Technology
Arts Sciences
TH Köln

# Course
# DSP - Digital Signal Processing

Version: 2 | Last Change: 11.09.2019 11:34 | Draft: 0 | Status: vom verantwortlichen Dozent freigegeben

## General information

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|---|---|
| **Long name** | Digital Signal Processing |
| **Approving CModule** | DSP_MaCSN, DSP_MaTIN |
| **Responsible** | Prof. Dr. Harald Elders-Boll<br>Professor Fakultät IME |
| **Level** | Master |
| **Semester in the year** | winter semester |
| **Duration** | Semester |
| **Hours in self-study** | 60 |
| **ECTS** | 5 |
| **Professors** | Prof. Dr. Harald Elders-Boll<br>Professor Fakultät IME |
| **Requirements** | No formal requirements, but students will be expected to be familiar with:<br>Basic Knowledge of Signals and Systems: Continuous-Time LTI-Systems and Convolution, Fourier-Transform<br>Basic Knowledge of Probability and Random Variables |
| **Language** | English |
| **Separate final exam** | Yes |

### Final exam

#### Details

In the written exam students shall demonstrate that they are able to solve problems dealing with the design, analysis and implementation of DSP systems in soft and hardware considering computational complexity and hardware resource limitation, by using their thorough understanding of the theoretical concepts, especially frequency domain analysis, and insights gained from the practical implementation of DSP systems in software using

Python and on microprocessors, such that they are able to design, select, use and apply actual and future DSP systems for various signal processing application in commercial products.

### Minimum standard

At least 24 of the 50 points that can be gained in total in the final exam and the two midterm tests during the semester.
In the final exam 40 points can be gained in total, in the two midterm test 5 points can be gained each yielding 10 points in total for the two tests.

### Exam Type

In the written exam students shall demonstrate that they are able to solve problems dealing with the design, analysis and implementation of DSP systems in soft and hardware considering computational complexity and hardware resource limitation, by using their thorough understanding of the theoretical concepts, especially frequency domain analysis, and insights gained from the practical implementation of DSP systems in software using Python and on microprocessors, such that they are able to design, select, use and apply actual and future DSP systems for various signal processing application in commercial products.

# Lecture / Exercises

## Learning goals

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### Knowledge

Signals, Systems and Digital Signal Processing
Basic Elements of DSP Systems
Classification of Signals
Continuous-Time and Discrete-Time Signals
Deterministic and Random Signals
Even and Odd Signals
Periodic and Aperiodic Signals
Energy and Power of Signals
Some Fundamental Signals

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Discrete-Time Linear Time-Invariant Systems
Difference Equations
Discrete-Time Convolution
Unit-Pulse and Impulse Response
Basic Systems Properties: Causality, Stability, Memory

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Ideal Sampling and Reconstruction
Ideal Sampling and the Sampling Theorem
Aliasing

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Fourier-Transform of Discrete-Time Signals
Eigenfunctions of Discrete-Time LTI Systems
Frequency response of Discrete-Time LTI Systems
The Fourier-Transform of Discrete-Time Signals
Ideal Continuous-Time Filters

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The z-Transform
The Two-sided z-Transform
Properties of the z-Transform

The Inverse z-Transform

Analysis of LTI Systems using the z-Transform

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Discrete Fourier-Transform

Sampling the DTFT

The DFT and the Inverse DFT

The Fast Fourier Transform

Radix-2 FFT Algorithms

Linear Convolution Using the FFT

Overlap-And-Add

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Design of Digital Filters

Design of FIR Filters

Design of IIR Filters

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Random Signals

Review of Probablity and Random Variables

Ensemble Averages

Correlation Functions

Stationary and Ergodic Processes

Power Spectral Density

Transmission of Random Signals over LTI Systems

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Advanced Sampling Techniques

Quantization and Encoding

Sampling of Bandpass Signals

Sampling of Random Signals

Sample Rate Conversion

Sample Rate Reduction by an Integer Factor

Sample Rate Increase by an Integer Factor

Sample Rate Conversion by a Rational Factor

Oversampling and Noise Shaping

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Optimum Linear Filters

Linear Prediction

The Wiener Filter

Orthogonality Principle

FIR Wiener Filter

IIR Wiener Filter

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Spectrum Estimation

The Periodogram

Window Functions

Eigenanalysis Algorithms

MUSIC Algorithm

ESPRIT Algorithm

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### Skills

Students understand the fundamentals of discrete-time signals and systems

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Students can analyse the frequency content of a given signal using the appropriate Fourier-Transform and methods for spectrum estimation

Analysis of discrete-time LTI Systems
Students can calculate the output signal via convolution
Students can determine the frequency response of a given system
Students can characterize a given system in the frequency domain and in the z-domain

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Implementation of discrete-time LTI systems
Students can implement the convolution sum in software
Students can implement different structures for IIR systems in software
Sudents can use the FFT to implement an FIR system

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Analyze effects of practical sampling
Quantization noise
Aliasing
Trade-off pros and cons of advanced implementations like noise shaping

## Expenditure classroom teaching

| Type | Attendance (h/Wk.) |
|---|---|
| Lecture | 2 |
| Exercises (whole course) | 2 |
| Exercises (shared course) | 0 |
| Tutorial (voluntary) | 0 |

## Separate exam

### Exam Type

solving exercises within limited functional / methodical scope under examination conditions

### Details

Two midterm tests with excercises dealing with the subjects from the lecture/tutorial that were covered up to that point, suich the by passing the midterm tests students demonstrate that they have the required skills to sucessfully participate in the corresponding labs and/or projects.

### Minimum standard

Two out of five points that can be scored in total per test.

# Practical training

## Learning goals

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### Knowledge

Review of Probablity and Random Variables

Moments, Averages and Distribution Functions

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Random Signals

Ensemble Averages

Correlation Functions

Stationary and Ergodic Processes

Power Spectral Density

Transmission of Random Signals over LTI Systems

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Sampling

Sampling and coding for speech and/or audio signals

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### Skills

Analysis of random variables by means of

Mean and moments

Distribution

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Analysis of random signals

Determine whether a given random signal is stationary or not

Analyse whether a random signal contains discrete harmonic components

by using the autocorrelation function

by using the power spectral density

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Combatting noise

Remove or suppress high-frequency noise from low-pass signals

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Abilty to trade-off different methods for digital coding of speech and audio signals

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Determine the quatization noise and the SNR for different sampling schemes

## Expenditure classroom teaching

| Type | Attendance (h/Wk.) |
|---|---|
| Practical training | 1 |
| Tutorial (voluntary) | 0 |

## Separate exam

### Exam Type

working on practical scenarion (e.g. in a lab)

### Details

Sucessful solution of the lab problems and/or projects in small groups consisting of two students, in general. The corresponding midterm test from the lecture/tutorial needs to be passed as a prerequisite for participation in the lab.

### Minimum standard

Successful participation of all labs and/or the corresponding small projects. To pass the corresponding midterm test 2 out of 5 points have to be gained.