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B.E / B.Tech (Full-Time) Degree End Semester Examinations, Nov/Dec 2012
Anna University, Chennai

Computer Science and Engineering
Sixth Semester

CS9351 – Digital Signal Processing
(Regulations 2008)

Time: 3Hrs | Max Marks: 100

Answer ALL Questions

Part A – (10 * 2 = 20 marks)

1. Represent unit step, unit impulse and unit-ramp signals graphically and state their expression
2. Prove that multiplication in Z-domain is the same as convolution in time domain.
3. Explain overlap-add method of linear filtering.
4. What is Discrete Cosine Transform? When do you prefer it?
5. State the advantages of IIR filters.
6. Convert the following analog filter to digital filter using approximation of derivatives with T = 1 sec.

$$H(s) = \frac{1}{(s+1)}$$

7. What is Gibb's oscillations?
8. Explain limit cycle oscillations
9. List the issues in recognizing speech
10. Draw the block diagram of an echo cancellation system using adaptive filter and explain.

Part B – (5 * 16 = 80 marks)

11. i. Check whether the following systems are linear, time invariant, causal and stable

a. $y(n) = \sin(n)\, e^{n}\, u(n)$ (4)

b. $y(n) = \alpha + \sum_{k=0}^{4} x(n-k)$, α is anon-zero constant (4)

ii. Determine the Z-transform and ROC for the following sequence (4)

$$x(n) = 3^n u(n+2) + (½)^{-n} u(-n)$$

iii. Determine h(n) given the following IIR filter system (4)

$$y(n) - (3/4)\, y(n-1) + (1/8)\, y(n-2) = x(n)$$

12. a i. Compute the FFT using DIF algorithm for the sequence given by

$x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ (16)

(OR)

b. If $x_3(n)$ is the circular convolution of $x_1(n)$ and $x_2(n)$, Determine $x_3(n)$ if

$x_1(n) = \{1, -1, 1, 2\}$ and $x_2(n) = \{1, 2, 3, 4\}$. (8+8)

Verify the computed value of $x_3(n)$ by means of DFT and IDFT.

13. a. Realize the following system using Cascade and Parallel realization and represent them using Direct form II transpose (16)

$$H(z) = \frac{(z+1)(z-2)(z+0.2}{(z-0.3)(z+2)(z+0.1)}$$

(OR)

b. Design a digital low pass Butterworth filter using bilinear transformation technique for the following specification with T = 1 sec. Realize the designed filter using Direct form II structure (12)

Passband gain – 0.9
Passband edge: - 0.3π rad/ sec
Stop band attenuation – 0.2
Stop band edge – 0.5π rad / sec

14. a. Design an FIR low pass filter for the following specification and realize it using Direct Form I structure (16)

$$H_d(\omega) = e^{-j3\omega} \quad |\omega| \le \pi/6$$
$$= 0 \quad (\pi/6) \le |\omega| \le \pi$$

Use Hamming window for terminating the desired frequency response

(OR)

b. i. Design a FIR filter using frequency sampling technique for the following specification (10)

Pass band edge	=	1200 Hz
Stop band edge	=	1800 Hz
Sampling frequency	=	15,000 Hz
Filter Length	=	9

ii. Discuss the effect of finite word length and the errors in a DSP system (6)

15. a. i. Explain the process of decimation and interpolation with a neat block diagram. (10)

ii. Write short notes on Adaptive filters. (6)

(OR)

b. Write short notes on the following

i. Speech Processing (8)

ii. Image Enhancement (8)