## ANNA UNIVERSITY COIMBATORE

B.E. / B.TECH. DEGREE EXAMINATIONS : MAY / JUNE 2010

REGULATIONS : 2007
FIFTH SEMESTER
070290037 - DIGITAL SIGNAL PROCESSING
(COMMON TO ECE / MEDICAL)
TIME : 3 Hours
PART - A
Max.Marks: 100
( $20 \times 2=40$ MARKS $)$

## ANSWER ALL QUESTIONS

Write down the equations for forward and inverse DFT.
When sequence is called circularly even?
Compare the DIT and DIF,radix-2 FFT.
What is zero padding? What are its uses?
The first five DFT coefficients of a real valued sequence of length 8 is $\{2$,
$3+2 \mathrm{j}, 1-6 \mathrm{j}, 4,-3\}$, determine the remaining DFT coefficients.
6. What are two methods which are used for the sectional convolution? How many multiplications and additions are required to compute N -point DFT using radix-2 FFT?

How can we calculate IDFT using FFT algorithms?
9. What is the main advantage of direct form II realization over direct form I realization? Also compare cascade and parallel realizations.
10. What is pre-warping? Why is it needed?
11. Find digital transfer function using approximate derivative technique for the analog transfer function $H(s)=1 / s+3$. Assume $T=0.1 \mathrm{sec}$

Why do we need pipelining?
PART - B

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(5 \times 12=60 \text { MARKS })
$$

## ANSWER ANY FIVE QUESTIONS

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

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

Compute the response $y[n]$ by doing circular convolution of $x[n]$ and $h[n]$.
21. b. Compute 4 -point IDFT of $X(k)=\{0,4 j, 0,-4 j\}$ using decimation in frequency fast 6 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.
23. a. Design a linear-phase low-pass FIR filter of order 7 and cutoff frequency 1 $\mathrm{rad} / \mathrm{sec}$. 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 $\quad y[n]+\frac{3}{4} y[n-1]+\frac{1}{8} y[n-2]=x[n]+x[n-1]$
25. For the constraints

$$
\begin{array}{ll}
0.8 \leq\left|H\left(e^{j w}\right)\right| \leq 1, & 0 \leq w \leq 0.2 \pi \\
\left|H\left(e^{j w}\right)\right| \leq 0.2, & 0.6 \pi \leq w \leq \pi
\end{array}
$$

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

## *****THE END*****

26. a. With help of suitable mathematical equation briefly explain impuise 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 $y[n]=x[n]+0.5 y[n-1]$ assume that the data register length is three bits plus a sign bit. The input $x[n]=0.875 \delta[n]$. Explain the limit cycle oscillations in the above filter, if quantization is performed by means of rounding and signed magnitude representation is used.

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27. With help suitable mathematical equations and diagram briefly explain various number representation and its truncation and rounding errors.

With neat sketch explain the architecture of TMS320C50

