Fifth Semester B.E. Degree Examination, December 2011 Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: 1. Answer any FIVE full questions, selecting atleast TWO questions each from Part – A and Part - B.

- 2. Use of normalized Butterworth and Chebyshev table is not permitted.
- 1 a. Let $x_p(n)$ be a periodic sequence with fundamental period N. Consider the following DFTs $x_p(n) \xleftarrow{DFT}_N X_1(k)$

 $x_p(n) \xleftarrow{\text{DFT}} X_3(k)$. What is the relationship between $x_1(k)$ and $x_3(k)$? (10 Marks)

- b. Compute the N point DFT of the sequence x(n) = an, $0 \le n \le N-1$. (10 Marks)
- 2 a. Given the 8 point sequence

$$X(n) = \begin{cases} 1, & 0 \le n \le 3 \\ 0, & 4 \le n \le 7 \end{cases}$$

Compute the DFT of the sequence $x_1(n)$ using properties of DFT.

$$X_{I}(n) = \begin{cases} 1, & n = 0 \\ 0, & 1 \le n \le 4 \\ 1, & 5 \le n \le 7 \end{cases}$$
 (08 Marks)

- b. What are the methods used to perform fast convolution? Explain any one method giving all the steps involved to perform fast convolution. (07 Marks)
- c. Given the sequences $x(n) = \cos \frac{\pi n}{2}$ and $h(n) = 2^n$. Compute the 4 point circular convolution. (05 Marks)
- a. How many multiplications and additions are needed for a 64-point sequence using FFT algorithm and DFT using Direct computation? Find i) speed improvement factor
 ii) number of real and complex registers needed. (06 Marks)

b. Prove the symmetry and periodicity property of a twiddle factor. (06 Marks)

- c. A designer has a number of eight point FFT chips. Show explicity how he should interconnect four such chips in order to compute a 32 point DFT. (08 Marks)
- 4 a. Describe Goertzel algorithm. Also Obtain direct form II realization of two pole resonator for computing the DFT. (10 Marks)
 - b. What is chirp signal? What are the applications of chirp Z transform? (04 Marks)
 - c. Let x(n) be the following 8 point sequence.

$$x(n) = \left\{ \frac{1}{\sqrt{2}}, 1, \frac{1}{\sqrt{2}}, 0, -\frac{1}{\sqrt{2}}, -1, \frac{-1}{\sqrt{2}}, 0 \right\}.$$
 Use DIT FFT algorithm to compute DFT of

the above sequence. Also show all intermediate values.

5 a. Define Chebyshev polynomial and list all its properties.

(05 Marks)

- b. Design a Chebyshev analog filter with ripple of 0.5dB in band $|\Omega| \le 1$ and at $\Omega = 3$, amplitude if down by 3dB. (10 Marks)
- c. A prototype low pass filter has the system response $H(s) = \frac{1}{s^2 + 2s + 1}$. Obtain a band pass filter with $\Omega_0 = 2$ rad/sec and $B_0 = 10$ rad/sec. (05 Marks)
- 6 a. A low pass filter is to be designed with the following desired frequency response:

$$H_d(e^{j\omega}) = \begin{cases} e^{-j2w}, & -\pi/4 \le \omega \le \pi/4 \\ 0, & \pi/4 < |\omega| \le \pi \end{cases}$$

Determine the filter coefficients h_d(n) if the window functions is defined as

$$\omega(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & \text{otherwise} \end{cases}$$
 (10 Marks)

b. Determine the filter coefficients h(n) obtained by sampling $H_d(\omega)$ given by,

$$H_{d}(\omega) = \begin{cases} e^{-j3\omega}, & 0 < \omega < \frac{\pi}{2} \\ 0, & \frac{\pi}{2} < \omega < \pi \end{cases}$$

Also, obtain the frequency response $H(\omega)$. Take N = 7.

(10 Marks)

7 a. Convert the analog filter into a digital filter whose system function is

$$H(s) = \frac{2}{(s+1)(s+3)}$$
 using bilinear transformation with T = 0.1 sec. (06 Marks)

- b. Explain how an analog filter is mapped on to a digital filter using backward difference method. Using this technique convert the analog filter with system function $H(s) = \frac{1}{s+2}$ into a digital filter. (14 Marks)
- 8 a. Obtain the parallel form realization of a system function:

$$H(z) = \frac{(z-1)(z+1)(z-2)z}{\left(z - \frac{1}{2} + j\frac{1}{2}\right)\left(z - \frac{1}{2} - j\frac{1}{2}\right)\left(z - \frac{j}{4}\right)\left(z + \frac{j}{4}\right)}$$
(08 Marks)

b. Obtain the cascade realization for the system function given by

$$H(z) = \frac{\left(1 + \frac{1}{4}z^{-1}\right)}{\left(1 + \frac{1}{2}z^{-1}\right)\left(1 + \frac{1}{2}z^{-1} + \frac{1}{4}z^{-2}\right)}$$
(06 Marks)

c. Realize the linear-phase FIR filter having the following impulse response:

$$h(n) = \delta(n) + \frac{1}{4}\delta(n-1) - \frac{1}{8}\delta(n-2) + \frac{1}{4}\delta(n-3) + \delta(n-4)$$
 (06 Marks)
