USN

NEW SCHEME

Fifth Semester B.E. Degree Examination, July 2007
Electrical and Electronics Engineering
Digital Signal Processing

Time: 3 hrs.] [Max. Marks:100

Note : ***Answer any FIVE full questions.***

1 a. Compute the DFT of the 8 point sequence

$$x[n] = \begin{cases} 1 & ; \quad 0 \le n \le 3 \\ 0 & ; \quad 4 \le n \le 7 \end{cases}$$

also determine the DFT of the following sequence without explicitly computing y(k) using basic equation.

$$y[n] = \begin{cases} 0 & ; \quad 0 \le n \le 1 \\ 1 & ; \quad 2 \le n \le 5 \\ 0 & ; \quad 6 \le n \le 7 \end{cases}$$

(10 Marks)

b. By means of DFT and IDFT, determine the response of the FIR filter with impulse response
$h[n] = [1, 2]$ with the input $x[n] = [1, 2, 3]$ take $N = 4$. **(10 Marks)**

2 a. Compute the quantity $\sum_{n=0}^{N-1} x_1[n]\, x_2[n]$ for the following sequences, using DFT properties. Take $N = 4$.

$$x_1[n] = x_2[n] = \cos\frac{2\pi}{N}n; \quad 0 \le n \le N-1$$

(12 Marks)

b. Prove the following properties of DFT
   i) Circular frequency shift
   ii) Circular time shift
   iii) Parseval's theorem **(08 Marks)**

3 a. 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)**

b. Determine the 8 point DFT for the signal $x[t] = \sin 314\, t$ using DIF FFT flow chart. **(10 Marks)**

4 a. Develop a radix 3 DIT FFT algorithm for evaluating the DFT for $N = 9$. **(10 Marks)**

b. Given
$x(k) = [20, -5.828 - j2.414,\ 0, -0.172 - j0.414,\ 0, -0.172 + j0.414,\ 0, -5.828 + j2.414]$
find $x[n]$ using IFFT algorithm. **(10 Marks)**

Contd.... 2

5 a. Obtain cascade and parallel structure for the system described by

$$y[n]+0.1y[n-1]-0.72y[n-2]=0.7x[n]-0.252x[n-2]$$ **(12 Marks)**

b. Obtain the direct form realization of the linear phase FIR system given by

$$H(Z)=1+\frac{3}{4}Z^{-1}+\frac{17}{8}Z^{-2}+\frac{3}{4}Z^{-3}+Z^{-4}$$ **(08 Marks)**

6 a. Explain impulse invariance method of transforming an analog filter into an equivalent digital filter. **(08 Marks)**

b. Apply bilinear transformation to obtain digital low pass filter to approximate $H(S)=\frac{1}{S^2+\sqrt{2}S+1}$. Assume cutoff frequency of 100 Hz and sampling frequency of 1 kHz. **(06 Marks)**

c. Explain the principle features of Harward architecture. **(06 Marks)**

7 a. Design a digital low pass filter using Butterworth approximation to meet the following specifications

Pass band edge = 120 Hz

Stop band edge = 170 Hz

Stop band attenuation = 16 dB.

Assume sampling frequency of 512 Hz. Use bilinear transformation. **(10 Marks)**

b. Design a digital Chebyshev filter using bilinear transformation to meet the following specifications.

i) 3 dB ripple in pass band $0 \le |\omega| \le 0.3\pi$

ii) 20 dB attenuation in the stop band $0.6\pi \le |\omega| \le \pi$

Use bilinear transformation. **(10 Marks)**

8 a. The desired frequency response of a low pass filter is given by

$$H_d(\omega)=e^{-j3\omega} \quad |\omega| \le \frac{3\pi}{4}$$

$$=0 \quad \frac{3\pi}{4} \le |\omega| \le \pi$$

Determine the frequency response of the FIR filter, if a Hamming window is used with N = 7. **(10 Marks)**

b. Design an ideal band pass filter with frequency response $H_d(\omega)=1$ for $\frac{\pi}{4} \le |\omega| \le \frac{3\pi}{4}$ Use rectangular window with N = 11 in your design. **(10 Marks)**

*****