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Reg. No. : | | | | | | | | | | | | |

Question Paper Code : 81051

M.E./M.Tech. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2013.

First Semester

Applied Electronics

AP 9211/DS 9311/UAP 9114/AP 911/10244 CM 104 — ADVANCED DIGITAL SIGNAL PROCESSING

(Common to M.E. Communication System, M.E. Computer and Communication M.E. Digital Signal Processing, M.E. Digital Electronics and Communication Engineering and M.Tech. Information and Communication Technology)

(Regulation 2009/2010)

Time : Three hours   Maximum : 100 marks

Answer ALL questions.

PART A — ($10 \times 2 = 20$ marks)

1. State Parseval's theorem.

2. Write the wiener — klintchine relation.

3. What is modified periodogram?

4. What is the difference between parametric and non - parametric method?

5. What do you mean by linear prediction?

6. Define mean - square error.

7. What are the applications of adaptive filters related to signal processing?

8. Draw the block diagram of a adaptive Channel equalizer.

9. What is meant by Interpolation and Decimation?

10. What are the applications of multirate signal processing?

PART B — (5 × 16 = 80 marks)

11. (a) (i) State and prove spectral factorization theorem. (8)

(ii) What is white noise and how it can be simulated? (8)

Or

(b) (i) The auto correlation sequence of a signal R(k) = [0.1, 0.2, 0.3, 0.4]. Obtain a third order AR model by solving Yule - walker equations. Assume the modeling error is 0.1. (12)

(ii) Write short notes on power spectral density. (4)

12. (a) (i) What are the advantages of parametric methods of spectrum estimation over the non- parametric methods? (6)

(ii) Explain the auto - regressive spectrum estimation method, with reference to selection of order of the model. (10)

Or

(b) (i) Write short notes on Welch estimation and Levinson — Durbin algorithm. (10)

(ii) Describe periodogram spectral estimation. (6)

13. (a) (i) Explain the least square error criterion applied to the Wiener filter. (8)

(ii) Discuss the solution of prony's normal equation. (8)

Or

(b) (i) Determine the least square FIR inverse filter of length 2 to the system having the impulse response:

$$h(n) = \begin{cases} 1 & n = 0 \\ -\alpha & n = 1 \\ 0 & otherwise \end{cases}$$

Where $|\alpha| < 1$. (8)

(ii) Write short notes on Kalman filter. (8)

14. (a) (i) Explain the steepest decent method of iterating of optimum FIR filter co-efficient for adaptation. (12)

(ii) Write short notes on normalized LMS algorithm. (4)

Or

(b) (i) Explain the method of echo - cancellation in data transmission over telephone channels. (8)

(ii) Derive the Weiner hoff equations by starting from fundamentals. (8)

15. (a) (i) Explain the process of reducing and increasing the sampling rates in digital signal processing. (8)

(ii) Discuss in detail wavelet expansion of signals. (8)

Or

(b) (i) Explain with an example the subband coding process. (8)

(ii) Discuss on sampling rate conversion by a rational factor. (8)
