frequency resolution of the Bartlett, Welch and Blackman-Tukey methods of power spectrum estimation for quality factor Q=10. Assume that overlap in Welch's method is 50%. Given the length of the sample sequence is 1000. **07**

(b) Give answer of following questions. **07**

(i) What is coefficient quantization error? Describe effects of finite word length in FIR digital filters.

(ii) List the characteristic of linear-phase filter.

Q.5 **(a)** Describe sampling rate reduction by an integer factor and corresponding effect in frequency domain. Also explain Polyphase implementation of decimation filters in brief. **07**

(b) Explain application of DSP to radar signal processing. **07**

OR

Q.5 **(a)** Describe modified Harvard Architecture. What are the different stages in pipelining? **07**

(b) Write the desirable features of DSP processors and compare the following: **07**

(i) DSP processor and General purpose microprocessor architecture.

(ii) Fixed point and floating point arithmetic for DSP Processors.

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