

# Lehrveranstaltungshandbuch Digital Signal Processing

## Lehrveranstaltung

Befriedigt Modul (MID)

Organisation

Kompetenznachweis

## Lehrveranstaltungselemente

Vorlesung / Übung

Praktikum

**Verantwortlich:** Prof. Dr.-Ing. Harald Elders-Boll

## Lehrveranstaltung

### Befriedigt Modul (MID)

- aktuelle
  - Ma CSN2012 DSP
  - Ma TIN2012 DSP

## Organisation

Version	
erstellt	2013-04-25
VID	2
gültig ab	WS 2012/13
gültig bis	

Bezeichnung	
Lang	Digital Signal Processing
LVID	F07_DSP
LVPID (Prüfungsnummer)	

Semesterplan (SWS)	
Vorlesung	2
Übung (ganzer Kurs)	1
Übung (geteilter Kurs)	
Praktikum	1
Projekt	
Seminar	
Tutorium (freiwillig)	

Präsenzzeiten	
Vorlesung	30
Übung (ganzer Kurs)	15
Übung (geteilter Kurs)	
Praktikum	15
Projekt	
Seminar	
Tutorium (freiwillig)	

max. Teilnehmerzahl	
Übung (ganzer Kurs)	15
Übung (geteilter Kurs)	
Praktikum	15
Projekt	15
Seminar	

**Gesamtaufwand:** 150

## Unterrichtssprache

- Englisch

## Niveau

- Master

## Notwendige Voraussetzungen

- No formal requirements, but students will be expected to be familiar with:
  - Basic Knowledge of Signals and Systems
    - Continuous-Time LTI-Systems and Convolution
    - Fourier-Transform
  - Basic Knowledge of Probability and Random Variables

## Literatur

- John G. Proakis and Dimitris K. Manolakis. Digital Signal Processing (4th Edition). Prentice Hall, 2006.
- Alan V. Oppenheim, Ronald W. Schafer. Discrete-Time Signal Processing (3rd Edition). Prentice Hall, 2007.
- Vinay Ingle and John Proakis. Digital Signal Processing using MATLAB. Cengage Learning Engineering, 2011.

## Dozenten

- Prof.Dr. Harald Elders-Boll

## Wissenschaftliche Mitarbeiter

- Dipl.-Ing.Martin Seckler

## Zeugnistext

Digital Signal Processing

## Kompetenznachweis

Form	
sMP	80% (mündliche Prüfung)

Aufwand [h]	
sMP	10

Intervall: 2-3/Jahr

## Lehrveranstaltungselemente

### Vorlesung / Übung

## Lernziele

### Lerninhalte (Kenntnisse)

- 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
- Discrete-Time Linear Time-Invariant Systems
  - Difference Equations
  - Discrete-Time Convolution
  - Unit-Pulse and Impulse Response
  - Basic Systems Properties: Causality, Stability, Memory
- Ideal Sampling and Reconstruction
  - Ideal Sampling and the Sampling Theorem
  - Aliasing
- 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
- 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
- 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
- Design of Digital Filters
  - Design of FIR Filters
  - Design of IIR Filters
- Random Signals
  - Review of Probability and Random Variables
  - Ensemble Averages
  - Correlation Functions
  - Stationary and Ergodic Processes
  - Power Spectral Density
  - Transmission of Random Signals over LTI Systems
- 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
- Optimum Linear Filters
  - Linear Prediction
  - The Wiener Filter
    - Orthogonality Principle
    - FIR Wiener Filter
    - IIR Wiener Filter
- Spectrum Estimation
  - The Periodogram
    - Window Functions
  - Eigenanalysis Algorithms
    - MUSIC Algorithm
    - ESPRIT Algorithm

## Fertigkeiten

- Students understand the fundamentals of discrete-time signals and systems
- 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
- Implementation of discrete-time LTI systems
  - Students can implement the convolution sum in software
  - Students can implement different structures for IIR systems in software
  - Students can use the FFT to implement an FIR system
- Analyze effects of practical sampling
  - Quantization noise
  - Aliasing
  - Trade-off pros and cons of advanced implementations like noise shaping

## Begleitmaterial

- elektronische Vortragsfolien zur Vorlesung (lecture slides as pdf-file)
- elektronische Übungsaufgabensammlung (lernmaterial slides)

## Besondere Voraussetzungen

## Besondere Literatur

## Besonderer Kompetenznachweis

Form	
bK	2-3 eTests je 20min (je 1x wiederholbar)
bÜA	Präsenzübung und Selbstlernaufgaben

Beitrag zum LV-Ergebnis	
bK	20%
bÜA	unbenotet

Intervall: 1/Jahr

## Praktikum

## Lernziele

### Lerninhalte (Kenntnisse)

- Review of Probability and Random Variables
  - Moments, Averages and Distribution Functions
- Random Signals
  - Ensemble Averages
  - Correlation Functions
  - Stationary and Ergodic Processes
  - Power Spectral Density
  - Transmission of Random Signals over LTI Systems
- Sampling
  - Sampling and coding for speech and/or audio signals

### Fertigkeiten

- Analysis of random variables by means of
  - Mean and moments
  - Distribution
- 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
- Combatting noise
  - Remove or suppress high-frequency noise from low-pass signals
- Ability to trade-off different methods for digital coding of speech and audio signals
- 
- Determine the quantization noise and the SNR for different sampling schemes

## Begleitmaterial

- elektronische Beschreibung der Praktikums-Versuche (Instructions for lab experiments as pdf-files)

## Besondere Voraussetzungen

## Besondere Literatur

## Besonderer Kompetenznachweis

Form
bSZ Praktikum (Lab Experiments)

Beitrag zum LV-Ergebnis
bSZ Voraussetzung für Modulprüfung (prerequisite for final exam)

**Intervall:** 1/Jahr

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