CBCS SCHEME

USN | | | | | | | | | | 18EC52

Fifth Semester B.E. Degree Examination, June/July 2024
Digital Signal Processing

Time: 3 hrs. Max. Marks: 100

Note: Answer any FIVE full questions, choosing ONE full question from each module.

Module-1

1 a. Determine the 6-DFT of the data sequence $x(n) = \{1, 1, 2, 2, 3, 3\}$ and Compute the corresponding amplitude and phase spectrum. **(10 Marks)**

b. State and prove the following properties of DFT's :
   (i) Linearity property
   (ii) Periodicity property **(10 Marks)**

OR

2 a. Find the N-point DFT of $x(n) = \cos\left(\frac{2\pi}{N}K_0 n\right)$; $0 \le n \le N-1$. **(05 Marks)**

b. Find the 4-point DFT of the sequence $x(n) = \{1, 2, 0, 1\}$ using matrix method. **(05 Marks)**

c. Find the circular convolution of given data sequence $x_1(n) = \{1, 3, 5, 7\}$ and $x_2(n) = \{2, 4, 6, 8\}$, using DFT-IDFT method. **(10 Marks)**

Module-2

3 a. Determine the output sequence of a FIR filter whose jmpulse response in $h(n) = \{1, 1, 1\}$ and input sequence $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ using overlap-add method. Assume length of block is 6. **(10 Marks)**

b. Determine 8-point DFT Sequence for given input signal $x(n) = n + 1$ using DIF-FFT algorithm. **(10 Marks)**

OR

4 a. Find the response of LTI system (Linear convolution) of input sequence $x(n) = \{1, 1, 1\}$ and impulse response $h(n) = \{-1, -1\}$ using DIT-FFT algorithm. **(12 Marks)**

b. What is total number of complex additions and multiplications required to compute N = 1024 point DFT using direct and FFT method and also calculate the percentage savings in multiplications and additions. **(08 Marks)**

Module-3

5 a. With necessary mathematical analysis, explain the frequency sampling technique of FIR filler design. **(10 Marks)**

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

$$H_d(w) = \begin{cases} e^{-j3w}; & |w| \le \frac{3\pi}{4} \\ 0 \quad ; & \frac{3\pi}{4} < |w| < \pi \end{cases}$$

Determine the frequency response of the FIR filter if Hamming window is used. **(10 Marks)**

OR

6 a. List the steps in the design of a FIR filter using window functions. **(05 Marks)**

b. Realize the Linear FIR filter having the following transfer function,
$H(z) = 1 + 0.25z^{-1} - 0.125z^{-2} + 0.25z^{-3} + z^{-4}$ **(05 Marks)**

c. Sketch the lattice realization for given FIR filter with the following difference equation,
$y(n) = x(n) + 3.1x(n-1) + 5.5x(n-2) + 4.2x(n-3) + 2.3x(n-4)$ **(10 Marks)**

Module-4

7 a. Derive an expression for order and cut off frequency of a low pass Butterworth filter. **(08 Marks)**

b. Design a butterworth digital low pass filter with maximum pass band attenuation of 3 db at 500 Hz, minimum attenuation of 15 db at stopband edge frequency of 750 Hz and sampling frequency $F_3 = 2$ KHz. Use bilinear transformation method. (Assume T= 1 sec) **(12 Marks)**

OR

8 a. Derive mapping function used in transforming analog filter to digital filter by bilinear transformation. **(08 Marks)**

b. Distinguish between FIR and IIR filters. **(04 Marks)**

c. Obtain the direct form I and direct form II realization for the following system :
$y(n) + 0.1y(n-1) - 0.2y(n-2) + 3x(n) + 3.6x(n-2) + 0.6x(n-2)$ **(08 Marks)**

Module-5

9 a. With neat diagrams, explain hardware units used in DSP processors. **(10 Marks)**

b. Find the signed Q-15 representation for the decimal number -0.160123. **(06 Marks)**

c. Convert the Q-15 signed number 0.100011110110010 to the decimal number. **(04 Marks)**

OR

10 a. With a neat diagram, explain the fixed point basic architecture of TMS 320C54X processor. **(10 Marks)**

b. Explain the IEEE double precision floating point format used in DSP processor. **(05 Marks)**

c. Describe fixed point representation of numbers used in DSP processor. **(05 Marks)**

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