L 1114

B.E./B.Tech. DEGREE EXAMINATION, MAY/JUNE 2006.

Fourth Semester

Information Technology

IT 1252 — DIGITAL SIGNAL PROCESSING

(Regulation 2004)

Time : Three hours

Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. What is meant by aliasing? How can it be avoided?

2. Is the system $y(n) = \ln[x(n)]$ is linear and time invariant?

3. Define DFT pair.

4. Differentiate between DIT and DIF FFT algorithms.

5. Find the transfer function for normalized Butterworth filter of order 1 by determining the pole values.

6. What does 'frequency warping' mean?

7. State the advantages and disadvantages of FIR filter over IIR filter.

8. List out the different forms of structural realizations available for realizing a FIR system.

9. Bring out the difference between fixed-point and floating-point arithmetic.

10. How will you avoid limit cycle oscillations due to overflow in addition?

PART B — ($5 \times 16 = 80$ marks)

11. (i) With a neat diagram, explain the analysis and synthesis part of a vocoder in detail. (8)

(ii) The system is characterized by the difference equation $y(n) = 0.75y(n-1) + 5x(n)$. The input signal $x(n)$ has a range of $-6V$ to $+6V$, represented by 8 bits. Find the quantization step size, variance of the error signal and variance of the quantization noise at the output. (8)

12. (a) (i) Find the output response of the system given the input signal $x(n) = \{1, -2, \underset{\uparrow}{3}, -2\}$ and impulse response $h(n) = \{2, \underset{\uparrow}{-3}, 4\}$. (10)

(ii) Define correlation and bring out the difference between convolution and correlation. (3)

(iii) Find the cross correlation of $x(n) = \{1, 2, 3\}$ and $y(n) = \{3, 2, 1\}$. (3)

Or

(b) (i) Determine the Z-transform of the signal $x(n) = a^n u(n) - b^n u(-n-1), b > a$ and plot the ROC. (5)

(ii) Find the steady state value given $Y(z) = \{0.5/[(1 - 0.75z^{-1})(1 - z^{-1})]\}$. (3)

(iii) Find the system function of the system described by $y(n) - 0.75\, y(n-1) + 0.125\, y(n-2) = x(n) - x(n-1)$ and plot the poles and zeroes of $H(z)$. (8)

13. (a) (i) Using DFT - IDFT method, perform circular convolution of the two sequences $x(n) = \{1, 2, 0, 1\}$ and $h(n) = \{2, 2, 1, 1\}$. (10)

(ii) State and prove the circular convolution property of DFT. (6)

Or

(b) (i) Determine the number of complex multiplications and additions involved in a N-point Radix-2 and Radix-4 FFT algorithm. (4)

(ii) Compute the 8-point DFT of the given data sequence $x(n) = \{\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0\}$ using radix-2 Decimation in Time FFT algorithm. (12)

14. (a) (i) Convert the analog filter with system function

$H_a(s) = \{(s+0.1)/[(s+0.1)^2+9]\}$ into a digital IIR filter using impulse-invariance method. (Assume T = 0.1 sec). (7)

(ii) Obtain the direct form I, canonic form and parallel form realization structures for the system given by the difference equation $y(n) = -0.1\, y(n-1) + 0.72\, y(n-2) + 0.7x(n) - 0.252\, x(n-2)$. (9)

Or

(b) Design and realize a digital Butterworth filter using bilinear transformation to meet the following requirements

$$0.707 \le |H(e^{j\omega})| \le 1, \; 0 \le |\omega| \le \pi/2$$
$$|H(e^{j\omega})| \le 0.2, \; 3\pi/4 \le |\omega| \le \pi.$$ (16)

15. (a) (i) Determine the filter coefficients $h(n)$ of length $M = 15$ obtained by sampling its frequency response as

$$H[(2\pi/15)k] = 1, \quad k = 0,1,2,3,4$$
$$= 0.4, \; k = 5$$
$$= 0, \quad k = 6, 7$$

Using Rectangular window. (10)

(ii) Obtain the transversal structure and linear phase realization structure for a filter given by $h(n) = \{0.5, 2.88, 3.404, 2.88, 0.5\}$. (6)

Or

(b) Design a digital filter with

$$H_d(e^{j\omega}) = 1, \; 2 \le |\omega| \le \pi$$
$$= 0, \; \text{otherwise}$$

Using Hamming window with N = 7. Draw the frequency response. (16)
