COURSE SYLLABUS: EE483 - INTRODUCTION TO DIGITAL SIGNAL PROCESSING

Instructor: Dr. Edgar Satorius

1. Schedule and Introduction

This class meets 6:00 PM - 9:10 PM every Tuesday evening in RTH 115 beginning May 24, 2011 and ending on August 9, 2011. The TA is Mr. Sachin Chachada and the grader is TBD. There will also be a discussion section for this class meeting every Thursday evening in OHE 100C from 5:30 – 6:20 PM beginning May 19, 2011. The discussion section will be conducted by the TA.

The objective of this course is to provide a basic introduction to the theory of digital signal processing (DSP). I assume a familiarity with the Fourier and Laplace transforms and concepts such as linearity and shift invariance that are used in the description and analysis of linear analog systems. Much of what we do extends these ideas to the field of discrete time systems. Major parts of the course will concentrate on signal analysis using Fourier transforms, linear system analysis, Filter design and a few more advanced topics. We will study the discrete Fourier transform and its properties. We will also study the sampling theorem and the relationship between continuous and discrete time transforms. We will see how discrete time, linear shift invariant systems can be characterized using linear difference equations and the impulse response and show how tools such as the z-transform and discrete Fourier transform can be used in the design and analysis of such systems. We will then study the design and implementation of digital filters. I will also include some topical material: what is bandpass sampling?, what are polyphase filters and filterbanks?, what are adaptive filters? While this course deals largely with the theory of DSP, we will use a powerful software package, MATLAB, to look at applications of this theory, particularly Fourier analysis and digital filter design.

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 you should have covered in a Linear Systems course. If you are not too familiar with this material or need to refresh your memory, I suggest you use the "Signals and Systems" book by Oppenheim and Wilsky referenced below or perhaps the “Signals and Systems Made Ridiculously Simple” book by Karu also referenced below (actually I have not read this book but its title sounds interesting!).

Policy on class attendance for on-campus students: There is no requirement to attend the class in the studio (though I appreciate in-studio attendance as it can get lonely !!!!).

2. Grading and Computers

Midterm:	30%
Final:	30%
Homeworks:	10%

Two Matlab computer
projects: 30% (15% each)

Throughout the semester I will assign 5-6 homework sets plus two Matlab computer projects. The homework sets will help prepare you for the midterm and final exams. The Matlab projects will help you learn the material by conducting practical computer experiments on real world problems. 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 I encourage you to discuss the problems with your friends). Likewise with the Matlab assignments: please discuss them, but write them yourself. The midterm will include all material covered up to the midterm and the final will cover the remainder of the course. Both exams will be closed book.

Policy on late assignment submittals: I will allow late submittals **provided you let me know in advance via e-mail**. However, once the solutions are posted (typically about a week after the assignments are due), no submittals will be accepted or graded. **One other important note regarding assignment submittals for on-campus students**: Do not interrupt the class to submit an assignment (I will not accept or grade assignments submitted during the class). You must submit your assignment either before or after class or during the break.

3. Office Hours

My office hours are 5:00-6:00 Tuesdays in PHE 414. TV students may call me during this time (213 740 7654), or arrange an appointment for Tuesday 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. If you can't reach me otherwise, my e-mail address is: **Edgar.H.Satorius@jpl.nasa.gov**.

Questions related to the homework, projects, Matlab, etc. should initially be addressed to either the TA or grader. The TA's e-mail address is: **chachada@usc.edu**. The TA's office hours are tbd. The grader's e-mail address is: **tbd**. The grader's office hour is TBD (you can reach the grader during his office hours at TBD). Please make use of our TA and grader – remember: they're located on-campus and I'm not !

COURSE OUTLINE:

4. Introduction to discrete linear systems (Class 1)
[Mitra, §§2.1-2.7]

[1] Discrete time signals.
[2] Special sequences.
[3] Shift invariance.
[4] Stability and causality.
[5] Impulse response.
[6] Difference equations.

5. Discrete-Time Fourier Transform and Linear Time Invariant Systems (Class 1-2)
[Mitra, §§3.1-3.9]

[1] Transform definitions.
[2] Theorems.
[3] Frequency response of linear time invariant systems.
[4] Phase and group delays.
[5] Matlab computations.

6. The Z transform (Class 2)
[Mitra, §§6.1-6.7]

[1] Z-transforms by summation of left, right, and two-sided sequences.
[2] Regions of convergence and Z-transform properties.
[3] Inverse Z-transform.

7. Properties of digital filters (Class 3)
[Mitra, §§8.1-8.12]

[1] Averaging filter.
[2] Recursive smoother.
[3] First-order notch filter.
[4] Second-order unity gain resonator.
[5] All-pass filters.
[6] Comb filters.
[7] Equalization filters.
[8] Group delay, linear phase, all-pass, minimum phase

8. Fourier transforms, sampling – Part I (Class 4-5)
[Mitra, §§4.1-4.6, 13.1]

[1] Fourier transform review.
[2] Sampling continuous-time signals: the sampling theorem.
[3] Aliasing.
[4] Re-sampling digital signals.

[5] Midterm review.

9. Midterm, Class 6: June 28, 2011

8. Fourier transforms, sampling – Part II (Class 7)
[Mitra, §§4.7-4.11, 13.2-13.6]

[1] A/D conversion and quantization
[2] D/A conversion
[3] Polyphase decomposition
[4] Polyphase DFT filterbanks
[5] Bandpass sampling

10. The discrete Fourier transform (Class 8-9)
[Mitra, §§5.1-5.9, 15.1, 15.2]

[1] Definition of DFT and relation to Z-transform.
[2] Properties of the DFT.
[3] Linear and periodic convolution using the DFT.
[4] Zero padding, spectral leakage, resolution and windowing in the DFT.

11. The fast Fourier transform (Class 9)
[Mitra, §§11.1, 11.3]

[1] Decimation in time FFT.
[2] Decimation in frequency FFT.

12. Digital filter design

12.1. Finite impulse response (FIR) filters (Class 10)
[Mitra, §§10.1, 10.2, 10.5]

[1] Window design techniques.
[2] Kaiser window design technique.
[3] Equiripple approximations.

12.2. Infinite impulse response (IIR) filters (Class 10-11)
[Mitra, §§9.1-9.6]

[1] Bilinear transform method.
[2] Examples of bilinear transform method.

13. Structures and properties of FIR and IIR filters and review (Class 11)
[Mitra, §§8.1-8.9]

[1] IIR - Direct, parallel and cascaded realizations.
[2] FIR – Direct and cascaded realizations.

[3] Coefficient quantization effects in digital filters
[4] Final review.

14. Final: 6:30-8:30 PM, August 9, 2011

15. References

15.1. Required Texts/Notes

[1] Digital Signal Processing: A Computer-Based Approach, S. K. Mitra, McGraw-Hill, Third edition, 2006.

[2] The Student Edition of MATLAB, Prentice-Hall, New Jersey.

[3] Supplementary class notes, available over the USC Distance Education Network.

All course materials will be distributed via the DEN website (http://den.usc.edu/). Access to the materials at the DEN website requires login with an individual i.d. and password. If you have problems accessing DEN please contact the folks at DEN directly - see http://den.usc.edu/contact/index.htm for contact info and telephone numbers. All students enrolled in the class should have access to all DEN materials, including the streamed lectures.

15.2. Recommended Reading and Some Comments:

There are a huge number of books on DSP, many of which are strongly influenced by Oppenheim and Schafer (see below) and in most cases are inferior. But do take a look in the bookstore or library and see what else is available:

[1] Schaum's Outline of Digital Signal Processing, M. Hays, McGraw-Hill, 1999: This complements Mitra with lots of worked examples and summaries of each topic as well as a large number of additional problems.

[2] Discrete-Time Signal Processing, A. Oppenheim and R. Schafer, Prentice Hall, Second edition, 1999: this is the classic textbook for DSP, and a model for most of the other introductory books on the subject. Highly recommended as a reference.

[3] MATLAB Reference Guide: High-Performance Numeric Computation and Visualization Software, The MathWorks, Inc., South Natick, MA, 1984-92.

[4] Computer-Based Exercises for Signal Processing Using MATLAB 5, J. McClellan (Ed.), Prentice Hall, 1997.

[5] Digital Signal Processing Using MATLAB (r), V. Ingle, J. Proakis, Brooks/Cole Pub. Co., 1999.

[6] A Course in Digital Signal Processing, B. Porat, J. Wiley and Sons, 1996: This is an excellent DSP book – It has been used by Prof. Leahy as a course text, but according to Prof. Leahy: “many students didn’t share my enthusiasm...”

[7] Understanding Digital Signal Processing, R. Lyons, Prentice-Hall, 1996: Excellent reference book with lots of interesting DSP “tricks.”

[8] Digital Signal Processing: Principles, Algorithms and Applications, J. Proakis, D. Manolakis, Prentice-Hall, 2006 (4-th edition): “The new version has moved away from its DSP basics background to give space more advanced topics.” (Amazon user comments)

[9] Digital Filter Design, T. W. Parks and C. S. Burras, J. Wiley & Sons, 1987: A little out of date but it's a useful book.

[10] The Fast Fourier Transform and its Applications, E. O. Brigham, Prentice-Hall, 1988: This is a popular book with a lot of graphic illustrations of discrete convolutions and

Fourier transforms. It is very useful for developing a better understanding of the DFT but it is probably a little too basic to be of long term value.

[11] Digital Signal Processing, R. Roberts and Cliff Mullis, Addison Wesley, 1987: This is a very good book on DSP – it covers a lot of ground but tends to be a little terse.

[12] Introduction to Signal Processing, S. Orfandis, Prentice Hall, 1995: A good book to read, lots of interesting examples.

15.3. Background Material

[1] Signals and Systems Made Ridiculously Simple, Z. Karu, ZiZi Pr Pub., 1995: I think you can download this for free.

[2] Signals and Systems, Oppenheim, Wilsky, et. al., Prentice Hall, 1996.

[3] Linear Algebra and Its Applications, G. Strang, International Thomson Publishing, 1988: See also Strang's lectures at: http://www.youtube.com/watch?v=ZK3O402wf1c

[4] Orthogonal Transforms for Digital Signal Processing, N. Ahmed, Springer-Verlag, 1975.