Advanced Digital Signal Processing

Part 1: Introduction

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Digital Signal Processing and System Theory

Christian-Albrechts-Universität zu Kiel

Contents of the Lecture – Part 1

- Introduction
- Digital processing of continuous-time signals
  - Sampling and sampling theorem (repetition)
  - Quantization
  - Analog-to-digital (AD) and digital-to-analog (DA) conversion
- Efficient FIR filter structures
- DFT and FFT
  - Leakage effect
  - Windowing
  - FFT structure
  - Spectral refinement
- Digital filters
  - FIR filters
  - IIR filters
  - Finite word-length effects

Contents of the Lecture – Part 2

- Multi-rate digital signal processing
  - Decimation and interpolation
  - Filters in sampling rate alteration systems
  - Polyphase decomposition and efficient structures
  - Digital filterbanks

Literature

Books:

- J. G. Proakis, D. G. Manolakis: ***Digital Signal Processing: Principles, Algorithms, and Applications***, Prentice Hall, 1996, 3rd edition
- S. K. Mitra: ***Digital Signal Processing: A Computer-Based Approach***, McGraw Hill Higher Education, 2000, 2nd edition
- A. V. Oppenheim, R. W. Schafer: ***Discrete-Time Signal Processing***, Prentice Hall, 1999, 2nd edition
- M. H. Hayes: ***Statistical Signal Processing and Modeling***, John Wiley and Sons, 1996

Boundary Conditions of the Lecture

Basics

- "3+1 lecture", what means three hours of lecture and one hour of exercise each week.
- 5 ECTS points
- Oral exam during the exam period (rather long for electrical engineering)
- Register via the official university system and book a slot for you via the DSS booking system
- All material (slides, examples) is available in the corresponding section of the DSS website

https://dss-kiel.de/

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Lecture "Advanced Digital Signal Processing"

Basic Information	
Lecturers:	Gerhard Schmidt (lecture) and Owe Wisch (exercise)
Room:	Building F, room F-SR-I and online (via zoom) for students that are not able to attend in presence
Language:	English
Target group:	Students in electrical engineering and computer engineering
Prerequisites:	Basic Knowlegde about signals and systems
Contents:	Students attending this lecture should be able to implement efficient and robust signal processing structures. Knowledge about moving from the analog to the digital domain and vice versa including the involved effects (and trap doors) should be acquired. Also differences (advantages and disadvantages) between time and frequency domain approaches should be learnt. Topic overview: • Digital processing of continuous-time signals • Efficient FIR structures • DFT and FFT • Digital filters (FIR filters/IIR filters) • Multirate digital signal processing
References:	J. G. Proakis, D. G. Manolakis: Digital Signal Processing: Principles, Algorithms, and Applications, Prentice Hall, S. K. Mitra: Digital Signal Processing: A Computer-Based Approach, McGraw Hill Higher Education, 2000 A. V. Oppenheim, R. W. Schafer: Discrete-time signal processing, Prentice Hall, 1999, 2nd edition M. H. Hayes: Statistical Signal Processing and Modeling, John Wiley and Sons, 1996

Boundary Conditions of the Lecture

People:

- Lecture
  - Gerhard Schmidt

    E-Mail: gus@tf.uni-kiel.de

    Web: Gerhard Schmidt (dss-kiel.de)

- Exercise
  - Christian Kanarski

    E-Mail: chk@tf.uni-kiel.de

    Web: Christian Kanarski (dss-kiel.de)

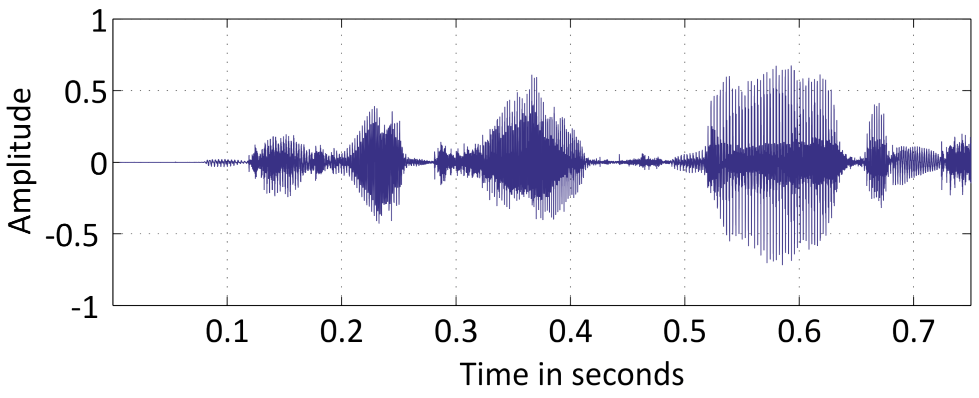
Further people that can be asked for help (screenshot from the DSS website)

What does "Digital Signal Processing" mean?

The term "Signal" in "Digital Signal Processing":

- Physical quantity that varies with time, space, or any other independent variable
- Mathematically: Function of one or more independent variables, $v(t) = 5t,\ v(n) = 20\,n^2,\ \ldots$
- Examples: Temperature over time $x(t)$, brightness (luminance) of an image $l(x,y)$, pressure of a sound wave over $p(x,y,z)$ or $p(x,y,z,t)$.

Speech signal:



What does "Digital Signal Processing" mean?

The term "Signal Processing" in "Digital Signal Processing":

- Passing the signal through a system
- Examples:
  - Modification of the signal (filtering, interpolation, noise reduction, equalization, …)
  - Prediction, transformation to another domain (e.g. Fourier transform)
  - Numerical integration and differentiation
  - Determination of mean value, correlation, probability density function, …
- Properties of the system (e.g. linear/nonlinear) determine the properties of the whole processing operation
- The definition of a system also includes:
  - ***Software*** realizations of operations on a signal, which are carried out on a digital computer (software implementation of the system),
  - digital ***hardware*** realizations (logic circuits) configured such that they are able to perform the processing operation, or
  - most general definition: a ***combination of both***.

Signals, Systems and Signal Processing – Part 3

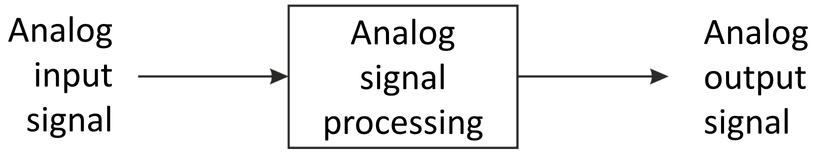
What does "Digital Signal Processing" mean?

Finally "Digital Signal Processing":

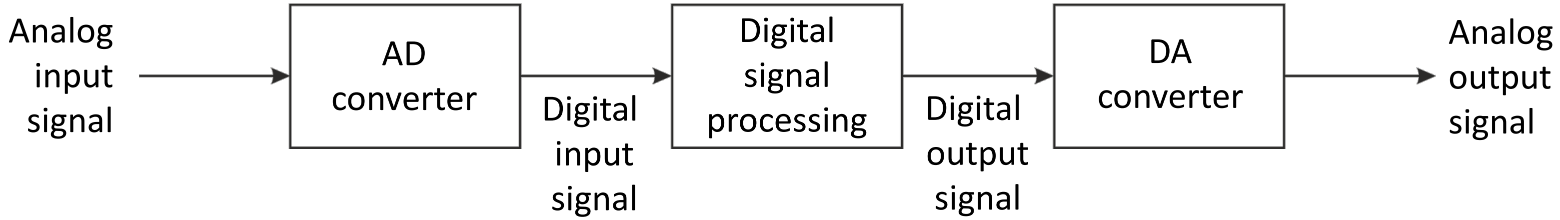
- Processing of signals by digital means (software and/or hardware)
- This includes:
  - ***Conversion*** from the analog to the digital domain and back (physical signals are analog)
  - Mathematical specification of the ***processing operations*** (Algorithm: method or set of rules for implementing the system by a program that performs the corresponding mathematical operations)
  - Emphasis on ***computationally efficient algorithms***, which are fast and easily implementable.

Basic Elements of a Digital Signal Processing System

Analog signal processing:



Digital signal processing:



Why has digital signal processing become so popular?

Advantages and disadvantages of digital processing compared to analog processing:

Property	Digital processing	Analog processing
Dynamics	Only limited by complexity	Generally limited
Precision	Generally unlimited (costs and complexity prop. to precision)	Generally limited (costs increase drastically with required precision)
Aging	Without problems	Problematic
Production costs	Low	Higher
Frequency range	Limited	Nearly unlimited
Linear-phase frequency responses	Exactly realizable	Approximately realizable
Complex algorithms	Realizable	Strong limitations

However, digital signal processing has always also analog components (amplifiers, etc.).

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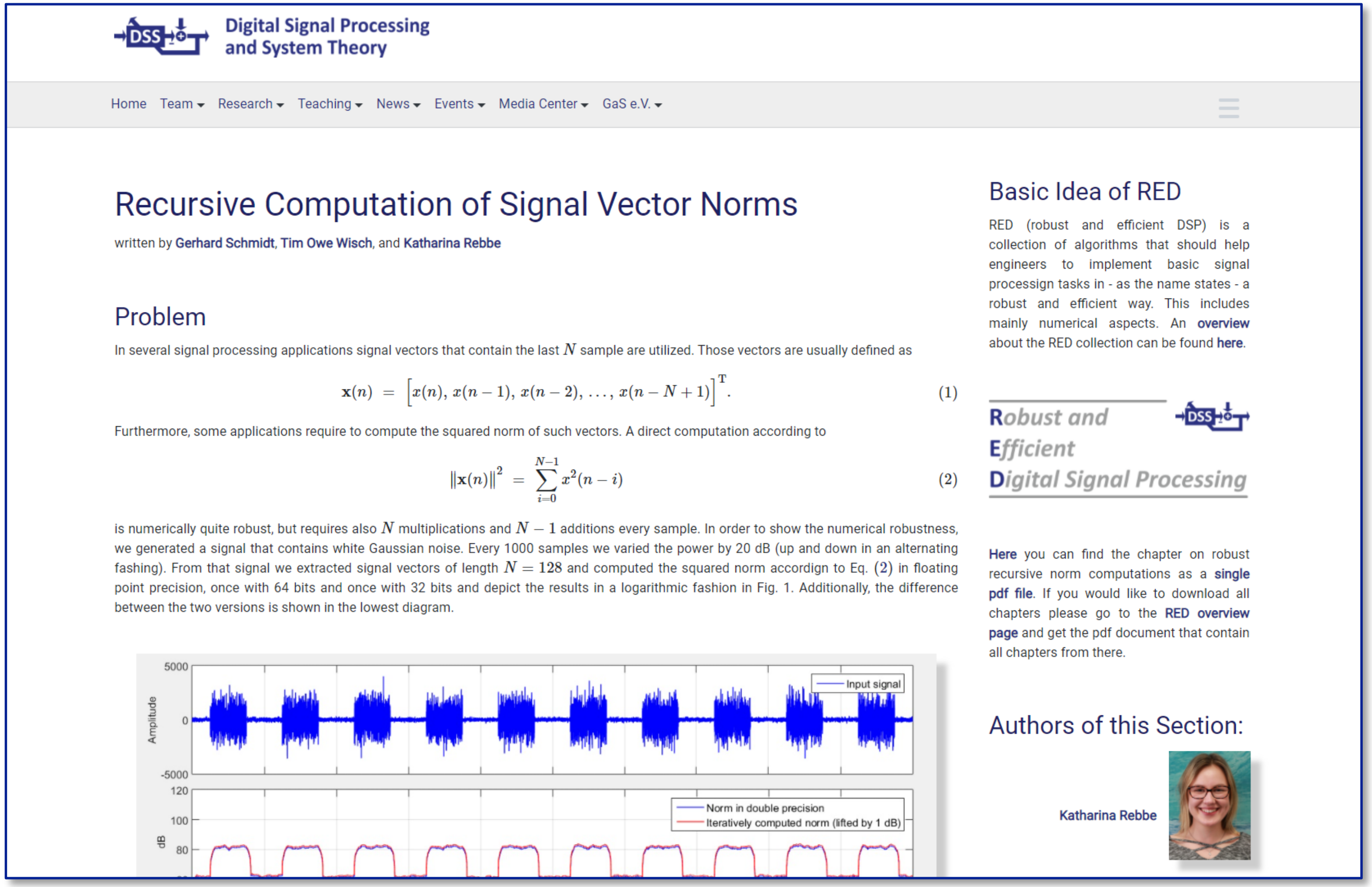
Examples for Understanding the Lecture Goals – Part 1

A simple smoothing filter:

(will be presented via the blackboard)

Examples for Understanding the Lecture Goals – Part 2

Vector norm computation:



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Recursive Computation of Signal Vector Norms

written by Gerhard Schmidt, Tim Owe Wisch, and Katharina Rebbe

Problem

In several signal processing applications signal vectors that contain the last N sample are utilized. Those vectors are usually defined as

$$\mathbf{x}(n) \;=\; \big[x(n),\, x(n-1),\, x(n-2),\, \ldots,\, x(n-N+1)\big]^{\mathrm{T}}. \qquad (1)$$

Furthermore, some applications require to compute the squared norm of such vectors. A direct computation according to

$$\|\mathbf{x}(n)\|^2 \;=\; \sum_{i=0}^{N-1} x^2(n-i) \qquad (2)$$

is numerically quite robust, but requires also N multiplications and $N-1$ additions every sample. In order to show the numerical robustness, we generated a signal that contains white Gaussian noise. Every 1000 samples we varied the power by 20 dB (up and down in an alternating fashing). From that signal we extracted signal vectors of length $N=128$ and computed the squared norm accordign to Eq. (2) in floating point precision, once with 64 bits and once with 32 bits and depict the results in a logarithmic fashion in Fig. 1. Additionally, the difference between the two versions is shown in the lowest diagram.

Basic Idea of RED

RED (robust and efficient DSP) is a collection of algorithms that should help engineers to implement basic signal processign tasks in - as the name states - a robust and efficient way. This includes mainly numerical aspects. An overview about the RED collection can be found here.

Robust and Efficient Digital Signal Processing

Here you can find the chapter on robust recursive norm computations as a single pdf file. If you would like to download all chapters please go to the RED overview page and get the pdf document that contain all chapters from there.

Authors of this Section:

Katharina Rebbe

(will be presented via the “RED” webpage)

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Summary

- ***Introduction***
  - ***Contents of the lecture***
  - ***Literature***
  - ***Analog versus digital signal processing***
- Digital processing of continuous-time signals
- Efficient FIR filter stuctures
- DFT and FFT
- Digital filters
- Multi-rate digital signal processing