Registration No. :

Total number of printed pages – 2

B. Tech

PCEC 4304 (New)

Sixth Semester (Back) Examination – 2013

DIGITAL SIGNAL PROCESSING

BRANCH : CSE, EC, ETC

QUESTION CODE : B249

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) What is aliasing effect ? How it can be avoided ?

   (b) The following analog signal is sampled at 10,000 samples per second :
   $$x(t) = \sin(2000\, t) + 2\cos(1550\, \pi t)$$
   What is corresponding discrete time signal after sampling ?

   (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 ?

   (e) Draw the basic structure of 1st order digital FIR filter.

   (f) Why IIR filters does not have Linear phase characteristics ?

   (g) State the final value theorem in Z-transform.

   (h) Give the mapping of S-plane to Z-plane using impulse invariance method.

   (i) What is the time shifting property of DFT ?

   (j) Why FIR filters are inherently stable ?

2. (a) Determine Z-transform of the following signal : 4

   (i) $x(n) = nu(n)$

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

   (b) Determine the response of the system 6
   $$y(n) = 0.8\, y(n-1) + 0.2\, Y(n-2) + X(n)$$
   to the input signal $x(n) = \delta(n) - \delta(n-1)$.

3. (a) Explain the design of linear phase FIR filter using windows. 5

(b) Convert the analog filter with the system function

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

Into a digital IIR filter using impulse invariance method. 5

4. 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

$$\mathrm{Hr}\left(\frac{2\pi K}{15}\right) = 1 \quad K = 0,1,2,3$$
$$= 0 \quad K = 4,5,6,7$$

5. (a) Consider the casual system

$$Y(n) = 0.75\,y(n-1) - 0.125\,y(n-2) + x(n) + 0.3\,x(n-1)$$

Obtain direct form-I and form-II structure. 5

(b) Explain how the IIR filter is designed from analog filter using bilinear transformation method. 5

6. (a) Perform the convolution of the following two sequence using Z-transform : 5

$$X1(n) = \{2, 1, 0, 1\}$$
$$X2(n) = \{1, 2, -1, 1)$$

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

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

(b) What is N-point DFT ? Find 4-pont DFT of the discrete signal, $X(n) = \{0, 1, 2, 3\}$. 5

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

(a) Linear phase FIR filter by frequency sampling method

(b) Stability of LTI system

(c) Adaptive Line Enhancer

(d) Minimum Mean Square Error Criterion.