**Reg. No. \_\_\_\_\_\_\_\_**

**Karunya University**

**(Karunya Institute of Technology and Sciences)**

(Declared as Deemed to be University under Sec.3 of the UGC Act, 1956)

**Supplementary Examination - June 2011**

**Subject Title: DIGITAL SIGNAL PROCESSING Time: 3 hours**

**Subject Code: EE260 Maximum Marks: 100**

#### **Answer ALL questions**

**PART – A (10 x 1 = 10 MARKS)**

1. Give the classification of Signals.

2. Define symmetric and anti-symmetric signal.

3. State the properties of DFT.

4. Define circular convolution.

5. What are the types of digital filter according to their impulse response?

6. What is warping effect?

7. List the errors which arise due to quantization process.

8. What are the methods used to prevent overflow?

9. State the different computer architectures for signal processing.

10. State some applications of Digital Signal Processors.

**PART – B (5 x 3 = 15 MARKS)**

11. What is the need of Digital Signal Processing? State its benefits.

12. First five DFT coefficients of a sequence X(0) = 20, X(1) = 5+j2, X(2) = 0, X(3) = 0.2+j0.4 and X(4) = 0. Determine the remaining DFT coefficients.

13. List the desirable characteristics of Window function.

14. List any four finite word length effects.

15. State the Characteristics of VLIW processors.

**PART – C (5 x 15 = 75 MARKS)**

16. a. What is meant by Discrete Time Signal? Explain in detail. (7)

b. Explain in detail about Anti Aliasing and Anti Imaging Filter. (8)

(OR)

17. State a typical DSP system. Explain the various processes involved in detail.

18. Compute linear and circular convolutions of the two sequences. x1(n) = {1,1,2,2} and x2(n) = {1,2,3,4}. Use DFT and IDFT for circular convolution.

(OR)

19. Given x(n) = {1,2,3,4,5,6,7}, find X(k) using DIT – FFT algorithm.

20. A filter is to be designed with the following desired frequency response

Hd (ejω) = . Determine the filter coefficients hd (n) if the window function is defined as ω (n) = . Also determine the frequency response H(ejω) of the designed filter.

(OR)

[P.T.O]

21. Design a FIR digital filter to approximate an ideal low pass filter with pass band gain of unity, cut off frequency of 850 Hz and working at a sampling frequency of fs= 5000 Hz. The length of the impulse response should be 5. Use Blackman window.

22. Find the round off noise power for the following transfer function H(z) = H1(z). H2(z); where H1(z) = 1 /(1-a1z-1), H2(z) = 1 /(1-a2z-1) and a1 = 0.5, a2 = 0.6.

(OR)

23. For the recursive filter with difference equation y(n) = 0.93 y(n-1) + x(n). The input x(n) has a peak value of 10V, represented by 6 bits. Compute the output noise power due to A/D conversion process.

24. Explain in detail the Van Neumann and Harvard architectures pipelining.

(OR)

25. Discuss in detail about Super Scalar Processor Architecture.