ADAPTIVE SIGNAL PROCESSING

B.E., VI Semester, Electronics & Communication Engineering/ Telecommunication Engineering

[As per Choice Based Credit System (CBCS) scheme]

Subject Code	15EC652	IA Marks	20
Number of Lecture Hours/Week	03	Exam Marks	80
Total Number of Lecture Hours	40 (8 Hours / Module)	Exam Hours	03

CREDITS – 03

Course Objectives: The objectives of this course are to:

- Introduce to the concept and need of adaptive filters and popular adaptive signal processing algorithms
- Understand the concepts of training and convergence and the trade-off between performance and complexity.
- Introduce to common linear estimation techniques
- Demonstrate applications of adaptive systems to sample problems.
- Introduce inverse adaptive modelling.

Module-1	RBT Level
Adaptive systems: Definitions and characteristics - applications – properties-examples - adaptive linear combiner input signal and weight vectors - performance function-gradient and minimum mean square error - introduction to filtering-smoothing and prediction - linear optimum filtering-orthogonality - Wiener – Hopf equation-performance surface(Chapters 1& 2 of Text).	L1, L2
Module-2	
Searching performance surface-stability and rate of convergence: Learning curve-gradient search - Newton's method - method of steepest descent - comparison - Gradient estimation - performance penalty - variance - excess MSE and time constants – mis-adjustments (Chapters 4& 5 of Text).	L1, L2
Module-3	
LMS algorithm convergence of weight vector: LMS/Newton algorithm - properties - sequential regression algorithm - adaptive recursive filters - random-search algorithms - lattice structure - adaptive filters with orthogonal signals (Chapters 6& 8 of Text).	L1, L2, L3
Module-4	
Applications-adaptive modeling and system identification: Multipath communication channel, geophysical exploration, FIR digital filter synthesis. (Chapter 9 of Text).	L1, L2, L3
Module-5	
Inverse adaptive modeling: Equalization, and deconvolution adaptive equalization of telephone channels-adapting poles and zeros for IIR digital filter synthesis(Chapter 10 of Text).	L1, L2, L3

Course Outcomes: At the end of the course, students should be able to:

- Devise filtering solutions for optimising the cost function indicating error in estimation of parameters and appreciate the need for adaptation in design.
- Evaluate the performance of various methods for designing adaptive filters

through estimation of different parameters of stationary random process clearly considering practical application specifications.

- Analyse convergence and stability issues associated with adaptive filter design and come up with optimum solutions for real life applications taking care of requirements in terms of complexity and accuracy.
- Design and implement filtering solutions for applications such as channel equalisation, interference cancelling and prediction considering present day challenges.

Question paper pattern:

- The question paper will have ten questions
- Each full question consists of 16 marks.
- There will be 2 full questions (with a maximum of Three sub questions) from each module.
- Each full question will have sub questions covering all the topics under a module.
- The students will have to answer 5 full questions, selecting one full question from each module.

Text Book:

Bernard Widrow and Samuel D. Stearns, “Adaptive Signal Processing”, Person Education, 1985.

Reference Books:

1. Simon Haykin, “Adaptive Filter Theory”, Pearson Education, 2003.
2. John R. Treichler, C. Richard Johnson, Michael G. Larimore, “Theory and Design of Adaptive Filters”, Prentice-Hall of India, 2002.

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