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B.E / B.Tech (Full-Time) Degree End Semester Examinations, April/May 2011

Anna University, Chennai

Computer Science and Engineering

Sixth Semester

CS9351 – Digital Signal Processing

(Regulations 2008)

Time: 3Hrs Max Marks: 100

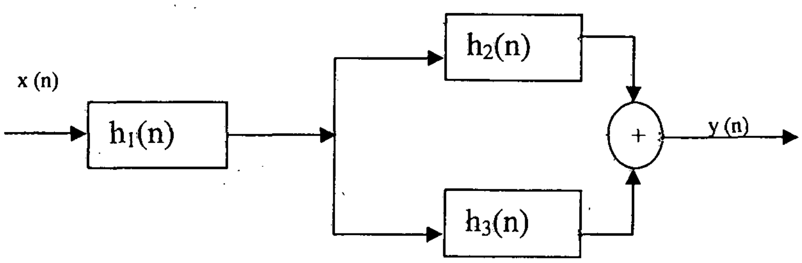
Answer ALL Questions

Part A – (10 * 2 = 20 marks)

1. Represent the following using unit step signal only
   $x(n) = 1, -\infty \leq n \leq \infty$
2. Prove that multiplication in Z-domain is the same as convolution in time domain.
3. Differentiate overlap-add and overlap-save methods of linear filtering?
4. What is Discrete Cosine Transform? When do you prefer it?
5. Compare IIR and FIR 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. Why do we not choose Rectangular window for designing FIR filters?
8. Determine the deadband of the first order filter system given by
   $y(n) = x(n) - 0.8y(n-1)$
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. Determine the response y(n) of the following system given that
   $h_1(n) = (1, -1, 2\}$, $h_2(n) = u(n) - u(n-5)$, $h_3(n) = \delta(n-2)$, $x(n) = \{1, 0, -1\}$ (8)



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 DIT algorithm for the sequence given by

$x(n) = \{1, 2, 3, 4, 4, 3, 2, 1\}$ (12)

ii. Derive the equation to compute FFT using DIT algorithm and explain the savings in computation time (4)

(OR)

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

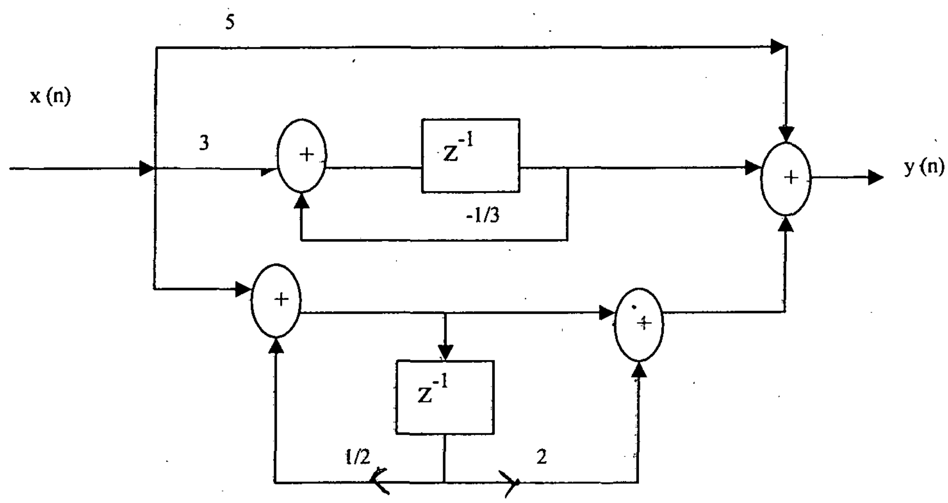
$x_1(n) = \{1, -1, 1, 2\}$ and $x_2(n) = \{1, 2, 3, 4\}$. (8+8)

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

13. a. i. Realize the following system using Cascade and Parallel realization and represent them using Direct form II transpose (8)

$$H(z) = \frac{(z-0.5)(z-2)}{(z-3)(z+2)(z+0.1)}$$

ii. Determine the system function H(z) of the following system (8)



(OR)

b. i. 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

ii. Explain Chebyshev approximation and discuss the procedure to compute Chebyshev poles (4)

14. a. i. Design an FIR low pass filter for the following specification and realize it using Direct Form structure (12)

$$H_d(\omega) = e^{-j3\omega} \quad |\omega| \leq \pi/3$$
$$= 0 \quad (\pi/3) \leq |\omega| \leq \pi$$

Use Hamming window for terminating the desired frequency response

ii. Determine the $h_d(n)$ for an ideal FIR band pass filter with cut-off frequencies ω_{c1} and ω_{c2} (4)

(OR)

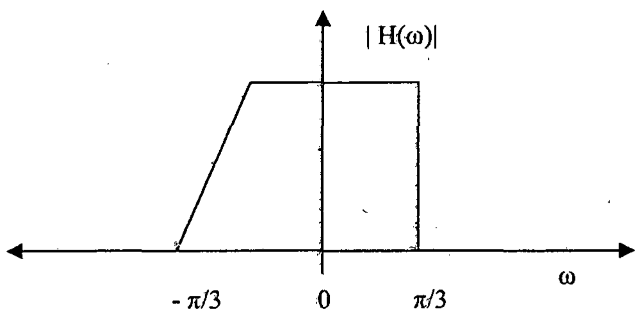
b. i. Design a FIR filter using frequency sampling technique for the following specification (10)

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. Consider the following diagram and show the effect of decimating by a value of 1.5 (6)



(OR)

b. Write short notes on the following

i. Speech Processing (8)

ii. Image Enhancement (8)