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B.E. (Electronics)

ADVANCED DIGITAL SIGNAL PROCESSING

(2003 Course) (404205)

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

Instructions to the candidates :

1) *Answer Q1 or Q2, Q3 or Q4, Q5 or Q6, from Section I and Q7 or Q8, Q9 or Q10, Q11 or Q12 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 electronic pocket calculator is allowed.*
5) *Assume suitable data, if necessary.*

SECTION - I

Q1) a) Obtain the spectral density, autocorrelation and signal energy when $x(t) = A \sin ct$. [8]

b) Explain with proof how autocorrelation and energy spectral density form FT pair and also explain Rayleighs energy theorm. [8]

OR

Q2) a) Explain sampling rate conversion by a rational factor I/D. [6]

b) Design a two stage decimator for the following specifications:

$D = 100$

Passband : $0 \leq F \leq 50$

Transitionband : $50 \leq F \leq 55$

Input sampling rate : 10,000 Hz

Ripple : $S_1 = 10^{-1}$, $S_2 = 10^{-3}$. [10]

Q3) a) Explain basic LMS adaptive algorithm along with a flow chart. [7]

b) Explain practical limitations of the basic LMS algorithm. [6]

c) Explain the main components of adaptive filter. [3]

OR

Q4) a) Explain how adaptive filters are useful for the following applications: [10]

i) Adaptive Jammer suppression.

ii) Adaptive telephone echo cancellation.

b) Explain RLS algorithm and compare its performance with LMS algorithm. [6]

P.T.O.

Q5) a) Define the following:

i) Autoregressive (AR) process.

ii) Moving average (MA) process.

iii) Autoregressive, moving average (ARMA) process. **[6]**

b) Explain prediction error filter with the help of neat diagram. **[6]**

c) Derive the optimum reflection coefficient for the lattice Forward & Backward predictors. **[6]**

OR

Q6) a) Explain the Levinson Durbin algorithm for the solution of the normal equations. **[8]**

b) Explain any two properties of the linear prediction error filters. **[6]**

c) Explain AR Lattice structure. **[4]**

SECTION - II

Q7) a) Compare the computational requirements of non parametric power spectrum estimate. **[10]**

b) Determine the mean and autocorrelation of the sequence $x(n)$ generated by the MA(2) process described by 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. **[8]**

OR

Q8) a) Explain with an application how to estimate power spectrum using autoregressive modelling. **[10]**

b) Explain the effect of spectral leakage and spectral smearing with an example. **[8]**

Q9) a) Explain SHARC architecture with neat block diagram. **[10]**

b) State and then discuss four key factors, apart from execution speed, that should be considered in choosing a DSP processor for each of the following application. **[6]**

i) High fidelity digital audio.

ii) Physiological signal processing for diagnosis.

OR

*Q10)*a) Explain how Harvard architecture as used by TMS 320 family differs from the strict Harvard architecture. Compare this with the architecture of a standard Von Neumann processor. **[8]**

b) Explain the concept of pipelining with an example and appropriate timing diagram. **[8]**

*Q11)*a) Explain the speech production mechanism with the help of a neat diagram. **[8]**

b) What is formant? **[2]**

c) Draw the block diagram of formant synthesizer and explain. **[6]**

OR

*Q12)*a) Explain the LPC model of speech. **[8]**

b) Explain the difference between Vowel and Consonant. **[4]**

c) Define the following terms: **[4]**

i) Bilabial.

ii) Glottal.