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Signal and Information Processing Laboratory

Prof. Dr. A. Lapidoth, Prof. Dr. H.-A. Loeliger, Prof. em. Dr. G.S. Moschytz
Prof. Dr. F. Eggimann, Dr. K. Heutschi

ANNUAL REPORT

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Research Period 2000

Teaching Period 1999/2000

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Foreword

One only becomes "senior" at an institute once an even more recent recruit joins, and I therefore suppose that with Professor Hans-Andrea Loeliger joining our Institute, I have at least become senior enough to write this foreword.

These are times of great change for our Institute, times that offer both promise and challenges. Professor George Moschytz has retired to start a new career in Israel, and we all wish him the best of luck in his new-found home and thank him for all his years of service to the ETH and to our Institute. Replacing him is Professor Loeliger, who has joined us in June 2000. He brings with him industrial and academic experience as well as great enthusiasm. We are very fortunate to have him with us, and are very much looking forward to benefiting from his expertise. Professor Loeliger brings a unique perspective to the fields of Signal Processing and Information Theory. Rather than setting up borders and emphasizing the difference between these fields, Professor Loeliger is exploring the fundamental ideas that are common to both.

It seems that Professor Moschytz is not the only one to flee our Institute. We thus bid farewell to the newest crop of Ph.D's including Markus Erne, Thomas von Hoff, Hanspeter Schmid, Peter Wellig, and Sigi Wyrsch. We very much hope that they will maintain contact with us, and that we shall be seeing them at our future Holidays Parties.

Contents

1. Personnel

Institute Director and Professor for Communication Engineering (Network Theory and Signal Processing):

Prof. Dr. George S. Moschytz
retired since 31.3.2000

Professor for Information Theory:

Prof. Dr. Amos Lapidoth

Professor for Signal Processing:

Prof. Dr. Hans-Andrea Loeliger
since 1.6.2000

Professor for Information Technology:

Prof. Dr. Fritz Eggimann

Adjunct Lecturer: **Dr. K. Heutschi**

Secretaries: **Mrs. Bernadette Röösli**
Mrs. Renate Agotai
Mrs. Heidi Schenkel

Administr.Supervisor: **Dr. Marcel Joho**

Technical Supervisor: **Dr. Max Dünki**

Research Assistants:	**Dieter Arnold**	Dipl.El.Eng.	
	Justin Dauwels	Dipl.Phys.Eng.	since 1.11.00
	Markus Erne	Dipl.El.Eng.	left on 31.12.00
	Qun Gao	Dipl.El.Eng.	
	Markus Hofbauer	Dipl.El.Eng.	
	Ralf Kretzschmar	Dipl.Phys.	
	Dani Lippuner	Dipl.El.Eng.	
	Felix Lustenberger	Dipl.El.Eng.	
	Heinz Mathis	Dipl.El.Eng.	
	Patrick Merkli	Dipl.Ing. Microtechn.EPF	since 16.10.00
	Stefan Moser	Dipl.El.Eng.	
	Hanspeter Schmid	Dipl.El.Eng.	left on 30.11.00
	Thomas von Hoff	Dipl.El.Eng.	left on 31.12.00
	Pascal Vontobel	Dipl.El.Eng.	
	Peter Wellig	Dipl.El.Eng.	
	Sigi Wyrsch	Dipl.El.Eng.	left on 29.2.00

Technical Staff: **Francesco Amatore**
Thomas Schaerer

Academic Guests: (see 6.1 for report of activities)

Aaron S. Cohen	MIT, Cambridge, USA	01.01. – 28.01.00
Ibrahim Abou Faycal	MIT, Cambridge, USA	01.01. - 28.01.00
Prof. Stuart Schwartz	Princeton, USA	20.06.00
Prof. Allen Lindgren	University of Rhode Island, Kingston, USA	01.09. – 30.09.00
Prof. Leon Chua:	University of California, Berkely, USA	15.05. – 14.08.00

2. Teaching

2.1 Lectures and Practica

Sem.	Instructors	Title	ETH-No.
5th	Dr. H.-A.Loeliger	Zeitdiskrete Systeme & stochastische Signale	35-405
6th	Dr. A. Kaelin	Digitale Signalverarbeitung und Filterung	35-416
5/7th	Prof. A.Lapidoth	Applied Digital Information Theory I	35-417
8th	Prof. A. Lapidoth	Applied Digital Information Theory II	35-418
7th	Prof. F. Eggimann M. Hofbauer	Adaptive Filter & neuronale Netzwerke	35-467
8th	H.P. Schmid	Analoge Signalverarbeitung und Filterung	35-468
7th	Dr. K. Heutschi	Acoustics I	35-477
8th	Dr. Heutschi	Acoustics II	35-478
5/ 6th	Prof. A. Lapidoth et al.	Laboratory for "Fundamentals in Electrical Engineering"	35-095/6
	Prof. F. Eggimann et al.	Colloquium on "Neuro-Informatics"	95-899 95-999
	Prof. F. Eggimann	Colloquium on "Material- und Werkstoffwissenschaften"	39-797
	Dr. K. Heutschi	Acoustics Colloquium	35-950

2.2 Semester Projects and Diploma Theses

During the winter semester 1999/00 and summer semester 2000, 6 Semester Projects (8 candidates) and 6 Diploma Theses (9 candidates) were carried out.

Candidates	Title	Supervisor
Semester Projects WS 99/00 (7th Semester)		
Philippe Messmer Michael Bircher	Fingerprint Recognition with Cellular Neural Networks: Image Preprocessing	Gao, Kretzschmar
Matthias Frey	The (24,14,6) Wagner Code	Proffs.Lapidoth, Blahut, & Massey,
Semester Projects SS 00 (8th Semester)		
Philippe Foerster Rainer Moebus	Image Enhancement with Cellular Neural Networks	Gao, Kretzschmar
Frank Herzog	Anwendung Neuronaler Netze: Bestimmung des Bewölkungsgrades	Kretzschmar, Gao
Rolf Sigg	Dekodierung langer Codes	Vontobel, Arnold
Singer Thomas	On the capacity of a very dirty tape	Prof. Lapidoth
Diploma Theses WS 99/00		
Gabriel Hauser Marco Dübendorfer	Robuste Spracherkennung am Beispiel einer Pflegebetten –Steuerung	Hofbauer Drs. Oberle & Kälin
Büeler Reto	Neuronale Netzwerke: Wahrscheinlichkeitstheorie versus Fuzzy-Logic	Kretzschmar Quarenghi
Eric Svensson Daniel Hösli (IBM)	Turbo-Kodes fuer magnetische Festplatten	Arnold Dr. Mittelholzer
Alex Koster Tobias Geyer	Time-scale and pitch modification of Speech	Joho Dr. Etter (Lucent)
Gennaro Lanzetta	Soft-Input / Soft-Output Sequential Decoding	Arnold, Vontobel & Prof. Costello
Bülent Aydin R.	Fingerprint Recognition with Cellular Neural Networks: Feature Extraction	Gao Kretzschmar
Post-Diploma Thesis		
Lustenberger Felix	Analog Probability Propagation Networks - Part II: Decoder Examples	

Mathis Heinz

Differential Detection of GMSK Signals with Low BT using the SOVA

3. Research

3.1 Research Areas

The Signal and Information Processing Lab focusses on research and teaching in the following areas:

Information Theory and Coding

Information theory, error correcting codes, and their application to communication systems. Current topics:

3. Fundamental limits on reliable communication over fading channels
4. Provably secure digital watermarking
5. Turbo codes and low density parity check codes
6. Reduced-complexity receivers for intersymbol interference channels
7. Coding for magnetic recording

Digital Signal Processing

8. Adaptive filters for equalization and related issues in communications and acoustics
9. Artificial neural networks and cellular neural networks
10. Processing of electromyograms

Analog Signal Processing

Prof. Moschytz and many of his students worked on linear filters, especially switched capacitor filters. The tradition of analog signal processing is continued by Prof. Loeliger and some of his students with a focus on new nonlinear networks, especially for the decoding and error correcting codes.

3.2 Current Research Projects

Information Theory and Coding

Universal Decoders of InterSymbol Interference Channels

We consider the design of a decoder for coded communication over an InterSymbol Interference channel of an unknown impulse response. Our previous results have demonstrated the existence of a decoder that does not require knowledge of the channel law and yet performs asymptotically as well as the best decoder that could have been designed had the impulse response been known. The current project is aimed at designing such a universal decoder under complexity constraints.

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In Collaboration with: Jacob Ziv

Keywords: Universal Decoding, InterSymbol Interference, Random Coding, Error Exponents.

Robust Decoding of Space-Time Codes

We consider digital communication with multi-antennas, and focus on robustness with respect to imprecise knowledge of the fading matrix. Of special interest is the performance of a maximum-likelihood decoder that is fed with imprecise measurements of the fading matrix. Rather than focusing on specific codes, our approach is information theoretic, focusing on the achievable rates with a given ensemble of codes.

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In Collaboration with: Shlomo Shamai

Keywords: Robust decoding, Mismatched decoding, Space-Time codes

The Capacity Region of the Poisson Multiple-Access Channel with Noiseless Feedback

The Poisson mutiple-access channel (MAC) models a any-to-one optical communication system. Its capacity region has recently been computed by Lapidoth & Shamai. The purpose of the present research is to investigate the gains (in capacity) afforded by noisless delayless feedback from the receiver to the transmitters.

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In Collaboration with: Shraga Bross

Keywords: Poisson Channel, Multiple-Access, Capacity region, Feedback

Models and Codes for the Magnetic Recording Channel

The magnetic recording channel possesses several characteristics that differentiate it from ordinary communication links. First, the input of the magnetic recording channel is constrained to be binary. This greatly complicates the computation of the ultimate transmission limit, i.e. capacity. But even for the much simpler problem of computing the information rate, i.e. the average mutual information, between the input and the output, there exists no exact algorithm.

Second, at high storage densities, the channel is nonlinear and the noise signal-dependent. This so called media noise can overpower the stationary electronics noise component by a ratio 9:1. The major source of media noise in high density magnetic recording is transition noise. Transition noise encompasses both pulse jitter and partial signal erasure which is a consequence of magnetization percolation at high linear densities.

In the report period, a new practical algorithm for computing the information rate was found. Its efficacy was demonstrated for various channel models and compared to existing bounding techniques. These new exact lower bounds are important for assessing the potential coding gain for different coding schemes. In particular, an optimal deterministic construction method for binary Low-Density-Parity-Check Codes that do not contain 4-cycles was developped. These codes provide a coding gain of around 2dB over hard-decoded RS codes and are at a sector error rate of 10^3 about 1.5 dB away from the information rate lower bound. Further, a new simple channel model was proposed that captures the essence of transition noise and that is suited for signal processing algorithms.

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Supported by: IBM Research, Zurich Research Lab.
In Collaboration with: Dr. E. Eleftheriou, IBM Research, Zurich Research Lab.

Keywords: Magnetic Recording, Coding, Information Theory

Bounds on the Capacity of Fading Channels

Fading channels (like Rayleigh or Ricean fading channels) with or without memory are frequently used to model mobile wireless communication links. With the discovery of Turbo-codes that are often capable of approaching channel capacity, the interest in computing the capacity of such channels has been renewed. However, it seems hopeless to compute capacity precisely.

The goal of this project is to compute upper and lower bounds to the capacity of fading channels that capture the behavior of the true channel capacity.

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Professor: Amos Lapidoth

Keywords: Channel capacity, high SNR, fading, flat-fading, Ricean Fading, Rayleigh fading

Algebraic Coding and Iterative Decoding

Channel coding deals with the problem of reliable communication over a noisy channel. In this project we aim at designing new channel codes based on algebraic principles and their representation by graphs. Rather than relying on algebraic decoding methods, we decode iteratively using the sum-product algorithm.

A main tool for this research are factor graphs, which are very well suited for representing codes and channels. Additionally, decoding by the sum-product algorithm can be interpreted as message passing along the edges of the graph.

The goal is to build a bridge between algebraic and turbo codes: finding factor graph representations of known codes and designing new codes with focus on block lengths no more than several thousand symbols. The construction of these codes will not only be based on ideas from algebra but also be in the spirit of Tanner‘s transform theory. We want to explore graph properties like cycles and connectivity; their impact on code parameters and especially on decoding performance are not well understood until now.

In the report period we have constructed codes derived from finite incidence structures (finite geometries) and codes derived from expander graphs. The resulting codes belong to the category of low-density parity-check codes, i.e. they have parity-check matrices which are sparsely filled with ones.

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Keywords: algebraic coding, iterative decoding, factor graphs

Signal Processing

Signal adaptive Audio Coding using Wavelets and Rate Optimization

Current audio compression techniques such as MPEG-1 and MPEG-2 are based on fixed filterbanks (Polyphase filterbanks or Modified Discrete Cosine Transforms). The compression-ratio of these algorithms can be fixed for a given application but severe degradation of the compressed signal will occur, if the selected channel bit-rate exceeds the momentary channel capacity.

In this research project, a new, Wavelet-based, embedded approach to audio compression has been investigated. The variety of existing musical instruments such as castanets, harpsichord or pitch-pipe exhibiting various coding requirements due to their completely different temporal and spectral fine-structure, suggests to use a filterbank with variable time-frequency resolution. Therefore, a signal-adaptive filterbank, offering almost arbitrary time-frequency tiling has been implemented in C++. The filterbank is controlled based on „rate-distortion“ analysis or on perceptual criteria. A psychoacoustic model taking care of frequency-domain and temporal masking has been implemented in C++. A „cost-function“ which controls the switching of the filterbank determines the switching based on rate-distortion or on perceptual criteria. The audio compression scheme has been extensively evaluated under critical listening test situations, and some ideas already have been implemented in the MPEG-4-Standard.

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In Collaboration with: Scopein Research

Keywords: MPEG, Audio Compression, Wavelets, Rate-Distortion Theory, Perceptual Modeling

Fingerprint Recognition Using Cellular Neural Networks

Personal identification by fingerprint recognition is a particularly interesting and challenging task in the area of image processing and pattern recognition. Fingerprint-based recognition systems are usually used for criminal identification and police work. But now, with the increasing power of computers and scanners, research on fingerprint-based recognition systems for civilian applications is becoming increasingly attractive.

A promising candidate for fingerprint-based personal identification in civilian applications is the Cellular Neural Network~(CNN). CNNs belong to the class of nonlinear, recurrent, dynamic, and analog systems. They carry out complex nonlinear signal processing in parallel. Their local connectivity and analog operation make them very suitable for VLSI implementations requiring low power consumption. This means that they provide the possibility of implementing a fingerprint-based recognition system on one chip.

This project is aimed at developing robust CNN algorithms for fingerprint recognition. To this end, a CNN Fingerprint Image Preprocessing Algorithm has been developed. It improves the contrast of an original gray-scale fingerprint, sharpens ridges and reduces the high frequency noise in the original fingerprint, recovers the destroyed connectivity in the ridges thus enhancing the fingerprint ridges, and transforms the original fingerprint image into a binary image. Finally, it reduces the width of ridges to one pixel. The resulting black lines contain essentially all the necessary characteristics of the original fingerprint image. A CNN Fingerprint Feature Extraction Algorithm is now under development. It is able to detect ridge endings and ridge bifurcations in a thinned fingerprint image, able to extract the information of directions of ridges leaving endings and bifurcations, and able to eliminate false features to facilitate the next processing stage: Fingerprint Feature Matching.

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In Collaboration with: Prof. em. Dr. G.S. Moschytz

Keywords: fingerprint recognition, cellular neural networks, image enhancement, feature extraction

Fast Algorithms for Adaptive Beamforming

In acoustical applications related to hearing aids or teleconferencing the signals received at a single microphone usually contain a mixture of several sound sources. Removing the disturbing noise sources is a difficult task, when only one microphone is used to pick up the sound, especially when the source signals overlap in their spectra. As the disturbing noise sources are usually coming from spatially different locations, additional microphones which are placed at different positions receive different mixtures of the sound sources involved. In the case where the microphones are located closely together (e.g. microphone array), the phase information between the received signals can be used to amplify or attenuate the signals coming from different angles. This technique is related to beamforming. In case of strong reverberation caused by acoustical reflections, standard beamforming techniques fail to work properly, as the assumption that each sound source impinges from a single direction on the microphone array is violated.

Blind source separation algorithms have shown their capability of solving the multi-path problem in a simulation environment and are therefore very promising for their use in real acoustical applications. Blind algorithms make only week assumptions on the signals involved, e.g. non-Gaussianity or non-stationarity, which both are the case for speech signals.

We have developed adaptive algorithms for blind signal separation (BSS), which are motivated by non-blind algorithms typically used for inverse modeling or system equalization of an instantaneous mixing system. Furthermore, we have found a systematic way to extend these BSS algorithms to the multichannel blind deconvolution task, which is used in case of an unknown convolutive mixing system. These blind deconvolution algorithms operate mainly in the frequency domain (filtering and adaptation), as they make use of fast convolution techniques and therefore have a low computational complexity. In a different project, we have investigated the blind signal separation task with a noisy mixing model and for the case where more sensors than source signals are used. We have examined an algorithm which comprises two stages, where the first stage consists of a principal component analysis (PCA) and the second one of an independent component analysis (ICA). The purpose of the PCA stage is to increase the input SNR of the succeeding ICA stage and to reduce the sensor dimensionality. The ICA stage is used to separate the remaining mixture into its independent components.

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Supported by: ETH

Keywords: Adaptive Beamforming, Blind Signal Separation

Prediction of local winds with neural networks

Weather prediction has always been a dream of mankind. In contrast to the success of the meso and large-scale prediction of the atmosphere during the last decades, little progress was made for small scale (or local) prediction. The main reason for this is the chaotic nature of the leading small scale terms in the governing equations of the atmosphere. As a consequence, numerical methods dealing with these terms tend to behave in an unstable manner and many statistical assumptions about the small scale behaviour are found to be inaccurate.

The goal of this project is to improve the prediction of local winds which are strongly influenced by small scale phenomena. For this purpose, several neural networks are under investigation. Neural networks are nonlinear function approximators that require no mathematical model and no prior assumptions of the underlying process.

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Supported by: MeteoSchweiz
In Collaboration with: Dr. N.B. Karayiannis, University of Houston, Houston, Texas and MeteoSchweiz

Keywords: nonlinear signal processing, neural networks, local wind prediction

Adaptive Filters for Nonstationary Environments

In this project, computationally efficient adaptive algorithms shall be developed for operating in an environment of unknown instationary statistics. Most adaptation algorithms converge rapidly in identifying an unknown time-invariant system in a stationary environment. However, in tracking a time-varying system, especially if the environment is nonstationary, the existing algorithms are rather unsatisfying. The estimation-error impairment may have two main reasons: either due to changes of the system parameters or due to the statistics of the measurement noise. It is important to distinguish between these two influences, because only in the former case the adaptive filter shall be readjusted. In order to reduce the computational complexity the optimum Kalman filter has been reduced to an LMS (Least Mean-Square)-like adaptive filter. The effects of a correlated excitation signal on the scalar time-varying step size has been investigated. Furthermore, it has been found that the excess mean-squared error of the adaptive estimation can be obtained by tracking the signs of the filter-update terms. These findings helped to develop a robust step-size control for an adaptive LMS filter. Its performance has been verified by means of simulations using real-world speech signals and impulse responses.

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Supported by: ETH

Keywords: nonstationary environment, adaptive FIR filters, Kalman filter, acoustic echo cancellation

Nonlinear Functions for Blind Signal Processing

Under the most general conditions, i.e., when second-order methods fail to work, nonlinear functions are an important part of algorithms solving blind problems such as blind separation and blind equalization. Roughly speaking, they take over the role of a proper training reference signal, which is not available, hence the term ‚blind'. The common idea shared by stochastic gradient-search algorithms to either separate or deconvolve signals (or both) is the cross-correlation of signals before and after a nonlinear function, which reveals any existing higher-order correlation among the signals or among different time-lags of the same signal. Such higher-order correlations indicate dependence, which is then formed to an error signal to drive the output signals into a state of higher independence. The underlying higher-order statistics are implicitly produced by nonlinear functions. These nonlinear function are essentially defined by the probability density function of the original source signals to extract and on the cost function (such as independence, maximum-likelihood, and so on). In cases where the original distributions are unknown, change over time, or are of different nature within the source signals, the nonlinearity has to adapt itself according to some estimate of the distribution, or be robust enough to cover a wide mismatch of the assumed model. Stability regions for different nonlinearities are derived and presented. Although the exact form of the nonlinearity might not matter for an algorithm to converge, it may have an impact on the convergence time or the separation/deconvolution performance. This impact of different nonlinearities is investigated, and robust, optimal, and universal nonlinearities are presented. Moreover, if complexity is an issue, simple nonlinearities are preferable to nonlinearities employing hyperbolic or polynomial functions. The threshold nonlinearity is such a simple nonlinearity, which works for sub-Gaussian signals such as typically used in digital data communications. Moreover, by adjusting the threshold, it may be used to separate and deconvolve any non-Gaussian signal.

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Keywords: Blind Separation, Blind Equalization, Nonlinear Functions

On the Convergence of Blind Separation and Deconvolution

In situations where only mixtures of independent desired signals can be picked up, one needs a blind algorithm to separate them and to recover the original signals. Such situations occur in acoustics, in biomedical applications, geophysics, and communications. Over the recent years, algorithms have been proposed that estimate the desired signals by adaptively finding a demixing system with independent outputs.

In this project, the convergence behavior of adaptive blind source separation algorithms is investigated, in particular stability and performance. Methods exist which turn an instable algorithm into a stable one. For a given learning rate and known source distribution, the performance of the algorithm can be determined. Those results were then extended to the case where, additionally, the mixing process introduces a temporal dependency in the sensor signals (Multichannel Blind Deconvolution (MCBD)).

If a fixed step size is employed in the adaptive algorithm for separation, a trade-off exists between speed and exactness. To achieve both at the same time a self-adjusting step size is proposed. This new procedure allows error-free convergence in a time-invariant mixing environment and an improved tracking behavior in the case of changing mixing conditions.

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Professor: Hans-Andrea Loeliger
Supported by: KTI (Noise reduction for a car mobile phone)
In Collaboration with: Siemens Schweiz AG

Keywords: blind source separation, multichannel blind deconvolution, on-line learning

Decomposition of Long-Term Intramuscular EMG Signals using Wavelets

For medical studies of chronic muscle pain, long-term measurements of EMG (electromyogram) signals need to be analyzed. Such signals represent the electrical activity in a muscle: A muscle fibre group, named Motor Unit (MU), is stimulated by a given nerve cell and radiates a specific waveform, called a Motor Unit Action Potential (MUAP). The repetitive activation of several individual Mus results in a superposition of pulse trains, which constitute the EMG signal. In contrast to short-term recordings, the number of active Mus and the MUAP shapes change during long-term measurements. Therefore, the main goals of a decomposition algorithm are: the evaluation of the number of active Mus at any time, the determination of MUAP shapes, the detection of MUAP shape changes, and the complete decomposition of overlapping MUAPs.

The decomposition of long-term recordings can be considered as a classification problem, where both unsupervised and supervised classification techniques have to be used. Beside white noise, high-frequency noise and low-frequency noise influence the classification performance. Low-frequency noise is caused by electrode movements and depolarisations of the muscle fibres lying farther away from the electrode placement. High-frequency noise is caused by time-offsets of the aligned waveforms and physiological jitter. Using selected wavelet coefficients, the classification performance of both the supervised classification and the unsupervised classification can be improved. Furthermore, a reduction of the number of the features can be achieved.

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In Collaboration with: Institute of Hygiene and Applied Physiology (IHA) and European Co-partners.

Keywords: EMG analysis, data compression, supervised and unsupervised classification techniques, wavelet analysis

Analog Circuits

Analog VLSI Decoders for Error Correcting Codes

This project aims at developing analog VLSI decoders for the iterative decoding of error-correcting codes. It was motivated by some recent developments both in analog VLSI (bio-inspired networks) and in coding theory (turbo coding) that suggested the possibility of building analog VLSI decoders that are much more efficient than traditional digital VLSI decoders in terms of operating speed and/or power consumption.

The main challenge of this project was to identify suitable computational primitives (elementary circuits) on the transistor level. This first goal was achieved in the first year of the project: a „natural“ mapping of the sum-product algorithm onto transistor circuits was found that applies, in particular, to turbo codes, to conventional trellis codes, and to low-density parity check codes. These circuits reveal an interesting connection between semiconductor physics and probability theory. The proof of concept was established by building a demonstration unit for a small binary trellis code using discrete transistors. Swiss and international patent applications have been filed.To demonstrate the advantages of the new decoding approach, a first test chip for a binary (18, 9, 5) tailbiting trellis code was designed and fabricated in AMS 0.8um BiCMOS technology. Simulation results show the chip‘s robustness against non-idealities such as transistor mismatch, finite output resistance of MOS transistors, and temperature effects. Furthermore, measurement results show that bit rates of over 100Mbit/s can be achieved at a single 5V power supply and a power consumption of 50mW.

In a next step towards a full-sized decoding system, a test-chip for a more complex turbo-style code with digital interfaces was designed using the AMS 0.8um BiCMOS technology. Our decoders can be designed by construction using a C program to convert the parity-check matrix representation of the code into a Verilog structural description. Subsequently, this file can be imported into the Cadence IC design environment, and digital place-and-route tools can be used to generate the final layout of the decoder. This chip was fabricated, but the first attempt for chip-on-board (COB) packaging failed. However, we expect to have measurement results available shortly.

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Supported by: Swiss National Science Foundation
In Collaboration with: Prof. em. Dr. G.S. Moschytz; Prof. em. Dr. J.L. Massey; Endora Tech AG, Basel

Keywords: error-correcting codes, analog signal processing, analog VLSI, bio-inspired circuits

Single-Amplifier Biquadratic MOSFET-C Filters

This dissertation discusses the theory of single-amplifier biquadratic filters (SABs) and their implementation as CMOS video-frequency filters. It shows that building filters as cascades of single-amplifier biquadratic MOSFET-C sections is a viable alternative to using biquadratic Gm-C filter sections. The advantage of MOSFET-C SABs is that they typically use less chip area than a Gm-C filter with equivalent speed, distortion, noise, and power consumption.

The first part of this dissertation discusses the theory of integrated amplifiers, provides a new perspective of the current-mode vs. voltage-mode debate, and discusses the theory of SABs and the effects that amplifier non-idealities have on them.

The second part discusses second-order MOSFET-C networks and how to design filters with them, presents perfectly symmetrical video-frequency current amplifiers, one with fixed gain and one with variable gain, and contains measurement results of test circuits from two chips.

The third part presents a brief comparison of the MOSFET-C SABs presented in this dissertation to other video-frequency filters, and finishes with a discussion of design trade-offs and ideas for future research on the topic.

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3.3 Completed Research Projects

JOHO Marcel

A Systematic Approach to Adaptive Algorithms for Multichannel System Identification, Inverse Modeling and Blind Identification

ETH-Diss. Nr. 13783 (Referee: Prof. Dr. G.S. Moschytz)

A Systematic Approach to Adaptive Algorithms for Multichannel System Identification, Inverse Modeling, and Blind Identification. In many situations related to acoustics and data communications we are confronted with multiple signals received from a multipath mixture, e.g., the famous cocktail-party problem.

A multipath mixture can be described by a mixing matrix, whose elements are the individual transfer functions between a source and a sensor. The mixing matrix is usually unknown, and so are sometimes also the source signals.

Depending on the application, different parameters are of interest the mixing matrix for system identification, the inverse mixing matrix for inverse modeling, or the source signals for system equalization.

This thesis gives a systematic approach to the aforementioned problems in a multipath mixing environment.

To this end, we investigate the multichannel-mixing problem and the single-channel multipath problem separately.

Based on a mean-squared-error (MSE) cost function, several stochastic-gradient update equations, which are related to the least-mean-square (LMS) and the recursive least-squares (RLS) algorithm, are derived for the instantaneous mixing case.

Thereby the matrix-inversion lemma has shown to be a very powerful tool to transform an algorithm which estimates the mixing matrix (system identification) into an algorithm which estimates the inverse mixing matrix (inverse modeling).

With the help of circulant matrices, the adaptive algorithms for the multichannel instantaneous mixing case are transformed to cope with the single-channel multipath case. Block processing techniques are used, allowing efficient implementation of the filtering and adaptation in the frequency domain. The Fast Fourier Transform (FFT) plays a crucial role, owing to its close relationship to circulant matrices.

We extend the algorithms to operate as multichannel adaptive filters, using the fact that a multipath mixture is the combination of instantaneous mixing and single-channel multipath convolution.

In addition, we investigate the situation where not only the multipath-mixing system, but also the source signals are unknown. This situation is referred to as blind identification. By exchanging the non-blind error criterion with a blind error criterion, we derive new algorithms for blind identification (blind source separation, single-channel and multichannel blind deconvolution). The same technique provides an alternative derivation of the well-known natural-gradient learning algorithm for blind source separation, revealing new insight.

Throughout the thesis, many simulation examples illustrate the performance behavior of the different adaptive algorithms.

VON HOFF Thomas

On the Convergence of Blind Separation and Deconvolution

ETH-Diss. Nr. 13846 (Referee: Prof. Dr. H.-A. Loeliger)

This thesis considers the problems of Blind Source Separation (BSS), Blind Deconvolution (BD) and Multichannel Blind Deconvolution (MBD). New results on stability and steady-state error levels are presented. Computationally efficient algorithms are given for all problems.

Blind separation is referred to as the task where unknown but mutually independent signals are to be recovered from observed mixtures. The mixing process is described by a matrix that is also unknown. The goal of blind separation is to determine an appropriate demixing matrix.

Methods for achieving blind separation include the maximization of an information-theoretically motivated objective function, the approach taken in this thesis. This yields an adaptive nonlinear update rule. This nonlinear characteristic together with the distributions of the source signals impacts the convergence and equilibrium conditions making the convergence behavior of the algorithm an important issue.

The nonlinear algorithm has more than one equilibrium and solutions may exist that do not correspond to a separation of the source signals. It is essential that the separating solutions are the stable solutions of the algorithms.

The local convergence behavior of the algorithm around a solution is considered by analyzing the linearized update equation. This permits a statement on the conditions for stability. For the case where separating solutions do not correspond to a stable equilibrium, procedures for stabilization are presented and compared. This analysis shows that in the vicinity of the solution the “rate of convergence” and the steady-state error are proportional to the step size. For the special case of identically distributed source signals and identical nonlinear functions in the update equation, a closed-form expression of the steady-state error is determined.

If a fixed step size is employed in the adaptive algorithm for separation, a trade-off exists between speed and exactness. To achieve both at the same time a self-adjusting step size is proposed. It is based on an estimated measure of the squared error. This new procedure allows error-free convergence in a time-invariant mixing environment and an improved tracking behavior in the case of changing mixing conditions.

Blind deconvolution is the problem to recover an unknown source signal where only a convolved version is available. The corresponding update rule is derived starting from the related problem of blind circular deconvolution. The update equation is obtained by tracing the problem back to the special case of blind separation where the mixing matrix is circular. The extension to blind deconvolution is then direct because deconvolution with an infinitely long filter is equivalent to its circular counterpart. Using fast Fourier transforms the algorithm implementation is computationally efficient. The local convergence is analyzed and the stability issues are detailed and an approximation to the steady-state error presented.

The problem where the observed signals are mixtures of convolved sources is referred to as multichannel blind deconvolution. It can be considered as a combination of blind source separation and blind deconvolution. First, the corresponding adaptive update rule for multichannel blind circular deconvolution is derived. Second, it is extended to linear convolution by making the length of the filter infinite. The implementation follows the techniques used for blind deconvolution. Local convergence is again analyzed for stability and steady-state error.

For the special situation where only separation of the sources and not full deconvolution is desired a two-stage architecture is proposed. Making use of the computational structure of the mixing system's inverse this approach involves the adjustment of fewer parameters than multichannel blind deconvolution. This results in a faster convergence.

The theoretical results presented are supported by extensive simulations that verify all assumptions employed in the derivations.

LUSTENBERGER Felix

On the Design of Analog VLSI Iterative Decoders

ETH-Diss. Nr. 13881 (Referee: Prof. Dr. G.S. Moschytz)

The rapidly growing electronic networking of our society has created the need for a high-speed and low-power data communications infrastructure. Both voice and data communications have been made available for the mobile user. Additionally, more complex coding schemes and decoding algorithms have been introduced to protect the user data from corruption during the transmission over a communications channel. The aim of all these new coding and decoding approaches is to meet the theoretical channel capacity limit to make a better use of the signal power and channel bandwidth. The iterative probability-propagation-type algorithms that are used to decode state-of-the-art codes such as Turbo codes and low-density parity-check codes create the need for a tremendous computational power. Often, the computational complexity can not be implemented with a traditional digital design approach and a given power budget.

This thesis discusses the efficient implemention of high-performance decoding algorithms in analog VLSI technology. The building blocks are very simple analog translinear circuits that implement vector multipliers with basically only one transistor per element of the outer product of two discrete probability distributions. The presented analog probability propagation networks made of these building blocks are a direct image of the underlying sum-product algorithm. The design of these analog networks follows a heavily semiconductor-physics-centered bio-inspired design approach, that exploits, rather than fights against, the inherent nonlinear behaviour of the basic semiconductor devices. By using such a bio-inspired design approach, the performance of these networks in terms of speed or power-consumption or both is increased by at least a factor of 100 compared to digital implementations. Despite the use of very-low-precision circuit devices, a remarkable system-level accuracy can be achieved by such a large, highly-connected analog network.

The first part of the thesis discusses the background of channel coding and decoding and the theoretical foundations of factor graphs and the sum-product algorithm, which operates by message passing on such graphs. This part provides a brief introduction to the information-theoretic aspects of the interdisciplinary research effort.

The second part of the thesis is devoted to the actual transistor level implementation of the sum-product algorithm using very simple analog-VLSI computational building blocks. This part discusses the design-oriented aspects of the research, however, it relies heavily on the information-theoretic concepts introduced in the first part. Finally, we present practical designs and design studies of several decoding networks. Algorithmic simulations, circuit simulations, and, where available, measurement results of the implemented decoding networks are presented. Two of the decoder examples were actually fabricated in a 0.8µm BiCMOS process. Additionally, application-specific design problems are discussed.

The thesis is finished with a summary of the achieved results and a presentation of future research propositions in the field of analog decoding.

Keywords: Iterative decoding, low-density parity-check (LDPC) codes, repeat-accumulate (RA) codes, trellis codes, Turbo codes, maximum-a posteriori probability (MAP) decoder, maximum-likelihood (ML) sequence detection, sum-product algorithm, Viterbi algorithm, probability propagation, factor graphs, analog VLSI technology, bio-inspired networks.

SCHMID Hanspeter

Single-Amplifier Biquadratic MOSFET-Cfilters

ETH-Diss. Nr. 13878 (Referee: Prof. Dr. G.S. Moschytz)

This dissertation discusses the theory of single-amplifier biquadratic filters (SABs) and their implementation as CMOS video-frequency filters. It shows that building filters as cascades of single-amplifier biquadratic MOSFET-C sections is a viable alternative to using biquadratic Gm-C filter sections. The advantage of MOSFET-C SABs is that they typically use less chip area than a Gm-C filter with equivalent speed, distortion, noise, and power consumption.

The first part of this dissertation discusses the theory of integrated amplifiers, provides a new perspective of the current-mode vs. voltage-mode debate, and discusses the theory of SABs and the effects that amplifier non-idealities have on them.

The second part discusses second-order MOSFET-C networks and how to design filters with them, presents perfectly symmetrical video-frequency current amplifiers, one with fixed gain and one with variable gain, and contains measurement results of test circuits from two chips.

The third part presents a brief comparison of the MOSFET-C SABs presented in this dissertation to other video-frequency filters, and finishes with a discussion of design trade-offs and ideas for future research on the topic.

WELLIG Peter

Zerlegung von Langzeit-Elektromyogrammen zur Prävention von arbeitsbedingten Muskelschäden

ETH-Diss. Nr. 13881 (Referee: Prof. Dr. G.S. Moschytz)

To study the development of work-related chronic neck pain, it is necessary to measure the muscle-fibre activities, i.e. intramuscular, multi-channel long-term measurements have to be considered. The analysis of the measured signals, so-called electromyograms, is based on the decomposition of the signals into their basic signal units called motor-unit action potentials. Existing decomposition tools are restricted to short registration periods and mainly to single-channel recordings detected under constant muscle effort. In contrast, this thesis deals with algorithms for the decomposition of multi-channel long-term recordings detected under slight muscle movements.

A decomposition concept was developed which allows a fast and accurate analysis of multi-channel long-term recordings. Based on the decomposition concept and the algorithms considered, the decomposition program EMG-LODEC ("Electro-MyoGram Long-Term DEComposition") was written which decomposes one-, two-, or three-channel long-term recordings.

A decomposition of an electromyogram consists of several signal processing stages. First, the electromyogram is separated into inactive segments with low activity and active segments containing motor-unit action potentials. Using the active segments and a cluster-analysis technique, the number of classes, i.e., the number of motor units, is estimated. The detection of outliers and the use of a local group structure improve the estimation of the number of motor units compared to other clustering algorithms. Furthermore, supervised classification algorithms are used to classify non-overlapping action potentials of motor units already detected. Using multi-channel signal information and the weighted averaging method to track action-potential shape changes, the classification performance is improved compared to other algorithms which are use for the decomposition of short-term electromyograms. Finally, detected outliers, e.g., segments containing overlapping action potentials, are decomposed into their units using class mean signals. To improve the recognition rate, the signal resolution is increased by signal interpolation.

Feature analysis criteria show that most of the signal energy and most of the classification information is concentrated in a few wavelet coefficients. This is the reason, why the performance of the electromyogram data compression, the cluster analysis, and the supervised classification is improved using extracted wavelet coefficients instead of using time samples. In the case of electromyogram data compression, the wavelet-based embedded zero-tree encoding algorithm shows better results compared to other algorithms. Using the wavelet-based encoding algorithm, an SNR of 25 dB was achieved for a compression ratio of eight and for the signals considered. In the case of the clustering, feature analysis criteria show that the clusters can be better separated and outliers can be better detected by using extracted wavelet coefficients than by using bandpass-filtered time samples. And finally, in the case of the supervised classification of active segments, the wavelet-based linear discriminant analysis classifier achieves a higher recognition rate compared to other linear time-based classifiers. Therefore, extracted wavelet coefficients are considered for the data compression, the clustering, and the supervised classification of active segments.

Measured multi-channel long-term recordings were used to test the algorithms. EMG-LODEC was capable of detecting and tracking the long-term motor-unit activity of those signals. Furthermore, simulated multi-channel electromyograms were considered. Recognition rates of 91% to 100% were achieved using EMG-LODEC.

Keywords. Decomposition of intramuscular electromyograms, feature extraction, wavelet transform, data compression, cluster analysis, supervised classification.

ERNE Markus

Signal Adaptive Audio Coding using Wavelets and Rate Optimization

ETH-Diss. 13883 (Referee: Prof. Dr. G.S. Moschytz)

Perceptual Audio Coding algorithms have become very popular, not only due to professional applications (DAB, DVB, DVD, etc.) but additionally have gained an increasing market potential for consumer applications e.g mp3-player, downloading of compressed files, music distribution over the Internet, etc.

In this thesis, a different perspective of perceptual audio coding is presented. Based on the notation of music, where the notes can be chosen almost arbitrarily by the composer, a flexible subband-filterbank algorithm is introduced which has almost the same flexibility as the composer, who has selected the notes or time-frequency atoms carefully. This signal-adaptive wavelet-packet transform allows to trade time- versus frequency resolution and each subband can be switched individually. No boundary distortion will occur during the switching due to careful selection of the block-boundaries although switching costs will occur, if the filterbank is changed from the highest frequency resolution at the very bottom of the wavelet-packet tree to the root of the tree which the highest temporal resolution.

In the second part of the thesis, a perceptual model is introduced which not only allows to shape the quantization noise in frequency such that it can not be perceived by the human auditory system but additionally, the model takes care of temporal masking effects in order to shape the coding distortion in time. Different options for the detection of tonal or more noise-like maskers have been testes and implemented. The perceptual model is explained in detail and several test signals highlight the different steps in order to compute the overall masking threshold for a given block of samples.

In the third part of the thesis, a cost-function for the switching of the filterbank is introduced.

Based on the analysis of the perceptual model, the shape and the level of the masking curve as well as the temporal energy-characteristics of the input signal are evaluated in the cost function. Applying a recursive "split-merge" algorithm and a bottom-up strategy, the full wavelet-packet tree is pruned such that the best coding quality with the lowest amount of bits can be achieved. Such a process is highly connected to non-linear approximation techniques where the *n-best* coefficients of a orthogonal expansion are used for the approximation of a signal. The allocation of bits is introduced and an embedded bit-stream syntax is presented which offers graceful degradation of the signal in case of momentary limited transmission bandwidth.

In the fourth part, the algorithm was evaluated using subjective listening tests in a professional studio environment. A method, recommended by the ITU (ITU-R BS.1116) with a "double blind triple stimulus with hidden reference principle" using expert listeners with headphones, has proven to the best one. The results are very promising and the wavelet-coder showed superior performance over many competing coding schemes at equal bitrates and only for very tonal signals, such a pitch-pipe or flute, the wavelet coder showed a lower performance in terms of audio transparency due to the limited maximum frequency resolution provided by a maximum tree-depth of L=6 and the presence of aliased components

In the fifth part, the overall implementation of the algorithm is briefly sketched. All algorithms have been evaluated in Matlab and then have been ported to C++. The encoder and decoder together, run in real-time on a 600 MHz Pentium PC.

Keywords: Wavelets, signal-adaptivity, perceptual model, audio coding, MPEG, embedded coding, non-linear approximation

3.4 Completed Dissertations

JOHO Marcel	A Systematic Approach to Adaptive Algorithms for Multichannel System Identification, Inverse Modeling and Blind Identification *ETH Diss. Nr. 13783* Referee: Prof. em. Dr. G.S. Moschytz Co-Referees: Prof. S.C. Douglas Prof. Dr. H.-A. Loeliger
VON HOFF Thomas	On the Convergence of Blind Separation and Deconvolution *ETH Diss. Nr. 13846* Referee: Prof. Dr. H.-A. Loeliger Co-Referees: Prof. Dr. A. G. Lindgren Dr. A. Kälin
SCHMID Hanspeter	Single-Amplifier Biquadratic MOSFET-C Filter *ETH Diss. Nr. 13878* Referee: Prof. em. Dr. G.S. Moschytz Co-Referee: Prof. Dr. Q. Huang
LUSTENBERGER Felix	On the Design of Analog VLSI Interative Decoders *ETHDiss. Nr. 13879* Referee: Prof. em. Dr. G.S. Moschytz Co-Referees: Prof. D.A. Johns Prof. Dr. H.-A. Loeliger
WELLIG Peter	Zerlegung von Langzeit-Elektromyogrammen zur Prävention von arbeitsbedingten Muskelschäden *ETH Diss. Nr. 13881* Referee: Prof. em. Dr. G.S. Moschytz Co-Referees: Dr. med. Th. Läubli Prof. Dr. H.-A. Loeliger
ERNE Markus	Signal Adaptive Audio Coding using Wavelets and Rate Optimization *ETH Diss. Nr. 13883* Referee: Prof. em. Dr. G.S. Moschytz Co-Referees: Prof. Dr. M. Vetterli Prof. Dr. J. Blatter Prof. Dr. A. Kündig

4. Congresses, Meetings and Committees

4.1 Congress Organization

Prof. Lapidoth

Organizer of the Third ETH-Technion Workshop on Information Theory, Zurich, Switzerland, January 2000.

Session Organizer, 2000 Conference on Information Sciences and Systems, Princeton, USA.

Member Program Committee, 2001 IEEE International Symposium on Information Theory, Washington, USA.

Co-Chair Program Committee, 2002 IEEE International Symposium on Information Theory, Lausanne, Switzerland.

Prof. Loeliger

Organizer of the 5th ETHZ-EPFL Summer School on Linear, Nonlinear, and Adaptive Circuits, Systems, and Signal Processing, together with Prof. M. Hasler, CIRC EPFL and Prof. Leon Chua, UC Berkeley.

Prof. Moschytz

Member of the Scientific Committee for EUSIPCO, Brussels.

International Zurich Seminar on Digital Communications: Steering Committee (as Chairman of the IEEE Switzerland Chapter on Digital Communications).

Program Committee and Steering Committee of ICECS (International Conference on Electronics, Circuits, and Systems).

PROCID Meeting, Zurich, Switzerland.

Steering Committee IEEE International Symposium on Circuits and Systems.

4.2 Participation in Congresses and Meetings

Arnold Dieter Vontobel Pascal	Third ETH-Technion Workshop on Information Theory, ETH Zurich, Switzerland, 19.-21.1.00.
Arnold Dieter Vontobel Pascal	Winter School on Coding and Information Theory, Schloss Reisensburg, Guenzburg, Germany, 17.-20.12.00.
Erne Markus	108th AES-Convention, Paris, France, 19.-22.2.00.
Erne Markus	SPIE International Conference on Wavelet Applications, Orlando, USA, 25.-28.4.00.
Erne Markus	52th ISO-MPEG-Meeting on MPEG4, Geneva, Switzerland, 31.5.-2.6.00.
Erne Markus	109th AES-Convention, Los Angeles, USA, 22.-25.9.00.
Erne Markus	COST G6 Conference on Digital Audio Effects, Verona, Italy, 7.-9.12.00.
Gao Qun	CNNA'2000, Cellular Neural Networks and their Applications, Catania, Italy, 23.-25.5.00.
Hofbauer Markus Lippuner Daniel Mathis Heinz von Hoff Thomas Vontobel Pascal	5th ETHZ-EPFL Summer School: Probabilistic and Adaptive Signal Processing, ETH Zurich, Switzerland, 13.-14.7.00.
Heutschi Kurt	DAGA 2000, 26. Deutsche Jahrestagung Akustik, Oldenburg, Germany, 20.3.00.
Heutschi Kurt	Tagung der Schweizerischen Gesellschaft fuer Akustik, Lugano, Switzerland, 25.-26.5.00.
Heutschi Kurt	Tagung der Fachkommission fuer Hochspannungsfragen, Olten, Switzerland, 4.10.00.
Heutschi Kurt	Tagung der Schweizerischen Gesellschaft fuer Akustik, Chur, Switzerland, 19.-20.10.00.
Kretzschmar Ralf	Forschungskolloqium 2000 MeteoSchweiz, MeteoSchweiz Zurich, Switzerland, 17.5.00.
Kretzschmar Ralf	IEEE-INNS-ENNS International Joint Conference on Neural Networks (IJCNN'2000), Como, Italy, 24.-27.7.00.
Kretzschmar Ralf	The Seventeenth National Conference on Artificial Intelligence (AAAI'2000), Austin, Texas, USA, 30.7.-3.8.00.
Kretzschmar Ralf	Research Stay at the University of Houston, Texas, USA, 1.7.-21.7.00 and 29.7.-25.8.00.

Kretzschmar Ralf	The Seventeenth National Conference on Artificial Intelligence (AAAI'2000), Workshop on Learning from Imbalanced Data Sets, Austin, Texas, 31.7.00.
Kretzschmar Ralf	2000 IEEE International Workshop on Neural Networks for Signal Processing (INNSP'2000), Sydney, Australia, 11.-13.12.00.
Lapidoth Amos	MIT, USA, 7.-15.2.00.
Lapidoth Amos	IZS 2000 International Zurich Seminar on Broadband Communications, Zurich, Switzerland, 15.-17.2.00.
Lapidoth Amos	CISS 2000 Princeton, USA, 14.-17.3.00.
Lapidoth Amos	21st IEEE Convention of the Electrical and Electronic Engineers in Israel, Tel Aviv, Israel, 11.-12.4.00.
Lapidoth Amos	MIT, USA, 5.-9.5.00.
Lapidoth Amos	2000 IEEE Int. Symposium on Information Theory, Sorrent, Italy, 25.-30.6.00.
Lapidoth Amos	MIT, USA, 31.7.-3.8.00.
Lapidoth Amos	Information Theory Summer Workshop, Ithaca, USA, 17.-20.8.00.
Lapidoth Amos	HP-Microsoft Workshop, Random matrices, percolation and queues, Bristol, UK, 11.-15.9.00.
Lapidoth Amos	MIT, USA, 28.-29.9.00.
Lapidoth Amos	ETH Zürich – Siemens Munich Workshop, Munich, Germany 6.10.00.
Lapidoth Amos	MIT, USA, 6.11.00.
Lapidoth Amos	Technion, Haifa, Israel, 24.-29.11.00.
Lapidoth Amos Moser Stefan	Winter School on Coding and Information Theory, University of Ulm, Reisensburg, Germany, 17.-20.12.00.
Lippuner Daniel	Research Stay at Technischen Universitaet Darmstadt, Germany, 15.-19.5.00.
Loeliger Hans-Andrea Arnold Dieter Vontobel Pascal	IEEE International Symposium on Information Theory, Sorrento, Italy, 25.-30.6.00.
Lustenberger Felix	Miniaturisierte Elektronik mit modernen Aufbautechniken – Entwicklung und Einsatz, IFE (ETHZ), Zurich, 5.-6.9.00.
Lustenberger Felix Mathis Heinz Schmid Hanspeter	ISCAS 2000 International Conference on Circuits and Systems, Geneva, Switzerland, 29.-31.5.00.

Mathis Heinz	IZS 2000 International Zurich Seminar on Broadband Communications, Zurich, Switzerland, 15.-17.2.00.
Mathis Heinz	Didaktik-Workshop, ETH Zurich, Switzerland, 7.-9.3.00.
Mathis Heinz	ICA 2000, Independent Component Analysis and Blind Signal Separation, Helsinki, Finland, 29.-22.6.00.
Mathis Heinz	e-Business Symposium, Hilton Zurich Airport, Zurich, Switzerland, 11.9.00.
Mathis Heinz	Technology Leadership Day, Rapperswil, Switzerland, 10.10.00.
Schmid Hanspeter	Norchip 2000, Turku, Finland, 6.-7.11.00.
von Hoff Thomas	Second International Workshop on Independent Analysis and Blind Signal Separation, Helsinki, Finland, 19.-22.7.00.
von Hoff Thomas	10th European Signal Processing Conference, Tampere, Finland, 4.-8.9.00.
Vontobel Pascal	International Zurich Seminar, ETH Zurich, Switzerland, 15.-17.2.00.
Vontobel Pascal	Research Stay with University of Notre Dame, South Bend, IL, USA, 20.9.-3.10.00.
Vontobel Pascal	38th Annual Allerton Conference on Communication, Computing and Control, Monticello, IL, USA, 4.-6.10.00.
Vontobel Pascal	Research Stay with University of California at Santa Cruz (UCSC), Santa Cruz, CA., USA, 15.7.-15.9.00.
Wellig Peter	PROCID Meeting, Zurich, Switzerland, 26.3.-28.3.00.

4.3 Service Activities and Society Memberships

Prof. Lapidoth

Senior Member of the IEEE New York

Member of Search Committee for the Professorship in Wireless Communications, ETHZ

Prof. Loeliger

Member of IEEE

Chairman of the IEEE Switzerland Chapter on Digital Communication Systems

Prof. Moschytz

Member of the Swiss Section of the IEEE

Member of the Planning Committee of the EE Dept., ETHZ

Chairman of the IEEE Switzerland Chapter on Digital Communication Systems

Member of the Editorial Board of the "International Journal of Circuit Theory and Applications", Publ. John Wiley & Sons, Chichester, GB

Member of the European Editorial Board of the journal: "Journal of Circuits, Systems and Computers," Scientific Publ. Co., Singapore, New Jersey, London, Hongkong

Member of the Editorial Board of the International Journal "Analog Integrated Circuits and Signal Processing", Kluwer Academic Publishers, Norwell MA, USA

Member of the international Editorial Board of the newly appearing "Annales des Télécommunications", Issy-les-Moulineaux, France

Swiss Committee of URSI, Member and Deputy of Commission C

Fellow of the IEEE, New York

Member, Swiss Electrical Engineering Society

Member, Swiss Academy of Engineering Sciences

Past-President of IEEE Circuits and Systems Society

TAB (Technical Acitvities Board) Committee of IEEE Circuits and Systems Society

Dr. Heutschi

Member, Acoustical Society of America

Member, Audio Engineering Society

Member, Swiss Acoustical Society (SGA)

4.4 Presentations by Institute Members

Arnold Dieter	“A New Model for Magnetic Recording: The Binary Jitter Channel”, International Symp. On Information Theory, Sorrento, Italy, 30.6.00.
Arnold Dieter	“Computing the Information Rate of the DICODE Channel”, 6th Winterschool on Coding and Information Theory, Schloss Reisensburg, Guenzburg, Germany, 19.12.00.
Erne Markus	“Nouveaux horizons dans le codage audio: au-déla de MP3”, 5ème Congrès Français d’Acoustique, EPFL Lausanne, Switzerland, 3.-6.9.00.
Erne Markus	“A Bit-Allocation Scheme for an Embedded and Signal Adaptive Audio Coder”, 108th AES-Convention, Paris, France, 19.-22.2.00.
Erne Markus	“Audio Coding Based on Rate-Distortion and Perceptual Optimization Techniques”, SPIE International Conference on Wavelet Applications, Orlando, USA, 25.-28.4.00.
Joho Marcel	“Combining Blind and Non-Blind Algorithms”, Daimler Chrysler Forschungslabor, Ulm, Germany, 3.8.00.
Joho Marcel	“Connecting Partitioned Frequency-Domain Filters”, Beckman Institute, University of Illinois at Urbana-Champaign, IL, USA, 13.9.00.
Joho Marcel	“A Systematic Approach to Multichannel Systemidentification, Inverse Modeling, and Blind Identification”, Technical University Eindhoven, Holland, 4.12.00.
Joho Marcel	“Blind Source Separation for Noise Reduction”, Workshop Auditory Scene Analysis for Noise Reduction, Phonak AG, Staefa, Switzerland, 15.12.00.
Kretzschmar Ralf	“A Comparison of Feature Sets and Neural Network Classifiers on a Bird Removal Approach for Wind Profiler Data”, IEEE-INNS-ENNS International Joint Conference on Neural Networks (IJCNN’2000), Como, Italy, 24.-27.7.00.
Kretzschmar Ralf	“NEURO-BRA: A Bird Removal Approach for Wind Profiler Data Based on Quantum Neural Networks”, IEEE-INNS-ENNS International Joint Conference on Neural Networks (IJCNN’2000), Como Italy, 24.-27.7.00.
Kretzschmar Ralf	“Quantum Neural Networks versus Conventional Feedforward Neural Networks: An experimental study”, 2000 IEEE International Workshop on Neural Networks for Signal Processing (NNSP’2000), Sydney, Australia, 11.-13.12.00.
Lapidoth Amos	“Fading Channels with Estimation Errors“, 3rd ETH-Technion Workshop on Information Theory, Zurich, Switzerland, 20.1.00.

Lapidoth Amos Telatar Emre	“Gaussian ISI Channels and the Generalized Likelihood Ratio Test“, CISS 2000, Princeton, USA, 15.-17.3.00.
Lapidoth Amos	“On the Capacity of Reduced Complexity Receivers for Intersymbol Interference Channels“, The 21st IEEE Convention of the Electrical and Electronic Engineers in Israel, Tel Aviv, Israel, 12.4.00.
Lapidoth Amos Telatar Emre	“Gaussian ISI Channels and the Generalized Likelihhood Ratio Test“, ISIT 2000, Sorrento, Italy, 25.-30.6.00.
Lapidoth Amos Cohen Aaron	“On the Gaussian Watermarking Game“, ISIT 2000, Sorrento, Italy, 25.-30.6.00.
Lapidoth Amos	“The Digital Watermarking Game“, ETH-Siemens Workshop, Munich, Germany, 6.10.00.
Lapidoth Amos	“Limits on Reliable Communication over Flat-Fading Channels“, Forschungszentrum Telekommunikaton, Vienna, Austria, 15.12.00.
Lapidoth Amos Moser Stefan	“Limits on Reliable Communication over Flat-Fading Channels“, Winter School on Coding and Information Theory, Schloss Reisensburg, Germany, 19.12.00.
Cohen Aaron Lapidoth Amos	“On the Gaussian Watermarking Game“, CISS 2000, Princeton, USA, 15.-17.3.00.
Abou Faycal Ibrahim Lapidoth Amos	“On the Capacity of Reduced-Complexity Receivers for InterSymbol Interference Channels, CISS 2000, Princeton, USA, 15.-17.3.00.
Lippuner Daniel	“Step-Size Control in Model-Based Adaptive Algorithms”, Technische Universitaet, Fachgruppe Theorie der Signale, Prof. Dr. E. Haensler, Darmstadt, Germany, 15.5.00.
Lippuner Daniel	“The Kalman Filter for Nonstationary Environments”, ETHZ-EPFL Summerschool, Zurich, Switzerland, 14.7.00.
Loeliger Hans-Andrea	“Adaptive Equalization: Factor Graphs and Message Passing Algorithms“, Forney Fest, Cambridge, MA, USA, 3.-4.3.00.
Loeliger Hans-Andrea	“Factor Graphs, Belief Propagation Algorithms, and Analog Computation”, MIT, Lids, USA, April 00.
Loeliger Hans-Andrea	“Probabability Propagation Networks – Theory and Analog Circuits”, INTELECT Summer School, Orebro, Sweden, 14.-16.8.00.
Lustenberger Felix	“Design of Analog VLSI Iterative Decoders (DAVID)”, Nachdiplomstudium BWI (ETHZ), Zurich, Switzerland, 10.1.00.
Lustenberger Felix	“Design of Analog VLSI Iterative Decoders (DAVID), CSEM Neuchâtel, Switzerland, 12.1.00.
Mathis Heinz	“A Simple Threshold Nonlinearity for Blind Separation of Sub-Gaussian Signals”, ISCAS 2000, Geneva, Switzerland, 31.5.00.
Mathis Heinz	“Blind Separation of Mixed-Kurtosis Signals using an Adaptive Threshold Nonlinearity”, ICA 2000, Helsinki, Finland, 20.6.00.

Mathis Heinz	"Blind Deconvolution in Communication Systems", ETHZ-EPFL Summer School, Zuerich, Switzerland, 14.7.00.
Mathis Heinz	"Blind Deconvolution in Communication Systems", Daimler-Chrysler Forschungszentrum, Ulm, Germany, 3.8.00.
Schmid Hanspeter	"A Charge-Pump-Controlled MOSFET-C Single-Amplifier Biquad", ISCAS 2000, Geneva, Switzerland, 28.-31.5.00.
Schmid Hanspeter	"8.25 MHz 7th-order Bessel filter built with MOSFET-C single-amplifier biquads", Norchip 2000, Turku, Finland, 6.-7.11.00.
von Hoff Thomas	"Step-Size Control in Blind Source Separation", Second International Workshop on Independent Component Analysis and Blind Signal Separation, Helsinki, Finland, 21.6.00.
von Hoff Thomas	"Blind Source Separation and Deconvolution: Stability and Performance", Daimler Chrysler AG, Ulm, Germany, 2.8.00.
von Hoff Thomas	"Stability and Performance of Adaptive Algorithms for Multichannel Blind Source Separation and Deconvolution", 10th European Signal Processing Conference, Tampere, Finland, 6.9.00.
Vontobel Pascal	"The Binary Jitter Channel: A New Model for Magnetic Recording", Dept. of Electrical Engineering, University of Notre Dame, South Bend, IL, USA, 28.9.00.
Wellig Peter Zennaro Daniel	"Zerlegung von Elektromyogrammen zur Praevention von arbeitsbedingten Muskelschaeden", Workshop ueber Biosignalverarbeitung und ihr Stellenwert in der medizinischen Informatik, Muenchen, Germany, 13.-14.7.00.

4.5 Organization of Lectures, Seminars, and Colloquia

Colloquium Speakers for the Colloquium “Electronics and Communications“ were:

Invited by Prof. Lapidoth:

19.06.00 **Prof. Robert M. Gray**, Information Systems Lab, Dept. of Electrical Engineering, Stanford University, USA,
“Gauss Mixture Vector Quantization**“**

Invited by Prof. Loeliger:

03.03.00 **Prof. Arie Arbel**, Technion – Israel Institute of Technology, Haifa, Israel,
“Selected Chapters from A/D Design“.

Invited by Dr. Heutschi

19.01.0 **Stefan Launer, Dr. rer. Nat.,** Phonak AG, Stäfa,
“Von der Physik des Hörens zur akustischen Wahrnehmung“.

02.02.00 **Robert Attinger, Dr. phil. Nat**., Grolimund & Partner AG, Bern,“Lärmmindernde Fahrbahnbeläge“.

17.05.00 **Colin McCulloch,** LMS International Leuven, Belgium,
“Vibro-Acoustic Modelling: Where are we, where should we go, and how?“.

30.11.00 **Peter Mapp,** Peter Mapp Associates, Colchester, UK,
“State of the Art Sound Reinforcement and PA Systems Design – defining, achieving and verifying the requirements“.

5. Publications

Arnold Dieter	"High-Rate Low-Density Parity-Check Codes: Construction and Application", IEEE Proceedings of the 2nd International Symposium on Turbo Codes, Brest, France, pp. 447-450 , September 00.
Arnold Dieter	"Computing the Information Rate of the DICODE Channel", IEEE Proceedings of the 6th Winterschool on Coding and Information Theory, Schloss Reisensburg, Guenzburg, Germany, pp. 15, December 00.
Arnold Dieter Kavcic Aleksander Kötter Ralf Loeliger Hans-Andrea Vontobel Pascal	"The Binary Jitter Channel: A New Model for Magnetic Recording", Proceedings of IEEE International Symposium on Information Theory, Sorrento, Italy, p. 433, June 00.
Erne Markus	"A Bit-Allocation Scheme for an Embedded and Signal Adaptive Audio Coder", AES 108th convention, Paris, France, Preprint 5083, February 00.
Erne Markus	"Audio Coding Based on Rate-Distortion and Perceptual Optimization Techniques", Proceedings of SPIE, Orlando, Wavelet Applications VII, vol. 4056, pp. 235-246 , April 00.
Erne Markus	"Noveaux horizons dans le codage audio: au-déla de MP3", Proceedings of the 5th French Congress on Acoustics, Lausanne, pp. XXIII-XXXIII, September 00.
Gao Qun	"Computer erkennen Dich", Bulletin: Magazin der ETH Zurich, Switzerland, No. 278, pp. 22-25, September 00.
Joho Marcel Moschytz George S.	"Connecting Partitioned Frequency-Domain Filters in Parallel or in Cascade", IEEE Transactions on Circuits and Systems-II, no. 8, vol. 47, pp. 685-698, August 00.
Kitahara Teruyo Schnoz Michael Laeubli Thomas Wellig Peter Krueger Helmut	"Motor-Unit Activity in the Trapezius Muscle during Rest, while Inputting Data, and during Fast Finger Tapping", European Journal of Applied Physiology, 2000, vol. 83, pp. 181-189.
Kretzschmar Ralf	"A Comparison of Feature Sets and Neural Network Calssifiers on a Bird Removal Approach for Wind Profiler Data", IEEE-INNS-ENNS International Joint Conference on Neural Networks (IJCNN'2000), Como, Italy, pp. 279-284, July 00.

Kretzschmar Ralf	“NEURO-BRA: A Bird Removal Approach for Wind Profiler Data Based on Quantum Neural Networks”, IEEE-INNS-ENNS International Joint Conference on Neural Networks (IJCNN’2000), Como, Italy, pp. 373-378, July 00.
Kretzschmar Ralf	“Quantum Neural Networks versus Conventional Feedforward Neural Networks: An Experimental Study”, 2000 IEEE International Workshop on Neural Networks for Signal Processing (NNSP’2000), Sydney, Australia, pp. 328-337, December 00.
Lapidoth Amos Telatar Emre	“Gaussian ISI Channels and the Generalized Likelihood Ratio Test”, Proceedings CISS 2000, Princeton, USA, vol. I, pp. TA-13-16, March 00.
Cohen Aaron Lapidoth Amos	“On the Gaussian Watermarking Game“, Proc. CISS 2000, Princeton, USA, vol. I, TA4-21-26, March 00.
Abou-Faycal Ibrahim Lapidoth Amos	“On the Capacity of Reduced Complexity Receivers for Intersymbol Interference Channels“, Proceedings CISS 2000, Princeton, USA, vol. I, WA4-32-37, March 00.
Lapidoth Amos Sallaway Peter J.	“Convolutional Encoders to Minimize Bit-Error-Rate“, ETT Vol. 11, No. 3, pp. 263-269, May/June 00.
Ganti Anand Lapidoth Amos Telatar Emre	“Mismatched Decoding Revisited: General Alphabets, Channels with Memory, and the Wide-Band Limit“, IEEE Transactions on Information Theory, Vol. 46, No. 7, pp. 2315-2328, November 00.
Cohen Aaron Lapidoth Amos	“On the Gaussian Watermarking Game“, Proceedings ISIT 2000, Sorrento, Italy, p. 48, June 00.
Lapidoth Amos Telatar Emre	“Gaussian ISI Channels and the Generalized Likelihood Ratio Test”, Proceedings ISIT 2000, Sorrento, Italy, p. 460, June 00.
Lapidoth Amos Moser Stefan	„Limits on Reliable Communication over Flat-Fading Channels“, Proceedings Winter School on Coding and Information Theory 2000, Schloss Reisensburg, Germany, December 00.
Lustenberger Felix	“On the Design of Analog VLSI Iterative Decoders”, PH.D. Dissertation, Series in Signal and Information Processing, vol. Hartung-Gorre Verlag Konstanz, ISBN 3-89649-622-0, 202 pages, November 00.
Mathis Heinz	“Differential Detection of GMSK Signals with Low BT Using the SOVA”, ETH Zurich, Switzerland, Nachdiplomarbeit, Tech. Rep. No. 200002, 21.2.00.
Mathis Heinz Joho Marcel Moschytz George S.	“A Simple Treshold Nonlinearity for Blind Separation of Sub-Gaussen Signals”, Proceedings of ISCAS 2000, Geneva, Switzerland, vol. IV, pp. 489-492, May 00.

Mathis Heinz von Hoff Thomas Joho Marcel	"Blind Separation of Mixed-Kurtosis Signals Using an Adaptive Threshold Nonlinearity", Proceedings of ICA 2000, Helsinki, Finland, pp. 221-226, June 00.
Mathis Heinz Vontobel Pascal	"Shape Optimization of a Rectangularly Constrained Small Loop Antenna", Proceedings Intrnational Zurich Seminar, Zurich, Switzerland, pp. 73-76, February 00.
Moschytz, George S. Hofbauer Markus	"Adaptive Filter, Eine Einfuehrung in die Theorie mit Aufgaben und MATLAB-Simulationen auf CD-ROM", Textbook Springer –Verlag, 246 Seiten, September 00.
Rosenthal Joachin Vontobel Pascal	"Construction of LDPC Codes Based on Ramanujan Graphs and Ideas from Margulis", Proc. 38th Annual Allerton Conference on Communication, Computing and Control, Monticello, Illinois, USA, October 00.
Schaerer Thomas	"Einschaltstrombegrenzung für Netzteile mit Ringkerntrafos (II)", MEGALINK, p. 38, 15.6.00.
Schaerer Thomas	"SC-Filter, nach wie vor aktuell: Switched-Capacitor-Filter, kurze Einführung und praktische Anwendung", p. 36, MEGALINK, 11.12.00.
Schmid Hanspeter	"Single-Amplifier Biquadratic MOSFET-C Filters", Dissertation , Hartung-Gorre Verlag, Konstanz, Series in Signal and Information Processing, ISBN 3-89649-616-6, ISSN 1616-671X, November 00.
Schmid Hanspeter	"Approximating the Universal Active Element", IEEE Transactions on Circuits and Systems-II, vol. 47, no. 11, pp. 1160-1169, November 00.
Schmid Hanspeter Moschytz George S.	"A 8.25 MHz 7th-order Bessel Filter built with MOSFET-C Single-Amplifier Biquads", Proceedings of the NORCHIP, Turku, Finland, pp. 217-224, November 00.
Schmid Hanspeter Moschytz George S.	"Active MOSFET-C Single-Amplifier Biquadratic Filters for Video Frequencies", IEE Proceedings on Circuits, Devices and Systems (Special Issue on Hifh-Frequency Analogue Filters), vol. 147, no. 1, pp. 35-41, February 00.
Schmid Hanspeter Moschytz George S.	"A Charge-Pump-Controlled MOSFET-C Single-Amplifier Biquad", Proceedings of the ISCAS, Geneva, vol. 2, pp. 677-680, February 00.
von Hoff Thomas Lindgren Allen G. Kaelin August	"Step-Size Control in Blind Source Separation", Proceedings of the Second International Workshop on Independent Component Analysis and Blind Signal Separation, Helsinki, Finland, pp. 509-514, June 00.
von Hoff Thomas Lindgren Allen G. Kaelin August	"Transpose Properties in the Stability and Performance of the Classic Adaptive Algorithms for Blind Source Separation and Deconvolution", Signal Processing, vol. 80, no. 9, pp. 1807-1822.

von Hoff Thomas Lindgren Allen G. Kaelin August	“Stability and Performance of Adaptive Algorithms for Multichannel Blind Source Separation and Deconvolution”, Proceedings of the tenth European Signal Processing Conference, Tampere, Finland, vol. 2, pp. 861-864, September 00.
Vontobel Pascal	“Using finite Geometries for Deriving Codes that can be Decoded Iteratively”, Proc. Winter School on Coding and Information Theory, Schloss Reisensburg, Guenzburg, Germany, December 00.
Wellig Peter	“Zerlegung von Langzeit-Elektromyogrammen zur Praevention von arbeitsbedingten Muskelschaeden”, Hartung-Gorre Verlag, ISBN 3-89649-632-9, November 00.

6. Guests, Visitors

6.1 Activities of Academic Guests at the Institute

Guests of Prof. Lapidoth:

Aaron S. Cohen	MIT, Cambridge, USA, studied and presented a talk on watermarking for data protection.	01.01. - 28.01.00
Ibrahim Abou Faycal	MIT,Cambridge,USA, studied and presented a talk on an information-theoretic approach to the design of reduced-complexiy receives for ISI channels.	01.01. - 28.01.00
Prof. Stuart Schwartz	Princeton, USA, presented a talk on “Adaptive Equalization Architectures for High Speed CDMA Networks“.	20.06.00

Guests of Prof. Loeliger:

Prof. Allen Lindgren	University of Rhode Island, Kingston, USA Collaboration with the Adaptive Filter Group.	01.09. - 30.09.00

Guests of Prof. Moschytz:

Prof. Leon Chua:	University of California, Berkely, USA held a lecture on “Exploiting Chaos in Wireless Communication Systems“.	15.05. - 14.08.00