USN | | | | | | | | | | | **10EC52**

Fifth Semester B.E. Degree Examination, June/July 2013
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: 1. ***Answer FIVE full questions, selecting at least TWO questions from each part.***
2. ***Use of Normalized filter tables not permitted.***

PART – A

1 a. Define DFT. Derive the relationship of DFT to the z-transform. **(05 Marks)**

b. An analog signal is sampled at 10 kHz and the DFT of 512 samples is computed. Determine the frequency spacing between the spectral samples of DFT. **(03 Marks)**

c. Consider the finite length sequence $x(n) = \delta(n) - 2\delta(n-5)$:
Find i) The 10 point DFT of x(n) ii) The sequence y(n) that has a DFT $y(K) = e^{-\frac{j4\pi}{10}K} X(K)$ where X(K) is the 10 point DFT of x(n) iii) The 10 point sequence y(n) that has a DFT Y(K)=X(K)W(K) where X(K) is the 10 point DFT of x(n) and W(K) is the 10 point DFT of u(n) – u(n-6). **(12 Marks)**

2 a. Determine the circular convolution of the sequences x(n)={1, 2, 3, 1} and h(n)={4, 3, 2, 2} using DFT and IDFT equations. **(08 Marks)**

b. Let X(K) be a 14 point DFT of a length 14 real sequence x(n). The first 8 samples of X(K) are given by: X(0) = 12, X(1) = –1 + J3, X(2) = 3 + j4, X(3) = 1–J5, X(4) = –2+J2, X(5) = 6+J3, X(6) = –2 – J3, X(7) = 10.
Determine the remaining samples of X(K). Also evaluate the following functions without computing the IDFT.
i) x(0) ii) x(7) iii) $\sum_{n=0}^{13} x(n)$ iv) $\sum_{n=0}^{13} |x(n)|^2$ **(12 Marks)**

3 a. Consider a FIR filter with impulse response, h(n) = {3, 2, 1, 1}. If the input is x(n) = {1, 2, 3, 3, 2, 1, –1, –2, –3, 5, 6, –1, 2, 1}, using the overlap save method and 8 point circular convolution. **(10 Marks)**

b. What are FFT algorithms? Prove the i) Symmetry and ii) Periodicity property of the twiddle factor W_N. **(06 Marks)**

c. How many complex multiplications and additions are required for computing 256 point DFT using FFT algorithms? **(04 Marks)**

4 a. Find the DFT of the sequence $x(n) = \{\underset{\uparrow}{1}, 2, 3, 4, 4, 3, 2, 1\}$ using the decimation in frequency FFT algorithm and draw the signal flow graph. Show the outputs for each stage. **(10 Marks)**

b. Given x(n) = {1, 0, 1, 0}, find x(2) using the Goertzel algorithm. **(05 Marks)**

c. Write a note on chirp z transform algorithm. **(05 Marks)**

PART – B

5 a. Given that $\left|H(e^{j\Omega})\right|^2 = \dfrac{1}{1+64\Omega^6}$, determine the analog Butterworth low pass filter transfer function. **(06 Marks)**

b. Design an analog Chebyshev filter with a maximum passband attenuation of 2.5 dB at $\Omega_P = 20$ rad/sec and the stopband attenuation of 30 dB at $\Omega_S = 50$ rad/sec. **(10 Marks)**

c. Compare Butterworth and Chebyshev filters. **(04 Marks)**

6 a. What are the conditions to be satisfied while transforming an analog filter to a digital IIR filter? Explain how this is achieved in Bilinear transformation technique. **(05 Marks)**

b. Design a Butterworth filter using the impulse invariance method for the following specifications: Take T = 1 sec,

$$0.8 \le \left|H(e^{jW})\right| \le 1 \quad 0 \le W \le 0.2\pi$$
$$\left|H(e^{jW})\right| \le 0.2 \quad 0.6\pi \le W \le \pi$$

(10 Marks)

c. Determine H(z) for the given analog system function $H(s) = \dfrac{(s+a)}{(s+a)^2 + b^2}$ by using Matched z-transform. **(05 Marks)**

7 a. A z-plane pole-zero plot for a certain digital filter shown in Fig. Q7 (a). Determine the system function in the $H(z) = \dfrac{(1+a_1z^{-1})(1+b_1z^{-1}+b_2z^{-2})}{(1+c_1z^{-1})(1+d_1z^{-1}+d_2z^{-2})}$ giving the numerical values for parameters a_1, b_1, b_2, c_1, d_1 and d_2. Sketch the direct form II and Cascade realizations of the system. **(10 Marks)**

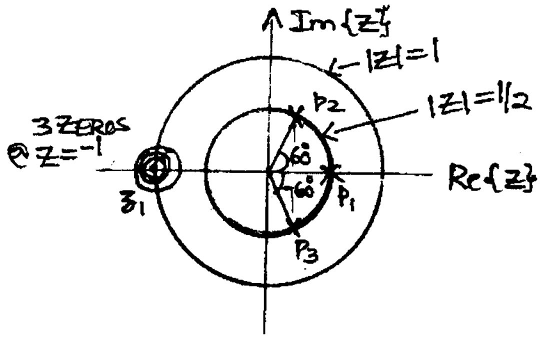

Fig. Q7 (a)

b. A FIR filter is given by,

$y(n) = x(n) + \dfrac{2}{5}x(n-1) + \dfrac{3}{4}x(n-2) + \dfrac{1}{3}x(n-3)$. Draw the direct form I and lattice structure. **(10 Marks)**

8 a. Design a FIR filter (low pass) with a desired frequency response,

$$H_d(e^{jW}) = e^{-j3W}; \quad -\frac{3\pi}{4} \le \omega \le \frac{3\pi}{4}$$
$$= 0; \quad \frac{3\pi}{4} < |\omega| < \pi$$

Use Hamming window with M = 7. Also obtain the frequency response. **(10 Marks)**

b. Design a linear phase low pass FIR filter with 7 taps and cutoff frequency of $\omega_C = 0.3\pi$ rad, using the frequency sampling method. **(10 Marks)**

* * * * *