Student Exam No._____

GANPAT UNIVERSITY

B. Tech. Semester: VII Electronics and Communication Engineering
Regular Examination November-December 2014

2EC702: Digital Signal Processing

Time: 3 Hours **Total Marks: 70**

Instructions:

1. Attempt all questions.
2. Answers to the two sections must be written in separate answer books.
3. Figures to the right indicate full marks.
4. Assume suitable data, if necessary.

SECTION-I

1 (A) Obtain the following structures for given transfer function **12**
(i) Direct form - I (ii) Direct form - II (ii) Cascade form (iii) Parallel Form

$$H(Z) = \frac{\left(1-\frac{1}{2}Z^{-1}\right)}{\left(1-\frac{1}{4}Z^{-1}\right)\left(1+\frac{1}{4}Z^{-1}\right)}$$

OR

1 (A) Obtain direct form structure of given FIR system **2**
$y(n) = x(n) + 2x(n-1) + 3x(n-2) + 4x(n-3) + 5x(n-4)$

(B) Compare the digital filter with analog filter. **4**

(C) Draw signal flow graph for Direct-I and Direct-II structure of the given system function: **6**

$$H(Z) = \frac{1-\frac{7}{6}Z^{-1}+\frac{1}{6}Z^{-2}}{1+Z^{-1}+\frac{1}{2}Z^{-2}}$$

2 (A) Normalized lowpass filter with a cutoff frequency of 1 rad/sec is given as $Hp(s)=\frac{1}{s+1}$. Design digital IIR Highpass filter using Bilinear transformation method with a cutoff frequency of 20 Hz and a sampling rate of 100Hz. **5**

(B) A low pass filter is designed with the following desired frequency response specifications **6**

$$H(e^{j\omega})=\begin{cases} e^{-3j\omega} & ; -\frac{\pi}{2} \le \omega \le \frac{\pi}{2} \\ 0 & ; \ otherwise \end{cases}$$

Determine the filter coefficient and transfer function if the window function is defined as

$$W(n)=\begin{cases} 1 & ; 0 \le n \le 6 \\ 0 & ; \ otherwise \end{cases}$$

OR

2 (A) Determine filter transfer function H(z) using the impulse invariant method if the sampling rate = 10Hz for the Laplace transfer function $H(s) = \frac{2}{s+2}$. **5**

(B) Design a Bandpass FIR filter for the following specification; sampling frequency 4000Hz, lower cutoff frequency 0.6KHz and higher cut off frequency 0.8KHz. Order of filter = 6. **6**

3 (A) Obtain and draw the structure of Linear phase FIR filter. **6**

(B) Find the impulse response of a digital system with following input output relation. Is it an IIR or FIR filter? **4**
(i) $y(n) = \frac{1}{2}x(n) + \frac{1}{2}x(n-1)$ (ii) $y(n) = x(n) + y(n-1)$

(C) What are the disadvantage of FIR and IIR filter? **2**

SECTION-II

4	(A)	Explain the cyclic property of Twiddle factor.	4
	(B)	Find the IDFT of X(K)={4,2,0,2}	4
	(C)	Find the linear convolution of the following sequences using DFT x(n)={1,2,1} h(n)={2,0,1}	4
		OR	
4	(A)	Explain the Periodicity and time shifting properties of DFT	4
	(B)	Find the 4-point DFT of x(n)={1,-2,3,2}	4
	(C)	Find the linear convolution of the following sequences using DFT x(n)={2,3,4} h(n)={3,7,0,5}	4
5	(A)	Why the result of linear convolution and circular convolution is not same? How to obtain same result from linear and circular convolution.	3
	(B)	Compute the DFT of the following sequence using DIT FFT algorithm X(n)={0,1,2,3,0,0,0,0}	6
	(C)	How the mathematical complexity is reduced in DIT FFT algorithm?	2
		OR	
5	(A)	Implement the decimation-in-Time FFT algorithm of N-point DFT where N=8.Aslo explain the steps involved in this algorithm.	6
	(B)	Compute the DFT of the following sequence using DIF FFT algorithm X(n)={0,1,2,3,0,0,0,0}	5
6	(A)	Explain the Multiplier-accumulator (MAC) unit for DSP processors and how we can control the overflow and underflow?	6
	(B)	Explain Impulse Invariant Method to design Digital IIR filter.	4
	(C)	Convert following point in the s-plane to z-plan using bilinear transformation. Assuming sampling period T=2 seconds. (i) s = 1 - j (ii) s = - j	2

END OF PAPER