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

Total number of printed pages – 3

B. Tech
PCEC 4304

Sixth Semester Regular / Back Examination – 2015

DIGITAL SIGNAL PROCESSING

BRANCH(S) : EC, ETC

QUESTION CODE : J 134

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) Which standard discrete signal is usually used to analyze discrete time system ? Justify with necessary mathematical expression using Z-transform.

   (b) What is time shifting property of z-transform ?

   (c) Find out the frequency response of a system if the impulse response of the system is $h(n) = u(n + 3)$.

   (d) What is Gibbs phenomenon ?

   (e) How many real multiplication and real additions are required to compute 64 point DFT using direct computation and DIT FFT algorithm ?

   (f) When DFT x (k) of a sequence x (n) is real ?

   (g) Draw the basic structure of 1^{st} order digital FIR filter.

   (h) Why aliasing occurs most of the time when mapping of s-plane to z-plane is done using impulse invariance sampling method ?

   (i) What type of filter best describes the LMS algorithm ? Give its schematic representation.

   (j) What is linear phase characteristics of FIR filter ? What is its importance ?

P.T.O.

2. (a) Find inverse Z-transform of 5

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

(b) Determine the transient and steady state response of the system described by 5

$$x(n) = y(n-1) + 2y(n-2)$$

When a unit step function is applied to the system.

3. (a) Perform the convolution of the following two sequence using DFT properties

$X^1(n) = \{1,1,1,0\}$ 5

$X^2(n) = \{1,1,0,1\}$

(b) Explain how DFT can be used in linear filtering the discrete signal. 5

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

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

Into a digital IIR filter using impulse invariance transformation method. The digital filter is to have resonant frequency of $\frac{\pi}{2}$. 5

(b) Using bilinear transformation method, Design a high pass filter, monotonic in pass band with cutoff frequency of 1000 Hz and down 10 dB at 350 Hz. The sampling frequency is 5000 Hz. 5

5. (a) Consider the casual system 6

$$Y(n) = -0.5y(n-1) - 0.12y(n-2) + 0.7x(n) - 0.252x(n-2)$$

Justify whether the system is FIR or IIR and then obtain a cascade structure of the system.

(b) Find the impulse response of LTI system whose frequency response is described as 4

$$\text{For } H(e^{jw}) = 1 \qquad \text{For } |w| < \Pi/4$$
$$= 0 \qquad \text{otherwise}$$

6. (a) Establish the relation between DFT and Z-transform. 5

(b) What are the differences in structure of FIR and IIR filter ? Explain with suitable example. 5

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

(b) Determine Z-transform of the following signal using properties of z-transform 5

(i) $x(n) = n^2 u(n)$

(ii) $x(n) = a^n u(n+2)$

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

(a) Mapping of S-plane into Z-plane using bilinear transformation Technique

(b) Stability of discrete system in Z-domain

(c) Adaptive Noise Cancelling

(d) FIR filter using windowing technique.