Introduction to Digital Signal Processing

Course description:

Digital Signal Processing (DSP) is concerned with the representation, transformation and manipulation of signals on a computer. After half a century advances, DSP has become an important field, and has penetrated a wide range of application systems, such as consumer electronics, digital communications, medical imaging and so on. With the dramatic increase of the processing capability of signal processing microprocessors, it is the expectation that the importance and role of DSP is to accelerate and expand.

Discrete-Time Signal Processing is a general term including DSP as a special case. This course will introduce the basic concepts and techniques for processing discrete-time signal on a computer. By the end of this course, the students should be able to understand the most important principles in DSP. The course emphasizes understanding and implementations of theoretical concepts, methods and algorithms.

Objectives:

- To give the students a comprehension of the concepts of discrete-time signals and systems
- To give the students a comprehension of the Z- and the Fourier transform and their inverse
- To give the students a comprehension of the relation between digital filters, difference equations and system functions
- To give the students knowledge about the most important issues in sampling and reconstruction
- To make the students able to apply digital filters according to known filter specifications
- To provide the knowledge about the principles behind the discrete Fourier transform (DFT) and its fast computation
- To make the students able to apply Fourier analysis of stochastic signals using the DFT
- To be able to apply the MATLAB programme to digital processing problems and presentations

Course outline:

Part I - Introduction to Digital Signal Processing (5 lectures)

- Discrete-time sequences and systems
- linear time-invariant (LTI) systems
- impulse response and convolution
- the Z-transform and its inverse
- difference equations and system functions
- signal flow graphs
- Fourier transforms and frequency response
- periodic sampling and reconstruction of band limited signals

Part II - Filter design and Fourier signal analysis (5 lectures)

- Up- and down sampling

- design of IIR- and FIR-filters
- digital filter structures (direct, cascade, parallel and lattice)
- filter transformations
- all-pass, minimum phase systems
- the discrete and fast Fourier transform
- circular convolution, block convolution
- Fourier analysis, the effect of windowing

Lecture plan

Lecture 1
Course overview and discrete-time signals and systems

- Discrete-time sequences, discrete-time systems, linear time-invariant systems (LTI), impulse response, convolution in time, properties of LTI systems.

Lecture 2
Difference equations and introduction to digital filters

- Linear constant-coefficient equations, stability, introduction to FIR-filters, introduction to IIR-filters.

Lesson 3
Fourier transform and frequency response

- Fourier transform of sequences, properties of the Fourier transform, frequency response of Linear Time-Invariant (LTI) systems, inverse Fourier transform, Fourier transform theorems.

Lecture 4
Sampling and reconstruction

- Periodic sampling, frequency-domain representation of sampling, reconstruction of band-limited signals, changing the sampling rate of discrete signals.

Lecture 5
The Z-transform and its inverse

- The bilateral Z-transform, properties of the Z-transform, inversion, system representation in the Z-domain, solutions to difference equations.

Lecture 6
Basic structures of IIR- and FIR filters

- Filter structures (direct form I & II), signal flow graph representations, IIR systems, transposed forms, FIR systems.

Lecture 7
The Discrete Fourier Transform (DFT)

- Discrete Fourier Series, sampling and reconstruction in the Z-domain, the DFT, properties, linear and circular convolution.

Lecture 8
Filter design techniques - IIR-filters

- Analog prototypes, impulse invariance, bilinear transformation.

Lecture 9
Filter transformations

- All-pass systems, minimum phase systems, linear phase systems, lowpass/highpass/bandpass/bandstop transformation.

Lecture 10
The Fast Fourier Transform (FFT) and FFT analysis

- Block convolution, the Goertzel algorithm, decimation-in-time & -in-frequency, FFT analysis.

Required text:

- Oppenheim, A.V., Schafer, R.W, "Discrete-Time Signal Processing", Second Edition, Prentice-Hall, New Jersey, 1999, ISBN 0-13-083443-2.

References:

- Steven W. Smith, “The Scientist and Engineer's Guide to Digital Signal Processing”, California Technical Publishing, 1997, ISBN 0-9660176-3-3. http://www.dspguide.com/pdfbook.htm (You can download the entire book!)
- Kermit Sigmon, "Matlab Primer", Third Edition, Department of Mathematics, University of Florida.
- V.K. Ingle and J.G. Proakis, "Digital Signal Processing using MATLAB", Bookware Companion Series, 2000, ISBN 0-534-37174-4.

Lecturer:

- Zheng-Hua Tan, zt@kom.aau.dk, Aalborg University.