Registration no:

Total Number of Pages: 02

B.TECH
PCEC4304

6^{th} Semester Regular / Back Examination 2015-16

DIGITAL SIGNAL PROCESSING

BRANCH(S): AEIE, ECE, EEE, EIE, ETC, IEE
Time: 3 Hours
Max Marks: 70
Q.CODE: W105

Answer Question No.1 which is compulsory and any five from the rest.

The figures in the right hand margin indicate marks.

Q1 Answer the following questions: **(2 x 10)**

a) How stability of a system is analysed by plotting pole and Zero of a system in Z-domain?

b) Determine the pole-Zero plot for the signal

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

c) With proper justification show that impulse function can be used as test signal for a DTS system

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 1st 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 is the importance of linear phase in filter design? State the conditions that must be fulfilled for FIR filter to be linear phase?

j) With necessary mathematical expressions, find the location of poles of a finite length causal FIR filter in the Z-plane.

Q2 a) Find inverse Z-transform of **(5)**

$$X(z) = \log(1 + az^{-1}) \qquad |z| > |a|$$

b) Find out the inverse Z transform of the discrete signal x(n) using Cauchy integral theorem. **(5)**

Q3 a) By means of DFT and IDFT determine the response of FIR filter with impulse response **(5)**

$$h(n) = \{1,2,3\} \text{ with input sequence}$$

$$x(n) = \{1,2,2,1\}$$

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

Q4 **(a)** 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 transformation method. The digital filter is to have resonant frequency of $\pi/2$.

(b) Design a single pole low pass digital filter with 3-dB bandwidth of 0.2π, using bilinear transformation applied to the analog filter $H(s) = \frac{\Omega}{s+\Omega}$ Where Ω is the 3-dB bandwidth of an analog filter **(5)**

Q5 **a)** Consider the casual system **(4)**

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

Obtain a cascade structure of the system

b) Find the impulse response of LTI system whose frequency response is described as **(4+2)**

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

Is such LTI system practically realizable? Justify your answer.

Q6 **a)** Establish the relation between DFT and Z-transform **(5)**

b) Compare FIR with IIR filter with suitable example. **(5)**

Q7 **a)** Explain Decimation in time FFT algorithm **(5)**

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

$$x(n) = n^3 u(n)$$

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

Q8 Write short notes on (Any two) **(5 x 2)**

a) Mapping of S-plane into Z-plane using impulse invariance technique
b) Symmetric and asymmetric condition of FIR filter
c) Adaptive filter
d) FIR filter using windowing technique