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

[Total No. of Pages : *4*]

T.E. 2008 (Electronics & Telecommunications)

Digital 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) ***Neat diagrams must be drawn wherever necessary.***
3) ***Figures to the right side indicate full marks.***
4) ***Use of Calculator is allowed.***
5) ***Assume Suitable data if necessary***

SECTION I

Q1)	a)	State any four advantages of Digital Signal Processing over Analog Signal Processing.	[4]
	b)	Obtain the direct form I, direct form II realization of the following system $y(n) - 0.2y(n-1) + 0.3y(n-2) = x(n) + 3.6x(n-1) + 0.6x(n-2)$	[8]
	c)	Determine the impulse response h(n) for the system described by the second –order difference equation $y(n) - 3y(n-1) - 4y(n-2) = x(n) + 2x(n-1)$	[6]
		OR	
Q2)	a)	Perform the convolution on the following sequence i) $x(n) = \{3\ 2\ 4\ 1\}$, $h(n) = (1\ 2\ 1\ 2)$ ii) $x(n) = a^n\ u(n)$, $h(n) = b^n\ u(n)$ if $a = b$	[8]
	b)	Comment on stability of Linear Time –Invariant systems	[6]
	c)	The impulse response of LTI system is $h(n) = \{1, 2, 1, -1\}$ Determine the response of the system to the input signal $x(n) = \{1, 2, 3, 1\}$	[4]

Q3)	a)	State and prove the following properties of Z transform i) Convolution of two sequences ii) Differentiation in Z domain	[6]
	b)	State and prove the relationship between Z transform and DFT	[2]
	c)	Determine the Z transform and sketch the ROC of the following signals i) $x(n) = -a^n u(-n-1)$ ii) $x(n) = a^n u(n) + b^n u(-n-1)$	[8]
		OR	
Q4)	a)	A linear time –invariant system is characterized by the system function $H(z) = \dfrac{3 - 4z^{-1}}{1 - 3.5z^{-1} + 1.5z^{-2}}$ Specify the ROC OF H(z) and determine h(n) for the following conditions: i)The system is stable ii) The system is causal iii)The system is anticausal	[8]
	b)	Determine the Inverse Z transform the following signals i) $H(z) = \dfrac{1}{1 - 1.5z^{-1} + 0.5z^{-2}}$ ROC : $\lvert z\rvert > 1$ ii) $X(z) = \dfrac{1}{(1+z^{-1})(1-z^{-1})^2}$ for the causal signal	[8]
Q5)	a)	State and prove any four properties of DFT	[8]
	b)	Perform the circular convolution of the following sequences $x1(n) = \{1, 2, 3, 4\}$ $x2(n) = \{2, 1, 2, 1\}$	[4]

	c)	Compute four point DFT of the following sequence x (n) ={ 1, 2 , 3, 4 }	[4]
		OR	
Q6)	a)	Find the DTFT of the following sequence of length L. x (n) = A for 0 < n < L-1 = 0 Otherwise	[8]
	b)	Compute the eight point DIT-FFT of the following sequence $x(n) = \begin{cases} 1 & 0 < n <= 7 \\ 0 & \text{otherwise} \end{cases}$	[8]
		SECTION II	
Q7)	a)	Use frequency sampling method to design a lowpass filter to meet the following specifications. N = 9. Sampling frequency = 18000 samples/sec. Passband = 0-5 KHz	[10]
	b)	Show that the impulse response coefficients of a linear phase FIR filter with positive symmetry, for N even, is given by $h(n) = \frac{1}{N}\left[\sum_{k=1}^{\frac{N}{2}-1} 2\|H(k)\|cos[2\pi k(n-\alpha)/N] + H(0)\right]$ where α=(N-1)/2 and H(k) are the samples of the frequency response of the filter taken in the frequency range of (0 – 2π).	[8]
		OR	
Q8)	a)	Design a digital low pass filter with a 3- db cutoff frequency of ω_c =0.2π by applying the bilinear transformation to the analog butterworth filter $H_a(s) = \frac{1}{1+s/\Omega_c}$	[4]
	b)	Show that the bilinear transformation maps jΩ-axis in the s-plane onto unit circle in z-	[4]

		plane, and maps the left half s-plane inside the unit circle in z-plane.	
	c)	Design a digital low-pass filter to meet the following specifications. Passband cutoff frequency = π/2 Stopband cutoff frequency = 3π/4 Minimum passband gain= 0.9 Maximum stopband gain= 0.2 Use Butterworth approximation and Bilinear transformation.	[10]
Q9)	a)	Explain sampling rate conversion by a non-integer factor	[8]
	b)	What is the need of antialiasing filter prior to down sampling and anti-imaging filter after up sampling a signal?	[8]
		OR	
Q10)	a)	What is the need of polyphase interpolation? Explain in detail polyphase interpolator.	[8]
	b)	Explain application of DAC in compact disc Hi-Fi systems.	[8]
Q11)	a)	Explain the desirable architectural features for selecting a digital signal processor.	[8]
	b)	Write short note on i) Pipelining ii) MAC Unit	[8]
		OR	
Q12)	a)	Explain five important salient features of TMS 320C6713 digital signal processor and draw its functional block diagram.	[8]
	b)	Write short note on i) Harvard Architecture ii) Barrel Shifter	[8]