

COURSE SYLLABUS: EE583 - ADAPTIVE SIGNAL PROCESSING

Instructor: Dr. Edgar Satorius

1. Introduction

This class meets 6:30 PM - 9:20 PM every Wednesday evening beginning January 14, 2009 and ending on April 29, 2009. The final exam for this course is on Wednesday May 6, 2009 from 7-9:00 PM. Our class room will be OHE 100B. The TA is Mr. Ashok Patel who will also serve as the grader.

The sections given below are an outline of the topics I hope to cover in this course. Section 4 is mainly a review of material on discrete random processes. Most of this material is covered in EE562a, which is a prerequisite for this course. To do well in this course, you should be thoroughly familiar with this material. If you need to refresh your memory, I suggest you use the 'Probability, Random Variables, and Stochastic Processes' book by Papoulis or the 'Discrete-Time Signal Processing' book (§2.10, App. A) by Oppenheim and Schaffer referenced below -- also the course text ('Adaptive Filter Theory' by Haykin; Chps. 2 and 3) has a good review on discrete random processes. Other related courses include: EE563 (Estimation Theory which covers Kalman filters); EE586L (Advanced DSP Design Lab); EE667 (Array Signal Processing) and EE668 (VLSI Processors).

2. Grading and Computers

Computer Project(s)	Midterm (open book & notes)	Final (open book & notes)	Homework
50%	15 %	25%	10 %

Throughout the semester I will assign 5-6 homework sets plus two or three computer projects. The computer projects will help you learn the course material by conducting practical computer experiments on real world problems. Each project will focus on a reasonably well defined problem so that you can concentrate on learning the techniques - not coming up with problems. The results of your studies should be well documented in a report with computer printouts/plots (no source listings required) to justify your conclusions. In doing the computer projects, you can use any computer language you wish; however, I encourage you to consider Matlab – especially if you have experience with it. Matlab is a very simple and powerful language that is particularly suitable for programming signal processing algorithms. It also has a very nice graphical display capability and includes a convenient mechanism for incorporating on-line help into the system. If you do use Matlab, make sure you have the Signal Processing toolbox.

If you do well on the homeworks and the projects, then you will be able to perform well in the class. Do the homeworks on your own (although you are free to discuss the problems with other classmates). Likewise with the computer projects: you can discuss

them with others, but write them yourself. The midterm and final exams will be open text/note exams that will cover the course material.

3. Office Hours

My office hours are 5:30-6:30 Wednesdays in EEB 102. TV students may call me during this time (213 740 2221), or arrange an appointment for Wednesday evenings. I strongly encourage you to make use of this time to discuss problems with the course material or any related aspects of digital signal processing which interest you. Alternatively, you can always reach me by e-mail (I try to answer all queries from students as soon as possible, including detailed technical questions). My e-mail address is: **esatoriu@usc.edu**. (Please note that I have traditionally used my JPL address, **Edgar.H.Satorius@jpl.nasa.gov**, for student e-mail but thought this year I would give my USC e-mail a tryout – I'll see how this works.)

Questions related to the homework, projects, Matlab, etc. should initially be addressed to the TA or grader. The TA's e-mail address is: **ashokpat@usc.edu**. The TA's office hours are Mondays 4:00-6:00 in EEB 420 as well as Tuesdays 4:00-5:00 to address grading questions. During these times you can reach the TA at 213 740-2356. Please make use of our TA – remember: he's located on-campus and I'm not !

COURSE OUTLINE:

4. Discrete random processes (Class 1)

[Papoulis; O&S-DTSP (§2.10, App. A); Haykin (Chp. 1)]

- [1] Random variables, random processes, filtered random processes.
- [2] Ensemble averages, correlation, covariance, power spectrum, cross power spectrum.
- [3] Ergodicity, time averages, biased & unbiased estimators, consistent estimators.

5. Linear prediction (Classes 1-3)

[Haykin (Chp. 3)]

- [1] Direct form linear prediction filtering.
- [2] Normal equations for linear prediction filtering.
- [3] Levinson algorithm.
- [4] Linear prediction lattice filtering.

6. Digital Wiener filtering (Classes 4-6)

[Haykin (Chp. 2)]

- [1] Wiener smoothing and prediction filters.
- [2] Application of Wiener smoothing to noise cancelling.
- [3] Application of Wiener prediction filters.

- [4] Constrained, linear MMSE filtering.
- [5] Minimum variance beamforming.

7. Midterm, Class 7: February 25, 2009

8. Least mean squares adaptive filter (Classes 8-12)

[Haykin (Chps. 4,5,6,13 and 14)]

- [1] LMS adaptive algorithm.
- [2] Properties of LMS adaptive filter.
- [3] Normalized forms.
- [4] Finite precision effects.
- [5] Adaptive beamforming.

9. Orthogonalized adaptive filters (Classes 12-13)

[Haykin Chps. 7,12]

- [1] Frequency domain adaptive filters.
- [2] Adaptive lattice filters.

10. Least squares adaptive filters (Classes 13-14)

[Haykin (Chps. 8,9,12)]

- [1] Godard algorithm.
- [2] Lattice.

11. Other adaptive filtering techniques (Classes 14-15)

[Haykin (Chps. 15,17)]

- [1] Neural networks and multi-layer perceptrons.
- [2] Adaptive IIR filtering.
- [3] The constant modulus algorithm.

12. Blind adaptive filtering (Class 15 – time permitting)

[Haykin (Chp. 16)]

- [1] Cost functions.
- [2] Higher-order statistics.
- [3] Examples.

REFERENCES:

13.1. Required

- [1] Class notes: I have prepared a large amount of supplementary class notes that are required for the course. These will be available at the DEN course website.

- [2] Adaptive Filter Theory, S. Haykin, Prentice-Hall, 4-th edition, 2001.

13.2. Recommended Reading

- [1] Fundamentals of Adaptive Filtering, Ali H. Sayed, John Wiley, 2003.
- [2] Statistical and Adaptive Signal Processing: Spectral Estimation, Signal Modeling, Adaptive Filtering and Array Processing, D. Manolakis, V. Ingle, S. Kogan, McGraw Hill, 1999.
- [3] Adaptive Signal Processing, B. Widrow, S. Stearns, Prentice-Hall, 1985.
- [4] Theory and Design of Adaptive Filters, J. Trierchler, C. Johnson, M. Larimore Prentice-Hall, 1995.
- [5] Adaptive Filtering: Algorithms and Practical Implementation, P. Diniz, Kluwer, 1997.
- [6] Adaptive Filters: Structures, Algorithms and Applications, M. Honig, D. Messerschmitt, Kluwer, 1984.
- [7] Adaptive Signal Processing, L. Sibul, Ed., IEEE Press, 1987.
- [8] Time Series Analysis: Forecasting and Control, G. Box, G. Jenkins, Holden-Day, 1976.
- [9] Time Series, D. Brillinger, Holt-Reinhart-Wilson, 1975.
- [10] Signal Processing: The Modern Approach, J. Candy, McGraw Hill, 1988.
- [11] Signal Processing: The Model Based Approach, J. Candy, McGraw Hill, 1986.
- [12] Digital Spectral Analysis, S. Marple, Prentice-Hall, 1987.
- [13] Blind Deconvolution, S. Haykin, ed., Prentice-Hall, 1994.

13.3. Background Material

- [1] Discrete-Time Signal Processing, A. Oppenheim and R. Schaffer, Prentice Hall, 1999 (Chps. 2 -- §2.10, 11 & App. A).
- [2] Probability, Random Variables and Stochastic Processes, A. Papoulis, McGraw-Hill, 1991.
- [3] Applied Linear Algebra, G. Strang, Academic Press, 1976; **OR**:
- [4] Applied Linear Algebra, Noble and Daniel, Prentice Hall, 1977.
- [5] Matrix Computations, G. Golub, C. Van Loan, Johns Hopkins University Press, 1983.
- [6] Numerical Recipes: The Art of Scientific Programming, W. Press, B. Flannery, S. Teukolsky, W. Vetterling, Cambridge University Press, 1993.
- [7] MATLAB Reference Guide: High-Performance Numeric Computation and Visualization Software, The MathWorks, Inc., South Natick, MA, 1984-92.