## TE(ELEX.) | Sem-I | R-191 'C' Scheme

**Duration: 3hrs** 

[Max Marks:80]

- N.B.: (1) Question No 1 is Compulsory.
  - (2) Attempt any three questions out of the remaining five.
  - (3) All questions carry equal marks.
  - (4) Assume suitable data, if required and state it clearly.
- 1 Attempt any FOUR

[20]

- Compute IDFT for the given DFT  $X(k) = \{2, 1+j, 0, 1-j\}$
- State and prove the circular time shift property of discrete time signal.
- Explain Decimation and Interpolation.
- Convert the following analog filter with system function  $H(s) = \frac{1}{(s+2)(s+0.6)}$ a digital filter by means of impulse invariant and BLT method.
- Compute the circular convolution of following discrete time signals  $x_1(n) = \{1, 3, 5, 3\}$  and  $x_2(n) = \{2, 3, 1, 1\}$
- Using DFT/IDFT method, find the response of the system with impulse response 2 [10]  $h(n) = 5\delta(n) - 2\delta(n-1)$  if the input to the system is x(n) = 3u(n) - 2u(n-2) - u(n-3)
  - Design a Butterworth digital IIR lowpass filter using Impulse Invariant [10] transformation method for the following specifications.  $0.707 \le |H(e^{iw})| \le 1.0$  for  $0 \le w \le 0.3\pi$  $|H(e^{jw})| \le 0.2$  for  $0.75 \le w \le \pi$ , (T= 1sec)
- Draw-architectural block diagram of DSP processor and explain functions of each block. [10]
  - Design a lowpass FIR filter using Hamming window to meet the following [10] specifications

 $H_d(w) = 1 \quad \pi/6 \le w \le \pi/6$ = 0 Otherwise

Use 10 tap filter and obtain impulse response of desired filter.

- Obtain the DFT of the sequence  $x(n) = (\frac{1}{2}, \frac{1}{2}, \frac{1}{2}, \frac{1}{2}, 0, 0, 0, 0)$  using DIF FFT [10] algorithm.
  - [10] Describe any one application of DSP.
- Find linear convolution using overlap-save method the following sequences: [10]  $x[n] = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$  and  $h[n] = \{1, 2, 3\}$ 
  - Determine impulse response h(n) of a filter having desired frequency response, [10]  $H_d(e^{jw}) = e^{-j(M-1)w/2} \quad 0 \le w \le \pi/2$  $\pi/2 \le w \le \pi$ , M=7, Use frequency sampling approach.
- [10] A difference equation describing a filter is given below  $y[n] - \frac{3}{4}y[n-1] + \frac{1}{8}y[n-2] = x[n] + \frac{1}{2}x[n-1]$ 
  - Draw direct form-I and direct form-II realizations. [10] Write a note on polyphase filters.

Prog. code: 1701135