# ANNA UNIVERSITY – 2006

B.E/B.TECH V SEMESTER DEGREE EXAMINATION

DIGITAL SIGNAL PROCESSING

(INFORMATION TECHNOLOGY)

TIME-3HOUR
MARK-100

ANSWER ALL QUESTIONS

**PART A (10 * 2 = 20)**

1. State and prove the convolution property of Z transform.

2. Check the system is linear or not y(n) = x(n)+ay(n-1)

3. Write equations for finding DFT and IDFT using Z transform.

4. Draw the radix 2 butterfly structure for DIF

5. Draw the implementation for the generalized for IIR filter using direct form II.

6. Explain how the addition and multiplication of (H1, H2) impulse responses implemented in filter design

7. Write equations for Hanning and Blackman window.

8. Why frequency prewarping procedure is adopted in the design of IIR filter?

9. Write two advantages of musical sound processing and briefly explain.

10. Explain the effects due to upsampling.

**PART B (5 * 16 = 80)**

11.i) The impulse response of a linear TI system is h(n) = {1, 0, 1, -1}. Find the response of the system to the input signal x(n) = {1, 0, 2, 1}.

ii) Check whether the system y(n) = x(n) – x(n-1) is LTI and stable.

12.a) Develop and draw the 8 point radix-2 DIT FFT algorithm for DFT computation.

(OR)

12.b) Compute the DFT of the following sequence

x(n) = 0 0£ n £ 2

= 1 3£ n £ 6

= 0 n=7 Plot magnitude and phase spectra

13.a) Design a LPF with following specifications. Use Hamming window and at least 8 points(OR)

13.b)i) Obtain H(z) from H(s) when T = 1 sec.

ii) Design a digital BPF using w1 & w2 as cutoff frequencies

14.a)i) Perform the following using Floating Point arithmetic.

1.5 x 1.75 and 1.5 x 1.75

ii) Realize the following H(z) given by using cascade and Parallel form with Direct form-I.

(OR)

14.b)i) What is meant by quantization error? Explain briefly.

ii) Realize the following filter using cascade technique with DF-I and DF-II.

15.a) Briefly explain

a. Interpolator
b. Decimator
c. Effects due to sampling rate conversion

(OR)

15.b)i) Write a note on Musical sound processing

ii) Explain how the data compression is achieved in speech signal and discuss a technique to check the quality.