Signal and Information Processing Laboratory

Prof. Dr. G.S. Moschytz (Director) / Prof. Dr. J.L. Massey
Prof. Dr. F. Eggimann / Prof. Dr. A. Kälin / Dr. K. Heutschi

ANNUAL REPORT

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Research Period 1997

Teaching Period 1996/97

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Foreword

It is with great pleasure that we at the Signal and Information Processing Laboratory (ISI) reach out once again to our many friends to present our annual report on the events at ISI in the past year.

As will be apparent from reading the contents of the annual report, 1996 was a successful and active year with the usual turnover of doctoral students completing their thesis and leaving ISI, and young new teaching assistants coming aboard to fill the resulting vacant slots. Thomas Ernst completed his thesis on “Adaptive Detectors for Data Communications over Recursive Channels“ and joined the Swiss Bank Corporation in Basel; Pius Estermann completed his thesis on “Adaptive Filters in the Frequency Domain: Analysis and Design Strategies“ and started work at Siemens Schweiz AG in Zurich; Carlo Harpes completed his thesis on “Cryptanalysis of Iterated Block Ciphers“ and returned to Luxembourg, where he is working at Weyderf in Fentage; Daniel Müller finished his thesis on “Hybrid Echocompensation with Applications in Digital Data Communications over Copper Wires“ and accepted a position with Philips Semiconductors AG in Zurich, and Christian Waldvogel completed his thesis entitled “On the Nature of Authentication Protocols“ and joined Eutelsat in Paris. As always, it was with mutual mixed feelings that these outstanding young researchers took their leave from ISI: on the one hand they were reluctant to leave what had become a close bond in research activities and friendship within ISI, and on the other they were eager to take on the challenge of applying their newly acquired expertise in an entirely different working environment.

Our freshly promoted PhDs were soon replaced at ISI by new assistants: Dani Lippuner and Thomas von Hoff, who had just graduated as ETH diplom engineers, and Madhu Reddy who joined us from the California State University at Long Beach, California, USA (and who, incidentally, is the son of our popular frequent guest and visiting professor, Dr. Hari Reddy). Another newcomer, dipl. El. Ing. Markus Erne, graduate of ETH, joined ISI as a research engineer after having spent nearly a decade in industry, during the last few years of which he ran his own company Scopein, which specializes in signal processing equipment for communications.

One very important activity of ISI is to host guests from academic and research institutes from all over the world. The last year was no exception; we had interesting visitors from the USA, UK, Israel, Spain and China. These contacts never fail to stimulate new ideas and, very often, result in ongoing joint research activities and formal projects. These visits, as well as most of our other activities, require support both financially and administratively. It is a pleasure, once again, to thank the EE department chairman and his staff, as well as the ETH administration, for providing this support generously and forthrightly at all times. Warmest thanks go also to all the members of ISI, whose constant cooperation, motivation and good will, in the final resort, are responsible for the fine accomplishments of our ISI.

Mai 1997

Prof. Dr. G.S. Moschytz Prof. Dr. J.L. Massey

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1. Personnel

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

Prof. Dr. George S. Moschytz

Professor for Digital Systems Engineering (Coding and Information Theory):

Prof. Dr. James L. Massey

Professor for Information Technology:

Prof. Dr. Fritz Eggimann

Adjunct Lecturer:

Dr. K. Heutschi

Assistant Professsor (Signal Processing):

Prof. Dr. August Kälin

Secretaries:	**Mrs. Bernadette Röösli**		
	Mrs. Renate Agotai		
	Mrs. Heidi Schenkel		
Administr.Supervisor:	**Dr. Markus Helfenstein**		
Technical Supervisor	**Dr. Max Dünki**,		
Teaching Assistants:	**Dieter Arnold**	Dipl.El.Eng.	since 1.11.97
	Kukorelly Zsolt	Dipl.Math	since 1.4.97
	Heinz Mathis	Dipl.El.Eng.	since 1.8.97
	Thomas von Hoff	Dipl.El.Eng.	
	Pascal Vontobel	Dipl. El.Eng.	since 1.4.97
Research Assistants:	**Richard De Moliner**	Dipl.El.Eng.	
	Markus Erne	Dipl.El.Eng.	
	Marcel Joho	Dipl.El.Eng.	
	Beat Keusch	Dipl.El.Eng.	
	Gerhard Krämer	Master of Sci.	
	Xuejia Lai	Dr.	
	Drahoslav Lim	Master of Sci.	
	Martin Hänggi	Dipl.El.Eng.	
	Dani Lippuner	Dipl.El.Eng.	
	Urs Loher	Dipl.El.Eng.	
	Hans-Andrea Löliger	Dr.	
	Felix Lustenberger	Dipl.El.Eng.	
	Merk Marcel	El.Eng. HTL	since 1.8.97
	Bahram Mirzai	Dipl. Phys.	
	Stefan Oberle	Dipl.El.Eng.	

	Andreas Poncet	Dipl.El.Eng.	left on 28.1.97
	Madhu Reddy	Master of Sci.	left 31.7.97
	Jossy Sayir	Dipl.El.Eng.	
	Hanspeter Schmid	Dipl.El.Eng.	
	Rolf Steiner	Dipl.El.Eng.	
	Jürg Stettbacher	Dipl.El.Eng.	
	Felix Tarköy	Dr.	
	Peter Wellig	Dipl.El.Eng.	
	Sigi Wyrsch	Dipl.El.Eng	
Technical Staff:	**Francesco Amatore**		
	Petro Gavriilidi	left on	
	Felix Frey	El.Eng.HTL	
	Thomas Schaerer		
	Schweizer Patrick	El.Eng.HTL	since 1.12.97

Fellowship Recipinets: (see 6.1 for report of activities:

A. Jurisic	University of Zagreb, Zagreb	15.10. - 31.12.97

Academic Guests: (see 6.1 for report of activities)

Dr. A. Carlosena	Universidad Publica de Navarra Pamplona, Spain	02.06. - 31.07.97
Prof.. A. Lindgren	The University of Rhode Island Kingston, USA	02.06. - 31.08.97
Prof. T. Hinamoto	Hiroshima University Higashi-Hiroshima, Japan	01.09. - 10.09.97
A. Lopez	Universidad Publica de Navarra, Pamplona, Spain	01.08. - 30.09.97
Prof. T. Inoue	Kumamoto University, Kurokami, Japan	11.10. - 31.12.97
Prof. H. Reddy	California State University, Long Beach, USA	05.12. - 16.12.97

2. Teaching

2.1 Lectures and Practica

Sem.	Instructors	Title	ETH-No.
5th	Prof. Massey	Zeitdiskrete Systeme & stochastische Signale	35-405
6th	Prof. Moschytz	Digitale Signalverarbeitung und Filterung	35-416
5/7th	Prof. Massey	Applied Digital Information Theory I	35-417
8th	Prof. Massey	Applied Digital Information Theory II	35-418
7th	Prof. Moschytz	Analoge Signalverarbeitung und Filterung	35-467
8th	Prof.Moschytz Prof.Eggimann	Adaptive Filter & neuronale Netzwerke	35-468
7th	Dr. Heutschi	Acoustics I	35-477
8th	Dr. Heutschi	Acoustics II	35-478
5/ 6th	Prof. Moschytz Prof. Massey et al.	Laboratory for "Fundamentals in Electrical Engineering"	35-095/6
	Prof. Moschytz Prof. Massey et al.	Colloquium on "Electronics and Communications	35-910
	Prof. Eggimann et al.	Colloquium on "Neuro-Informatics"	95-899 95-999
	Prof. Eggimann	Colloquium on "Material- und Werkstoffwissenschaften"	35-797
	Prof. Rathe	Acoustics Colloquium	35-950

2.2 Semester Projects and Diploma Theses

During the winter semester 1996/97 and summer semester 1997, 10 Semester Projects (18 candidates) and 7 Diploma Theses (11 candidates) were carried out.

Candidates	Title	Supervisor
Semester Projects WS 96/97 (7th Semester)		
Giuseppe Mazza Gérard Salzgeber	Steuerung für TACTAID Vibrator- Array	Hänggi
Zhenlan Cheng	Adaptive CNNs	Mirzai
Massimo Ferrari Roberto Grandi	Digital generierte Schallfelder	Stettbacher
Christian Franchi Fabio Solari	Effiziente Filterbank für digitales Hörgerät	Wyrsch
Michael Bengtson Magnus Hilding	Part-and-Try Algorithms for (un)blocked Access with q-ary Feedback	Loher
Semester Projects SS 97 (8th Semester)		
Rico Schwendener Oliver Lamparter	Integrierte, abstimmbare Hoch- frequenzfilter (Bandgrenze 1MHz)	Schmid
Martin Diergardt Pascal Freuler	Elektronisch abstimmbare Hoch- frequenzfilter (bis 100 Mhz)	Schmid Frey
Christoph Randazzo Balz Odermatt Knecht/ AG	Störgeräuschunterdrückung mit Mehrmikrophon-Horgerät	Joho Dr. Phonak
Marius Portmann Heutschi Marc Rennhard	Almost perfect nonlinear permutations	Dr.
Rolf Wildberger Canteaut	Verwendung von Musiksignalen zur Bestimmung raumakustischer Parameter	Dr.

Diploma Theses WS 96/97

Urs Fleisch	PC-integriertes Mulimedia-Modem	Erne

Eric Suter	Lokalisierung von bewegten Schall-quellen mit der akustischen Holographie	Steiner
Ralph Tonezzer Pascal Vontobel	Esploring the Ziv-Lempel Algorith	Sayir
David Perels Andreas Leuenberger	Estimating Permanents with Competitive Lists	Sayir
Matthias Giger Olivier Lalive d'Epinay	Kodierung für Kanäle mit Rückkopplung	Krämer
Domenico Mignone Massey/	Attacking a New Type of	Prof.
Christoph Mäder	Stream Cipher	De Moliner

Diploma Theses SS 97

Michael Gastpar	Kapazität und Kodierung für Kanäle mit Rückkopplung	Krämer

3. Research

3.1 Research Areas

The Institute for Signal and Information Processing engages in teaching and research in those aspects of communication engineering that deal with the processing of electrical signals and digital information. This includes:

Signal Processing

Analog and digital signal processing as applied to analog signals (e.g. speech or biological signals) and to digital signals (e.g. digitally transmitted data or coded speech signals). Current research topics include:

- Neural Networks and Cellular Neural Networks (CNNs) for Signal Processing (Speech, Acoustical Alarm Signals, Recognition of Handwriting)
- Switched-Capacitor (SC) and Switched Current (SI) Filters and Networks; Application to Mixed Mode Circuits for High-Speed Communication Systems
- CAS Tools for the Design and Layout of Analog, SC and SI Filters for the Realization of VLSI Technology
- Processing of Electromyograms (EMG's), EMG Modeling and Analysis using Wavelets and related Algorithms
- Acoustical Signal Detection and Recognition
- Compression Techniques for Acoustical Signals
- Adaptive Filters and Systems for Communications
- Signal Processing Algorithms (e.g. Noise Suppression, Beam Forming, Adaptive Gain Control and Filters) for Hearing Aids and Freehands Phones
- Measurement of Sound Propagation in Open Spaces
- Sound Localization in Audiology
- Adaptive Filterss for Nonstationary Environments

Information Theory

Information Theory as applied to problems in communications and data processing. Current research topics include:

- Codes over Rings and over Groups
- Spread-Spectrum Multiple-Access Techniques
- Coding for Spread-Spectrum Systems
- Design and Testing of Secret-Key Ciphers
- Complexity of Cryptographic Functions

3.2 Current Research Projects

Section 1: Signal Processing

Section Leader: Prof. Dr. G.S. Moschytz

Group 1: Analog and Digital Signal Processing

Group Leader: Prof. Dr. G.S. Moschytz

Fast Algorithms for Adaptive Beamforming

Hearing-impaired people often complain about the conventional hearing aids not only amplifying desired signals (targets, e.g. speaker in front) but also spatially distributed noise sources (jammers, e.g. engine noise, speakers from behind). It is usually difficult to separate targets from jammers with a single-microphone hearing aid because both are similar in nature.

Array signal processing provides a technique of discriminating between different sound sources (speaker in front, machine noise from aside) because of the different spatial locations. This allows to distinguish between the signals to be amplified and the ones to be attenuated (spatial filtering).

We have investigated the behaviour of the partitioned frequency-domain adaptive LMS which is known for its smaller computational complexity if the application requires long filters and a small processing delay. It could also be shown, that frequency-domain adaptive block algorithms applied to a Griffiths-Jim beamformer converge faster than time-domain block adaptive algorithms, independently of the input spectra of the involved input signals.

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Supported by: ETH

Keywords: linear adaptive filters, array processing for hearing aids

Design of Active CMOS Current-Mode Filters for the Video Frequency Range

Although most of today's signal processing is done digitally, it is the analogue part of an IC which is difficult to build. One problem of analogue filters is their precision: switched capacitor and switched current filters can be very precise, but they are sampled data filters and therefore comparatively slow. Time-continuous analogue filters can be much faster, but they are normally less precise. However, the concept of on-chip tuning makes it possible to tune a filter during its operation and therefore eliminate all errors which come from process tolerances, temperature and ageing.

After some theoretical work about current-mode single-amplifier biquadratic filters (biquads) and fast CMOS current-mode amplifiers with unprecise, but precisely reproducible gain, we showed how tunable biquads can be implemented in CMOS. The results were video-frequency filters (around 16 MHz) which, compared to gm-C filters, consume less power at the same signal-to-noise ratio and harmonic distortion.

The project now goes into its final stage, in which sample filters are integrated and tested first using external, later on-chip tuning circuitry.

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Keywords: current-mode amplifiers, current-mode filters, current conveyor, analogue integrated circuits, CMOS

Decomposition of long-term intramuscular EMG signals using Wavelets

Computer work is characterized by repetitive finger or arm movement and by constrained or immobile head or arm posture. A prevailing hypothesis postulates that specific motor units are continuously active during this repetitive, low-level contraction. The motor units may become metabolically overloaded which cause chronic (neck) muscle pain.

The goal of this project is to develop an advanced decomposition algorithm for intramuscular, long-term electromyographic (EMG) signals and to evaluate the motor units which are firing during low-level activation.

The decomposition of the EMG signal requires advanced signal processing techniques. The main problems are data compression, feature extraction, classification, and decomposition of overlapping signals. Wavelet techniques are used to compress the data and to suppress the noise of the non-stationary EMG signal.

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

Keywords: EMG analysis, data compression, classification, Wavelet analysis

Systematic Template Design for Cellular Neural Networks Using Stochastic Optimization Techniques

Cellular neural networks (CNNs) are analog, time-continuous, nonlinear dynamical systems and formally belong to the class of recurrent neural networks. Since their introduction in 1988, they have been the subject of intense research.

The operation of a simple, but important class of CNNs is defined by a set of 19 parameters, the so-called template set. One of the unsolved problems in CNN theory is how to find these template values for a specific processing task. The non-differentiable output function impedes the use of standard learning algorithms such as steepest-descent methods. Additional constraints with regard to these parameters emerge from the properties of the CNN chip, mainly from its limited precision. Hence, robust parameters are a prerequisite for hardware implementation.

Due to the difficulty of deriving CNN parameters analytically (which is related to non-differentiable functions, a high dimensional parameter set, and nonlinear constraints), Genetic Algorithms (GAs), which have proven to be a robust optimization methods in other fields, were explored for use in CNN template design.

GAs have shown to be capable not only of finding templates for given tasks, but also of optimizing them with respect to robustness; however, the optimization for robustness needs considerable computational effort, whereas large sets of different templates for the same tasks can be found comparatively fast. It has been demonstrated that the average of such a population of solutions is a perfect candidate for a robust template. In combination with a hill climbing algorithm, this approach produces even better results, while still being computationally cheap.

The algorithms have been implemented on the Intel Paragon, a massively parallel supercomputer. Both theoretical investigations and CNN template learning necessitate a fast and numerically accurate simulator. Such a tool has been realized on the supercomputer mentioned above and has proven its usefulness in different applications.

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Keywords: cellular neural networks, genetic algorithms

Design and Applications of Robust Cellular Neural Networks

Cellular neural networks (CNNs) constitute a class of two dimensional, coupled arrays of identical dynamical systems (cells). The underlying equation governing the dynamics of each cell is non-linear and the cells are assumed to operate in parallel. The connectivity among the cells is determined by a set of parameters denoted as a template set. A specific task is implemented by determining the appropriate template set.

This project is concerned with two different, but related aspects of CNNs: robustness and applications. Signal processing via CNNs only becomes efficient if the network is implemented in analog hardware. In view of the physical limitations that analog implementations entail, robust operation of a CNN chip with respect to parameter variations has to be insured. To quantify the robustness degree, we define a measure that allows to compare the (relative) robustness of different template sets performing the same task. Furthermore, given a specific application, this measure is used to optimize the boundary value of the CNN with respect to its robustness.

Design of CNN templates and robust design of them in particular are considered by applying different approaches. Exact design rules are derived to obtain robust parameters in the case of uncoupled CNNs. These rules and their generalizations are found to apply to certain types of coupled CNNs as well. An algorithmic approach is used in particular for those uncoupled CNNs that require a large number of neighbor cells to interact with. This allows a more robust implementation of such tasks. Furthermore, we introduce learning algorithms to parameter synthesis as an alternative approach requiring less a priori knowledge of the number and symmetry of parameters.

In view of possible applications in pattern recognition, we investigate the dynamic behavior of symmetric and anti-symmetric CNNs and provide a classification of the equilibria in the latter case. Discrete-time CNNs are investigated in the framework of the delta operator. This enables us to apply and compare the corresponding results obtained for the time-continuous case by taking an appropriate limit.

Another goal pursued is to investigate the capability of the CNN paradigm in speech processing applications. The speech signals first have to be transformed into two dimensional representations. A CNN-based encoding is then utilized to represent the utterances as bipolar images. Bipolar encoding reduces the computational cost of the recognition. The results obtained are comparable to the state-of-art recognition systems based on hidden Markov models.

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Keywords: nonlinear circuit theory, cellular neural networks

Design of Analog VLSI Iterative Decoders (DAVID)

This joint research project by the signal processing group of Prof. Dr. G.S. Moschytz and the information theory group of Prof. Dr. J.L. Massey aims at developing an analog VLSI design technique for iterative decoding of error-correcting codes. It is motivated by some recent developments both in analog VLSI (`neuromorphic' networks) and in coding theory (turbo coding) that suggest the possibility of building analog VLSI decoders that are much more efficient compared to traditional digital VLSI decoders in terms of operating speed and/or power consumption.

The challenge of this project was to identify suitable computational primitives (elementary circuits) on the transistor level. This first goal has now been achieved: a `canonical' mapping of the sum-product algorithm into 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 match between semiconductor physics and probability theory. A patent application is being filed.

In a next step towards a working decoder chip which demonstrates the advantages of the new approach, a test-chip will be integrated to verify the basic concepts.

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In Collaboration with: Prof. Dr. J.L. Massey, Endora Tech AG, Basel

Keywords: error-correcting codes, analog signal processing, analog VLSI

Embedded Audio Compression based on Wavelets and improved psychoacoustic models

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

In this research project, an new, Wavelet-based, scalable approach to audio compression will be investigated. Wavelet-based filterbanks can offer an almost arbitrary time-frequency-tiling and therefore can easily adapt to the spectral and temporal behaviour of the signal to be compressed. The scalability of the audio coder will offer the ability to trade bitrate versus quality on an adaptive basis and therefore graceful degradation can be achieved although transmission bandwidth may be temporarely limited. An improved psychoacoustic model will be integrated into the framework of the adaptive filterbank, the adaptive quantizer and the adaptive entropy-coding stage. The audio compression scheme will be extensively evaluated under critical listening test situations and an integration into forthcomming standards such as MPEG-4 is planned.

The problem of the analysis- and synthesis-filterbank has been adressed, resulting in a Wavelet-based filterbank, offering a flexible time-frequency-tiling and perfect reconstruction. Additional studies on the psychoacoustic- model as well as on the adaptive quantizer have been carried out.

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Keywords: MPEG, Audio Compression, Wavelets, Scalability, Psychoacoustic Models

Distortion Analysis of Switched-Current Circuits

In the switched current (SI) field, not much work has yet been done in the area of large-signal and distortion analysis of the basic switched-current building blocks. This is due to the fact that even a single MOS transistor, when operating as a current sample-and-hold circuit, leads to non-linear differential (or difference) equations, which are not easy to solve. However, practical guidelines and rules of thumb are needed prior to SPICE simulations to circumvent trial-and-error approaches. The aim of this project was to give insight into the large-signal behavior of the basic SI memory cell as well as of the current mirror.

The non-linear behavior due to the finite settling properties of the basic SI memory cell was reviewed, extended using additional circuit elements, and applied to distortion analysis. It was shown that distortion arises due to residual errors. Based on the envelope of the difference equation of an approximated settling error, a closed-form solution for the harmonics was found. The resulting Fourier series has terms consisting of constants multiplied by a sum weighted by the factorial of the index. Therefore, the terms decrease very rapidly. It was shown that the predicted results were closer to the actual solution (simulated with SPICE) than the bound previously derived. In addition, we found that for current mirrors used in high frequency applications the interconnection between drain and gate should be implemented in low resistivity material, i.e. metal, to circumvent extra settling delay.

In a second part, distortion caused by clock-feedthrough was analysed. The analysis included the signal-dependent term of the error and was extended to Vt and Beta mismatch studies. For practical applications, it was shown that harmonic terms above the fifth can be neglected in both cases. Using the proposed formulas, a better prediction of the distortion was achieved. For practical applications, the results lie within 2dB of SPICE simulations and measured data.

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Keywords: switched-current filters, analog signal processing, distortion analysis

A Mixed Analog/Digital Implementation of a Programmable Cellular Neural Network

A Cellular Neural Network (CNN) is a dynamic system based on parallel operation of simple units. In the literature CNNs often appear embedded inside a digital system, forming the so-called "CNN Universal Machine". Such a system is suitable for processing many types of 2-D or transformed 1-D data. The circuit is "neural" only in a formal sense; it is used in the manner of an analog computer. Virtually all application of the system have been based on simulations due to the lack of useable CNN analog hardware.

This project is concerned with developing techniques for implementing CNNs as integrated circuits, and with the design of CNN circuits programmable to a degree sufficient to be capable of a wide range of processing tasks. The system must incorporate elements of continuous-time analog circuits, as well as digital circuits for data input/output, programming and control. In contrast to mixed analog/digital circuits encountered in, e.g., communication circuits, the analog parts (essentially gm-C circuits) and the digital parts of the CNN are closely linked and cannot be separated. Care must be taken so that the digital circuits do not interfere with the analog circuits' operation.

After designing and testing several smaller chips, a medium-sized chip containing six CNN cells along with controlling logic was designed and manufactured in a 0.7 micron CMOS process. The chips can be connected to form an arbitrarily large network. Twenty chips were connected to form a 10x12 fully programmable network. The chips were tested statically to obtain representative data on accuracy and matching. Such data is important for CNN program ("template") design purposes (see related projects). Operation was tested on numerous CNN processing tasks of the uncoupled, coupled and propagating types. The ability to set so-called "spatially-variant template" programs was used to test the effects of different network sizes. This could be accomplished without having to physically re-configure or re-wire the circuit.

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Supported by: Swiss National Science Foundation

Keywords: nonlinear circuit theory, cellular neural networks, programmable analog VLSI

Group 2: Adaptive Systems

Group Leader: Prof. Dr. A. Kälin

Compensation of Loudness Recruitment in Hearing Aids with Adaptive Feedback Cancellation

People with loudness recruitment have a compressed dynamic range between the sound-pressure levels corresponding to threshold and discomfort. This highly frequency-dependent phenomena requires a nonlinear filter for compensation. A DFT-based approach which compensates for recruitment of loudness and cancels echos has been investigated. The echo canceler is adapted using only the available (e.g., speech) input signal. The closed-loop hearing aid system and the adaptive adjustment law has been analysed. The problems resulting from the use of a nonlinear hearing-loss compensator and stability problems caused by a rapidly changing echo path have been solved. The proposed solutions have been implemented and tested on a dummy head using a behind-the-ear hearing device. The result is a device that runs more than 20dB above the critical gain without any audible artifacts.

In further increasing the computational efficency, transformations (e.g., the modulated lapped transforms) allowing for non-uniformly spaced frequency bands will be considered. Therein, the requirements for the echo canceler and the nonlinear compensator have to be matched.

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In Collaboration with: Phonak AG

Keywords: Loudness Recruitment, Digital Hearing Aid, Feedback Cancellation, Multiband Compression

Noise Reduction for a Car Mobile Phone

In Switzerland, one is not allowed to use a cellular phone without hands-free equipment while driving a car. Today's hands-free phones work in a half-duplex way. This prevents a potential system instability which could be noticed as a whistling sound. Unfortunately, this results in not being able to send sound through both directions at the same time. In hands-free equipment, the distance between the speaker's mouth and the microphones is typically large and requires a large gain of the picked-up signal. The fact that besides the desired speech signals, noise and echo are amplified as well, calls for counter-measures to reduce these distortions. As the environmental noise in a moving car is high, noise and echo distort the speech signal and diminish the intelligibility. In order to improve the communication situation, this project will design an overall system consisting of an echo canceller and a noise reduction algorithm.

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Keywords: echo cancellation, noise reduction, car mobile phone

Adaptive Filters for Nonstationary Environments

Adaptive filters are very useful tools in the field of statistical signal processing, because they can operate in an environment of unknown statistics. The well-known Recursive Least Squares (RLS) algorithm is designed for an optimum estimation of the unknown - but constant - parameters of a Finite Impulse Response (FIR) filter, assuming stationary input and output signals. However, if these parameters are time-dependent and/or the signals possess nonstationary statistics, the RLS algorithm shows unsatisfactory tracking capabilities.

To reduce this serious drawback, the basic RLS algorithm is usually modified by incorporating a so-called forgetting factor that allows to continuously neglect older data values. This increases the capability to track unknown filter parameters, but reduces the estimation accuracy considerably. It already has been shown that such a modified RLS algorithm can be considered to be a special case of the Kalman filter. Therein, a time-independent state-space model describing the stochastic behaviour of the unknown filter parameters is inherently assumed. If this model corresponds to the real behaviour of the filter parameters, the RLS algorithm has a far better tracking behaviour than the one of the more common used model-independent Least Mean Squares (LMS) algorithm. However, for many practical problems (e.g. hands-free phone, hearing aid or public address system) this statistical model does not coincide with the real behaviour of the unknown parameters, which in turn results in an unsatisfactory tracking performance that can even be worse than the one of the model-independent LMS algorithm.

In this project we want to develop improved adaptation algorithms considering acoustical applications, such as a hands-free phone system. Here, based on signals that change their statistics at any moment (e.g., by a person who stops speaking), we have to estimate long and suddenly changing echo impulse responses. Assuming such a nonstationary environment, we aim at finding adaptive filters which can track fast and accurately hundreds or thousands of time-varying coefficients of a FIR filter. We want to improve the current, more intuitive approaches by using an explicitely formulated model of the statistical behaviour of the filter parameters. An extended Kalman filter for estimating both the filter parameters and the underlying statistical model will be explored. An important step consists of simplifying the obtained optimum filter in order to get algorithms which can be implemented in realtime (on digital signal processors) for FIR filters of large order. Therefore, our algorithms will possibly be formulated in a transform domain (e.g., in the frequency domain), in order to reduce the complexity of the computations.

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Keywords: adaptive signal processing, RLS, Kalman Filter

Noise Reduction with Hidden Markov Models

Enhancement of speech degraded by additive background noise plays an important role in communication systems such as mobile telephones and modern hearing aids. As opposed to noise reduction schemes which are based on a noise reference signal, we assume a scheme where only the noisy speech signal is available for processing (monophone noise reduction).

In this research project, a noise reduction scheme is investigated that uses two separate hidden markov models (HMM) to describe the statistical properties of speech and noise. The speech HMM models the clean speech and is trained using typical speech sequences from different speakers; the training of the noise HMM is based on typical noise sequences. Given the separate HMM's for the clean speech and the noise, a composite model for the noisy speech can be obtained. For every frame of the noisy speech signal, the composite model gives an estimate of the power spectra of the clean speech and noise signal within that frame. Using these power spectra, a Wiener filter is computed and applied to the noisy signal.

Although the HMM-based approach yields reasonably well enhanced speech, voiced speech segments often sound rough or hoarse. It was shown that this effect occurs because the noise between the harmonics of voiced segments is not removed by the Wiener Filter. An algorithm was proposed, which uses pitch period information, and which is based on least square (LS) estimation, to remove these noise components. Moreover, it was shown that the estimation involving low-energy states of the speech HMM is not reliable, and therefore a noise floor is inserted during low-energy speech segments instead of filtering the signal. To evaluate the performance of the proposed scheme and to compare it with other approaches, a paired-comparison listening test was carried out.

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Keywords: noise reduction, speech enhancement, hidden markov models

Level Gauge for Liquids using Ultrasonic Technology (TUB-US)

The start-up company mechatronica has developed within a several year work a level measuring system for liquids. Together with the firm Kübler AG it's been successfully introduced to the international market. Especially the chemical and the food industry have shown attention to the system because of their demand for high precision measurements with a millimeter resolution.

For the first time a combination of ultrasonic and float technology has been used. The level acquisition is performed with a float body which lies on the surface of the liquid and is guided by a vertical tube. Inside the tube there is a moveable cylindric reflector which is magnetically coupled through the tubewall to the float. By the measurement of the propagation time of an incidented ultrasonic soundwave at the top end of the tube, the distance between the reflector and the transducer will be estimated and therefore the filling level can be obtained. At very high temperatures (120 degrees C max.) the estimation of the distance becomes very inaccurate because of the temperature gradient along the tube and the large sound attenuation.

Within the scope of this project, the mentioned problems shall be investigated and suggestions for solutions should be offered. The Institute for Electrical Measurement Techniques at the Technical University of Linz performed a finite-element simulation which has shown that errors caused by a temperature gradient could be sufficiently reduced by a simple model, based on three temperature sensors along the tube. At the ISI (Institute for Signal and Information Processing, ETH Zurich) a measurement system shall be developed for the verification of this model and further on to deliver measuring data which are required for the development of an appropriate estimation algorithm. Even with noisy or distorted echos, this algorithm should allow to make a precise and reliable estimation of the propagation time.

This interdisciplinary project with its complex problems requires a close co-operation between the TU Linz (experts for sound field modeling), the ISI (experts for signal processing and measurement technique) and the firm mechatronica as the industrial partner which provides the mechanical system to perform measurements.

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In Collaboration with: mechatronica, TU Linz

Keywords: level gauge, ultrasonic, distance estimation

Active Noise Control

Conventional methods of suppressing acoustic noise using passive sound absorbers generally do not work well at low frequencies (below about 500 Hz). An approach to overcome this problem is to cancel the disturbing sound field with the help of a second interfering field, typically generated by coil loudspeakers. The generation and control of its input signals is the task which is usually associated with active noise control. Mostly, the acoustic system which describes the generation and transmission of the disturbing sound field is varying. This in turn asks for an adaptive controler. In general, the design of such a controler is still an open problem. The goal of this project is to find simple and efficient controlers for typical configurations by systematically using appropriate a-priori knowledge about the acoustic system and its excitation.

Two adaptive controlers have been developed; namely a computationally efficient DFT-based controler for periodic sound fields and a new controler for stochastic sound fields. Based on the available estimation of the system the latter uses a fixed control filter (high-order FIR filter) which is on-line adapted to the true system and to the true input process by means of a low-order FIR filter. In this research period the fixed control filter design has been further improved and the whole controler has been implemented in a test tube in cooperation with the Ingenieurschule Burgdorf.

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In Collaboration with: Sulzer Innotec, Ingenieurschule Burgdorf

Keywords: active noise control, frequency-domain adaptive filters, LMS algorithm

Group 3: Applied Acoustics

Group Leader: Prof. Dr. E. J. Rathe

Audiometry to Evaluate Binaural Hearing

Most of the hearing functions are based on the binaural perception. But today's audiometry possesses only rough tools to measure binaural abilities; and so far no scale or metric to judge a subject's useful hearing has been established.

This research project aims at a greater understanding of this complex field. Equipment as well as a measurement procedure and rules are called for.

During a first step a comprehensive measuring device was developed and built. It consists of an array of loudspeakers and a multi DSP computer. Each loudspeaker is driven by one processor. The system can be accessed and used from a workstation or pc.

The device is now being used to generate three-dimensional, dynamic acoustical situations like moving sound sources for example. It isparticularly interesting to compare different algorithms and signals with the (psychoacoustic) impression they create. By using results from these experiments procedures for the clinical testing of the binaural hearing are now derived.

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Keywords: binaural hearing, sound localization, audiology, audiometry, DSP

Group 4: Information Technology

Group Leader: Prof. Dr. F. Eggimann

Slant in Handwriting Recognition

Off-line Handwriting recognition has a wide range of applications but has to deal with problems such as the slant of a word. Since, slant can vary greatly between writers, it is important to have a robust method to deal with this problem. There are two methods of dealing with slant: (1) remove the slant in the preprocessingthrough a deslanting algorithm (2) train the recognition algorithm to be slant-invariant. Both methods were tested using our neural network based recognition system. We trained three different types of networks: (1) a normal network (2) a deslanted network trained with the deslanting algorithm implemented in the preprocessing and (3) a slant-invariant network. Each of the networks reactions to different slants and their recognition performance as part of the handwriting system were tested; first results were obtained.

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Keywords: off-line handwriting recognition, neural networks, pre-processing, word slant correction, slant tolerant networks

Section 2: Digital Information Theory

Section Leader: Prof. Dr. J.L. Massey

Statistical Tests for Block Ciphers

The aim of this project is to develop testing methods for deciding, when given a black box containing a block cipher, whether that cipher is secure or not. The goal is a theory of statistical testing that would be applicable to any block cipher. The testing should in principle detect any cryptographically significant weakness with high probability. In the past year, the main issues to be confronted by such a theory have been identified. Several statistical tests have been designed and/or considered and then applied to the ciphers DES, IDEA, SAFER, RC2 and RC5

(IDEA and SAFER were developed in our laboratory). Statistical dependencies between plaintext and ciphertext have been found for DES reduced to six rounds, for IDEA reduced to one round, for SAFER reduced to two rounds, for RC2 reduced to six rounds and for RC5 reduced to five rounds.

For computationally intensive testing, software has been written to support distributed computing of such testing by utilizing idle machines in a network.

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Keywords: cryptography, cryptanalysis, statistical tests

Information Theory and Strategies for Channels with Feedack

The role played by feedback for many-user channels is not very well understood. In a first part of this project, new information theoretic quantities are defined to obtain expressions for the capacity regions of the two-way channel and the multiple-access channel with feedback. These expressions yield easily computable inner bounds to the capacity region but have so far failed to show how to easily calculate outer bounds.

The second part of the project is to design feedback strategies for a class of multiple-access channels for which the capacity region is known. The information theoretic characterization of the capacity region is used to aid the design and results in several simple strategies which can achieve any rate point inside the capacity region of these channels.

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Keywords: feedback, capacity, strategies

Adaptive Coded Automatic-Repeat-Request (ARQ) Strategy: Telescoping Codes

Major research interest has recently been focused on the development of a high-speed wireless access system to an ATM network. The Medium Access Control (MAC) protocol is based on a contention-and-reservation scheme and supports traffic allocation according to the traffic contract of each ATM connection, mobility features as well as an error control functionality. The combined design of the access protocol (scheduler) and the error control scheme is aimed to provide multi-media services with guaranteed Quality of Service (QoS) to the mobile users. The scheduling algorithm specifies the time frame structure and, thus, the (variable) length of the packet stream consisting of fixed-length packetsof a specific service. An adaptively coded Automatic-Repeat-Request (ARQ) strategy is applied on this packet stream: the ARQ protocol operates on the individual packets whereas the code is applied on the entire stream. The code over the packet stream, the telescoping code, is an adaptively selected unequal-error-protection code which provides the robustness and flexibility to cope with the system limitations (finite buffers, etc.), to satisfy the service requirements (delay control, channel quality) and which is to be matched to the in-sequence constraint on the released packets. Its design is aimed to achieve a high bandwidth efficiency and, thus, to maximize the throughput under specific constraints imposed by the system (limited buffers and feedback capacity) and the required low complexity of the high-speed processing (decoding). Using this scheme, for instance, for the case of a loss-sensitive service and a limited (or, even, zero) capacity of the receiver buffer, we can achieve significant improvements on the performance compared to the uncoded case even when we apply a (very) low-complexity encoding and erasure decoding algorithm on packet level.

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Keywords: error control, ARQ, packet communication, ATM

The Linear Cryptanalysis of Iterated Block Ciphers

All present practical block ciphers are iterated ciphers, in which a relatively simple round function is applied many times in succession to the plaintext input block to produce the ciphertext output block. The round key used within each round is determined from the user-selected key by application of a key-schedule algorithm. Until recently, iterated ciphers had to be analysed separately and individually to determine their security against attacks. The recent development of powerful and general attacks, which can be used against any iterated cipher, has made possibe a more objective determination of the security of an iterated cipher. One of these attacks is Linear Cryptanalysis.

Until now, the resistance of iterated block ciphers against linear cryptanalysis has been studied on single ciphers and, more rarely, on families of ciphers. In this project, all iterated ciphers of a given block size are considered and their resistance against linear cryptanalysis studied, beginning with small ciphers and a small number of rounds. It has been possible to show that, for ciphers with only two rounds of encryption, the proportion of ciphers that have a given level of resistance against this attack increases as the block size increases and that, if the latter is large enough, almost all ciphers resist the attack.

Presently two goals are pursued: the study of the resistance of block ciphers with more than two rounds of encryption against linear cryptanalysis and the resistance of block ciphers of a given size to the group generalization of linear cryptanalysis (GGLC). Comparing the results for linear cryptanalysis and for its group generalization should give some information about the increased efficiency of GGLC with respect to linear cryptanalysis itself.

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Keywords: Cryptography, Iterated Block Ciphers, Linear Cryptanalysis

The Role of Information in Random-Accessing

It is rather surprising that, after almost fifty years since its birth as a science, information theory has only rarely been applied to communication systems that incorporate feedback. What is known about feedback in information theory is more qualitative than quantitative. For instance it is knwon that the existence of feedback from the base station in a mobile communications networks to the users increases the maximum stable throughput of random access systems and simplifies the communication protocols necessary for operation near this maximum. However, this maximum is not known to date for any kind of non-trivial feedback.

In this project we tried to attack the problem of determing the maximum achievable throughput depending on the feedback whereas we focused ourselves on binary and ternary feedback. This incorporated to identify precisely what information must be received by the base station from the users attempting to access it before the users can be granted access. In a cellular packet-radio system (and in many other forms of random accessing), this question is complicated by the fact that the packets that are used for gaining access to the base station also contain the information that is transmitted when the accessing is successful. It will be necessary to separate somehow the accessing information from the transmitted information. In this project we identified successfully the information that must be exchanged within a set of infinitely many, uncoordinated users in order to get access to a system. Moreover, information-theoretic bounds were applied to yield an upper bound on the maximal stable throughput of a random access broadcast channel. Although the best upper bound of 0.5683 for ternary feedback is tighter than our upper bound of 0.6484, the presented concept still offers possibilities to tighten the bound and shows aspects to find the capacity of the collision channel with feedback.

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Keywords: random accessing, collision resolution, multiple access protocols

Source Coding with Competitive Lists

Most of the considerable progress achieved over the past years by new universal source coding algorithms trades better compression ratio for a greatly increased storage requirement. In an aim to reverse this trend, a coding method was developed which achieves a good compression ration using very little storage.

The method relies on the use of so-called competitive lists to estimate the probabilistic model of the source. The competitive list has the property that it orders the alphabet of an unknown source by order of decreasing probabilities. A universal coding distribution is applied using arithmetic coding on the output of each competitive list.

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Keywords: nonlinear signal processing, adaptive models, statistical inference

3.3 Completed Research Projects

HELFENSTEIN Markus

Analysis and Design of switched-Curren Networks

ETH-Diss. Nr. 12257 (Referee: Prof. Dr. G.S. Moschytz)

Switched-current (SI) techniques represent a new method of designing analog sampled-data circuits in the current domain. Circuits using SI-techniques can be implemented without the need for floating capacitors. This makes them ideally suited for integration using a standard VLSI (CMOS) process with mixed analog/digital ICs. In this thesis new design and analysis methods for SI circuits are presented.

In the first part of this thesis, a by-inspection analysis and synthesis method for multiphase switched-current circuits using signal-flow graph (SFG) techniques is presented. The SFG is derived on the transistor level and the method is primarily useful for the hand analysis and design of small and medium-size SI circuits (e.g., SI filters, decimators, interpolators). Tables of commonly used SI circuits, in which the corresponding SFGs and circuits are given, make the derivation easy and fast. With the proposed method, it is straightforward to include non-ideal effects such as finite output resistance of MOS transistors, clock-feedthrough and settling error. The method is also a useful tool for the synthesis of new SI circuits. It will be shown that every low-sensitivity switched-capacitor (SC) circuit can be mapped directly into a low-sensitivity SI circuit with a corresponding topology. Examples of transformed SC circuits are given and two new double sampling integrators are introduced.

The second part of the thesis deals primary with circuits on the transistor level. The non-idealities of switched-current circuits such as the mismatch of MOS transistors, the finite output resistance of the basic sample-and-hold circuit, clock-feedthrogh (CLF), distortion due to the finite settling time of the memory cell and clock-feedthrough, are addressed. Circuit techniques, such as two new clock-feedthrough compensation schemes and gain-enhancement techniques, are suggested to improve the behavior of the elementary cells, and a basic understanding of the theoretical background is given. Whenever possible, the results are compared with simulations and/or measured data.

The third part of the thesis deals with circuits designed for broadband signals, such as video. Video signal processing is a key driver for today's research in analog IC design. Although the main part of present-day signal processing chips is digital, it is in the analogue section in which the most significant performance improvements can be made, primarily in the interface part. For the interfacing between the analogue and the digital circuit parts, video systems usually comprise components of an anti-aliasing filter followed by an analogue-digital converter. To relax the specifications of these devices, the design of switched-current decimators for wide bandwidth video filtering applications is presented. Two new polyphase SI decimator structures employing active-delay blocks to implement the delay lines are presented both analytically and graphically. A linear phase FIR filter structure with reduced complexity of the input commutator, and a FIR filter for relaxed timing requirements is described. The latter was implemented and designed for an amplitude response tailored for video interface applications.

Finally, in the application part, based on the design and measurements of two IC chips, a comparison between switched-capacitor and switched-current bandpass filters is presented. The filters are sampled with clock frequencies up to 10MHz and the nominal center frequency fp is 500kHz with a pole-Q of 10. The focus is on implementation issues such as low voltage supply operation and PSRR. It is demonstrated that in the SI case the design can be made independently of the modulation index, which results in superior circuit behavior for large input signals.

PONCET Andreas

Analysis and Design of Switched-Current Networks

ETH-Diss. Nr.12211 (Referee: Prof. Dr. G.S. Moschytz

The problem of designing an adaptive model from data is addressed in a unified probabilistic framework. The applications include system identification, signal prediction, and pattern classification. Usually the performance of a model is measured by the precision with which it can make predictions for new patterns based on training patterns. The precision is quantified by the generalization error (risk). The risk can be, e.g., the mean squared error, or the mean relative entropy, or the misclassification rate, depending on the loss criterion used.

A mathematical analysis of the ``generalization phenomenon'' in adaptive models (predictors) is undertaken. Since a model is trained with random data, the risk is a random variable. An approximation of the probability distribution of the risk is derived explicitly. This general result clarifies why (and by how much) the performance of an adaptive model tends to degrade as soon as more parameters ``than needed'' are used. The influence of regularization terms is also explained.

The main benefit of the analysis, however, is to suggest systematic solutions to the three main pragmatic issues of model design. First, a set of appropriate regressors has to be selected. For this purpose, a method based on kernel density estimation is developed to infer the Bayes (optimum) risk from data. The second issue involves the choice of basis functions. A method to select (from a set of candidate basis functions) the smallest subset needed to reach a given performance is presented. The third issue consists in evaluating the generalization error of the designed model. It is shown how Bayesian inference can solve this problem: by combining the probability distribution for the risk with the training data, one obtains the conditional distribution of the generalization error. The design methodology is illustrated with real-world data.

STEINER Rolf

Sound Field Reconstruction of Moving Noise Sources by Means of Acoustical Holography

ETH-Diss. Nr. 12451

Noise pollution of the environment is increasing daily due to growing traffic. In contrast to stationary sound emission reduction measures (noise barriers), this pollution can be reduced efficiently and in an inexpensive way by measures at the noise producing sources (e.g.~train, car). However, that requires the knowledge of the exact position, radiation behavior and power of the acoustical sources on these vehicles, which has to be obtained by reconstructing the sound field in the form of the intensity field. Since a moving measurement system produces turbulence noise and requires a unrealistically high number of microphones, a static measurement system, separated from the moving object, is preferred. The microphone signals are therefore recorded during the passing by of the object emitting sound, and the intensity field is reconstructed by means of subsequent signal processing.

The work presented here deals with this reconstructing task. Acoustical holography is applied to localize the sound sources, rather than the commonly used technique of beamforming. By correctly mathematically modeling the moving object, the applicability of acoustical holography, was extended from stationary to non-stationary situations. Furthermore, acoustical holography allows the reconstruction of the three dimensional sound field, which is not possible with existing beamforming methods.

In order to describe the physical process in mathematical terms, the process is mapped onto a mathematical model. Therefore, the process is treated by means of the system concept, which shows that the system can be split into three subsystems. These are, in turn, a velocity transform, the acoustical propagation, and the sampling of the sound field by the microphones. The velocity transform maps the moving signal of the moving sound source onto a stationary coordinate system. This is the key for continuing the subsequent analysis, since it permits the signals to be processed in a stationary system. The acoustical propagation describes the mapping of the sound field in a specific plane onto the sound field in another plane. This mapping is based on acoustical holography and constitutes the kernel of the sound field reconstruction.

Since the process to be modeled, is relatively complex, in a first step a continuous indexed model is developed. Thereby, the model can be described in an elegant and simple way by means of functional analysis. When the microphone are known, a discrete model can be derived from the continuous model in a second step. This allows processing by computer. Only the comprehensive and structured mathematical model admits a thorough analysis for the purpose of optimized signal processing as well as optimized microphone positioning. Furthermore, by means of the modeling discussed, other problems can be treated, e.g. sound field measurements in a wind channel or in a stationary situation.

The sampling of the sound field with a finite number of microphones leads to a finite measurable area of the sound field. Furthermore, the acoustical propagation introduces a spatial bandwidth limitation, which results in a resolution limitation of the sound sources to be reconstructed. Additionally measurement noise contaminates the measurement. The estimate of the sound field is found by a novel estimation algorithm which takes all aspects into account an in a statistically optimal way. The special merit of the estimation is that it does not need intuitive analysis steps, e.g. windowing of the samples, but uses physically meaningful parameters (e.g. SNR) to obtain the optimal estimate. By means of simulations it is showed that, in contrast to existing methods, the optimal estimation enables a strongly improved sound field reconstruction, especially at low frequencies. The mentioned improvement manifests itself in an improved reconstruction accuracy and in an improvement of the resolving power of a point source by a factor of up to two. By means of practical measurements of a series Ae 4/7 locomotive the applicability is demonstrated in practice.

MIRZAI Bahram

Robustness and Applications of Cellular Neural Networks

EHT-Diss. Nr. 12483 (Referee: Prof. Dr. G.S. Moschytz

Cellular neural networks (CNNs) constitute a class of two dimensional, coupled arrays of identical dynamical systems (cells).The underlying equation governing the dynamics of each cell is non-linear and the cells are assumed to operate in parallel. The connectivity among the cells is determined by a set of parameters denoted as a template set. A specific task is implemented by determining the appropriate template set.

This thesis is concerned with two different, but related aspects of CNNs: robustness and applications. Signal processing via CNNs only becomes efficient if the network is implemented in analog hardware. In view of the physical limitations that analog implementations entail, robust operation of a CNN chip with respect to parameter variations has to be insured. To quantify the robustness degree, we define a measure that allows to compare the (relative) robustness of

different template sets performing the same task. Furthermore, given a specific application, this measure is used to optimize the boundary value of the CNN with respect to its robustness. Design of CNN templates and robust design of them in particular are considered by applying different approaches. Exact design rules are derived to obtain robust parameters in the case of uncoupled CNNs. These rules and their generalizations are found to apply to certain types of coupled CNNs as well. An algorithmic approach is used in particular for those uncoupled CNNs that require a large number of neighbor cells to interact with. This allows a more robust implementation of such tasks. Finally, we introduce learning algorithms as an alternative approach requiring less a priori knowledge of the number and symmetry of parameters.

In view of possible applications in pattern recognition, we investigate the dynamic behavior of symmetric and anti-symmetric CNNs and provide a classification of the equilibria in the latter case. Discrete-time CNNs are investigated in the framework of the delta operator. This enables us to apply and compare the corresponding results obtained for the time-continuous case by taking an appropriate limit.

Another goal pursued is to investigate the capability of the CNN paradigm in speech processing applications. The speech signals first have to be transformed into two dimensional representations. A CNN-based encoding is then utilized to represent the utterances as bipolar images. Bipolar encoding reduces the computational cost of the recognition. The results obtained are comparable to the state-of-art recognition systems based on hidden Markov models.

3.4 Completed Dissertations

PONCET Andreas: Design of Adaptive Models for Identification, Signal Prediction, and Pattern Classification

ETH Diss. Nr. 12211
Referee: Prof. Dr. G.S. Moschytz
Co-referee: Prof. Dr. J.L. Massey
Prof. Dr. M. Hasler

HELFENSTEIN Markus: Analysis and Design of Switched-Current Networks

ETH Diss. Nr. 12257
Referee: Prof. Dr. G.S. Moschytz
Co-referee: Prof. Dr. Q. Huang

MIRZAI Bahram Robustness and Applications of Cellular Neural Networks

ETH Diss. Nr. 12483
Referee: Prof. Dr. G.S. Moschytz
Co-referee: Prof. Prof. Leon Chua

STEINER Rolf Sound Field Reconstruction of Moving Noise Sources by Meansof Acoustical Holography

ETH Diss. Nr. 12451
Referee: Prof. Dr. A. Kälin
Co-referee: Dr. J. Hald, Brüel & Kjaer, Dänemark
Dr. A. Stirnemann, Sulzer Innotec, W'thur

3.5. Internal Reports

9701	Reddy Madhu	Handwriting Recognition of Off-line Scanned Data.
9702	Kukorelly Zsolt	On Two Hypotheses in Cryptanalysis.
9703	Lim Drahoslav	Robust CNN Algorithms for High-Connectivity Tasks.

4. Congresses, Meetings and Committees

4.1 Congress Organization

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).

Member of ESTA (European Scientific and Technical Assembly, Brussels).

Member of Board of Governors, IEEE Circuits and Systems Society.

Organization of 1st ETHZ-EPFL Summer school on Linear, Nonlinear, and Adaptive Circuits, Systems and Signal Processing together with Prof. M. Hasler, CIRC EPFL and Prof. L.O.Chua, UC Berkeley.

Member of Board of Governors, IEEE Circuits and Systems Society.

Prof. Massey

Program Committee, Fast Software Encryption, 1997 4th Int. Workshop, Haifa, Israel.

Program Committee, 1997 IEEE Int. Symp. on Information Theory, Ulm, Germany.

Program Committee, CRYPTO'97, Santa Barbara, USA.

Advisory Board, SSC 97, Lausanne, Switzerland.

Technical Program Chairman, 1998 International Zurich Seminar on Broadband Communications, Zurich, Switzerland.

Technical Program Committee, 1998 IEEE Int. Telecommunications Symposium, Sao Paulo, Brazil.

Advisory Board, MMT'97, Melbourne, Australia.

Organizing Committee, 1998 Workshop on Cryptographic Protocols, Monte Verità, Switzerland.

4.2 Participation in Congresses and Meetings

Group: Analog and Digital Signal Processing

Moschytz George S.	Research Stays with Globespan Technologies, New Jersey, USA, 9.2.-4.3.97 and 13.7.-26.8.97.
Moschytz George S.	IEEE Board of Governors & ICCAD'97 Meeting, San Jose, USA, 5.-12.11.97.
Moschytz George S. Lustenberger Felix Hänggi Martin Helfenstein Markus	ISCAS'97, Hongkong, 9.-12.6.97.
Joho Marcel Erne Markus	International Conference on Acoustics, Speech, and Signal Processing, München, 21.-24.4.97.
Joho Marcel	IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, USA, 19.-22.10.97.
Erne Markus	103rd Convention of the Audio Engineering Society in New York, USA, 26.-19.9.97.
Erne Markus	MPEG-4 Standardisierungsmeeting in Freiburg, 27.-31.10.97.
Mirzai Bahram Hänggi Martin	ETHZ-EPFL Summer School on Linear and Nonlinear Adaptive Signal Processing, Lausanne, 7.-11.7.97.
Moschytz George Mirzai Bahram Hänggi Martin Schmid Hanspeter Lim Drahoslav	ECCTD'97, Budapest, Hungary, 30.8.-3.9.97.
Helfenstein Markus Wireless	Transmitter and Receiver Circuit Design for Communiclations, Workshop, Barcelona, Nov. 97.
Lim Drahoslav	First Electronic Circuit and Systems Conference, Bratislava, Slovakia, 4.-5.9.97.

Group: Adaptive Systems

Kälin August	ICASSP'97, München, 20.-24.4.97.
Kälin August	Intern. Workshop on Acoustic Echo and Noise Control, London, 11.-12.9.97.
Kälin August	Visit Brüel & Kjaer, Copenhagen, 28.10.97.

Oberle Stefan Steiner Rolf	ISCAS'97, Hongkong, 9.-12.6.97.
Wyrsch Sigi Processing to Audio	IEEE Workshop on Applications of Signal and Acoustics, New Paltz, USA, 19.-22.10.97.
Steiner Rolf	Inter-Noise'97, Budapest, Hungary, 25.-27.8.97.

Group: Applied Acoustics

Heutschi Kurt Physics,	ALFA Colloquium, Fraunhofer Institute for Building Stuttgart, 17.-18.7.97.
Heutschi Kurt	Swiss Acoustical Society Annual Meeting, Bern, 30.-31.10.97.
Stettbacher Jürg	AES 102nd Convention, Munich, 22.-25.3.97.
Stettbacher Jürg	DSP Germany'97, Munich, 30.9.-1.10.97.

Group: Digital Information Theory

Massey James L.	LATSIS-Preis, Berne, Switzerland, 31.1.97.
Massey James L.	Seminar, Univ. of Munich, Germany, 14.2.97.
Massey James L.	ETT Editors' Meeting, Liège, Belgium, 10.3.97.
Massey James L. .	4th ACM Conference on Computer and Comm. Security, Zurich, Switzerland, 1.-4.4.97.
Massey James L. Grant Alex Keusch Beat Krämer Gerhard Sayir Jossy	1997 IEEE Int.Symp. on Information Theory, Ulm, Germany, 29.6.-4.7.97.
Massey James L.	Spring Int. School, Politechnica Bucarest, Romania, 29.- 30.4.97.
Massey James L.	University of Budapest,Hungary, 1.-8.5.97.
Massey James L.	1997 IEEE Information Theory Workshop, Longyearbyen, Norway, 6.-12.7.97.
Massey James L.	4th Int.Symp. on Communication Theory and Applications, Ambleside, UK, 13-.7.97.
Massey James L. Grant Alex Keusch Beat Krämer Gerhard Kukorelly Zsolt Sayir Jossy Agotai Renate	Int. Seminar on Coding Theory in honor of R. Varshamov, Thahkadzor, Armenia, 2.-6.10.97.

Massey James L.	XV Brazilian Telecommuncations Congress, Recife, Brazil, 8.-11.9.97.
Massey James L.	ITG Meeting, Univ. of Erlangen, Germany, 31.10.97.
Massey James L.	5th Telecommunications Forum TELFOR'97, Belgrade, Yugoslavia, 22.-26.11.97.
Keusch Beat Krämer Gerhard Loher Urs Sayir Jossy	Seminar on Coding Theory,IPPI, Moscow, Russia, 18.-25.9.97.
Krämer Gerhard	University of Essen, Germany, 25.11.97.
Kukorelly Zsolt	ats Seminar on Cryptography, Engelberg, Switzerland, 1.-4.9.97.
Loher Urs	SMG2 UMTS, Lulea, Sweden, 7.-10.4.97.
Loher Urs	SMG2, Düsseldorf, Germany, 5.-6-6-97.
Loher Urs	SMG2, London, UK, 7.6.97.
Loher Urs	SMG2 UMTS, Rennes, France, 4.-8.8.97.
Loher Urs	ITU, NMT-2000 Workshop, Toronto, Canada, 9.-12.9.97.
Loher Urs	TAE Techn. Lehrgang, Sarnen, Switzerland, 16.9.97.
Loher Urs	SMG2 UMTS, Helsinki, Finland,17.-21.11.97.
Loher Urs	SMG2, Cork, Ireland, 1.-5.12.97.
Sayir Jossy	Compression and Complexity SEQUENCES'97, Positano, Italy, 11.-13.6.97.
Grant Alex	WAND Meeting Mägenwil, Switzerland, 27.2.97.
Grant Alex	47th IEEE Vehicular Technology Conference, Phoenix, USA, 5.-7.5.97.
Grant Alex	WAND Meeting, Rhodes, Greece, 20.-24.5.97.
Grant Alex	WAND Meeting Amsterdam, The Netherlands, 13.6.97.
Grant Alex	WAND Meeting Tampere, Finland, 7.-10.9.97.
Grant Alex	WAND Meeting Utrecht, The Netherlands, 15.-16.12.97.

4.3 Service Activities and Society Memberships

Prof. Moschytz

Member of the Swiss Section of the IEEE

Member of the Planning Committee of the EE Dpt., 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

Präsident des AGEN-Rates (Arbeitsgemeinschaft für elektr. Nachrichtentechnik) der Stiftung Hasler-Werke, Berne

Fellow of the IEEE, New York

Member, Swiss Electrical Engineering Society

Member, Swiss Academy of Engineering Sciences

Member of ESTA (European Science and Technology Assembly)

Member of the Board of Governors; IEEE Circuits and Systems Society

External Ph.D. Examiner, Swiss Federal Institute of Technology, Lausanne

Prof. Massey

Co-Editor, Book Series: Communications and Control Engineering, Springer-Verlag

Member, Advisory Board, Lecture Notes in Control and Information Sciences, Springer-Verlag

Member, Editorial Board, European Transactions on Telecommunication

Member, Editorial Board, Journal of Information and Optimization Sciences

Member, Editorial Board, Journal of Cryptology

Member, Editorial Board, AAECC Journal of Applicable Algebra in Engineering, Communication and Computing

Guest Co-Editor, 50th Anniversary Issue of the IEEE IT-Society Newsletter

Fellow of the IEEE New York

Member, Swiss Academy of Engineering Sciences

Member, Swiss Electrical Engineering Society

Member, IEEE Education Medal Committee

Member, Board of Governors, IEEE Information Theory Society

Member, U.S. National Academy of Engineering

Member, European Academy of Arts and Sciences

Member, International Asociation for Cryptologic Research

Honorary Member of the Hungarian Academy of Sciences

Member, Selection Committee of the Marconi Award

Member, Scientific Advisory Board, THESEUS Institute for Advanced Studies in Communications Strategy, Sophia Antipolis, France

Member, ComSoc Awards Board for 1996-98

Member, Board of Electors, Chair in Communications, Univ. of Cambridge, UK

Member, Election Committee, Professorships in Communications, EPFL

Member, Election Committee, Professorship in Computer-Vision, ETHZ

Dr. Heutschi

Member , Acoustical Society of America

Member, Audio Engineering Society

Member, Swiss Acoustical Society (SGA)

4.4 Presentations by Institute Members

Groups: Analog and Digital Signal Processing and Information Technology

Lim Drahoslav	"An Overview of CNN Implementation in VLSI" at the Second ETHZ-EPFL Summer School on Linear and Nonlinear and Adaptive Circuits, Systems and Signal Processing, 9.7.97.
Joho Marcel	"Adaptive Beamforming with Partitioned Frequency-Domain Filters", IEEE Worskshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, USA, 19.-22.10.97.
Mirzai Bahram	On the Dynamics and Stable Equilibria of Anti-Symmetiric CNNs", ISCAS'97, Hongkong, 12.6.97.
Mirzai Bahram	"Robustness of Delta Operator Based Cellular Neural Networks", ISCAS'97, Hongkong, 12.6.97.
Mirzai Bahram	On Applications of CNNs: Isolated Word Recognition", ETHZ-EPFL Summer School on Linear and Nonlinear Adaptive Signal Processing, Lausanne, 9.7.97.
Mirzai Bahram	"Isolated Word Recognition Utilizing a CNN Encoder", ECCTD'97, Budapest, 2.9.97.
Martin Hänggi	"CNN Template Design and Learning", ETHZ-EPFL Summer School on Linear and Nonlinear Adaptive Signal Processing, Lausanne, 9.7.97.
Schmid Hanspeter	"Tunable CCII-MOSFET-C Filter Biquads for Video Frequencies", ECCTD'97, Budapest, Hungary, 1.9.97.

Group: Adaptive Systems

Kälin August	"On a Digital Hearing Aid with Recruitment of Loudnmess Compensation on Acoustic Echo Cancellation", Intern. Workshop on Acoustic Echo and Noise Control, London, 11.-12.9.97
Oberle Stefan	"HMM-Based Speech Enhancement Using Pitch Period Information", ISCAS'97, Hongkong, 6.12.97.
Wyrsh Sigi	"A DSP Implementation of a Digital Hearing Aid with Recruitment of Loudness Compensation and Acoustic Echo Cancellation", IEEEWorkshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, USA, 19.-22.10.97.

Group: Applied Acoustics

Jürg Stettbacher	"DSP-Technik im Dienste von Surround Sound, Psycho-akustik und audiologie, DSP Germany'97 Munich, 30.9.-1.10.97.

Group: Digital Informations Theory

Massey James L.	"Simple Principles for Maximum Likelihood Estimation and for Sampling, with Applications", Colloquium lecture, Tech. Univ. of Munich, Germany,14.2.97.
Massey James L.	"The Discrete Fourier Transform and its Generalization with Applications to Coding", Spring Int.School on Multi-dimensional Signal Processing and Analysis, Methods, Algorithms,Technologies, Applications, Bucarest, Romania,21.-22.4.97.
Massey James L.	"The Invertibility Principle for Maximum- Likelihood Estimation with Application to Multiple Accessing", Colloquium lecture, Budapest, Hungary, 6.5.97.
Massey James L.	"On the Entropy Bound for Optimum Homophonic Substituion", ISIT'97, Ulm, Germany, 29.6.-4.7.97.
Massey James L.	"Codes and Ciphers -- What's the Difference ?" IEEE Info.Theory Workshop, Longyearbyen, Norway, 6.-12.7.97.
Massey James L.	"The Invertibility Principle for Maximum-Likelihood Estimation", 4th Int. Symp. Comm. Theory and Appl., Charlotte Mason College, Lake District, UK, 13.-18.7.97.
Massey James L.	"The Discrete Fourier Transform in Coding and Cryptography", Plenary lecture, XV Brazilian Telecomm. Congress,Recife, Brazil, 8.-9.97.
Massey James L.	"The Discrete Fourier Transform over Commutative Rings", Coll. lecture, Kath. Univ. Leuven, Belgium, 9.10..97.
Massey James L.	"The Discrete Fourier Transform over Commutative Rings", Seminar lecture, Lund Univ. Sweden, 17.10. 97.
Massey James L.	"On the Limits of Mobile Radio Communications", ITG Nürnberg, Germany, 31.10.97.
Massey James L.	"Maximum-Likelihood Estimation, Invertibility and Multiple Accessing", Keynote lecture,TELFOR'97, Belgrade, Yugoslavia, 25.-27.11.97.
Keusch Beat	"Telescoping Codes for a Packet Frame Automatic-Repeat-Request (ARQ)", IPPI, Moscow, Russia, 9.9.97.

Keusch Beat	"Adaptive Coded Automatic-Repeat-Request (ARQ) Strategy Applying Telescoping Codes", Int. Seminar on Coding Theory, Thahkadzor, Armenia, 3.10.97.
Krämer Gerhard	"A Sequential Strategy for the Two-User Noiseless Binary Adder Channel with Feedback", ISIT'97, Ulm, Germany, 29.6.-4.7.97.
Krämer Gerhard	"Directed Information for the Multiple-Access Channel with Feedback and the Common-Output Two-Way Channe", Int. Seminar on Coding Theory, Thahkadzor, Armenia, 2.-6.10.97.
Krämer Gerhard	"Feedback Strategies for a Class of Two-User Multiple-Access Channels", IPPI, Moscow, Russia, 18.-25.9.97.
Krämer Gerhard	"Feedback Strategies for a Class of Two-User Multiple-Access Channels", Univ. of Essen, Germany, 25.11.97.
Kukorelly Zsolt	"The Hypothesis of Fixed-Key Equivalence", Int. Seminar on Coding Theory, Thahkadzor, Armenia, 4.10.97.
Loher Urs	"The Role of Information in Random Accessing", ISIT'97, Ulm, Germany, 29.6.-4.7.97.
Loher Urs	"Universal Mobile Telecommunications System: The Wave of the Future?", TAE, Sarnen, 16.12.97.
Sayir Jossy	"Ordering the Probabilities of an Unknown Discrete Memoryless Source", ISIT'97, Ulm, Germany, 29.6.-4.7.97.
Sayir Jossy	"Source Transformation and Competitive Lists", IPPI, Moscow, Russia, 26.9.97.
Sayir Jossy	"Optimal Arithmetic Coding for Monotone Sources", Int. Seminar on Coding Theory, Thahkadzor, Armenia, 2.-6.10.97.
Grant Alex	"Fast Decoding of Peak-to-Average Power Ratio Limiting Codes for OFDM", Int. Seminar on Coding Theory, Thahkadzor, Armenia, 2.-6.10.97.

4.5 Organization of Lectures, Seminars, and Colloquia

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

Invited by Prof. Moschytz:

10.04.97 **Prof. S. Haykin,** McMaster University, Hamilton / Canada
Distinguished Lecturer for IEEE Signal Processing Society
"Adaptive Signal Detection".

24.11.97 **Prof. T. Inoue,** Kumamoto University, Kumamoto / Japan
"Development of Intelligent Chips for an Agricultural Robot for Havesting Water Melons: An Architecture and it’s Algorithm".

26.06.97 **Prof. A. Carlosena ,** Universidad Publica de Navarra, Pamplona, Spain, gave a seminar on his recent work.

30.06.97 **Prof. Leon O. Chua,** University of California, Berkeley, USA
"The first operating CNN Universal Chip: A tera ops Supercomputer for Image and Video Processing".

27.10.97 **Helmut Boelcskei, TU Wien**
"Noise-Shaping in ueberabgetasteten Filterbänken".

05.11. - 17.12.97 **Prof. M. Vetterli,** EPFL Lausanne
Advanced Signal Processing Post-Graduate Course:
"Wavelets and Applications".

Invited by Prof. Massey:

06.01.97 **Dr. P. Janson,** IBM Research Lab.Rüschlikon Switzerland:
"Internet Security and Electronic Payments".

21.02.97 **Prof.J.S.Byrnes,** Univ. of Massachusetts at Boston, USA:
"An Improvement of the Walsh Functions".

07.03.97 **Dr. V. Zyablov,** Inst. for Problems of Info. Transmission, Moscow, Russia:
"Generalized Concatenation Construction for Correcting Multiple Blots of Errors
in Array Codes".

14.04.97 **Dr. R. Anderson,** Univ.of Cambridge, UK:
"Talk on Steganography".

26.05.97 **Prof. Shu Lin,** Univ. of Hawaii, USA:
"A Recursive Maximum-Likelihood Decoding Algorithm for Linear Block Codes".

02.06.97 **Dr. J. Ruprecht,** Swiss Telecom, Berne, Switzerland,
G. Krämer and U. Loher, ISI, ETH-Zürich, Switzerland:
"Code Time Division Multiple Access - The Marriage of CDMA and TDMA".

07.07.97 **Dr. L. Rasmussen,** National University of Singapore:
"Complexity-Constrained Maximum-Likelihood Detection in CDMA Systems".

13.11.97 **Dr. V. Blinovsky,** IPPI, Moscow, Russia:
"Some Problems of Combinatorial Coding Theory".

17.11.97 **Dr. V.V. Prelov,** IPPI, Moscow, Russia:
"Information-Theoretic Methods in Filtering Problems".

20.11.97 **Dr. G. Kabatianski,** IPPI, Moscow, Russia:
"A Digital Signature Scheme Based on Error-Correcting Codes".

27.11.97 **Dr. M.S. Pinsker,** IPPI, Moscow, Russia:
"The Entropy of Certain Sets in Hamming Space".

15.12.97 **A. van Wijngaarden,** Univ.of Essen, Germany:
"Distributed Synchronization Sequences".

18.12.97 **Prof. N. Kuznetsow,** IPPI, Moscow, Russia:
"On the Lack of Synchronization in Linear Sampled-Data Systems".

Invited by Dr. Heutschi:

15.1.97 **Dipl. Ing. Markus Erne,** Scopein Research, Aarau
"Analoge und digitale Tonstudioeinrichtungen im Wandel der Zeit".

5.2.97 **Prof. Dr.-Ing. Helmut Fuchs,** Fraunhofer-Inst. für Bauphysik, Stuttgart
"Alternative faserfreie Schallabsorber".

40.4.97 **Dipl.-Ing. Rudi Volz,** Institut für Technische Akustik, Technische Universität Berlin
"Schallschirme mit zylinderförmigen Aufsätzen - Erste Ergebnisse".

4.6.97 **Dipl.-Phys. Detlef Englich,** Carl von Ossietzky Universität, Oldenburg
Fachbereich Physik
"Bestimmung meteorologischer Parameter mit Schall in der athmospärischen Grenzschicht".

18.6.97 **Dipl.-Ing. Kurt Eggenschwiler,** EMPA, Abt. Lärmbekämpfung & Akustik
"Aktuelle Werkzeuge in der Raumakustik".

3.12.97 **Karlheinz Stegmaier,** Ing.büro für Bauakustik - Raumakustik, Berlin
"Akustische Grundlagen beim Tonstudiobau".

5. Publications

Group: Analog and Digital Signal Processing

Moschytz George	Encyclopedia "Analog Filters".
Schmid Hanspeter Moschytz George S.	"Tunable CCII-MOSFET-C Filter Biquads for Video Frequencies", Proceedings of ECCTD'97, Budapest, vol. 1, pp. 82-87,
Helfenstein Markus Muralt Arnold Fischer Gody Zbinden Paul Pfaff Dieter Frey Felix Moschytz George S.	"SC and SI Filters in baseband applications: A Comparison". Proceedings IEEE Intern. Symposium on Circuits and Systems, Hong Kong, Juni 1997.
Joho Marcel Moschytz George S.	"Adaptive Beamforming with Partitioned Frequency-Domain Filters", IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, USA, 19.-20.10.97, (4 pages)
Mirzai Bahram Moschytz George S.	"On the Dynamics and Stable Equilibria of anti-symmetric CNNs", Proceedings of ISCAS'97, Hong Kong, vol. 1, pp. 573-576.
Mirzai Bahram Lim Drahoslav Moschytz George S.	"Robustness of Delta Operator Based Cellular Neural Networks", Proceedings of ISCAS'97, Hong Kong, vol. 1, pp. 761-764.
Bahram Mirzai Reutemann Robert Rüegg Michael Moschytz George S.	"Isolated Word Recognition utilizing A CNN Encoder", Proceedings of ECCTD'97, Budapest, vol. 2, pp. 615-620
Hänggi Martin Moschytz George S.	"Visualization of CNN Dynamics", Electronic Letters, Sept. 1997, vol. 22, nr. 20, pp. 1714-1716.
Schaerer Thomas	"Entwicklung: Einschaltstrombegrenzung für Netzteile mit Ringkerntrafos", MEGALINK, no. 1, 27.1.97, pp. 14-15.
Schaerer Thomas	"Forum: Ohm, Kirchhoff und die Warm-/Kalt-Anzeige", MEGALINK, nr. 2, 2.2.97, pp. 10.
Schaerer Thomas	"Workshop: Selbstbau-Anleitung: PC-Lüfterüberwachungs-Schaltung", MEGALINK, nr. 11, 16. 6.97, pp. 12.
Schaerer Thomas	"Workshop: Selbstbau-Tip: PC-Ueberwachungsschaltung (II)", MEGALINK, nr. 12, 30.7..97, pp. 12.

Group: Adaptive Systems

Kälin August Müller Daniel	"On the Joint Adaptation of Memory Based Nonlinear Adaptive Filters used in Echo Canceling for HDSL-Modems", 9th Thyrrenian International Workshop on Digital Communications: A perspective of broadband wireless communications in the last mile, Villa Marigola, Lerici, Italy, 7.-10.9.97.
Kälin August Lindgren Allen Wyrsch Sigi	"A Digital Frequency-Domain Implementation of a very high Gain Hearing Aid with Compensation for Recruitment of Loudness and Acoustic Echo Cancellation", accepted for publication in February 1998 in Signal Processing, Image, Communication.
Oberle Stefan Kälin August	"HMM-Based Speech Enhancement using Pitch Period Information", Proceedings of ISCAS'97, Hong Kong.
Wyrsch Sigi Kälin August	"On a Digital Hearing Aid with Recruitment of Loudness Compensation and Acoustic Echo Cancellation", Inter. Workshop on Acoustic Echo and Noise Control, London, Sept. 97, pp. 21-24.
Wyrsch Sigi Kälin August	"A DSP Implementation of a Digital Hearing Aid with Recruitment of Loudness Compensation and Acoustic Echo Cancellation", IEEE Workshop on Applications of Signal Processing to Audio and Acoustics, New Paltz, USA, October 1997.
Steiner Rolf Kälin August	"Spectral Estimation of Moving Noise Sources Based on Acoustic Holography", Proc. of ISCAS'97, Hong Kong.

Group: Digital Information Theory

Massey James L. V.C. da Rocha, Jr.	"On the Entropy Bound for Optimum Homophonic Substitution", in Proc. IEEE Int. Symp. on Info. Th., 1997, p. 93.
Massey James L.	"Codes and Ciphers--What’s the Difference?", in Proc. IEEE Information Theory Workshop, Longyearbyen, Norway, 1997, p. 37.
Massey James L.	"The Invertibility Principle for Maximum-Likeelihood Estimation", in Proc. 4th Int. Symp. Commun. Theory & Appl.,1997, p. 273.
Ganz Jürg	"Factoring Polynomials Using Binary Representations of Finite Fields", IEEE Trans.Inform. Theory, vol.IT-43, pp. 147-153, Jan. 1997.
Harpes Carlo Massey James L.	"Partitioning Cryptanalysis", in Fast Software Encryption (Ed. E. Biham) Lecture Notes in Computer Science, No. 67. Heidelberg and New York: Springer 1997, pp. 13-27.

Krämer Gerhard	"A Sequential Strategy for the Two-User Noiseless Binary Adder Channel with Feedback", in Proc. ISIT’97,p. 131.
Loher Urs	"The Role of Information in Random Accessing", in Proc. ISIT’97, p. 321.
Grant Alex Schlegel Christian	"Collision-Type Multiple-User Communications", IEEE Trans. Info. Theory, pp. 1725-1736, Sept. 97.

6. Guests, Visitors

6.1 Activities of Academic Guests at the Institute

Guests of Prof. Moschytz:

Prof. Alfonso Carlosena,
Universidad Publica de Navarra, Pamplona, Spanien
worked together with analog circuit group; gave a
seminar on his recent work. 01.06.-31.07.97

Prof. Allen Lindgren,
University of Rhode Island, Kingston, USA
Collaboration with the Adaptive Filter Group. 01.06.-31.08.97

Prof. Takao Hinamoto,
Hiroshima University, Higashi-Hiroshima, Japan
Exchange of ideas and research activities with
Prof. Moschytz. 01.09.-10.09.97

Prof. Takahiro Inoue,
Kumamoto University, Kumamoto, Japan
Exchange of ideas and research activities with
Prof. Moschytz, and gave 2 seminars. 11.10.-31.12.97

Prof. H. Reddy,
California State University, Long Beach, USA
Ongoing research on Delta-Operator Signal Processing. 05.12.-16.12.97

Guests of Prof. Massey:

Dr. Ann Canteaut,
INRIA, Rocquencourt, France 01.01.-30.09.97

Prof. John Kieffer,
Univ. of Minnesota, Minneapolis, USA 15.03.-15.07.97

Dr. Vladimir Blinovskii,
Dr. Viatcheslav Prelov,
IPPI, Moscow, Russia 09.11.-23.11.97

Dr. Mark Pinsker,
Dr. Grigori Kabatianski,
IPPI, Moscow, Russia 16.11.-30.11.97

Prof. Nikolai Kuznetsov,
IPPI, Moscow, Russia 07.12.-21.12.97

7. Honors and Awards