PART C — (1 × 15 = 15 marks)

16. (a) Compute the characteristics of a limit cycle oscillation with respect to the system described the difference equation $x(n) = 0.95\,y(n-1) + x(n)$. Determine the dead band of the filter. Assume 4 bit sign magnitude representation including sign bit and the input as $x(n) = \begin{cases} 0.875, & \text{for } n = 0 \\ 0, & \text{otherwise} \end{cases}$ (15)

Or

(b) (i) Perform Circular convolution of the two sequences: (7)

$x_1(n) = \{2, 1, 2, 1\}$ $x_2(n) = \{1, 2, 3, 4\}$

(ii) Find the 4 point DFT of the sequence $x(n) = \cos\left(\frac{\pi}{4} n\right)$ using Decimation in Frequency algorithm. (8)

Reg. No. :

Question Paper Code : 53236

B.E./B.Tech. DEGREE EXAMINATIONS, APRIL/MAY 2019.

Fifth/Sixth Semester

Information Technology

IT 6502 — DIGITAL SIGNAL PROCESSING

(Common to Computer Science and Engineering/Mechatronics Engineering)

(Regulation 2013)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. For the discrete time signal $x(n)$ shown in the Fig. 1 below, sketch the signal $x(n-3)$ and $x(n+2)$.

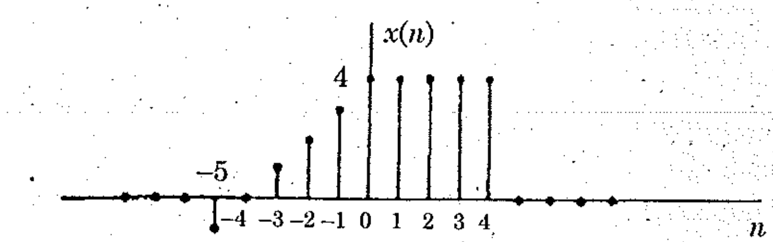

Fig. 1

2. Define correlation of two different signals.

3. Mention the number of computations involved in direct computation of DFT.

4. State the circular frequency shift property of DFT.

5. Mention the characteristics of the Butterworth and Chebychev analog filters.

6. Mention two advantages and disadvantages of IIR filters.

7. Define the Hamming and Hanning window functions.

8. Sketch the direct form structure for the FIR filter with the difference equation:

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

9. Mention the three ways of representing negative numbers. Express –7/8 in the three forms.

10. What is the advantage of scaling compared to saturation arithmetic?

PART B — (5 × 13 = 65 marks)

11. (a) (i) Consider the periodic sampling of a continuous time signal, establish the relation between analog and digital signal frequencies. (7)

(ii) Consider the analog signal $x_a(t) = 3\cos 100\pi t$.

(1) Determine the minimum sampling rate required to avoid aliasing. (2)

(2) Suppose that the signal is sampled at the rate $F_s = 300$ Hz and 75 Hz. What is the discrete time signal obtained after sampling? (4)

Or

(b) (i) Determine the power and energy of the unit step signal. (3)

(ii) Determine the Z-transform of the signal $x(n) = -a^n u(-n-1)$. Sketch its ROC. (5)

(iii) Compute the convolution of the two signals $x_1(n) = \{1, -2, 1\}$ and $x_2(n) = \begin{cases} 1, & 0 \le n \le 5 \\ 0, & \text{otherwise} \end{cases}$ (5)

12. (a) By means of DFT and IDFT, determine the response of the filter with impulse response $h(n) = \{1, 2, 3\}$ to the input sequence $x(n) = \{1, 2, 2, 1\}$. Assume N = 8. (13)

Or

(b) (i) Sketch the flow graphs of the basic butterfly computation and the 8 point Decimation in time FFT. (6)

(ii) Using the flow graph, determine the 8 point DFT of the sequence $x(n) = \{1, 2, 2, 2, 1, 0, 0, 0\}$. (7)

13. (a) A digital IIR low pass filter is required to meet the following frequency domain specifications :

3 dB ripple (maximum) in the passband $0 \le \omega \le 0.3\,\pi$ rad.

At least 20 dB (minimum) attenuation in the stopband $0.6\pi \le \omega \le \pi$

The digital filter is to be designed by applying bilinear transformation. (13)

Or

(b) A digital low pass filter is to be designed to have a maximally flat frequency response with the following specifications.

$20\log|H(\omega)|_{\omega=0.2\pi} \ge -1.9328\ dB$

$20\log|H(\omega)|_{\omega=0.6\pi} \le -13.9794\ dB$

Find the transfer function of the filter to meet the above specifications using impulse invariant transformation method. (13)

14. (a) The desired frequency response of a low pass filter is given by

$H_d(\omega) = \begin{cases} e^{-j3\omega}, & |\omega| < \frac{3\pi}{4} \\ 0, & \frac{3\pi}{4} < |\omega| < \pi \end{cases}$. Determine the frequency response of the FIR filter if Hamming window is used with N = 7. (13)

Or

(b) Design a 17 tap linear phase FIR low pass filter with cut off frequency $\omega_c = \frac{\pi}{2}$. The design is to be done using frequency sampling technique. (13)

15. (a) Consider the recursive filter shown in the Fig. 2 below. The input $x(n)$ has a range of values ± 100 V, represented by 8 bits. Compute the variance of the output of the A/D conversion process. (13)

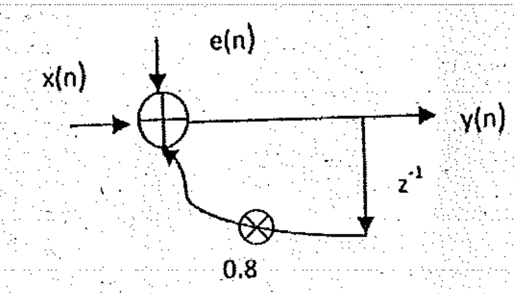

Fig. 2

Or

(b) Find the effect of coefficient quantization on pole locations of the given second order IIR system, when it is realized in direct form I and in cascade form. Assume a word length of 4 bits through truncation. $H(z) = \dfrac{1}{1 - 0.9z^{-1} + 0.2z^{-2}}$. (13)