Roll No. ..... Total No. of Pages : 3

BT-6/JX 8727

Digital Signal Processing

Paper : ECE-306 E

Time : Three Hours] [Maximum Marks : 100

Note :– Attempt **FIVE** questions in all, selecting at least **ONE** question from each Unit.

UNIT–I

1. (a) Determine if the system having the following system function is stable :

$$H(z) = \frac{1}{1 - \frac{7}{4}z^{-1} - \frac{1}{2}z^{-2}}$$ 10

(b) Determine the causal signal x(n) having the z-transform

$$X(z) = \frac{1}{(1-2z^{-1})(1-z^{-1})^2}$$ 10

2. (a) Compute the N-point DFT of the signal $x(n) = \sin\left(\frac{2\pi}{N}k_0 n\right)$; $0 \le n \le N-1$, and show the magnitude and phase spectra. 10

(b) Describe the 8-point, radix-2 decimation in time algorithm of FFT computation and obtain its signal flow graph. 10

UNIT–II

3. (a) Obtain the direct form I and direct form II structures for the following system :

$y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.252x(n-2)$ 10

(b) Describe the effect of finite word length on the performance of FIR filter structures. 10

4. (a) Obtain the cascade and parallel structures for the following system :

$$y(n) = y(n-1) - \frac{1}{2}y(n-2) + x(n) - x(n-1) + x(n-2)$$ 10

(b) A bandpass filter is to be used for digital clock recovery at 4.8 k baud and a sampling frequency of 153.6 kHz. The filter is characterized by the following transfer function :

$$H(z) = \frac{1}{1 + a_1 z^{-1} + a_2 z^{-2}}$$

where $a_1 = -1.957558$ and $a_2 = 0.995913$. Assess the effect of quantizing the coefficients to 8 bits on the pole positions and hence on the centre frequency. 10

UNIT–III

5. (a) Describe various parameters used to characterize the window functions for the design of FIR filters. 10

(b) Obtain the coefficients of an FIR low-pass filter to meet the specifications given below, using the Hamming window :

Pass-band edge frequency = 1.5 KHz

Transition width = 0.5 KHz

Stop-band attenuation > 50 dB

Sampling frequency = 8 KHz. 10

6. (a) Describe the procedure for design of optimum equiripple linear phase FIR filters. 10

(b) Determine the coefficients of a linear-phase FIR filter of length M = 15 which has a symmetric unit sample response and a frequency response that satisfies the conditions

$$H\left(\frac{2\pi k}{15}\right)=\begin{cases}1, & k=0,1,2,3\\ 0.4; & k=4\\ 0; & k=5,6,7\end{cases}$$ 10

UNIT–IV

7. (a) Enumerate and explain the factors influencing the choice between FIR and IIR filters for a given application. 10

(b) Determine the system function H(z) of the lowest-order Chebyshev digital filter that meets the following specifications :

(i) 1-dB ripple in the pass band $0 \le |w| \le 0.24\pi$

(ii) At least 60 dB attenuation in the stopband $0.35\pi \le |w| \le \pi$.

Use the bilinear transformation. 10

8. (a) Convert the analog filter with following system function into a digital IIR filter by means of the impulse invariance method :

$$H(s)=\frac{s+0.1}{(s+0.1)^2+9}$$ 10

(b) Describe the Pade approximation least squares method of designing a digital filter. 10