

***B. Tech Degree VI Semester (Supplementary) Examination,
October 2009***

**CS 601 COMPILER CONSTRUCTION
(2006 Scheme)**

Time : 3 Hours

Maximum Marks : 100

**PART A
(Answer all questions)**

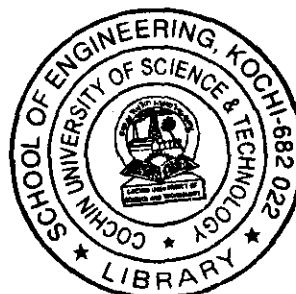
(8 x 5 =40)

- I. a. Briefly explain the role of a lexical analyzer.
- b. Write short notes on the different compiler construction tools.
- c. Explain the role of the parser.
- d. Write short note on the parser generator YACC.
- e. Describe synthesized attributes and inherited attributes.
- f. What are activation records? Explain.
- g. What are the different criteria for code improving transformations?
- h. Write short notes on
 - i. Copy propagation
 - ii. Dead code elimination

PART B

(4 x 15 =60)

- II. What is a compiler? With a neat diagram explain the different phases of a compiler.
OR
- III. With a suitable example explain how do you recognize tokens.
- IV. What is top down parsing? Explain any two top down parsing methods.
OR
- V. Explain Operator-Precedence Parsing.
- VI. Explain the different storage allocation strategies.
OR
- VII. Explain the different parameter passing methods.
- VIII. What is three address code? Explain the types of three address statements and their implementations.
OR
- IX. Explain the different issues in the design of a code generator.



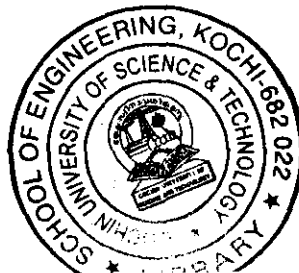
B.Tech. Degree VI Semester (Supplementary) Examination, October 2009

CS/EC/EI/EE 601 DIGITAL SIGNAL PROCESSING (1999 Scheme)

Time: 3 Hours

Maximum Marks: 100

- I. (a) Define
(i) linearity (ii) causality (iii) time invariance of discrete systems (8)
- (b) Find the inverse z -transform of
- $$X(z) = \frac{1 + 2z^{-1} + z^{-2}}{1 - \frac{3}{2}z^{-1} + \frac{1}{2}z^{-2}} \quad x[n] \text{ causal} \quad (12)$$
- OR**
- II. (a) What is system function? What is its significance? (6)
- (b) Why can't ideal filters be realized? (4)
- (c) Find the inverse z -transform of $x[n]$ of $X(z) = \frac{1}{1 - az^{-1}}$ by long division, where $x[n]$ is an anticausal sequence. What is its R.D.C? (10)
- III. (a) Describe the block convolution method using overlap add and overlap save schemes. (12)
- (b) Find the total multiplications and additions (complex as well as real) for computing an N point DFT. What is the computational saving when N is a power of 2 and radix 2 FFT is used. (8)
- OR**
- IV. (a) Show using a numerical example that circular convolution is linear convolution followed by time aliasing. (10)
- (b) Draw the signal flow graph for an 8 point Radix – 2 DIT FFT. (10)
- V. (a) Show that FIR filters can be designed with constant phase delay and constant group delay. (10)
- (b) Discuss the windowing method of FIR filter design. What is Gibb's phenomenon? (10)
- OR**
- VI. (a) Compare IIR and FIR filters. (10)
- (b) Discuss the frequency sampling technique for FIR filter design. (10)
- VII. (a) Implement the following filter in Direct form I, Direct form II and Cascade (12)
- $$H(z) = \frac{1 + 0.75z^{-1} + 0.125z^{-2}}{1 - 1.75z^{-1} + 0.875z^{-2} - 0.125z^{-3}}$$
- (b) Discuss the Bilinear transformation method. What is frequency warping. (8)



(Turn Over)

OR

- VIII. (a) Compare the characteristics of direct form 1, direct form 2, cascade and parallel forms of realizing IIR filters. (10)
- (b) Determine the order and poles of a Low pass Butter worth filter that has a 3 dB attenuation at 500 Hz and an attenuation of 40 dB at 1000 Hz. (10)

- IX. (a) Draw and explain the block diagram of a typical DSP processor. (10)
- (b) What do you mean by limit cycle oscillation. Illustrate with example. (10)

OR

- X. (a) Discuss any two DSP applications. (12)
- (b) Explain the need for scaling the input signal in saturated arithmetic. (8)

**B.Tech. Degree VI Semester (Supplementary) Examination,
October 2009**

CS/EI/EE 601 DIGITAL SIGNAL PROCESSING

(2002 Scheme)

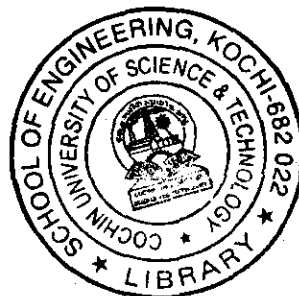
Time: 3 Hours

Maximum Marks: 100

- I a) (i) Find the convolution of the signals
$$x(n) = \begin{cases} 1 & \text{for } n=0,1 \\ 2 & \text{for } n=2,3 \\ 0 & \text{elsewhere} \end{cases}$$
$$h(n) = \delta(n) - \delta(n-1) + \delta(n-2) - \delta(n-3) \quad (5)$$
- (ii) Determine the response of the initially relaxed system characterized by the impulse response $h(n) = (1/2)^n u(n)$ to the input signal $x(n) = 2^n u(n)$. (5)
- b) Check the linearity, time invariance, causality and stability of the following systems
- i) $y(n) = nx(n)$
- ii) $y(n) = \cos \omega_0 n$ (10)
- OR**
- II a) (i) Write short notes on system function. (5)
- (ii) State and explain the time reversal and differentiation property of z -transform. (5)
- b) Find the inverse z -transform of $X(z) = \frac{z(z^2 - 4z + 5)}{(z-3)(z-1)(z-2)}$ for ROC
- (i) $2 < |z| < 3$ (ii) $|z| > 3$ (iii) $|z| < 1$ (10)
- III a) (i) Perform the circular convolution of the following sequences
 $x(n) = \{1, -1, 2, -2\}$ $h(n) = \{1, 2, 3, 4\}$ (5)
- (ii) Explain the relationship of DFT to Z -transform. (5)
- b) Find the linear convolution of the sequences
 $x(n) = \{1, -1, 2, -2, 3, -3, 4, -4, 5, -5, 6, -6\}$ and $h(n) = \{1, 1\}$
using overlap add method. (10)
- OR**
- IV a) Explain DIF FFT algorithm. (10)
- b) Determine the DFT values of the sequence $x(n) = \{2, 2, 2, 2, 1, 1, 1, 1\}$ using radix 2 DIT FFT algorithm. (10)
- V a) (i) Explain Gibbs oscillations. (5)
- (ii) Write short notes on windowing. (5)
- b) Explain frequency sampling method of FIR filter design. (10)

OR

(Turn over)



- VI a) Design an ideal low pass filter with a frequency response
- $$Hd(e^{j\omega}) = 1 \text{ for } -\pi/3 \leq \omega \leq \pi/3$$
- $$= 0 \text{ elsewhere}$$
- Use Fourier series method for the design choosing $N = 11$. (10)
- b) Obtain the direct form, cascade form and lattice structure realization of the FIR systems given by $H(z) = 1 + 2z^{-1} + 3z^{-2} + 4z^{-3} + 3z^{-4} + 2z^{-5} + z^{-6}$. (10)
- VII a) (i) Write short notes on prewarping. (5)
(ii) Compare FIR and IIR filters. (5)
- b) Obtain the direct form I, direct form II, cascade and parallel form realization for the following system. (10)
- $$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2).$$
- OR**
- VIII a) Using the bilinear transform, design a high pass filter, monotonic in pass band with cut off frequency of 1000 Hz and down 10dB at 350 Hz. The sampling frequency is 5000 Hz. (10)
- b) Design a chebyshev low pass filter with the following specifications. (10)
- $$\alpha_p = 1 \text{ dB ripple in the pass band } 0 \leq \omega \leq 0.2\pi \quad \alpha_s = 15 \text{ dB in the stop band}$$
- $$0.3\pi \leq \omega \leq \pi \text{ using Impulse invariance.}$$
- IX a) Draw and explain the architecture of typical DSP Processor. (10)
- b) Explain any two applications of DSP. (10)
- OR**
- X a) (i) Write short notes on product quantization error. (5)
(ii) Write short notes on signal sealing. (5)
- b) With an example explain limit cycle oscillations. (10)
