Register No. | | | | | | | | |

Anna University End Semester Examinations Nov 2011
B.E. (Full Time) Arrear Examination
Electronics and Communication Engineering
Regulations 2004
EC373 – Digital Signal Processing
V Semester

18

Time: 3 Hours **Max.marks: 100**

Answer All Questions

PART A (10x2=20marks)

1. Test whether the system y(n)= Ax(n)+B is causal or non-causal.
2. Draw the butterfly structure for decimation in frequency FFT algorithm.
3. What is the mapping procedure between s-plane and z-plane in the bilinear transformation method?
4. Write short note on prewarping.
5. Realize the system function $H(z)=\frac{1}{2}+\frac{1}{3}z^{-1}+z^{-2}+\frac{1}{4}z^{-3}+z^{-4}+\frac{1}{3}z^{-5}+\frac{1}{2}z^{-6}$ with minimum number of multipliers.
6. Distinguish between FIR and IIR filters.
7. Why is rounding preferred to truncation in realizing digital filter?
8. What is meant by saturation arithmetic? What is its disadvantage?
9. What is the need for anti-imaging filter?
10. How multistage sampling rate converters are implemented?

PART B (16x5 = 80 marks)

11. Design a linear phase FIR filter using a hamming window with N=7.

$$H_d(e^{jw}) = e^{-j3w} \quad -\frac{\pi}{4} \le w \le \frac{\pi}{4}$$

$$0 \qquad \frac{\pi}{4} \le |w| \le \pi$$

12a.Compute the DFT of the sequence x (n) = {1, 2, 3, 4, 4, 3, 2, 1} using decimation in time FFT algorithm.

(OR)

12b. Find the output y(n) of a filter whose impulse response is h(n)={1,1,1} and input signal x(n)={ 3,-1, 0, 1, 3, 2, 0, 1, 2, 1} using overlap add method.

13a. Design a Butterworth filter using the impulse invariance method for the following specifications

$$0.8 \le |H(e^{jw})| \le 1 \quad 0 \le w \le 0.2\pi$$

$$|H(e^{jw})| \le 0.2 \quad 0.6\pi \le w \le \pi$$

(OR)

13b (i) Obtain the direct form I , direct form II, and cascade realization of the system

$$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2) \qquad (12)$$

(ii)Apply bilinear transformation to $H(s) = \frac{2}{(s+1)(s+2)}$ with T=1 sec and find H(z) (4)

14a. Explain the characteristics of a limit cycle oscillation with respect to the system described by the difference equation $y(n) = 0.95y(n-1) + x(n)$. Determine the dead band of the filter. Use 4 bit sign-magnitude representation (excluding sign bit).

(OR)

14b. Find the steady state variance of the noise in the output due to quantization of input for the first order filter $y(n) = ay(n-1) + x(n)$

15a. What is a down sampler? Explain.and determine the input-output relation of the spectrum of the down sampled signal

(OR)

15b.Explain subband coding of speech signals with a neat block diagram.