KING FAHD UNIVERSITY OF PETROLEUM AND MINERALS
ELECTRICAL ENGINEERING DEPARTMENT
First Semester (091)

EE 562 Digital Signal Processing I

- ## Course Outline:

1. **Introduction, Review and Background** (1.0 week)
2. **Discrete time signals and systems** (1.0 week)
3. **Z-Transform and applications** (2.0 weeks)
4. **Frequency analysis of signals and systems** (2.5 weeks)
5. **Discrete Fourier transform** (1.5 weeks)
6. **Fast DFT Algorithms** (1.0 week)
7. **Implementing discrete time systems** (1.0 week)
8. **Design of digital filters** (3.0 weeks)
9. **Sampling and Reconstruction** (1.0 week)
10. **Multirate DSP** (1.0 week)

- **Textbook:** John G. Proakis and Dimitris G. Manolakis, "Digital Signal Processing: Principles, Algorithms, and Applications," 3rd Ed., Prentice Hall, 1996.

- ## References:

1. S. K. Mitra, *Digital Signal Processing: A Computer-Based Approach*, 3rd Edition, McGraw Hill, 2006.
2. A. V. Oppenheim, R. W. Schafer and J. R. Buck, *Discrete-Time Signal Processing*, 2nd Ed., Prentice Hall, 1999.
3. V. K. Ingle and J. G. Proakis, *Digital Signal Processing using MATLAB*, PWS Publishing Company, 1997.
4. J. H. McClellan, R. W. Schafer and M. A. Yoder, *DSP First: A Multimedia Approach*, Prentice Hall, 1998.
5. K. G. Beauchamp, *Transforms for Engineers: A Guide to Signal Processing*, Clarendon Press, 1987.

- **Homework Assignments** will be issued about once every two weeks.

- ## Grading Policy

1. Homework 25 %
2. Midterm Exam 20 %
3. Project 25 %
4. Final Exam 30 %

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KFUPM

EE 562 **EE Department** October 3, 2009

FINAL PROJECT

For the final project, every student will be assigned an "advanced topic" in digital signal processing and use one or two references to help you get started. Each student will then:

- Read up on the topic to understand the main issues,
- Perform matlab simulations to investigate these issues,
- Write a report that describes the topic (in a somewhat tutorial manner) and include relevant matlab simulations.

There will be an oral presentation. While I will be available for advice, it will be your responsibility to identify these main issues, since I see this as an important component of performing research in a new area or examining a new technique for practical application.

Given the review process, it is important that each report is clearly written and understandable by no-expert in the topic area.

Finally, to maximize the usefulness of the project, I would like each student to think about topics that might be in line with his research interest.

Below is a list of potential project topics with a short explanation to get you started.

1. **DTMF decoder**: Convert a recording of 'touch tones' from a real telephone into the corresponding set of digits. Mitra's book includes some simple code for DTMF detection (at the start of chapter 11), but this doesn't deal with the case of multiple tones in a single soundfile, or necessarily handle the varying signal quality and spurious sounds (such as dialtone) in some of the examples.

2. **Channel equalization**: This is a classic signal processing problem, where the signal has been subject to some unknown filter, and the goal is to infer what that filter was, then invert its effects.

3. **Signal denoising**: Signals get corrupted by noise all kinds of ways - by electrical interference, by mechanical damage of recording media, or simply because there were unwanted sounds present during the original recording. Some approaches to noise reduction involve estimating the steady-state noise spectrum (by looking during 'gaps' in the speech), then designing a Wiener filter to optimize the signal-to-noise ratio.

4. **Pitch extraction**: This is a widespread problem in speech and music processing; identifying the local periodicity of a sound with a perceived pitch. Autocorrelation is the classic method, but it makes a lot of common errors, so there are many approaches to improving it.

5. **Watermarking**: There are various motivations for 'hiding' data in a soundfile without making the alteration audible. One is watermarking, so that a sound can be recognized as 'valid' without knowing its content ahead of time. Another is embedding copyright markers that cannot easily be removed by counterfeiters.

6. **Compression**: Audio signal compression is a current hot topic. This project would involve implementing one or more simple compression schemes, and investigating how they perform for different kinds of signals, as well as the kinds of distortion they introduce. One of the classic, high-efficiency algorithms is ADPCM.

7. **Time-delay angle-of-arrival estimation**: We use our two ears to be able to detect the direction from which a sound arrives. The strongest cue is probably the slight time differences that occur due to the finite speed of sound traveling to each side. Cross-correlation can reveal this time difference, and indicate the azimuth from which sounds occur.

8. **Doubletalk detection**: In speech processing we frequently assume that the signal contains just a single voice; in many cases this is not true, for instance when people interrupt one another on a telephone call or in a meeting. Separating these voices is hard, but we would like at least to be able to detect when it is happening, so we know not to attempt normal processing.

9. **Image/video coding and transmission**: Still image coding, video coding, distributed source coding, image/video transmission, wavelet image processing.

10. **Other topics of interest**:

    • MIMO-OFDM (Channel Estimation), (Low Complexity Zero-Padding Zero-Jamming DMT Systems), (A phase noise compensation scheme for OFDM wireless systems).

    • Frequency Domain Filtering (DFE), (Block LMS algorithm and its FFT based Fast Implementation), (Adaptive Subband Filtering), (Subspace Approach for Mobile Positioning), (Subspace-based Estimation of DOA with know directions).

    • Digital Filter Design (Optimal and non-optimal designs).