(4)

Total No of Questions: [12] **SEAT NO. :**

[Total No. of Pages : 3]

T.E. 2008 (Electronics)

Discrete Time Signal Processing

(Semester - II)

Time: 3 Hours *Max. Marks : 100*

Instructions to the candidates:

1) *Answers to the two sections should be written in separate answer books.*
2) *Answer any three questions from each section.*
3) *Neat diagrams must be drawn wherever necessary.*
4) *Figures to the right side indicate full marks.*
5) *Use of Calculator is allowed.*
6) *Assume Suitable data if necessary*

SECTION I

Q1)	a)	Give the advantages and disadvantages of digital signal processing over analog signal processing.	[6]
	b)	Determine the zero-input response of the system described by the homogeneous second order difference equation: $y(n)-3y(n-1)-4y(n-2)=0$	[6]
	c)	Write a short note on BIBO stability of Linear Time Invariant (LTI) system.	[6]
		OR	
Q2)	a)	A digital communication link carries binary coded words representing samples of an input signal, $x(t) = 3\cos 600\pi t + 2\cos 800\pi t$. The link is operated at 10,000 bit/sec and each input sample is quantized into 1024 different voltage levels. Determine: i. what is the sampling frequency and folding frequency? ii. What is the nyquist rate for the same signal? iii. What is the resolution? What is the use of anti-aliasing filter in DSP system?	[9]
	b)	Why realization process is required to design digital filter? Explain direct form-I and direct form-II and obtain the direct form-II for LTI system described by $y(n) = -0.1y(n-1) + 0.72\ y(n-2) + 0.7\ x(n) - 0.252\, x(n-2)$.	[9]
Q3)	a)	Explain the difference between DTFT and DFT? Explain the cyclic property of twiddle factor.	[8]
	b)	Compute the circular convolution for following sequence and sketch the final response. $x1=\{1,1,1,1,0,0,0,0\}$ and $x2(n) = \sin(3\pi n/8) : 0 \le n \le 7$	[8]
		OR	
Q4)	a)	Write briefly about overlap save and overlap add methods; state the difference between the same.	[8]

	b)	Given, $x(n)=2^n$ and $N=8$, find $X(k)$ using DIT FFT algorithm.	[8]
Q5)	a)	What are the properties of Region of Convergence (ROC)?	[4]
	b)	State and prove the time scaling property of z-transform.	[4]
	c)	A LTI system is described by equation, $2y(n)+3y(n-1)+y(n-2)=u(n)+u(n-1)-u(n-2)$; find the response when initial conditions are given by, $y(-1)=2$, $y(-2)=-1$ and when unit step is applied at the input.	[8]
		OR	
Q6)	a)	State the initial value theorem and final value theorem. And find initial and final values of the function $X(z)=\frac{2z^2+0.25}{(z+0.25)(z-1)}$	[8]
	b)	A causal LTI system has transfer function: $H(z)=\frac{(1-0.5z^{-1})(1-z^{-1})}{(1+0.2z^{-1})(1+0.8z^{-1})(1-0.8z^{-1})}$ Give the ROC condition, Show pole zero diagram of system and find the response of the system, comment on stability.	[8]
		SECTION II	
Q7)	a)	Explain in detail warping effect and its cause.	[4]
	b)	Explain, how S-Plane is mapped to z-plane in impulse invariance method.	[4]
	c)	The system transfer function of analog filter is given by, $H(s)=\frac{(s+0.1)}{(s+0.1)^2+16}$ find the system function of digital filter using BLT which is resonant a $\omega r=\pi/2$	[8]
		OR	
Q8)	a)	Distinguish between FIR and IIR filter.	[4]
	b)	Write a short note on frequency sampling structure.	[4]
	c)	Design a chebyshev analog filter with maximum pass band attenuation of 2.5dB at $\Omega_p=20rad/\sec$ and stop band attenuation of 30 dB at $\Omega_s=50rad/\sec$	[8]
Q9)	a)	Write a note on sampling rate conversion by rational factor I/D.	[8]
	b)	A signal $x(n)$ at a sampling frequency of 2.048 KHz is to be decimated by factor of 32 to yield a signal at sampling frequency of 64Hz. The signal band of interest extends from 0 to 30 Hz, the antialiasing should satisfy the following specifications. Band pass Deviation: 0.01dB Stop band Deviation: 80dB Pass band: 0-30 Hz Stop band 32-64 Hz The signal components in the range from 30 to 32 Hz should be	[8]

		protected from aliasing, design suitable one stage decimator.	
		OR	
Q10)	a)	What is the principle of interpolation? Derive the expression for interpolated signal at the output.	[8]
	b)	What is Multirate signal processing? Design a Multirate LPF for the following specifications, Pass Band: 0 to 40 Hz Stop Band: 50 to 250 Hz $\delta_p = 0.01$, $\delta_s = 0.001, F_s = 500 Hz$	[8]
Q11)	a)	Explain the applications of DSP for DC motor control, AC phase control and proportional controller.	[9]
	b)	Give the comparison of Microprocessor and DSP in terms of their parameters.	[9]
		OR	
Q12)	a)	Give the different addressing formats of DSP processor.	[9]
	b)	Explain important features of TMS320C28XX DSP processor.	[9]