Total No. of Questions : 12] [Total No. of Pages : 4

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B.E. (Electronics)

ADVANCED DIGITAL SIGNAL PROCESSING

(404205) (2003 Course) Sem I. Elective-I

Time : 3 Hours] *[Max. Marks : 100*

Instructions to the candidates :

1) *Answer Q. No. 1 or Q. No. 2, Q. No. 3 or Q. No. 4, Q. No. 5 or Q. No. 6 from Section - I and Q. No. 7 or Q. No. 8, Q. No. 9 or Q. No. 10, Q. No. 11 or Q. No. 12 from Section - II.*
2) *Answers to the two sections should be written in separate answer books.*
3) *Neat diagrams must be drawn wherever necessary.*
4) *Use of electronic pocket calculator is allowed.*
5) *Figures to the right indicate full marks.*
6) *Assume suitable data, if necessary.*

SECTION - I

Q1) a) Explain the following : **[4]**

i) Deterministic signals.

ii) Random signals.

b) Explain the term power spectral density. Explain the properties of PSD. **[10]**

c) What do you mean by the term random process? **[2]**

OR

Q2) a) Design one stage and two stage interpolators to meet the following specifications : **[10]**

$I = 20$

Input sampling rate	:	10,000 Hz
Passband	:	$0 \leq F \leq 90$
Transition band	:	$90 \leq F \leq 100$
Ripple	:	$\delta_1 = 10^{-2}, \delta_2 = 10^{-3}$

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b) Explain sampling rate conversion by a rational factor I/D. [6]

Q3) a) Explain the basic architecture of adaptive filter and explain the situation in which adaptive filters are used. [8]

b) Explain LMS adaptive filtering with the help of flowchart. [4]

c) Explain the practical limitations of basic LMS algorithm. [4]

OR

Q4) a) Explain how adaptive filters can be used in Telephone echo cancellation. [8]

b) Explain RLS (Recursive Least Square Algorithm) and their practical limitations. [8]

Q5) a) Explain how the Levinson - Durbin algorithm can be used for the solution of normal equations. [10]

b) Explain Forward and Backward linear prediction with the help of block diagram. [8]

OR

Q6) a) The power density spectrum of an AR process $[x(n)]$ is given as

$$\tau_{XX}(w) = \frac{\sigma_w^2}{|A(w)|^2} = \frac{25}{\left|1 - e^{-jw} + \frac{1}{2}e^{-j2w}\right|^2}$$

Where σ_w^2 is the variance of the input sequence.

i) Determine the different equation for generating the AR process, when the excitation is white noise.

ii) Determine the system function for the whitening filter.

[9]

b) Explain ARMA processes and Lattice ladder filters. [9]

SECTION - II

Q7) a) Explain any two non-parametric methods of power spectrum estimation. [12]

b) Determine the mean & the autocorrelation of the sequence $x(n)$ generated by the MA(2) process described by the difference equation.

$x(n) = w(n) - 2w(n-1) + w(n-2)$

Where w(n) is a white noise process with variance σ_w^2. [6]

OR

Q8) a) Explain the ARMA model of power spectrum estimation. [6]

b) Compare parametric and non-parametric method of power spectrum estimation. [6]

c) Explain how DFT can be used for power spectrum estimation. [6]

Q9) a) Explain how FIR filtering can be implemented on general purpose DSP processor. [8]

b) Explain the criteria for selecting digital signal processor for an application. [6]

c) Explain what is floating point DSP. [2]

OR

Q10) a) Draw and explain the block diagram of fixed point DSP processor. TMS320C10. [8]

b) Explain the following terms w.r.t. DSP architecture. [8]

i) SIMD (Single instruction, Multiple data)

ii) VLIW (Very long instruction word)

iii) Static superscalar processing.

Q11) a) Draw the schematic diagram of the human speech production mechanism and explain its operation. [8]

b) Define and explain the characteristics of the following : [4]

i) Vowels.

ii) Consonants.

c) Explain how short time spectrum analysis can be used for analysis of speech. [4]

OR

Q12) a) Define the following : [6]

i) Voiced sound.

ii) Unvoiced sound.

iii) Pitch.

b) Draw the block diagram of pitch period estimation algorithm and explain the same. [10]

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