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

Fifth Semester B.E. Degree Examination, June/July 2017
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 Butterworth table is permitted.*

PART – A

1 a. Define DFT. Establish the relation between DTFT and DFT. **(04 Marks)**

b. Find the 5-point DFT of $x(n) = \{1, 2, 3, 1\}$. And also draw magnitude and phase plots. **(08 Marks)**

c. Find the IDFT for the sequence: $x(K) = [5, 0, (1-j), 0, 1, 0, (1+j), 0]$ **(08 Marks)**

2 a. State and prove the following properties of DFT's
(i) Circular time shift.
(ii) Circular convolution in time. **(08 Marks)**

b. For the sequences, $x_1(n) = \cos\left(\frac{2\pi}{4}\right)n$, $x_2(n) = \sin\left(\frac{2\pi}{4}\right)n$, $0 \le n \le 3$. Find $x_1(n) \circledN x_2(n)$ using DFT and IDFT. **(07 Marks)**

c. Consider the sequence, $x(n) = [4\delta(n) + 3\delta(n-1) + 2\delta(n-2) + \delta(n-3)]$. Let X(K) be the six point DFT of x(n), find the finite length sequence y(n) that has six point DFT, $Y(K) = W_6^{4K} X(K)$. **(05 Marks)**

3 a. A long sequence x(n) is filtered through a filter with a impulse response h(n) to yield the output y(n).
If x(n) = [1, 4, 3, 0, 7, 4, -7, -7, -1, 3, 4, 3] and h(n) =[1, 2]. Compute y(n) using overlap add technique. **(08 Marks)**

b. Develop the DIF-FFT algorithm for N = 8. Using the resulting signal flow graph. Compute the 8-point DFT of the sequence $x(n) = \sin\left(\frac{\pi}{2}n\right)$ $0 \le n \le 7$. **(12 Marks)**

4 a. Determine the IDFT of X(K) = [4, 1-j2.414, 0, 1-j0.414, 0, 1+j0.414, 0, 1+j2.414]. Using inverse-Radix-2 DIT-FFT algorithm. **(10 Marks)**

b. What are chirp signals? What are the applications of chirp-Z transform? **(04 Marks)**

c. Write a note on Goertzel algorithm. **(06 Marks)**

PART – B

5 a. A Butterworth lowpass filter has to meet the following specifications:
(i) Passband gain, Kp = -1 dB at $\Omega_p = 4$ rad/sec.
(ii) Stopband attenuation greater than or equal to 20 dB at $\Omega_S = 8$ rad/sec.
Determine the transfer function $H_a(s)$. **(12 Marks)**

b. Explain analog-to-analog frequency transformation. **(08 Marks)**

6 a. Explain the structures used for realizing FIR filters by illustrations. **(10 Marks)**

b. Realize the system function,

$$H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$$

using linear phase. **(04 Marks)**

c. Obtain the cascade form realization for the given difference equation.

$$y(n) = -\frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$$

Also, draw the signal flow graph. **(06 Marks)**

7 a. A filter has to be designed with the following desired frequency response:

$$H_d(W) = \begin{cases} 0, & -\frac{\pi}{4} < |\omega| < \frac{\pi}{4} \\ e^{-j2\omega}, & \frac{\pi}{4} < |\omega| < \pi \end{cases}$$

Find the frequency response of the FIR filter designed using a rectangle window defined below.

$$W_R(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & \text{Otherwise} \end{cases}$$ **(12 Marks)**

b. List the steps in the design procedure of a FIR filter using window functions. **(05 Marks)**

c. List the advantages of a FIR filter. **(03 Marks)**

8 a. Derive mapping function used in transforming analog filter to digital filter by bilinear transformation preserves the frequency selectivity and stability properties of analog filter. **(12 Marks)**

b. Transform the analog filter,

$H_a(s) = \frac{(s+1)}{s^2 + 5s + 6}$ into H(z) using impulse invariant transformation. Take T = 0.1 sec. **(08 Marks)**

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