Reg. No. :

Question Paper Code : 60073

M.E. DEGREE EXAMINATION, MAY/JUNE 2017.

First Semester

Applied Electronics

AP 7101 — ADVANCED DIGITAL SIGNAL PROCESSING

(Common to M.E. Communication Systems/M.E. Communication and Networking/M.E. Digital Signal Processing/M.E. Electronics and Communication Engineering/M.E. Mobile and Pervasive Computing and M.E. Optical Communication)

(Regulations 2013)

Time : Three hours Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. State the Wiener-Khinchine relation.

2. Define Gaussian random variable.

3. Differentiate between ARMA and AR models.

4. Differentiate parametric and non-parametric method of estimating the power spectrum.

5. State the properties of linear prediction error filters.

6. Write the Wiener-Hopf equation.

7. State the advantages of adaptive filters.

8. Mention the rules to be followed to select the mean of an adaptive filter.

9. List out the applications of sampling rate converters.

10. Design a decimator that down samples an input signal $x(n)$ by a factor D = 2. Use the Remez algorithm to determine the coefficients of the FIR filter that has 0.1dB ripple in the pass band and is down by at least 60dB in the stop band. Also determine the polyphase filter structure in a decimator realization that employs polyphase filters.

PART B — (5 × 13 = 65 marks)

11. (a) Explain the process of filtering in random process with relevant expressions. (13)

Or

(b) A signal whose first six values are $X = [11.25\ 0.55\ 0.375\ 0.1875\ 0.0938]^T$. Design the pade approximation to find a second-order all pole model $(p = 2 \text{ and } q = 0)$. (13)

12. (a) Compute the energy density spectrum of a finite sequence. (13)

Or

(b) Derive the barlett method of estimating the spectrum with neat diagram. (13)

13. (a) With neat diagrams and relevant expressions, briefly, explain how a Wiener filter can be used as a predictor? (13)

Or

(b) Derive the recursive predictor coefficients for optimum lattice filter using herinson-durbin algorithm. (13)

14. (a) Explain the method of working of adaptative filters based on steepest descent method. (13)

Or

(b) Explain LMS algorithm with neat diagram and relevant equations. (13)

15. (a) With neat sketches, explain the time domain and frequency domain characteristics of a decimeter with a factor of D. (13)

Or

(b) Obtain the poly phase realization structure of a 11-tap FIR filter with a decomposition factor of 2. (13)

PART C — (1 × 15 = 15 marks)

16. (a) Design a linear phase FIR filter that satisfies the following specifications: sampling frequency : 6000 Hz

Passband : $0 \leq F \leq 75$

Transition band : $85 \leq F \leq 80$

Stopband : $80 \leq F \leq 4000$

Passband ripple : $\delta_1 = 10^{-3}$

Stoppband ripple : $\delta_2 = 10^{-6}$. (15)

Or

(b) Let us consider a signal $x(n) = s(n) + w(n)$, where $s(n)$ is an AR (1) process that satisfies the equation $s(n) = 0.6s(n-1) + v(n)$. Where $\{v(n)\}$ is a white noise sequence with variance of 0.54 and $\{w(n)\}$ is a white noise sequence with variance of 1. Design a 2nd order wiener filter to estimate $\{s(n)\}$. (15)
