Digital Signal Processing

Prof. Nizamettin AYDIN

naydin@yildiz.edu.tr

http://www.yildiz.edu.tr/~naydin

Course Details

- Course Code : 0113620
- Course Name: Digital Signal Processing
  (Sayısal İşaret İşleme)
- Instructor : Nizamettin AYDIN

Assesment

• Midterm 1	:	20%
• Midterm 2	:	20%
• Homework	:	20%
• Final	:	40%

Course Outline

1. Introduction.
Mathematical Representation of Signals. Mathematical Representation of Systems.
2. Sinusoids.
Review of Sine and Cosine Functions. Sinusoidal Signals. Sampling and Plotting Sinusoids. Complex Exponentials and Phasors. Phasor Addition. Time Signals.
3. Spectrum Representation.
The Spectrum of a Sum of Sinusoids. Beat Notes. Periodic Waveforms. Fourier Series Analysis and Synthesis. Time-Frequency Spectrum. Frequency Modulation.
4. Sampling and Aliasing.
Sampling. Spectrum View of Sampling and Reconstruction. Discrete-to-Continuous Conversion. The Sampling Theorem.
5. FIR Filters.
Discrete-Time Systems. The Running Average Filter. The General FIR Filter. Implementation of FIR Filters. Linear Time-Invariant (LTI) Systems. Convolution and LTI Systems. Cascaded LTI Systems. Example of FIR Filtering.
6. Frequency Response of FIR Filters.
Sinusoidal Response of FIR Systems. Superposition and the Frequency Response. Steady State and Transient Response. Properties of the Frequency Response. Graphical Representation of the Frequency Response. Cascaded LTI Systems. Running-Average Filtering. Filtering Sampled Continuous-Time Signals.
7. z-Transforms.
Definition of the z-Transform. The z-Transform and Linear Systems. Properties of the z-Transform. The z-Transform as an Operator. Convolution and the z-Transform. Practical Bandpass Filter Design. Properties of Linear Phase Filters.

Course Outline

8. IIR Filters.
The General IIR Difference Equation. Time-Domain Response. System Function of an IIR Filter. Poles and Zeros. Frequency Response of an IIR Filter. The Inverse z-Transform and Some Applications. Second-Order Filters. Frequency Response of Second-Order IIR Filter. Example of an IIR Lowpass Filter.
9. Continuous-Time Signals and LTI Systems.
Continuous-Time Signals. The Unit Impulse. Continuous-Time Systems. Linear Time-Invariant Systems. Impulse Responses of Basic LTI Systems. Convolution of Impulses. Evaluating Convolution Integrals. Properties of LTI Systems.
10. The Frequency Response.
The Frequency Response Function for LTI Systems. Response to Real Sinusoidal Signals. Ideal Filters. Application of Ideal Filters. Time-Domain or Frequency-Domain?
11. Continuous-Time Fourier Transform.
Definition of the Fourier Transform. The Fourier Transform and the Spectrum. Examples of Fourier Transform Pairs. Properties of Fourier Transform Pairs. The Convolution Property. Basic LTI Systems. The Multiplication Property.
12. Filtering, Modulation, and Sampling.
Linear Time-Invariant Systems. Sinewave Amplitude Modulation. Sampling and Reconstruction.
13. Computing the Spectrum.
Finite Fourier Sum. Time-windowing. Analysis of a Sum of Sinusoids. Discrete Fourier Transform. Spectrum Analysis of Finite-Length Signals. Spectrum Analysis of Periodic Signals. The Spectrogram. The Fast Fourier Transform (FFT).

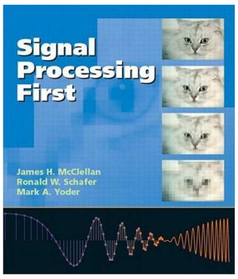
Main course book

Signal Processing First

by James H McClellan,
Ronald W. Schaffer
and Mark A. Yoder.

Published by Prentice Hall.

Isbn: 0-13-120265-0



some recommended books

- **Understanding Digital Signal Processing** by Richard G. Lyons.
- **The Scientist and Engineer's and Guide to Digital Signal Processing** by Steven W. Smith.
- **Digital Signal Processing and the Microcontroller** by Dale Grover and John R. (Jack) Deller with illustrations by Jonathan Roth.
- **Discrete-Time Signal Processing** by A. V. Oppenheim and R. W. Schafer.
- **Digital Signal Processing: Principles, Algorithms, and Applications** by J. G. Proakis and D. G. Manolakis.
- **Digital Signal Processing in Communication Systems** by Marvin E. Frerking.
- **Multirate Digital Signal Processing** by R. E. Crochiere and L. R. Rabiner.
- **Theory and Application of Digital Signal Processing** by Rabiner and Gold. A comprehensive, industrial-strength DSP reference book.
- **Digital Signal Processing** by Alan V. Oppenheim and Ronald W. Schafer. Another industrial-strength reference.
- **Disrete-Time Signal Processing** by Alan V. Oppenheim and Ronald W. Schafer **Digital Signal Processing** by William D. Stanley.

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Digital Signal Processing (DSP)

Basics:

What is DSP?

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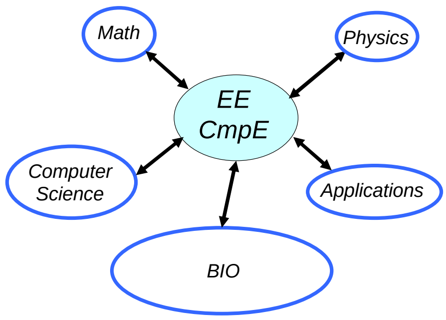
CONVERGING FIELDS



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COURSE OBJECTIVE

- Students will be able to:
- Understand **mathematical** descriptions of signal processing **algorithms**
- Express those algorithms as computer **implementations** (MATLAB)

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Digital Signal Processing (DSP)

Dictionary definitions of the words in DSP:

- ***Digital***
  - ***operating by the use of discrete signals to represent data in the form of numbers***
- ***Signal***
  - ***a variable parameter by which information is conveyed through an electronic circuit***
- ***Processing***
  - ***to perform operations on data according to programmed instructions***
- **So a simple definition of DSP could be:**
  - **changing or analysing information which is measured as discrete sequences of numbers**
- **Unique features of DSP as opposed to ordinary digital processing:**
  - **signals come from the real world**
    - **this intimate connection with the real world leads to many unique needs such as the need to react in real time and a need to measure signals and convert them to digital numbers**
  - **signals are discrete**
    - **which means the information in between discrete samples is lost**

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WHY USE DSP ?

- **Versatility:**
  - digital systems can be reprogrammed for other applications
  - digital systems can be ported to different hardware
- **Repeatability:**
  - digital systems can be easily duplicated
  - digital systems do not depend on strict component tolerances
  - digital system responses do not drift with temperature
- **Simplicity:**
  - some things can be done more easily digitally than with analogue systems

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DSP is used in a very wide variety of applications

- Radar, sonar, telephony, audio, multimedia, communications, ultrasound, process control, digital camera, digital tv, Telecommunications, Sound & Music, Fourier Optics, X-ray Crystallography, Protein Structure & DNA, Computerized Tomography, Nuclear Magnetic Resonance: MRI,Radioastronomy
- All these applications share some common features:
  - they use a lot of maths (multiplying and adding signals)
  - they deal with signals that come from the real world
  - they require a response in a certain time
- Where general purpose DSP processors are concerned, most applications deal with signal frequencies that are in the audio range

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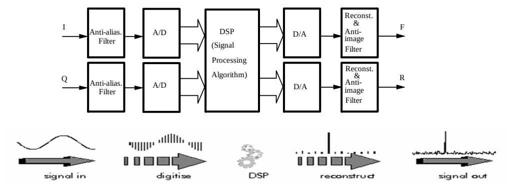
Fundamental concepts in DSP

- DSP applications deal with analogue signals
  - the analogue signal has to be converted to digital form



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- The analogue signal - a continuous variable defined with infinite precision - is converted to a discrete sequence of measured values which are represented digitally
- Information is lost in converting from analogue to digital, due to:
  - inaccuracies in the measurement
  - uncertainty in timing
  - limits on the duration of the measurement
- These effects are called quantisation errors

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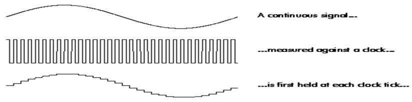
- The continuous analogue signal has to be held before it can be sampled



- Otherwise, the signal would be changing during the measurement
- Only after it has been held can the signal be measured, and the measurement converted to a digital value

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Sampling related concepts

- Over/exact/under sampling
- Regular/irregular sampling
- Linear/Logarithmic sampling
- Aliasing
- Anti-aliasing filter
- Image
- Anti-image filter

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Steps for digitization/reconstruction of a signal

- Band limiting (LPF)
- Sampling / Holding
- Quantization
- Coding

These are basic steps for A/D conversion

- D/A converter
- Sampling / Holding
- Image rejection

These are basic steps for reconstructing a sampled digital signal

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Digital data: end product of A/D conversion and related concepts

- Bit: least digital information, binary 1 or 0
- Nibble: 4 bits
- Byte: 8 bits, 2 nibbles
- Word: 16 bits, 2 bytes, 4 nibbles
- Some jargon:
  - integer, signed integer, long integer, 2s complement, hexadecimal, octal, floating point, etc.

Sampling

- The sampling results in a discrete set of digital numbers that represent measurements of the signal
  - usually taken at equal intervals of time
- Sampling takes place after the hold
  - The hold circuit must be fast enough that the signal is not changing during the time the circuit is acquiring the signal value
- We don't know what we don't measure
- In the process of measuring the signal, some information is lost