15/12/14/FN

Reg. No. [ ]

Question Paper Code : 11029

M.E. DEGREE EXAMINATION, NOVEMBER/DECEMBER 2014.

First Semester

Applied Electronics

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

(Common to M.E. Communication Systems/M.E. Computer and Communication Engineering/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. Define stationary random process.

2. Give examples for random signals.

3. State the various non parametric methods used for power spectrum estimation.

4. What is the computation time required to implement Levinson Durbin algorithm?

5. Write the least mean squared error criterion.

6. List the application of discrete Kalman filter.

7. State the recursive algorithm based on the method of steepest descent.

8. State any two properties of the LMS algorithm.

9. Define decimation.

10. What is sub band coding?

PART B — (5 × 16 = 80 marks)

11. (a) (i) State Wiener-Khintchine theorem and determine the energy density spectrum of the signal $x(n) = a^n u(n)$ where $-1 < a < 1$ using the theorem. (8)

(ii) State Parseval's theorem and prove it. (8)

Or

(b) (i) Define auto correlation and auto covariance. (6)

(ii) Let the power spectral density of the stationary random process is a rational function. Derive the relation between output and input for the linear system with rational system function considering AR, MA and ARMA random processes. (10)

12. (a) Obtain the power density spectrum of a random signal using periodogram and discuss the problems associated with the spectrum.

Or

(b) List the various parametric methods for power spectrum estimation and explain any two of them.

13. (a) (i) Explain backward linear prediction method. (11)

(ii) List the properties of linear prediction-error filters. (5)

Or

(b) Consider a signal $x(n) = f(n) + w_1(n)$, where $f(n) = 0.7s(n-1) + w_2(n)$ and $w_1(n)$ and $w_2(n)$ are white noise sequences with variances 0.64 and 1 respectively. Determine the optimum IIR Wiener filter for the signal $x(n)$. Also obtain the minimum MSE.

14. (a) (i) Define adaptive filters. (3)

(ii) Describe the application of adaptive filtering to adaptive channel equalization. (13)

Or

(b) Discuss in detail, the RLS algorithm for adaptive direct form filters.

15. (a) Illustrate the advantage of multistage interpolation with suitable example.

Or

(b) (i) Discuss the need for sampling rate conversion. (6)

(ii) Explain the application of wavelet transform in signal processing with suitable example. (10)

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