Technology
Arts Sciences
TH Köln

Lehrveranstaltung
DSP - Digital Signal Processing

Version: 2 | Letzte Änderung: 11.09.2019 11:34 | Entwurf: 0 | Status: vom verantwortlichen Dozent freigegeben

Allgemeine Informationen

Langname	Digital Signal Processing
Anerkennende LModule	DSP_MaCSN, DSP_MaTIN
Verantwortlich	Prof. Dr. Harald Elders-Boll Professor Fakultät IME
Niveau	Master
Semester im Jahr	Wintersemester
Dauer	Semester
Stunden im Selbststudium	60
ECTS	5
Dozenten	Prof. Dr. Harald Elders-Boll Professor Fakultät IME
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
Unterrichtssprache	englisch
separate Abschlussprüfung	Ja

Abschlussprüfung

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.

Mindeststandard

Mindestens 24 der möglichen 50 möglichen Gesamtpunkte aus der Klausur und den zwei Tests während des Semesters.
In der Klausur können maximal 40 Punkte in den zwei Tests während des Semesters können maximal jeweils 5 in der Summe also 10 Punkte erreicht werden.

Prüfungstyp

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.

Vorlesung / Übungen

Lernziele

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 Probablity 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

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

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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

Analyze effects of practical sampling

Quantization noise

Aliasing

Trade-off pros and cons of advanced implementations like noise shaping

Aufwand Präsenzlehre

Typ	Präsenzzeit (h/Wo.)
Vorlesung	2
Übungen (ganzer Kurs)	2
Übungen (geteilter Kurs)	0
Tutorium (freiwillig)	0

Separate Prüfung

Prüfungstyp

Übungsaufgabe mit fachlich / methodisch eingeschränktem Fokus unter Klausurbedingungen lösen

Details

Zwei semesterbegleitende Tests in Form von Aufgaben, die den bis zum jeweiligen Zeitpunkt in der Vorlesung/Übung behandelten Stoff aufgreifen und so bei Bestehen sicherstellen, dass die Grundlagen zur erfolgreichen Teilnahme an den entsprechenden Praktikumsversuchen und/oder Projekten gegeben ist.

Mindeststandard

Mindestens 2 von maximal 5 erreichbaren Punkten pro Test.

Praktikum

Lernziele

Kenntnisse

Review of Probablity 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

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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

Aufwand Präsenzlehre

Typ	Präsenzzeit (h/Wo.)
Praktikum	1
Tutorium (freiwillig)	0

Separate Prüfung

Prüfungstyp

praxisnahes Szenario bearbeiten (z.B. im Praktikum)

Details

Erfolgreiche Bearbeitung dfer Parktikumsversuche oder Projekte in Kleingruppen von in der Regel zwei Studierenden. Das Bestehen des entsprechenden Tests aus der Vorlesung/Übung ist Zugangsvoraussetzung um am Praktikum teilnehmen zu können.

Mindeststandard

Erfolgreiche Teilnehme an allen Versuchen und/oder erfolgreiche Bearbeitung von kleinen Projekten. Im entsprechenden Test in der Vorlesung/Übung müssen zum Bestehen 2 von 5 möglichen Punkten erreicht werden