2002 SCHEME

USN | | | | | | | | | | |

EE55

Fifth Semester B.E. Degree Examination, May/June 2010
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***1. Answer any FIVE full questions.***
2. Use of Butterworth and Chebyshev tables are not permitted.

1 a. Determine the following:
   i) N pt DFT of $x[n] = W_N^{-mn}$ **(05 Marks)**
   ii) 4 pt DFT of $x[n] = \cos\left(\frac{n\pi}{4}\right)$ **(05 Marks)**
   iii) DFT of $x[n] = \delta[n]$. **(02 Marks)**
b. Determine the 8 point DFT of the sequence x [n] = [1, 1, 1, 1] and plot the magnitude and phase angle spectra. **(08 Marks)**

2 a. Consider the finite length sequence $x[n] = \delta[n] + 2\delta[n-5]$.
   i) Find the 10 point DFT X (K)
   ii) Find the sequence y [n] that has a DFT given by $Y(K) = e^{j\frac{4\pi}{10}k} X(K)$. **(08 Marks)**
b. Let $x[n] = 2\delta[n] + \delta[n-1] + \delta[n-3]$. If X (K) is the 5 point DFT of x [n] and if $Y(K) = X^2(K)$, find IDFT of Y (K). **(08 Marks)**
c. Prove the time shifting property of DFT. **(04 Marks)**

3 a. Find the 4 point DFTs of the following sequences, using a single 4 point DFT $x_1[n] = [1, 2, 0, 1]$ and $x_2[n] = [2, 2, 1, 1]$. **(10 Marks)**
b. The sequence x [n] = [1, 2, 3, 3, 2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1] is filtered through a filter whose impulse response is h [n] = [3, 2, 1, 1]. Compute the output of the filter y [n] using overlap and save method. Use 9 point circular convolution. **(10 Marks)**

4 a. If $x_1[n] = [1, 2, 0, 1]$ and $x_2[n] = [1, 3, 3, 1]$, obtain $x_1[n]$ ④ $x_2[n]$ using DIT – FFT algorithm. **(12 Marks)**
b. Develop a DIF – FFT algorithm for decomposing the DFT for N = 6. Draw the flow diagram. **(08 Marks)**

5 a. A system is specified by the relation $y[n] = \frac{1}{2}y[n-1] + \frac{1}{4}y[n-2] + x[n] + x[n-1]$. Realize the system in the following forms:
   i) Direct form
   ii) Cascade form and
   iii) Parallel form. **(12 Marks)**
b. Realize the FIR filter whose transfer function is $H(Z) = 1 + \frac{3}{4}z^{-1} + \frac{17}{8}z^{-2} + \frac{3}{4}z^{-3} + z^{-4}$ in
   i) Direct form and ii) Cascade form. **(08 Marks)**

6 a. Explain bilinear transformation method of digital filter design. **(08 Marks)**

b. Obtain the digital filter equivalent of the analog filter shown in Fig.6(b), using
   i) Impulse invariance transformation and
   ii) Bilinear transformation methods.

Assume the sampling frequency $f_s = 2$ fo, where fo is the cutoff frequency of the analog filter shown

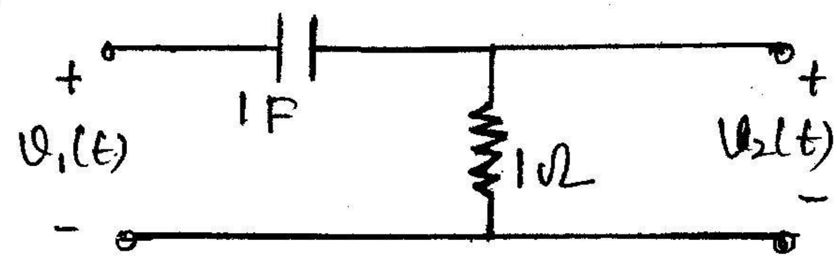

Fig.6(b)

(12 Marks)

7 a. Design a digital filter that meets the following specifications:

$$0.8 \le |H(w)| \le 1 \quad ; \quad 0 \le w \le 0.3\pi$$
$$|H(w)| \le 0.2 \quad ; \quad 0.6\pi \le w \le \pi$$

The filter must have a monotonic pass band and stop band frequency response. Use impulse invariance transformation. **(14 Marks)**

b. Compare the characteristics of IIR and FIR digital filters. **(06 Marks)**

8 a. Determine the filter coefficient $h_d[n]$ for the desired frequency response of a low pass filter given by,

$$Hd(ejw) = e^{-j2w} \quad \text{for} -\frac{\pi}{4} \le w \le \frac{\pi}{4}$$
$$= 0 \quad \text{for } \frac{\pi}{4} \le w \le \pi$$

If we define the new filter coefficients h [n] = h_d [n] w [n], where

$$w[n] = 1 \quad \text{for } 0 \le n \le 4$$
$$= 0 \quad \text{otherwise}$$

determine h [n] and also the frequency response H (e^{jw}) and compare with H_d (e^{jw}). **(12 Marks)**

b. Describe in detail, the architecture of general DSP processor. **(08 Marks)**

* * * * *