Seat No.	

T.E. (Computer Engineering) (Semester – I) Examination, 2013
DIGITAL SIGNAL PROCESSING
(2008 Course)

Time : 3 Hours   Max. Marks : 100

Instructions : *1) Answer **three** questions from **each** Section.*
*2) Answers to the **two** Sections should be written in **separate** answer books.*
*3) Figures to the **right** indicate **full** marks.*
*4) Assume suitable data, **if necessary**.*

SECTION – I

1. a) With example explain the time scaling and time reversal operations performed on a discrete time signal. **6**

b) Define ∂ (n) and u(n). Prove that $u(n) = \sum_{k=0}^{\infty} \partial(n-k)$. **6**

c) Explain ADC conversion process. **6**

OR

2. a) Define Nyquist rate. For an analog signal, $x_a(t) = 3\cos 50\pi t + 10\sin 300\pi t - \cos 100\pi t$. Calculate Nyquist rate. **6**

b) Calculate linear convolution of a sequences $x(n) = (\frac{1}{2})^n u(n)$ and $h(n) = (\frac{1}{4})^n u(n)$. **6**

c) Prove with example the operations of folding and time delaying (or advancing) a signal is not commutative. **6**

3. a) State and prove different properties of twiddle factor. **6**

b) State and prove convolution property of DTFT. Determine convolution of the sequences $x_1(n) = x_2(n) = \{1, \underset{\uparrow}{1}, 1\}$ **10**

OR

4. a) State and prove time reversal property of DFT. **6**

b) Determine DTFT of following sequences :

i) $x(n) = (-1)^n u(n)$

ii) $x(n) = (\cos\omega_0 n)\, u(n)$. **10**

5. a) Determine X(K) using DIT FFT algorithm for a sequence x(n) = {4 2 0 2 6 4 2 6}. **8**

b) State and prove :

i) Initial Value Theorem of ZT

ii) Final Value Theorem of ZT **8**

OR

6. a) Obtain z transform of signal $x(n) = -a^n u(-n-1)$. Specify ROC and find out another sequence having same ZT. **8**

b) Derive the first stage of DIF FFT algorithm. Draw the basic butterfly structure of the same. **8**

SECTION – II

7. a) Determine H(z) and draw a pole zero plot for a system

$y(n) + 3y(n-1) + 2y(n-2) = 2x(n) - x(n-1)$ **8**

b) Determine the output y(n) of a system with impulse response $h(n) = (0.5)^n u(n)$ to input signal $x(n) = 2^n u(n)$. **8**

OR

8. a) The system function of a causal LTI system is $H(z) = \frac{z-a^{-1}}{z-a}$ where a is real.

Determine the value of 'a' for which the system is stable. **6**

b) Determine impulse response of a causal system $y(n) - y(n-1) = x(n) + x(n-1)$. **6**

c) Define all pole system and all zero system. **4**

9. a) Explain Gibbs phenomenon observed in FIR filter design. State the desirable features of window functions. **8**

b) Find out the order of filter for the following specifications : **8**

$A_p = 1$ dB $\quad w_p = 0.2\Pi$

$A_s = 15$ dB $\quad w_p = 0.3\Pi$ and $T = 1$

OR

10. a) Explain the mapping of s-plane to z-plane of IIR filter design by using impulse invariance method. How stable analog filter is converted into stable digital filter ? **8**

b) Calculate the filter coefficient if the window function is **8**

$$w(n) = 1 \quad \text{for } 0 \le n \le 4$$
$$= 0 \quad \text{otherwise}$$

and desired frequency response is $H_d(w) = \begin{cases} e^{-j2\omega} & -\frac{\pi}{4} \le \omega \le \frac{\pi}{4} \\ 0 & \text{otherwise} \end{cases}$.

11. a) Compare the DSP processors and general purpose processor. **8**

b) Obtain and realize direct form-II IIR filter structure for a system **10**

$$H(z) = \frac{z^2 + \frac{1}{3}z}{z^2 - \frac{3}{4}z + \frac{1}{8}}$$

What is the advantage of this form over Direct Form – I structure ?

OR

12. a) Describe FIR filter by means of system function H(z). Explain how it is realized for direct and cascade form. **10**

b) What are the different features of ADSP 21xx processor ? What is the use of DAG1 and DAG2 ? **8**

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