ANNA UNIVERSITY COIMBATORE	
B.E. / B.TECH. DEGREE EXAMINATIONS : MAY / JUNE 20)10
REGULATIONS: 2007	
FIFTH SEMESTER	
070290037 - DIGITAL SIGNAL PROCESSING	
(COMMON TO ECE / MEDICAL)	
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DADT A	

ANSWER ALL QUESTIONS

The first five DFT coefficients of a real valued sequence of length 8 is {2,

How many multiplications and additions are required to compute N-point

What are two methods which are used for the sectional convolution?

12.

Marks : 100

 $(20 \times 2 = 40 \text{ MARKS})$

Convert the low-pass filter with system function $H(s) = \frac{\Omega_p}{s + \Omega_p}$ into a

band-pass filter with upper and lower cutoff frequencies Ω_{u} and Ω_{l} respectively.

- 13. State Gibbs phenomenon with respect to window based FIR filter design.
- 14. Find length of FIR filter which has normalized magnitude oscillations of 0.95 in both stop and pass bands and has relative transition width of 0.1.
- 15. The first 4 values of the impulse response of the linear-phase FIR filter are $\{4,-1, 7, 0\}$. If order of the filter is 7, then determine the remaining values.
- 16. Mention various methods available for the design of FIR filter. Also list a few window for the design of FIR filters.
- 17. Identify the various factors, which degrade the performance of the digital filter implementation when finite word length is used.
- 18. What are the results of truncation for positive and negative numbers?
- 19. List out a few addressing modes of the TMS320C50.
- 20. Why do we need pipelining?

PART – B

(5 x 12 = 60 MARKS)

ANSWER ANY FIVE QUESTIONS

21. a. Suppose we have two 4-point sequences x[n] and h[n] as follows

 $x[n] = \cos\left(\frac{\pi n}{2}\right), n = 0, 1, 2, 3$ $h[n] = 2^n, n = 0, 1, 2, 3$

Compute the response y[n] by doing circular convolution of x[n] and h[n].

How can we calculate IDFT using FFT algorithms? What is the main advantage of direct form II realization over direct form I realization? Also compare cascade and parallel realizations.

Write down the equations for forward and inverse DFT.

3+2j, 1-6j, 4, -3}, determine the remaining DFT coefficients.

When sequence is called circularly even?

What is zero padding? What are its uses?

Compare the DIT and DIF, radix-2 FFT.

What is pre-warping? Why is it needed?

DFT using radix-2 FFT?

Find digital transfer function using approximate derivative technique for the analog transfer function H(s) = 1/s + 3. Assume T=0.1 sec

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TIME : 3 Hours

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- 21. b. Compute 4-point IDFT of X(k) = {0,4j,0,-4j} using decimation in frequency fast
 Fourier transformation.
- 22. a. From the first principle derive the decimation in frequency FFT algorithm
 - b. Compute the 8-point DFT of the sequence x(n)≈(1,1,1,1,1,1,1,1) by using 6 decimation in time FFT algorithm.

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- a. Design a linear-phase low-pass FIR filter of order 7 and cutoff frequency 1 rad/sec. Use rectangular window. Also plot the magnitude response of the filter.
 - b. Briefly explain the procedure of designing FIR filter using frequencysampling technique.
- 24. a. From the first principles derive the conditions for achieving linear-phase in 6 symmetric FIR filters.
 - b. Obtain direct form II realization for the system characterized by the 6 difference equation $y[n] + \frac{3}{4}y[n-1] + \frac{1}{8}y[n-2] = x[n] + x[n-1]$
- 25. For the constraints

$$0.8 \le \left| H\left(e^{jw}\right) \right| \le 1, \quad 0 \le w \le 0.2\pi$$
$$\left| H\left(e^{jw}\right) \right| \le 0.2, \qquad 0.6\pi \le w \le \pi$$

with T = 1s determine system function H(z) for a Butterworth filter using bilinear transformation.

- 26. a. With help of suitable mathematical equation briefly explain impulse invariant design technique, which is used to convert analog filters into digital filters. Also, explain the mapping between s-plane and z-plane during this transformation.
 - b. Consider a first order filter with difference equation y[n]= x[n]+0.5y[n-1] assume that the data register length is three bits plus a sign bit. The input x[n]=0.875δ[n]. Explain the limit cycle oscillations in the above filter, if quantization is performed by means of rounding and signed magnitude representation is used.
- 27. With help suitable mathematical equations and diagram briefly explain various number representation and its truncation and rounding errors.
- 28. With neat sketch explain the architecture of TMS320C50.

*****THE END*****