Registration No. : | | | | | | | | | | |

Total number of printed pages – 3

B. Tech
PCEC 4304

Sixth Semester Regular / Back Examination – 2015

DIGITAL SIGNAL PROCESSING

BRANCH : EEE

QUESTION CODE : J 283

Full Marks – 70

Time : 3 Hours

*Answer Question No. 1 which is compulsory and any **five** from the rest.*
The figures in the right-hand margin indicate marks.

1. Answer the following questions : 2×10

(a) State and prove State the scaling property of the z transform.

(b) How delay and advance elements are expressed in discrete z-domain ?

(c) What is approximate transition width of main lobe in the rectangular window ? What happens to it if you double the filter length ?

(d) How many real multiplication and real additions are required to compute 16 point DFT using decimation in frequency (DIF) algorithm ?

(e) Find out the magnitude of transfer function of the system whose impulse response (at n = 0) is described as h(n) = {1.1, 0, 2, 1}

(f) Under what conditions the pair of zeros of FIR filter will be complex conjugate ?

(g) How ripples in the pass band of FIR filters can be eliminated ?

(h) What do you mean by frequency warping in digital filter ? How this effect can be eliminated ?

(i) Express Unit step function U(n) in terms of in terms of impulse functions $\delta(n)$.

(j) State and proof linearity property of DFT.

2. (a) Consider the LTI system described by the equation

$$x(n) = a^n u(n) - b^n u(-n-1)$$

What conditions must hold on a and b for Z-transform to exist. 4

(b) Find inverse Z-transform of 6

$$X(z) = \log(1-2z) \qquad |z| > |a|$$

3. (a) Find 4-point IDFT of the signal, X(k) = {1,1,0,1} and sketch magnitude response. 5

(b) The DFT of x(n) is described as X(k) = {1, −1 + 2j, −1, 1+2j,}. Find the DFT of $x^2(n)$. 5

4. (a) Convert the analog filter with the system function 5

$$H(s) = \frac{s+0.1}{(s+0.1)^2+9}$$

Into a digital IIR filter using impulse invariance method.

(b) How frequency wrapping occurs in designing IIR filters using bilinear transformation method ? Explain. 5

5. (a) Derive two conditions that must be fulfilled for FIR filter to behave as linear phase. 5

(b) Consider the casual system 5

$$Y(n) = 0.75y(n-1) - 0.125y(n-2) + x(n) + 0.25x(n-2)$$

Obtain parallel structure of the system.

6. Determine the coefficient of linear phase FIR filter length M=15, which has a symmetric unit sample response and frequency response that satisfies the condition 10

$$H_r\left(\frac{2\pi K}{15}\right) = \begin{cases} 1 & K = 0,1,2,3 \\ 0.4 & K = 4 \\ 0 & K = 5, 6, 7 \end{cases}$$

7. (a) Explain Decimation in time FFT algorithm. 5

(b) Find 4-pont DFT of the discrete signal, X(n) = {0, 1, 2, 1}. using DIT algorithm. 5

8. Write short notes on any **two** of the following : 5×2

(a) The LMS Algorithm

(b) Overlap add method in linear filtering

(c) System Identification

(d) Difference in Structure of FIR and IIR filters.

———