Q. Code: 846548

Reg. No.

B.E. / B.TECH. DEGREE EXAMINATIONS, DEC 2019

Fifth Semester

EC16502 – PRINCIPLES OF DIGITAL SIGNAL PROCESSING

(Electronics and Communication Engineering)

(Regulation 2016)

Time: Three Hours **Maximum : 100 Marks**

Answer **ALL** questions

PART A - (10 X 2 = 20 Marks)

		CO	RBT
1.	How many multiplications and additions are required to compute N point DFT using radix 2 FFT?	**1**	**R**
2.	State the convolution property of DFT.	**1**	**U**
3.	What is meant by Gibbs Phenomenon?	**2**	**U**
4.	Draw the direct form structure of FIR filter.	**2**	**R**
5.	What are the properties of Chebyshev filters?	**2**	**U**
6.	Why do we go for analog approximation to design a digital filter?	**2**	**AN**
7.	What are the methods used to prevent overflow?	**3**	**R**
8.	Compare fixed point and floating point representations.	**3**	**AN**
9.	What is anti imaging filter?	**4**	**R**
10.	List the applications of multi rate signal processing.	**4**	**R**

PART B - (5 X16 = 80 Marks)

11.	(a)	(i)	Compute 8-point DFT of the sequence x[n]={2,1,2,1,1,2,1,2} using Decimation in Frequency (DIF) FFT algorithm.	**(12)**	**1**	**AP**
		(ii)	Find the inverse DFT of Y(K)={1,0,1,0}	**(4)**	**1**	**AP**
			(OR)			
	(b)	(i)	Perform the linear convolution of the finite duration sequences h(n) = {2,1,-1} and x(n) = {1,2,3,-1,-2,-3,4,5,6} using overlap add method.	**(10)**	**1**	**AP**
		(ii)	Consider two sequences x[n] = cos(nπ/2) and h[n] = 2n. Determine the output sequence y[n] by circular convolution using concentric circle method. Assume N = 4	**(6)**	**1**	**AP**

12.	(a)		Design a linear phase FIR High Pass filter with a cut off frequency of $\omega_c = 0.8\pi$ rad/sample by taking 7samples of Hamming window sequence.	**(16)**	**2**	**AP**
			(OR)			
	(b)		Design a linear phase FIR low pass filter with a cut-off frequency of $(\pi/4)$ rad/sample by taking 9 samples using frequency sampling technique.	**(16)**	**2**	**AP**
13.	(a)	(i)	Realize the difference equation using Direct form-I, Direct form-II and Cascade realization structure. $y(n) = (3/4)\, y(n-1) - (1/8)\, y(n-2) + x(n) + (1/3)\, x(n-1)$.	**(12)**	**2**	**AP**
		(ii)	Using Impulse invariant technique, obtain H(z) Given the analog transfer function $H(s) = \frac{2}{(s+1)(s+2)}$ and T=1 sec.	**(4)**	**2**	**AP**
			(OR)			
	(b)		Design a digital butterworth IIR filter using bilinear transformation method satisfying the constraints. Assume T=0.5 sec $0.707 \le \lvert H(e^{j\omega})\rvert \le 1.0$; for $0 \le \omega \le 0.45\pi$ $\lvert H(e^{j\omega})\rvert \le 0.2$; for $0.65\pi \le \omega \le \pi$	**(16)**	**2**	**AP**
14.	(a)	(i)	Derive the quantization input noise power and determine the signal to noise ratio of the system.	**(8)**	**3**	**AP**
		(ii)	State the need for scaling and derive the scaling factor for a second order IIR filter.	**(8)**	**3**	**U**
			(OR)			
	(b)		Explain the characteristics of a limit cycle oscillation with respect to the system described by the difference equation $y(n)=0.95y(n-1)+x(n)$ when the product is quantized to 5 bits including sign bit by rounding, The system is excited by an input $x[n]=0.75$ for $n=0$ and $x[n]=0$ for $n\neq0$. Also, determine the dead band of the filter.	**(16)**	**3**	**AP**
15.	(a)		How does the sampling rate decrease by an integer factor D? Derive the input-output relationship in both time and frequency domain.	**(16)**	**4**	**AP**
			(OR)			
	(b)	(i)	Explain how DSP can be used for sub band coding of signals.	**(8)**	**4**	**U**
		(ii)	Explain the application of adaptive filtering in equalization.	**(8)**	**4**	**U**