**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: EC245 Maximum Marks: 100**

#### **Answer ALL questions**

**PART – A (10 x 1 = 10 MARKS)**

1. Define the impulse response of a discrete system.

2. Define discrete linear convolution.

3. What are the properties of FIR filter?

4. Why is direct Fourier series method not used in FIR filter design?

5. What are the methods used to convert analog to digital filter?

6. What is Gibb’s Oscillation?

7. What do you understand by a fixed point number?

8. What are the assumptions made concerning the statistical independence of various noise sources that occur in realizing the filter?

9. Why are FIR filters used in adaptive filter application?

10. What is adaptive noise cancellation?

**PART – B (5 x 3 = 15 MARKS)**

11. What are the advantages of FFT algorithm?

12. What are the methods used to design FIR filter?

13. Compare Butterworth filter and chebyshev filter.

14. What is meant by (zero input) limit cycle oscillation?

15. Describe the adaptive filtering.

**PART – C (5 x 15 = 75 MARKS)**

16. Compute the circular convolution of the following sequences and compare it with linear convolution x(n) =[1,-1,1,-1]; h(n) =[1,2,3,4]

(OR)

17. a. What are the properties of radix-2 DIT-FFT?

b. Describe the reduction of an 8-point DFT to 2-point Daft’s by DIF.

18. Design the symmetric FIR low pass filter for which desired frequency response is expressed as

**** The length of the filter should be 7 and wc=1radians/sample. Make use of rectangular window.

(OR)

19. a. For the linear phase FIR filter, prove that the impulse response satisfies the condition

h(n)=h(N-1-n).

b. Describe the design procedure of FIR filters using windows and compare the performance of various windows.

[P.T.O]

20. a. Convert the analog filter with system function Ha(s) = into a digital filter by means of the impulse invariance method.

b. Design a single pole low pass digital filter with a 3db bandwidth of 0.2π by use of bilinear transformation applied to the analog filter H(s) = whereΩc is the 3db bandwidth of the analog filter.

(OR)

21. Using bilinear transformation, design a Butterworth filter which satisfies the following conditions:

0.8 ≤ ⎜H(ejω) ⎜ ≤ 1, for 0 ≤ w ≤ 0.2π

⎜H(ejω) ⎜ ≤ 0.2 , for 0.6π**≤**w**≤** π

22. a. Consider the truncation of negative fraction numbers represented in(b+1)bits fixed point binary form including sign bit. Let (b-s) bits be truncated. Obtain the range of truncation errors for signed magnitude, 2’s complement and 1’s complement representations of the negative numbers.

b. An 8-bit ADC feeds a DSP system characterized by the following transfer function H(z)=1/(z+0.5). Estimate the steady state quantization noise power at the output of the system.

(OR)

23. Explain the characteristics of a limit cycle oscillation with respect to the system described by the difference equation y(n) =0.95y(n-1) + x(n). Determine the dead band effect of the filter.

24. Explain in detail the LMS adaptive algorithm for noise cancellation in communication Systems.

(OR)

25. a. Explain how Harvard architecture as used by the TMS 320 family differs from the strict Harvard architecture. Compare this with the architecture of a standard von Neumann processor. (8)

b. Sketch a block diagram of a suitable configuration for the MAC. With the aid of a timing diagram, explain how the MAC works. (7)