Seat No.: ________ *Enrolment No.__________*

GUJARAT TECHNOLOGICAL UNIVERSITY

M. E. - SEMESTER – I • EXAMINATION – WINTER • 2014

Subject code: 2712910 **Date: 12-01-2015**

Subject Name: Discrete Time Signal Processing

Time: 02:30 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 Do as directed : (02 marks each) **14**

(i) State equation for a forward difference system. Will it be causal?
(ii) Define DFT of a discrete time sequence. State its two applications.
(iii) Discuss significance of Interpolation for sampling.
(iv) Evaluate ŭ(n-1)* ŭ(n+1) . Comments on result obtained.
(v) Prove that ŭ(n) = u(n) ï u(n-1).
(vi) State relation between Z-transform and DFT.
(vii) Sketch various tolerance limits to approximate an ideal low pass filter.

Q.2 **(a)** Draw and explain the block diagram of basic generic hardware architecture for a digital signal processor. **07**

(b) Discuss (i) All-pass systems and (ii) Minimum phase system. **07**

OR

(b) For linear phase FIR filters, how constant group and phase delay is achieved? Enlist design techniques for the same. **07**

Q.3 **(a)** A continuous ï time signal is given as : x(t) = 3cos(100´)t. Determine **07**

(i) Nyqist rate to avoid aliasing.
(ii) If the signal is sampled at the rate f_s= 200Hz, What will be discrete time signal x(n) after sampling.?
(iii) If the signal is sampled at the rate f_s= 75Hz, What will be discrete time signal x(n) after sampling.?
(iv) What will be the difference between sampling period and sampling rate.

(b) An LTI system has impulse response $h(n) = 5(-1/2)^n u(n)$. Determine Fourier Transform to find the output of this system when the input is $x(n) = (1/3)^n u(n)$. **07**

OR

Q.3 **(a)** State the sampling theorem. Given $x(t) \xleftrightarrow{FT} X(w)$.For the spectrum of the continuous-time signal, shown in Fig.1, consider the three cases $f_s = 2f_x$; $f_s > 2f_x$ and $f_s < 2f_x$; draw the spectra, indicating aliasing. **07**

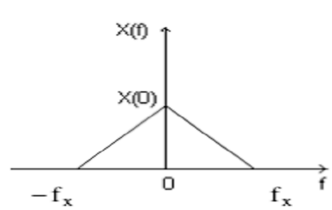

Fig.1

(b) Determine the Inverse Fourier transform for the first order recursive filter $H(w) = (1- ae^{\text{ï} jw})^{-1}$. State result in terms of unit step function. Do not use direct formula. **07**

Q.4 **(a)** Find inverse Z ï transform of **07**

$$X(z) = \frac{1}{(1- a z^{-1})} \quad ; \text{For both } |z| > |a| \text{ and } |z| < |a|.$$

(b) Discuss properties of ROC for the Z-transform. **07**

OR

Q.4 **(a)** Describe Kaiser window filter design method for a low-pass filter. **07**

(b) For $H(z) = (1- 2z^{-1})(1-4z^{-1}) / z(1-0.5z^{-1})$, sketch Direct form -II and its transposed realization. **07**

Q.5 **(a)** Perform circular convolution of the two sequences; $x_1(n) = \{\underline{\mathbf{2}},1,2,1\}$ and $x_2(n) = \{\underline{\mathbf{1}},2,3,4\}$. **07**

(b) Describe Implementation of a DSP A algorithm. **07**

OR

Q.5 **(a)** Explain DIT- FFT Algorithm using signal flow graphs for N=4. **07**

(b) For sequences $x(n) = \{ \mathbf{1}, 0.5\}$ and $h(n) =\{ \mathbf{0.5} , 1\}$, obtain linear convolution using DFT. **07**

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