Student Exam No.________

GANPAT UNIVERSITY

B. Tech. Semester: VII Electronics and Communication Engineering
Regular Examination November-December 2013
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) What are advantages of Digital filter over Analog Filter? 3

(B) Obtain direct form-II for Given a second order transfer function 4

$$H(z) = \frac{0.5(1 - z^{-2})}{1 + 1.3z^{-1} + 0.36z^{-2}}$$

(C) Obtain the parallel form via the first order sections of given a second order transfer function 5

$$H(z) = \frac{1 - 0.9z^{-1} - 0.1z^{-2}}{1 + 0.3z^{-1} - 0.04z^{-2}}$$

OR

1 (A) Obtain and draw the structure of Linear phase FIR filter having order of filter (M) is even. 3

(B) Obtain signal flow graph for Direct-II structure of the given system function: 4

$$H(Z) = \frac{1 + 2Z^{-1}}{1 - 1.5Z^{-1} + 0.9Z^{-2}}$$

(C) Obtain parallel form for given a second order transfer function $H(z) = \frac{0.5(1-z^{-2})}{1+1.3z^{-1}+0.36z^{-2}}$ 5

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

$H(e^{j\omega}) = \begin{cases} e^{-2j\omega} & ; -\frac{\pi}{4} \le \omega \le \frac{\pi}{4} \\ 0 & ; \ otherwise \end{cases}$ Determine the filter coefficient and transfer function if the window function is defined as $W(n) = \begin{cases} 1 & ; \ 0 \le n \le 4 \\ 0 & ; \ otherwise \end{cases}$

(B) Design a Highpass FIR filter using Fourier transform method for the following specifications. Sampling frequency = 2KHz, Cut off frequency = 0.5KHz, Order of filter = 6. 5

OR

2 (A) Calculate the filter coefficient for 5 tap (filter coefficient) FIR Bandpass filter with lower cutoff frequency =2000Hz, Higher cutoff frequency =2400Hz and Sampling frequency = 8000Hz. Make use of blackman window. 6

(B) Derive the impulse response formula of FIR Kaiser Window used for FIR lowpass filter design with the following specifications: $\omega_p=0.35\pi$, $\omega_s=0.5\pi$, $\delta_1=\delta_2=\delta = 0.021$ 5

3 (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) The normalized low pass filter with a cut off frequency of 1 rad/sec is given as $H_P(s) = \frac{1}{s+1}$. Use the given $H_p(s)$ and the Bilinear transformation method to design a corresponding digital IIR highpass filter with cutoff frequency of 15 Hz and a sampling rate of 90Hz. 5

(C) Briefly explain mapping between the s-palne and the z-plane by the bilinear transformation. 2

SECTION-II

4 (A) Explain following properties of DFT. 4
(i) Time reversal (ii) Periodicity

(B) Find the circular convolution of the following sequence using graphical method. 4
x(n) = {0, 1, 2, 3} ; h(n) = { 2, 1, 1, 2}

(C) How the computational complexity for computing DFT will reduce using radix-2 DIT FFT algorithm? 4

OR

4 (A) Explain Overlap save method for linear filtering of long data sequence. 2

(B) Find the linear convolution of the following sequences using DFT. 4
x(n) = { 1, 2, 1} ; h(n) = { 2, 0, 1}

(C) Calculate the 8 point DFT of x(n) = {1, 2, 1, 2} 6

5 (A) Explain Radix-2 DIT FFT algorithm. 6

(B) Determine the 8 point DFT of the following sequence using radix-2 DIF FFT algorithm. 5
x(n) = { -1, 0, 2, 0, -4, 0, 2, 0}

OR

5 (A) Determine the 8 point DFT of the following sequence using radix-2 DIT FFT algorithm. 6
x(n) = { -1, 0, 2, 0, -4, 0, 2, 0}

(B) Prove following relation of twiddle factor. 5
(i) $W_N^K = W_N^{K+N}$ (ii) $W_N^2 = W_{N/2}$

6 (A) Explain the Multiplier-accumulator (MAC) unit for DSP processors and how we can control the overflow and underflow? 6

(B) Explain different window function of FIR filter design and Compare the main lobe width, peak of side lobe and Minimum stopband attenuation for rectangular, Bartlett, hanning, hamming and blackman window function. 6

END OF PAPER