Roll No.:......................

National Institute of Technology, Delhi

Name of the Examination: B. Tech. / M. Tech. / Ph.D.

Branch	: M.Tech (VLSI)	Semester	: I
Title of the Course	: Digital Signal Processing	Course Code	: EVL 511

Time: 3 Hours   Maximum Marks: 50

Section A (10 x 1 = 10 marks)

All questions are compulsory.

A.1 What is Gibbs phenomenon?

A.2 What is frequency warping?

A.3 Compare the Impulse invariant method and bilinear transformation method of IIR digital filter design.?

A.4 What is recursive and non recursive systems? Give an example.

A.5 What is the advantage in a cascade and parallel realization of IIR system?

A.6 Compare the number of multiplications required to compute the DFT of a 64-point sequence using direct computation and that using FFT.

A.7 Compute the N point DFT of $x(n) = a^n,\ \ 0 \leq n \leq N\text{-}1$

A.8 Realize a linear phase FIR filter with the impulse response h(n) = 2δ(n) - 4δ(n-1) + 2δ(n-2)

A.9 Find 10 point DFT of **x(n) = δ(n) + 2δ(n-5)**

A.10 Find circular convolution of x(n) and y(n), if **x(n) = y(n) = 3δ(n) + 2δ(n-2) + 2δ(n-3)**

Section B (4 x 5 = 20 marks)

Attempt any four questions.

B.1 What is Goertzel algorithm? What are the advantages of Goertzel algorithm? Find DFT coefficient **X(k)** at **k = 1** for the sequence **x(n) = {1,2,3,4}** using the Goertzel algorithm.

B.2 Design the symmetric FIR low pass digital filter whose desired frequency response is as

$$H_d(w)\begin{cases} e^{-jw\tau}, & for\ |w| \leq w_c \\ 0, & otherwise \end{cases}$$

Consider the length of filter is 9 and w_c = 1 radian/sample. Use Blackman window function.

B.3 What is Multirate Digital Signal Processing? Why Multirate DSP is required? Discuss about the polyphase implementation of 1-to-3 interpolators with suitable diagrams.

B.4 Discuss about the process of decimation by integer factor? Why an anti-aliasing filter is used in this process. Explain the process of decimation with suitable waveforms in time as well as in frequency domain. Assume input and output sampling frequency of decimator is 16 kHz and 4 kHz, respectively.

B.5 The analog transfer function of LPF is H(s) and its bandwidth is 1 rad/sec. Design the digital filter using BLT method whose cut-off frequency is 10π and sampling time is 0.16 Sec by considering the warping effect.

$$H(s) = \frac{1}{(s+2)(s+3)}$$

Section C (2 x 10 = 20 marks)

Attempt any two questions.

C.1 Design a high-pass digital chebyshev filter using bilinear transformation to satisfy the following constraints

$$0.707 \le |H(e^{jw}) \le 1 \qquad 0 \le w \le 0.2\pi$$

$$|H(e^{jw}) \le 0.2 \qquad 0.5\pi \le w \le \pi$$

Consider T_s=1 sec and cut-off frequency of H.P.F is twice of L.P.F.

C.2 **(a)** In an LTI system, the input ***x(n)*** = **{1, 0.5, 0}** and impulse response ***h(n)*** = **{-2, -2}.** Determine the response of LTI system by Radix-2, DIT FFT.

(b) Show that bilinear transformation maps jΩ axis (imaginary axis) in the s-plane into the unit circle, |z| = 1 and maps the left half s-plane Re(s)<0 inside the unit circle, |z| < 1.

C.3 **(a)** Obtain the direct form-II realization of the LTI system governed by the equation

$$y(n) = -(13/12)\,y(n-1) - (9/24)y(n-2) - (1/24)y(n-3) + x(n) + 4x(n-1) + 3\,x(n-2)$$

(b) Design a linear phase FIR filter using Kaiser window to meet the following specification:

$$0.97 \le |H(e^{j\omega}) \le 1.03, \qquad 0 \le |\omega| \le 0.19\pi$$

$$|H(e^{j\omega}) \le 0.01, \qquad 0.21\pi \le |\omega| \le \pi$$