Digital Speech Processing

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UCSB
Dept. of Electrical and Computer Engineering

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Course Description

This course covers the basic principles of digital speech processing:

- ***Review of digital signal processing***
- ***Fundamentals of speech production and perception***
- ***Basic techniques for digital speech processing:***
  - short - time energy, magnitude, autocorrelation
  - short - time Fourier analysis
  - homomorphic methods
  - linear predictive methods
- ***Speech estimation methods***
  - speech/non-speech detection
  - voiced/unvoiced/non-speech segmentation/classification
  - pitch detection
  - formant estimation
- ***Applications of speech signal processing***
  - Speech coding
  - Speech synthesis
  - Speech recognition/natural language processing

A MATLAB-based ***term project*** will be required for all students taking this course for credit.

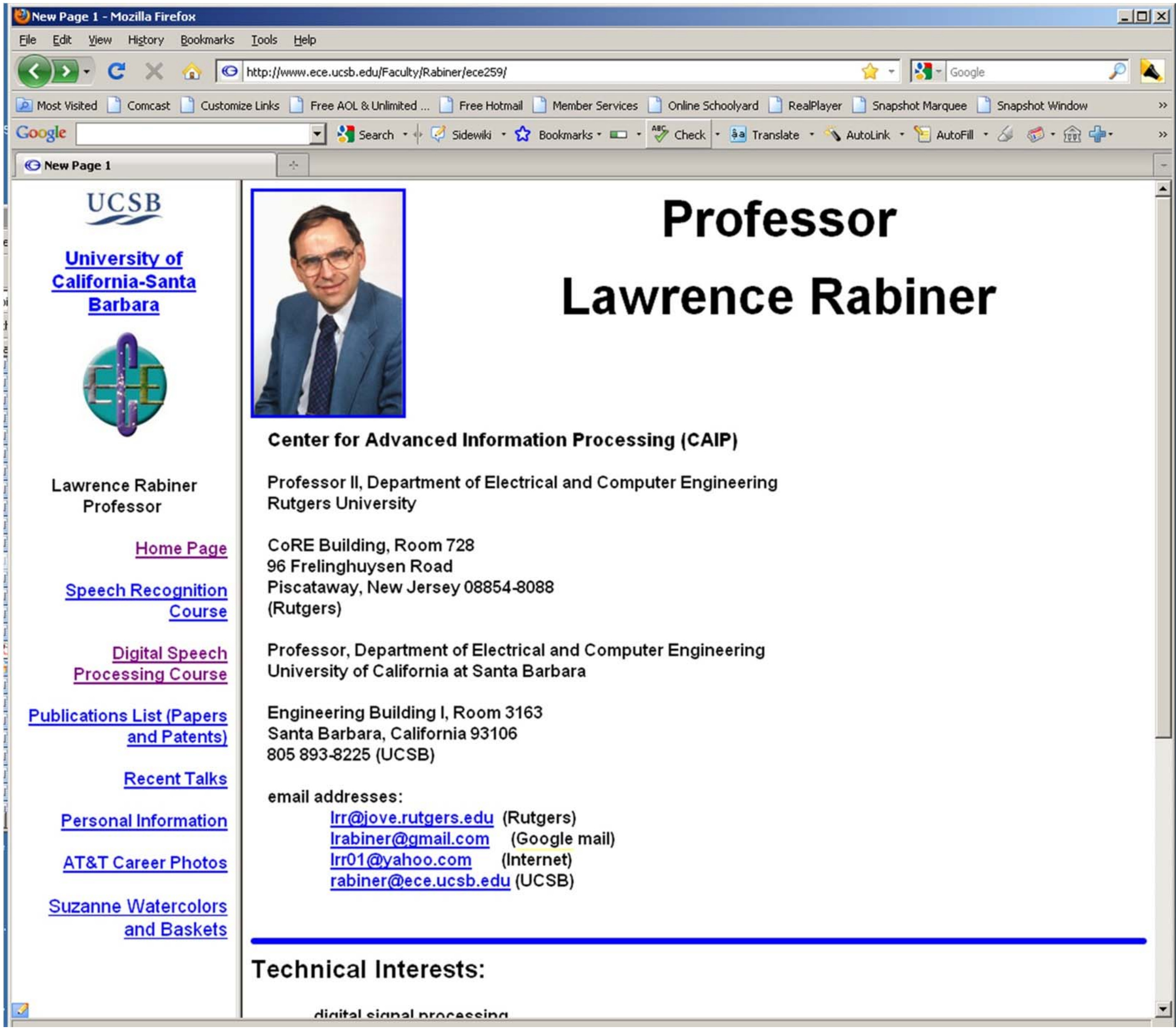
Course Information

- ***Textbook***: L. R. Rabiner and R. W. Schafer, Theory and Applications of Digital Speech Processing, Prentice-Hall Inc., 2011
- ***Grading***:

– Homework	20%
– Term Project	20%
– Mid - Term Exam	20%
– Final Exam	40%

- ***Prerequisites***: Basic Digital Signal Processing, good knowledge of MATLAB
- ***Time and Location***: Tuesday, Thursday, 10:00 am to 11:20 am, Phelps 1437.
- ***Course Website:*** www.ece.ucsb.edu/Faculty/Rabiner/ece259
- ***Office Hours:*** Tuesday, 1:00-3:00 pm

Web Page for Speech Course



Click on **Digital Speech Processing Course** on left-side panel

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Lawrence Rabiner
Professor

Home Page

Speech Recognition Course

Digital Speech Processing Course

Publications List (Papers and Patents)

Recent Talks

Personal Information

AT&T Career Photos

Suzanne Watercolors and Baskets

Digital Speech Processing Course (Winter 2011)

No Cheating Policy:

No cheating declaration: (No Cheating Policy.pdf)

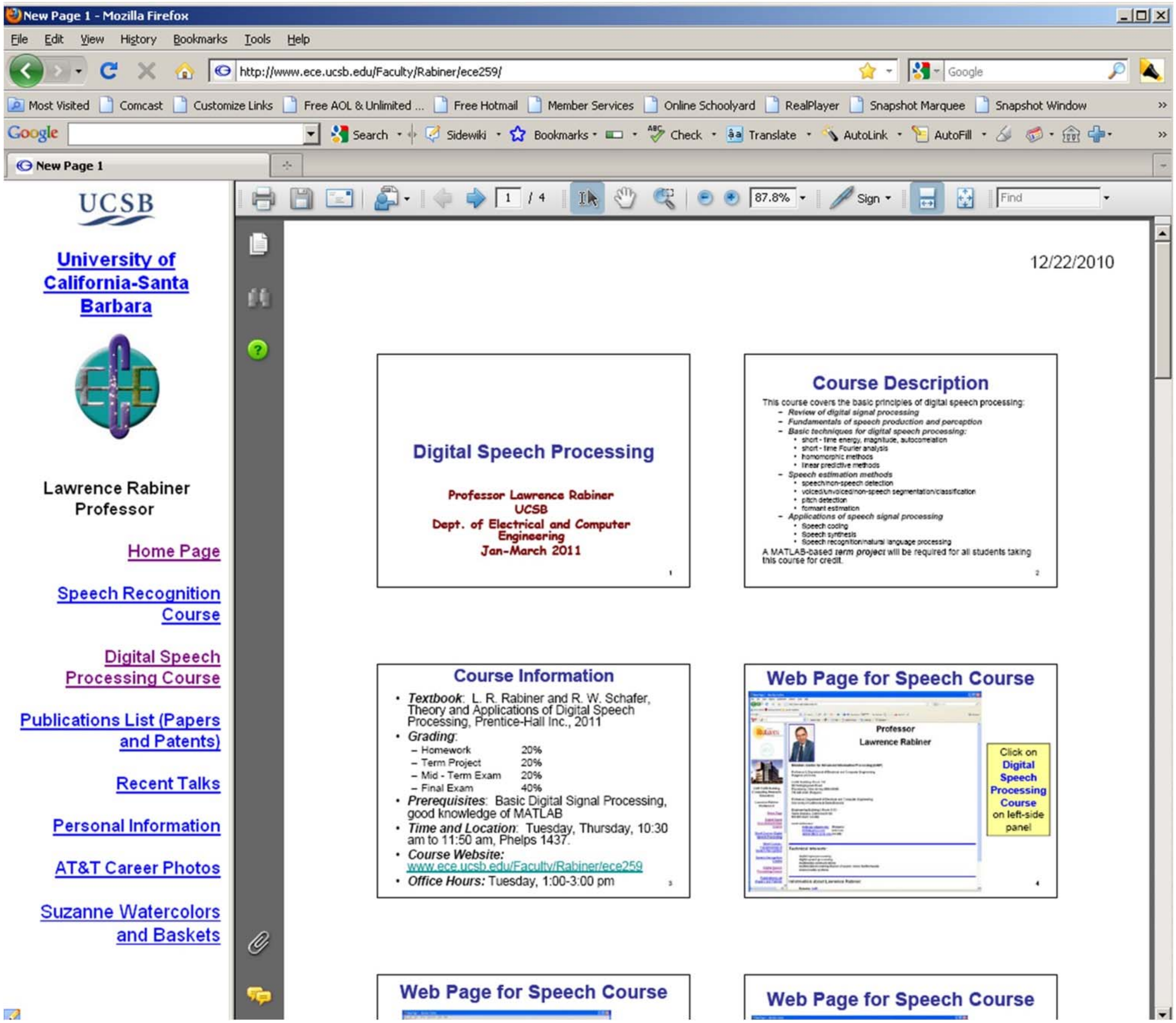
Monograph on Digital Speech Processing: Introduction to Digital Speech processing.pdf

Lectures:

Introductory Material: (basic course material_winter_2011.pdf), 6-to-a-page: (basic course material_winter_2011_6tp.pdf)
Lecture 1: Introduction to Digital Speech Processing: (Lecture 1_fall_2010.pdf) : 6-to-a-page:(Lecture 1_fall_2010_6tp.pdf)
Lecture 2: Review of DSP Fundamentals:(Lecture 2_fall_2010.pdf) : 6-to-a-page:(Lecture 2_fall_2010_6tp.pdf)
Lecture 3: Acoustic Theory of Speech Production:(Lecture 3_fall_2010.pdf) : 6-to-a-page:(Lecture 3_fall_2010_6tp.pdf)
Lecture 4: Speech Perception--Auditory Models, Sound Perception Models, MOS Methods:(Lecture 4_fall_2010.pdf): 6-to-a-page: (Lecture 4_fall_2010_6tp.pdf)
Lectures 5-6:Sound Propagation in the Vocal Tract: (Lectures 5-6_fall_2010.pdf) : 6-to-a-page: (Lectures 5-6_fall_2010_6tp.pdf)
Lectures 7-8:Time Domain Methods in Speech Processing: (Lectures 7-8_fall_2010.pdf) : 6-to-a-page: (Lectures 7-8_fall_2010_6tp.pdf)
Methods of Pitch Period Estimation:
Lecture 9: Short-Time Fourier Transform (STFT) Concepts: (Lecture9_fall_2010.pdf) : 6-to-a-page: (Lecture9_fall_2010_6tp.pdf)
Lecture 10: Short Time Fourier Analysis Methods--Filter Bank Summation and Overlap Add: (Lecture 10_fall_2010.pdf): 6-to-a-page: (Lecture10_fall_2010_6tp.pdf)
Lecture 11: Speech Representations Based on STFT Analysis-Synthesis Methods:(Lecture 11_fall_2010.pdf) : 6-to-a-page: (Lecture 11_fall_2010_6tp.pdf)
Lecture 12: Homomorphic Speech Processing:(Lecture 12_fall_2010.pdf) : 6-to-a-page: (Lecture 12_fall_2010_6tp.pdf)
Lecture 13: Linear Predictive Coding (LPC) Methods:(Lecture 13_fall_2010.pdf) : 6-to-a-page:(Lecture 13_fall_2010_6tp.pdf)

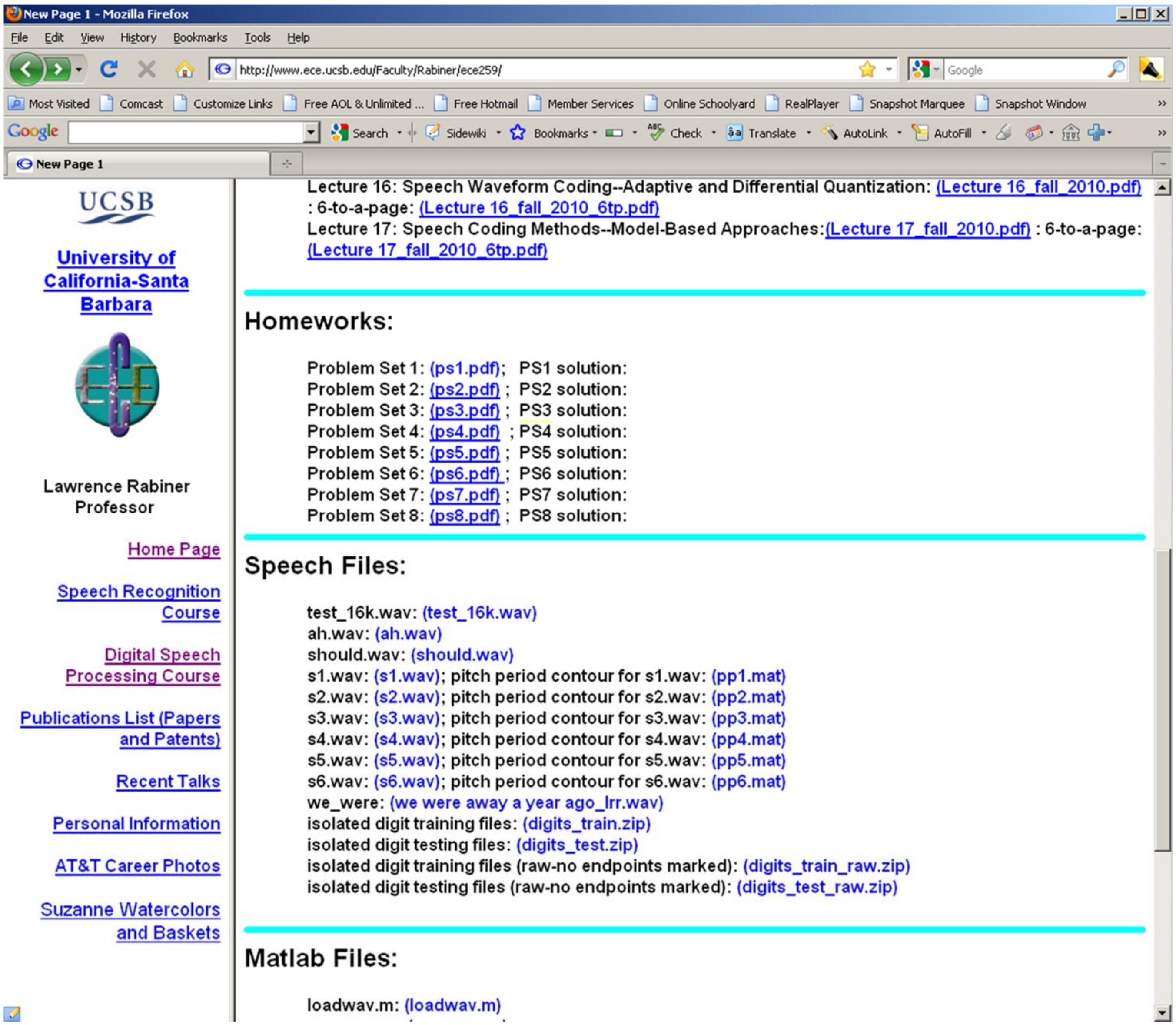
Download course lecture slides

Web Page for Speech Course



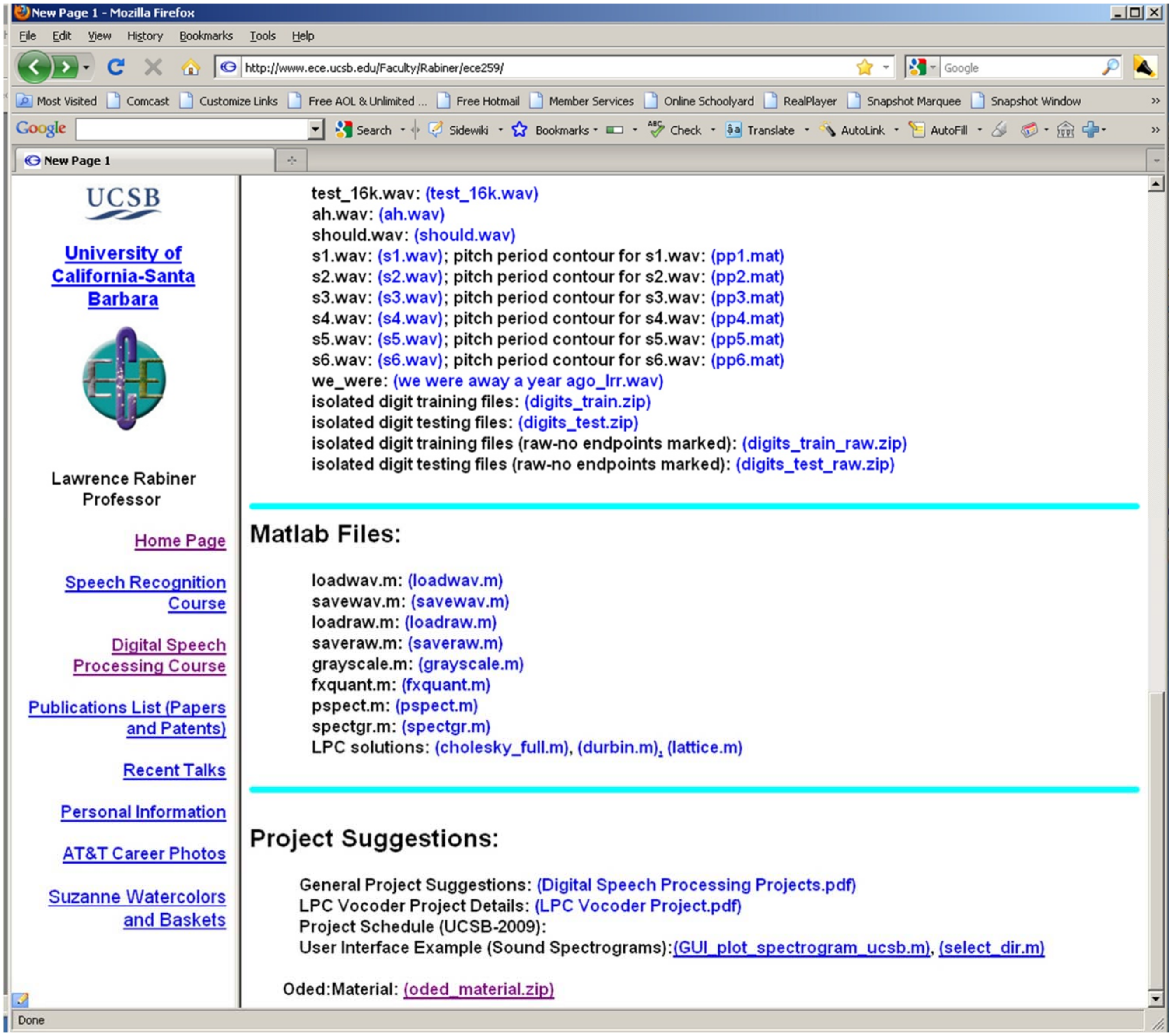
Course lecture slides (6-to-page)

Web Page for Speech Course



Download homework assignments, speech files

Web Page for Speech Course



Download MATLAB (.m) files; Examine Project Suggestions

Course Readings

Required Course Textbook:

- L. R. Rabiner and R. W. Schafer, *Theory and Applications of Digital Speech Processing,* Prentice-Hall Inc., 2011

Recommended Supplementary Textbook:

- T. F. Quatieri, *Principles of Discrete - Time Speech Processing,* Prentice Hall Inc, 2002

Matlab Exercises:

- C. S. Burrus et al, *Computer-Based Exercises for Signal Processing using Matlab*, Prentice Hall Inc, 1994
- J. R. Buck, M. M. Daniel, and A. C. Singer*, Computer Explorations in Signals and Systems using Matlab,* Prentice Hall Inc, 2002

Recommended References

- J. L. Flanagan, *Speech Analysis, Synthesis, and Perception,* Springer -Verlag, 2nd Edition, Berlin, 1972
- J. D. Markel and A. H. Gray, Jr., *Linear Prediction of Speech,* Springer-Verlag, Berlin, 1976
- B. Gold and N. Morgan, *Speech and Audio Signal Processing,* J. Wiley and Sons, 2000
- J. Deller, Jr., J. G. Proakis, and J. Hansen, *Discrete - Time Processing of Speech Signals,* Macmillan Publishing, 1993
- D. O'Shaughnessy, *Speech Communication, Human and Machine,* Addison-Wesley, 1987
- S. Furui and M. Sondhi, *Advances in Speech Signal Processing,* Marcel Dekker Inc, NY, 1991
- R. W. Schafer and J. D. Markel, Editors, *Speech Analysis,* IEEE Press Selected Reprint Series, 1979
- D. G. Childers, *Speech Processing and Synthesis Toolboxes,* John Wiley and Sons, 1999
- K. Stevens, *Acoustic Phonetics,* MIT Press, 1998
- J. Benesty, M. M. Sondhi and Y. Huang, Editors, *Springer Handbook of Speech Processing and Speech Communication,* Springer, 2008.

References in Selected Areas of Speech Processing

Speech Coding:

- A. M. Kondoz, *Digital Speech: Coding for Low Bit Rate Communication Systems-2nd Edition,* John Wiley and Sons, 2004
- W. B. Kleijn and K. K. Paliwal, Editors, *Speech Coding and Synthesis,* Elsevier, 1995
- P. E. Papamichalis, *Practical Approaches to Speech Coding,* Prentice Hall Inc, 1987
- N. S. Jayant and P. Noll, *Digital Coding of Waveforms,* Prentice Hall Inc, 1984

References in Selected Areas of Speech Processing

Speech Synthesis:

- T. Dutoit, *An Introduction to Text - To-Speech Synthesis,* Kluwer Academic Publishers, 1997
- P. Taylor, *Text-to-Speech Synthesis,* Cambridge University Press, 2008
- J. Allen, S. Hunnicutt, and D. Klatt, *From Text to Speech,* Cambridge University Press, 1987
- Y. Sagisaka, N. Campbell, and N. Higuchi, *Computing Prosody,* Springer Verlag, 1996
- J. VanSanten, R. W. Sproat, J. P. Olive and J. Hirschberg, Editors, *Progress in Speech Synthesis,* Springer Verlag, 1996
- J. P. Olive, A. Greenwood, and J. Coleman, *Acoustics of American English,* Springer Verlag, 1993

References in Selected Areas of Speech Processing

Speech Recognition:

- L. R. Rabiner and B. H. Juang, *Fundamentals of Speech Recognition,* Prentice Hall Inc, 1993
- X. Huang, A. Acero and H-W Hon, *Spoken Language Processing,* Prentice Hall Inc, 2000
- F. Jelinek, *Statistical Methods for Speech Recognition,* MIT Press, 1998
- H. A. Bourlard and N. Morgan, *Connectionist Speech Recognition-A Hybrid Approach,* Kluwer Academic Publishers, 1994
- C. H. Lee, F. K. Soong, and K. K. Paliwal, Editors, *Automatic Speech and Speaker Recognition,* Kluwer Academic Publisher, 1996

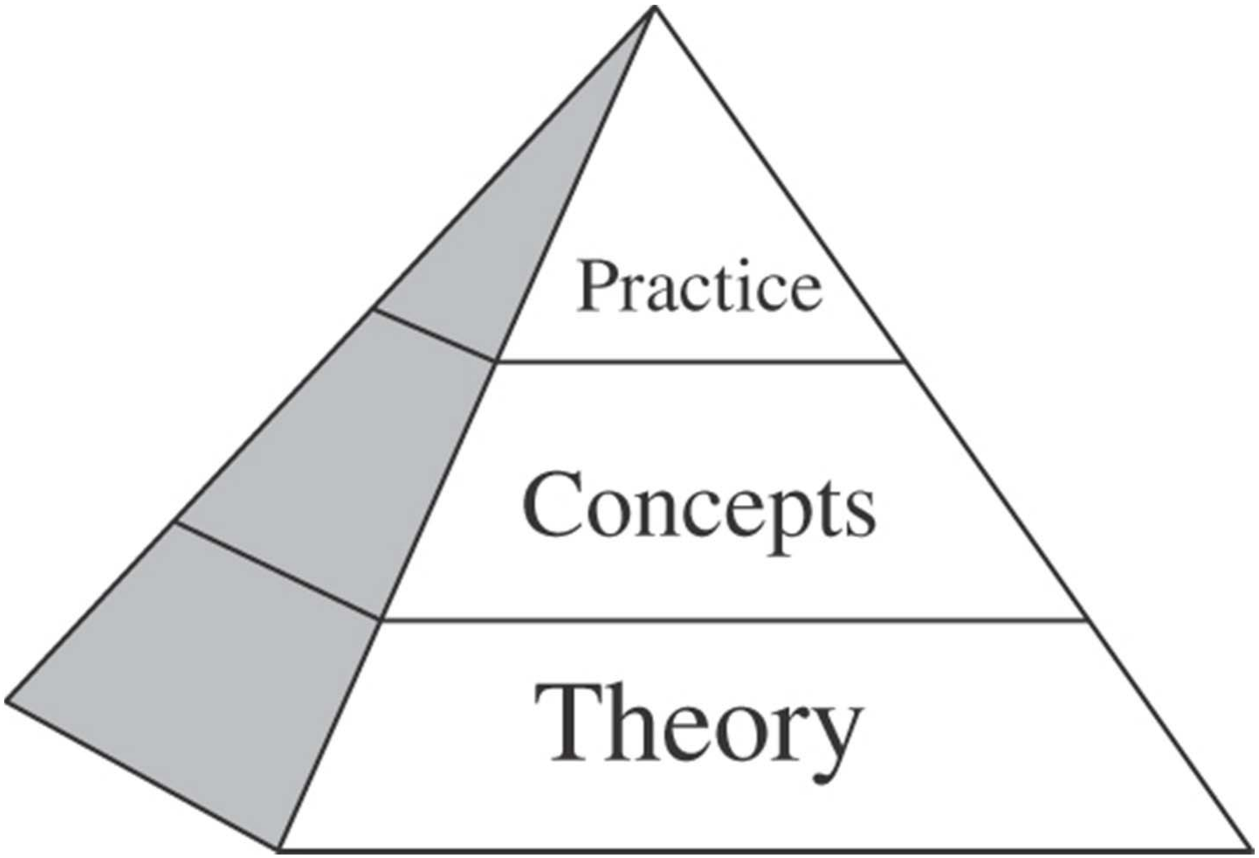
References in Digital Signal Processing

- A. V. Oppenheim and R. W. Schafer, *Discrete - Time Signal Processing, 3rd Ed.,* Prentice-Hall Inc, 2010
- L. R. Rabiner and B. Gold, *Theory and Application of Digital Signal Processing,* Prentice Hall Inc, 1975
- S. K. Mitra, *Digital Signal Processing-A Computer-Based Approach,* Third Edition, McGraw Hill, 2006
- S. K. Mitra, *Digital Signal Processing Laboratory Using Matlab,* McGraw Hill, 1999

The Speech Stack

Speech Applications **— coding, synthesis, recognition, understanding, verification, language translation, speed-up/slow-down**
Speech Algorithms **— speech-silence (background), voiced-unvoiced, pitch detection, formant estimation**
Speech Representations **— temporal, spectral, homomorphic, LPC**
Fundamentals **— acoustics, linguistics, pragmatics, speech production/perception**

Digital Speech Processing



Ability to implement theory and concepts in working code (MATLAB, C, C++)

Basic understanding of how theory is applied

Mathematics, derivations, signal processing

Need to understand speech processing at all three levels

Course Outline – ECE 259A – Speech Processing

- Jan 10 - Lecture 1,Basic Course Material; Introduction to Digital Speech Processing
- Jan 12 - Lecture 2a, Review of DSP Fundamentals
- Jan 17 - Lecture 2b, Review of DSP Fundamentals
- Jan 19 - Lecture 3a, Acoustic Theory of Speech Production
- Jan 24 - Lecture 3b, Lecture 4, Speech Perception—Auditory Models
- Jan 26 - Lecture 5, Sound Propagation in the Vocal Tract -- Part 1
- Jan 31 - Lecture 6, Sound Propagation in the Vocal Tract -- Part 2
- Feb 2 - Lecture 7, Time Domain Methods -- Part 1
- Feb 7 - Lecture 8, Time Domain Methods -- Part 2
- Feb 9 - Lecture 9, Frequency Domain Methods -- Part 1
- Feb 14 - Lecture 10-11, Frequency Domain Methods -- Part 2
- Feb 16 - Mid - Term Exam
- Feb 21 - Lecture 12a, Homomorphic Speech Processing -- Part 1
- Feb 23 - Lecture 12b, Homomorphic Speech Processing -- Part 2
- Feb 28 - Lecture 13, Linear Predictive Coding (LPC) -- Part 1
- Mar 1 - Lecture 14, Linear Predictive Codeing (LPC) -- Part 2
- Mar 6 - Lecture_Algorithms
- Mar 8 - Lecture 15, Speech Waveform Coding -- Part 1
- Mar 13 - Lecture 16, Speech Waveform Coding -- Part 2
- Mar 15 - Term Project Presentations (8-12 noon)
- Mar 20 - Final Exam (8 am-11 am)

Other Potential Topics for Discussion/Term Projects

- Sinusoidal modeling of speech
- Speech modification and enhancement—slowing down and speeding up speech, noise reduction methods
- Speaker verification methods
- Music coding including MP3 and AAC standards-based methods
- Pitch detection methods

Term Project

- All registered students are required to do a term project. This term project, implemented ***using Matlab,*** must be a speech or audio processing system that accomplishes a simple or even a complex task—e.g., pitch detection, voiced-unvoiced detection, speech/silence classification, speech synthesis, speech recognition, speaker recognition, helium speech restoration, speech coding, MP3 audio coding, etc.
- Every student is also required to make a 10-minute Power Point *presentation* of their term project to the entire class. The presentation must include:
  - A short description of the project and its objectives
  - An explanation of the implemented algorithm and relevant theory
  - A demonstration of the ***working*** program – i.e., results obtained when running the program

Suggestions for Term Projects

1. Pitch detector – time domain, autocorrelation, cepstrum, LPC, etc.
2. Voiced/Unvoiced/Silence detector
3. Formant analyzer/tracker
4. Speech coders including ADPCM, LDM, CELP, Multipulse, etc.
5. N-channel spectral analyzer and synthesizer – phase vocoder, channel vocoder, homomorphic vocoder
6. Speech endpoint detector
7. Simple speech recognizer – e.g. isolated digits, speaker trained
8. Speech synthesizer – serial, parallel, direct, lattice
9. Helium speech restoration system
10. Audio/music coder
11. System to speed up and slow down speech by arbitrary factors
12. Speaker verification system
13. Sinusoidal speech coder
14. Speaker recognition system
15. Speech understanding system
16. Speech enhancement system (noise reduction, post filtering, spectral flattening)

MATLAB Computer Project

The requirements for this project are a short description of the problem containing relevant mathematical theory and objectives of the project, a listing (with sufficient documentation and comments) of the program, and a demonstration that the program works properly.