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

Total number of printed pages – 2

B. Tech

PCEC 4304

Sixth Semester Back Examination – 2015

DIGITAL SIGNAL PROCESSING

BRANCH : EEE

QUESTION CODE : M 176

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 delay and advance element ? Write their transfer functions ? What is their importance ?

   (b) State the time shifting property of the z transform.

   (c) What is a Gibbs phenomenon ?

   (d) State the magnitude response of the system described by $y(n) = 0.3Y(n-2) + 0.5X(n-1)$

   (e) How many real multiplication and addition is required for computation of 32-point DFT ?

   (f) Why FIR filters are inherently stable ?

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

   (h) State the limitations of impulse invariance transformation method for realizing IIR filter.

   (i) Why an ideal lowpass digital filter is non-casual ?

   (j) What is adaptive in adaptive filter ? How it is different from classical filter ?

2. (a) Find the Z-transform of the following signal : 6

   (a) $x(n) = \delta(n)$

   (b) $x(n) = nu(-n)$

   (b) Find the inverse Z-transform of the following casual system. Then find its stability region. 4

$$h(z) = \frac{1}{1 - az^{-1} + bz^{-2}}$$

P.T.O.

3. (a) Explain the design procedure for IIR filter using bilinear transformation method. 5

(b) Find the impulse response of LTI system whose frequency response is described as 5

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

4. (a) Establish the relation between ω and Ω using bilinear transformation. And then, bring out a mapping between them. 5

(b) Convert the analog filter with system function 5

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

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

5. (a) Consider the casual system 6

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

Obtain direct form I and form-II structure.

(b) Explain the frequency sampling structure of FIR filter. 4

6. (a) Find the circular convolution of the following two sequence. 5

$X_1(n) = \{1,2,3,4\}$

$X_12(n) = \{2,1,2,1$

(b) Determine the impulse response for the two cascaded LTI system having impulse responses. 5

$h1(n) = 0.5^n u(n)$

$h2(n) = 0.25^n u(n)$

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

(b) What is adaptive filter ? How it can be used as channel equalization ? 5

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

(a) Pole Zero pattern of FIR filter

(b) Overlap add filtering using DFT method

(c) Windows used in designing FIR filter

(d) Use of DFT in linear filtering.
