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Name:	
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Inviailator's Sianature :	

CS/M.Tech (ECE VLSI)/SEM-2/MVLSI-202/2013 2013 ADVANCED DIGITAL SIGNAL PROCESSING

Time Allotted: 3 Hours Full Marks: 70

The figures in the margin indicate full marks.

Candidates are required to give their answers in their own words as far as practicable.

Answer *five* questions taking at least two questions from each Group.

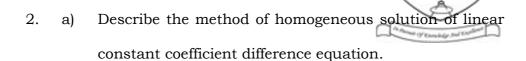
GROUP - A

- 1. a) Establish the relationship between input x (n) and output y (n) of an LTI system with impulse response h (n).
 - b) Derive the condition on impulse response h (n) for an LTI system to be (i) causal (ii) stable.
 - c) The impulse response of relaxed LTI system is h(n) = u(n). Determine the value of the step response of the system. 5 + 5 + 4

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- b) Determine the response y (n) for $n \ge 0$ of the system described by the 2nd order difference equation y(n) 3y(n-1) 4y(n-2) = x(n) + 2x(n-1) with $x(n) = 4^n u(n)$.
- 3. a) Determine the pole zero plot of the signal $x(n) = \begin{cases} a^n, \ 0 \le n \le M 1 \\ 0, \ else \ where \end{cases}$ where a > 0.
 - b) Determine the system function and unit sample response of the system y(n) = 0.5 y(n-1) + 2x(n).
 - c) Discuss the method of inverse z-transform using contour integration. 4 + 5 + 5
- 4. a) Show that the circular shift of a *N* point sequence is equivalent to linear shift of its periodic extension.
 - b) How is linear convolution obtained using circular convolution? Explain.
 - c) Find the output y (n) of a filter whose impulse response is h (n) = { 1, 1, 1 } and input x (n) = { 3, -1, 0, 1, 3, 2, 0, 1, 2, 1 } using overlap save method.

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- 5. a) Describe the Radix-2 DIT-FFT algorithm.
 - b) What is bit reversal? Compute the complexity of 4 point Radix-2 DIF-FFT algorithm.
 - c) Compute the 8-point DFT of the sequence $x(n) = \{0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0, 0\}$ using the radix-2 DIF algorithm. 7 + 3 + 4
- 6. a) For the given specification design an ananalog Butterworth Low Pass Filter : $0.9 \leq |H(j\Omega)| \leq 1 \text{ for } 0 \leq \Omega \leq 0.2\pi \text{ and } |H(j\Omega)| \leq 0.2 \text{ for } 0.4 \text{ } \pi \leq \Omega \leq \pi$
 - b) Discuss the bilinear transformation technique of digital IIR filter design.
 - c) A filter is to be designed with the following desired frequency response :

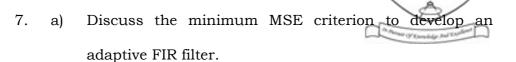
Determine the filter coefficient $h_d(n)$ if the window

function is defined as

$$w(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & \text{elsewhere} \end{cases}$$

4 + 4 + 6

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- b) Explain with block diagram the general polyphase framework for decimators and interpolators.
- c) What are the characteristics of a comb filter? 6 + 6 + 2
- 8. a) Discuss briefly the finite word length effects in IIR digital filters.
 - b) An 8 bit ADC feeds a DSP system characterized by following transfer function H (z) = 1/(z + 0·5). Estimate the steady state quantization noise power at the output of the system.
 - c) Explain the application of DSP in adaptive echo cancellation. 6+3+5