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 1998

Teaching Period 1997/98

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Foreword

Once again it is a pleasure to send the Annual Report, this time for 1998, of the Signal and Information Processing Laboratory (or ISI for "Institut für Signal- und Informationsverarbeitung) to our many friends and past and present associates.

Looking back on 1998 the most memorable and nostalgic event was without doubt the retirement on the 31st March of Professor James L. Massey, or Jim, as he is most commonly known. Jim Massey had been with ISI since the beginning of 1980, and spent the following 18 prolific years in Zurich with his wife Lis. He made an indelible mark on the Swiss and European communications scene by building up an outstanding group of researchers in the field of Information Theory whose achievements were respected and admired well beyond our national borders. Jim and Lis have retired to Copenhagen (Lis' birthplace) from where they continue to have close contacts with ISI and other centers of Information Theory worldwide.

Another "partial leave" from ISI last year was that of Professor August Kälin. August Kälin who joined ISI as a teaching assistant in June 1983, got his PhD degree in May 1990, and became an assistant professor in the E.E. Department in November 1992. Since that time he worked with, and supervised a group of graduate students working in the field of adaptive filtering, active noise control, automatic equalization and related topics. Some half dozen students successfully completed their PhD degrees under his guidance. Since October 1998 Dr. Kälin is leading a forward-looking research group at the Zurich-based Siemens-Schweiz Company; the group includes a number of his former PhD students. Fortunately, Dr. Kälin still has close ties to ISI and will, for the time being, continue part-time as a group leader of a small number of PhD students.

With Jim Massey's retirement, i.e. throughout 1998 and 1999, most of his doctoral students were able to complete their PhD theses. Urs Loher completed his PhD thesis on "Information-Theoretic and Genie-Aided Analysis of Random-Access Algorithms", and now works at Swisscom in Berne, and Gerhard Krämer completed his PhD on "Directed Information for Channels with Feedback" and presently works for an ETH-spawned start-up company. Dr. Krämer's outstanding thesis was awarded an ETH silver medal. The remaining "Massey students" will complete their theses in the course of 1999.

Other PhD theses completed at ISI in 1998 were those of Stefan Oberle on "Detection and Estimation of Acoustical Signals Using Hidden Markov Models"; he now works in Dr. August Kälin's research group at Siemens Schweiz. Jürg Stettbacher completed his thesis on "Beitrag zur Audiometrie fuer binaurales Hören" and still works part-time at ISI. Rolf Steiner who completed his thesis in 1997, but stayed on at ISI for another year, now works at Mc Kinsey Inc. Switzerland. Filling up the ISI vacancies are three new ISI members, namely Markus Hofbauer, diploma graduate of the EE department ETH, Ms. Qun Gao, MSc. graduate from Shanghai University, and Ralf Kretzschmar, diploma graduate of the department of Physics ETH. Marcel Merk, who spent a year at ISI assisting in the streamlining of some complex DSP algorithms for speech detection, left ISI for the USA where he is continuing his studies towards an MSc degree in Portland, Orgeon. The successful research work carried out at ISI was rewarded by a relatively large number of research papers presented at International Conferences and Symposia, and published in various papers and reputable journals, as listed in this Annual Report.

1998 was a year in which ISI was preoccupied with the organization of what turned out to be two exceptionally successful events. In June, members of ISI, under the leadership of Dr. Markus Helfenstein, organized the 1st IEEE Circuits-and-Systems (CAS) Workshop on "Emerging Technologies with emphasis on Circuits and Systems for Wireless Communications". The workshop was held in the beautiful town of Lucerne and was successful in every way: from the high quality of the lectures, to the social events high-lighting the charm and beauty of Lucerne, right up to the weather which was uncharacteristically beautiful from the first day to the last. The second event was the third annual summer school organized by three institutions ISI, the EPFL (Professor Martin Hasler) and Berkeley – in the person of Professor Leon Chua. This year the summer school, which is held alternately in Zurich and Lausanne, was devoted to "Introduction to Wavelet Theory and Applications for Signal Processing". It was organized by Markus Erne and Peter Wellig together with other ISI members at the ETH in Zurich. The summer school coincides with an annual summer visit of Professor Leon Chua from Berkeley University. Prof. Chua's expertise and contagiously motivating enthusiasm has been a driving motor for this valuable annual event, and for ongoing research at both EPFL and ISI-ETHZ in the field of Cellular Neural Networks (CNNs) and Nonlinear Circuits and Networks, for several years. Other guests at ISI in 1998 were Proffs A. Feuer (Technion), A Lindgren (University of Rhode Island), A. Arbel (Technion), D. Tosic and M. Lutovac (University of Belgrade), L. Bassolygo (Moscow) and Dr. S. Serkonek from Brazil. As usual, these visits were extremely stimulating and interesting for the research groups in question. In particular, we enjoyed the lively discussions on new approaches to filter synthesis with Professors Tosic and Lutovac at a time when the terrible trauma enveloping their country could not have been even dreamt of…

In closing, it is a pleasure to express our gratitude to the many administrative bodies at ETH for their encouragement and constant support for the running of our Institute. Most of all, though, we are grateful for the fine accomplishments of our talented ISI research team, and to the dedication and motivation of all the ISI members who make working at this laboratory a constant source of pride and satisfaction.

May 1999 Prof. Dr. G.S. Moschytz

Contents

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
retired on 31.3.98

Professor for Information Technology:

Prof. Dr. Fritz Eggimann

Adjunct Lecturer:

Dr. K. Heutschi

Assistant Professsor (Signal Processing):

Prof. Dr. August Kälin
left on 30.9.98

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.	
	Qun Gao	Dipl.El.Eng.	since 15.5.98
	Markus Hofbauer	Dipl.El.Eng.	since 15.3.98
	Zsolt Kukorelly	Dipl.Math	
	Heinz Mathis	Dipl.El.Eng.	
	Ralf Kretzschmar	Dipl.Phys.	since 1.11.98
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.	left on 30.6.98
	Xuejia Lai	Dr.	left on 31.3.98
	Drahoslav Lim	Master of Sci.	
	Martin Hänggi	Dipl.El.Eng.	
	Dani Lippuner	Dipl.El.Eng.	
	Hans-Andrea Löliger	Dr.	
	Felix Lustenberger	Dipl.El.Eng.	
	Marcel Merk	El.Eng. HTL	left on 31.8.98
	Bahram Mirzai	Dipl. Phys.	left on 31.3.98
	Stefan Oberle	Dipl.El.Eng.	left on 31.7.98

	Jossy Sayir	Dipl.El.Eng.	
	Hanspeter Schmid	Dipl.El.Eng.	
	Rolf Steiner	Dipl.El.Eng.	left on 31.8.98
	Jürg Stettbacher	Dipl.El.Eng.	
	Felix Tarköy	Dr.	
	Thomas von Hoff	Dipl.El.Eng.	
	Pascal Vontobel	Dipl.El.Eng.	
	Peter Wellig	Dipl.El.Eng.	
	Sigi Wyrsch	Dipl.El.Eng	
Technical Staff:	**Francesco Amatore**		
	Felix Frey	El.Eng.HTL	
	Thomas Schaerer		
	Patrick Schweizer	El.Eng.HTL	

Fellowship Recipients: (see 6.1 for report of activities)

A. Jurisic	University of Zagreb, Zagreb	01.01. - 31.12.98

Academic Guests: (see 6.1 for report of activities)

Prof. Arie Feuer	Technion – Israel Institute of Technology, Haifa, Israel	19.04. – 27.04.98
Prof. Leon Chua	University of California, Berkeley, USA	15.05. – 15.08.98
Prof. Allen Lindgren	University of Rhode Island, Kingston, USA	01.07. – 31.08.98
Prof. Arie Arbel	Technion – Israel Institute of Technology, Haifa, Israel	24.08. – 25.08.98
Prof. Dejan V. Tosic	University of Belgrade, Belgrade, Serbia	15.12. – 18.12.98
Dr. M.D. Lutovac	Telecommunications & Electronics Institute IRITEL Belgrade, Serbia	12.15. – 18.12.98
Dr. Shirlei Serconek	Brazilian Defense Ministry Goiania, Brazil	01.01. - 28.02.98
Prof. L. Bassalygo	IPPI, Moscow, Russia	22.02. - 08.03.98

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
6th	Dr. Mittelholzer	Applied Digital Information Theory II	35-418
7th	Prof. Moschytz	Adaptive Filter & neuronale Netzwerke	35-467
8th	Prof. Moschytz Prof. Eggimann	Analoge Signalverarbeitung und Filterung	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
	Dr. Heutschi	Acoustics Colloquium	35-950

2.2 Semester Projects and Diploma Theses

During the winter semester 1997/98 and summer semester 1998, 17 Semester Projects (29 candidates) and 9 Diploma Theses (12 candidates) were carried out.

Candidates	Title	Supervisor
Semester Projects WS 97/98 (7th Semester)		
Stöckli Michael Steger Marcus	Adaptive Filter basierend auf der Wavelet-Transformation	Wyrsch Lippuner
Arangeh Dordaneh	Adaptive Filter für nichtstationäre Umgebungen	Lippuner von Hoff
Flury Roger	Simulation alter Aufzeichnungsgeräte	Dr. Heutschi
Fengels Dirk Lukie Dejan	Timestretching und Timesqueezing von Audiosignalen im Frequenzbereich	Erne Wellig
Faller Christof	Audiodatenkompression unter Verwendung von Wavelets	Erne Wellig
Bucher Andreas Würsch Albert	Sprecherseparierung bei Videokonferenzen	Joho
Semester Projects SS 98 (8th Semester)		
Höneisen Bernhard	Recheneffizientes digitales Hörgerät mit MLT	Wyrsch Lippuner
Bürki Adrian Tschopp David	Stabilisation einer Verstärkungsanlage für einen Hörsaal mittels adaptivem Equalizer und adaptivem Echokompensator	von Hoff Lippuner
Fischer Beat Ferrat Pascal	Geräuschunterdrückung und akustische Echokompensation bei einer Freisprech-einrichtung für ein Autotelefon	von Hoff Lippuner
Hertach Peter Handlery Marc	Adaptive Filter für nichtstationäre Umgebungen	Lippuner von Hoff
Pfaffhauser Eric Moser Simon	JAVA-Simulator für Cellular Neural Networks	Hänggi
Würsch Albert Huber Marcel	Sprecher Trennung bei Videokonferenzen	Joho Mathis
Pagani Stefano Quarenghi Stefano	Akustische virtuelle Realität	Stettbacher
Ferrari Nicola Schait Claudio	Graphische Benutzeroberfläche für biomedizinische Daten	Wellig

Schinca Ermanno Fromherz Georg	Datenkompression von biomedizinischen Messdaten mit der Wavelet-Transformation	Wellig
Kalberer Gregor Spielmann Beat	Einstellbarer elektronischer Hybrid für duplexe Datenübertragung	Lím Muralt / GlobeSpan
Stefan Moser	Die Hypothese der festen Schlüssel bei der Gruppenverallgemeinerung der linearen Kryptoanalyse	Kukorelly

Diploma Theses WS 97/98

Semling Michael Cheng Zhenlan	Datenkompression von biomedizinischen Messdaten mit der Wavelet-Transformation	Wellig Erne Schnoz
Gençyilmaz Cenk Hofbauer Markus	Geräuschunterdrückung und akustische Echokompensation bei einer Freisprecheinrichtung für ein Autotelefon	von Hoff Lippuner
Salzgeber Gérhard	Stabilisation einer Verstärkungsanlage für einen Hörsaal mittels adaptivem Equalizer und adaptivem Echokompensator	von Hoff Lippuner
Notari Ruben Balmelli Pio	Fitnessfunktionen für genetische Algorithmen	Hänggi
Dober Peter	Echtzeit-Implementierung eines adaptiven Richtmikrophons	Joho
Rennhard Marc	Turning Ciphertext into Plain English Via Contextual Templates	Profos Massey Davida
Bengtson Michael	Minimum-Mean-Square-Error Receivers for Multi-User Fading Channels	Profos Massey Milstein
Walser Bernd	Impulsdetektion bei Laser-Distanz-Vermessung	Lippuner Kostyak/ Leica

Diploma Theses SS 98

Kenneth De Spiegeleire	Die Hypothese der festen Schlüssel in der linearen Kryptoanalyse	Kukorelly

Post-Diploma Thesis

Hänggi Martin	An Analysis of CNN Settling Time	Moschytz

Schmid Hanspeter	The ideal Amplifier Myth, and a Complete Amplifier Classification	Moschytz

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 Freehand Phones
- Measurement of Sound Propagation in Open Spaces
- Sound Localization in Audiology
- Adaptive Filters 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

Design of MOSFET-C 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. Switched-capacitor and switched-current filters can be very precise, but the bandwith of continuous-time analogue filters having the same power consumption is an order of magnitude higher, at the cost of precision. However, on-chip tuning makes it possible to tune a filter during its operation, and therefore to eliminate most of the errors which come from process tolerances, temperature and ageing.

In the beginning of the project, we investigated current-mode single-amplifier biquadratic filters (biquads) and high-frequency CMOS current-mode amplifiers with imprecise, but precisely reproducible gain. Then we showed how biquads with tunable pole frequency and pole Q can be implemented in CMOS. A first test chip contained a 25-MHz lowpass filter with tunable pole frequency and two current amplifiers suitable for building 30-MHz filters, one with fixed, the other linearly tunable gain. All test circuits have a comparatively low power consumption and a spurious-free dynamic range of 45dB (filter) or 60dB (amplifiers).
The final stage of the project is the integration of a tunable filter cascade of 6th order. A typical application for such filter could be pulse equalizing in harddisk or DVD read channels.

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

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 due, for example, to acoustical echos. 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 usually come 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, because the assumption that each sound source impinges from a single direction on to the microphone array is violated.

Blind source separation algorithms have shown the capability of solving multi-path problems in a simulated environment and therefore appear to be very promising for use in real acoustical applications. Blind algorithms make only weak assumptions on the signals involved, such as non-gaussianity, which is the case for speech signals.

We have investigated a new blind adaptive algorithm for blind source separation for the instantaneous mixing case which shows fast convergence behavior combined with low computational complexity. The algorithm has been modified to solve the blind deconvolution problem as well as the multichannel blind deconvolution problem, where the transfer functions between the sound sources and microphones are modeled by unknown filters.

In some applications, e.g. teleconferencing or hearing aids, some of the source signals involved are known. Different methods have been proposed that incorporate this additional information into a blind algorithm leading to an improved semi-blind algorithm.

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Keywords: adaptive beamforming, blind source separation

Embedded Audio Compression based on Wavelets and Improved Psychoacoustic Models

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, scalable 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 the use of 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 is to be extensively examined under critical listening test conditions. Some of the results obtained from this work have already been implemented in the upcoming MPEG-4-Standard.

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In Collaboration with: Scopein Research

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

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. em. 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 been achieved within the first year of the project: a "natural" 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. The proof of concept has been established by a demonstration unit for a small binary trellis code using discrete transistors. A Swiss patent application has been filed. (International patent applications are beeing prepared).

To demonstrate the advantages of the new decoding approach, a first test chip for a binary (18,9,5) tailbiting trellis code has been designed and fabricated in AMS 0.8um BiCMOS technology. Simulation results have shown a very robust behaviour with regard to non-idealities such as transistor mismatch, finite output resistance of MOS transistors, and temperature effects. Furthermore, measurement results show that bitrates of over 100MBit/s can be achieved with a single 5V power supply and power consumption of 100mW.

In a next step towards a full-sized decoding system, a test-chip for a more complex code with digital interfaces will be designed and fabricated.

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

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

High-Speed Interfaces for Analog, Iterative VLSI Decoders

Recently, we have introduced a new type of analog computing network (patent pending), that is especially suitable for decoding state-of-the art error correcting codes, such as turbo codes. The main advantages over digital implementation include higher speed and/or lower power consumption. These networks comprise simple computing blocks, which are densely connected, and operate at high speed and/or low-to-medium accuracy. The overall performance is achieved at the system level. In an effort to compare the performance of different interface architectures for connecting these decoding networks to off-chip circuitry, a case study has been carried out. Due to the parallelism of the problem, area and power efficiency are the crucial design parameters, in addition to the desired speed and accuracy. The parallelisation of the analog task has been studied, as well as the requirements for the analog memories with respect to the application under consideration, namely a decoder for a binary (18,9,5) tail-biting trellis code. Furthermore, the mapping from the soft decision output into a digital bit stream has been addressed. It was found that thanks to the current mode nature of the decoder, the switched-current approach in connection with the MOSFET-only ladder D/A converter makes true current mode operation possible, thereby circumventing additional V/I conversions. Using standard design techniques, an overall conversion time of less than 50ns has been achieved, and transmission rates in the 100MS/s range are possible.

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Supported by: ETH
In Collaboration with: Endora Tech AG

Keywords: high-speed interfaces, analog VLSI, switched-current networks

Decomposition of long-term intramuscular EMG signals using Wavelets

To analyse the causes of chronic muscle pain, intramuscular long-term measured EMG singals have to be decomposed. The measured signal, the so-called electromyogram or EMG signal, represents 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) along the muscle fibre. 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 depolarisations of the muscle fibres lying further away from the electrode placement. High-frequency noise is caused by time-offsets of the aligned waveforms and by physiological jitter. As we have recently shown, selected wavelet coefficients improve the SNR and the clustering performance. Furthermore, a reduction of the number of characteristic features can be achieved. Because of the achievable reduction of the error classification probability, the supervised classification of long-term recordings in the wavelet domain, including the MUAP shape tracking, appears to be very promising.

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

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

Analysis, Design and Optimization of Cellular Neural Networks

Cellular neural networks (CNNs) are analog, time-continuous, nonlinear dynamical systems and formally belong to the class of recurrent neural networks. Applications of CNNs include image processing, solving partial differential equations and pattern recognition, and nonlinear signal processing in general.

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. 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. In this project, it has been proved that optimally robust templates can be designed analytically by simple matrix algebra, and that a theoretical upper bound for the degree of robustness can be derived.

The time it takes a stable CNN to reach its equilibrium state is crucial for applications of CNN chips. The investigation of this so-called settling time as a function of the template parameters reveals that for some tasks, precise analytical expressions can be calculated, while for others, tight upper bounds can be given.

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Supported by: ETH

Keywords: cellular neural networks, robustness, settling time

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 analog processing units. CNNs are often 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 and functions in the manner of an analog computer. Virtually all applications 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, medium-sized chips containing six CNN cells along with controlling logic were designed and manufactured in a 0.7 micron CMOS process. The chips can be connected to form an arbitrarily large network and have provisions for automatic tuning of electrical parameters. Chips from different manufacturing runs were tested in order to collect representative data on accuracy and matching. Such data is important for CNN program ("template") design purposes. Twenty chips were connected to form a 10x12 fully programmable network. 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 successfully test the effects of different network sizes, network connectivities and cell missmatch, all without having to physically re-configure or re-wire the circuit. The operation of the circuit was found to conform well with previous simulations and to verify the feasibility of the design approach taken.

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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

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

The start-up company mechatronica has developed within a several years a level measuring system for liquids. 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. Another problem which made measurements at high temperatures nearly impossible was the large sound attenuation. To handle this problem the used sound frequency had to be reduced. Since most of the ultrasonic transducers are narrow-band devices, a new one was developed which operates at the desired frequency.

At our institute the whole system will be verified by means of measurements and, furthermore, an appropriate estimation algorithm will be developed. 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

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. Dealing with several sensors allows for the approach to the noise reduction problem by blind source separation algorithms. "Blind" means that neither the source signals (described by speech and noise) nor the mixing process (described by the multipath propagation of the sources in the car interior) are known. Blind source separation is the task to find a demixing system in order to recover the original signals.

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In Collaboration with: Siemens Schweiz AG

Keywords: blind source separation, echo cancellation, noise reduction, car mobile phone

Adaptive Filters for Nonstationary Environments

Most adaptation algorithms have excellent convergence properties in identifying an unknown time-invariant system in a stationary environment. However, in tracking a time-variant system, especially if the environment is nonstationary, the existing algorithms are rather unsatisfying. We describe the real-world system by a state-space model. Typically such a model includes process noise responsible for unknown changes of the system parameters on the one hand and measurement noise describing additive disturbances at the system output on the other hand. In tracking such a system we restrict ourselves to adaptive FIR filters using only the available input signal and the available, disturbed output signal. In readjusting the FIR filter coefficients with the help of the adaptation error, it is important to distinguish between process noise and measurement noise. Only in the former case the coefficients need to be readjusted.

An adaptation algorithm based on the optimum Kalman filter requires known noise statistics. In the report period, a new method for estimating these statistics has been developed. For a decorrelated input signal (white noise) the overall algorithm can be reduced to a simple 'LMS-like' adaptive filter with time-variant step size. The obtained linear complexity in the number of adjusted coefficents is important, since we would like to consider adaptive FIR filters with possibly thousands of coefficients. Future work has to be done to obtain a similar algorithm for colored input signals.

We will verify our algorithm on a real-world problem, namely the compensation of acoustic echos. For this purpose a prototype system with a programmable time-varying distance between the loudspeaker and the microphone has been constructed at our institute.

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

Subband Signal Processing for Hearing Instruments

People with loudness recruitment have a compressed dynamic range between the sound-pressure levels corresponding to threshold and discomfort. The audible frequency range from 20Hz to 16kHz is subdivided into 24 abutting non-uniformly spaced critical bands. The critical bands are closely related to several characteristics of the hearing system although the auditory filters become wider for hearing impaired people.

To compensate for the highly frequency-dependent compressed dynamic range, a nonlinear filter in each subband has to be used. An echo canceler is further needed to provide large gains in the compensator.

Different transformations such as the modulated lapped transform have been investigated to allow signal processing in the critical bands. The echo canceler is adapted in the frequency-domain using only the available speech input signal.

The problems resulting from the use of a nonlinear hearing-loss compensator and stability problems caused by a rapidly changing echo path have been investigated and solved. The computational efficient system shows excellent simulation results and allows an overall gain more than 10dB above the critical gain.

Further investigations have to be done to design an artifact-free hearing loss compensator.

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In Collaboration with: Phonak AG

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

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 was 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. The designs have been verified on test tubes at Sulzer Innotec and at the Ingenieurschule Burgdorf with the help of DSP-based controlers.

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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

Many of the most important 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 a circular 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 has been used to generate three-dimensional, dynamic acoustical situations like moving sound sources for example. In audiometric experiments these situations serve as stimulus to measure the useful hearing abilities under realistic binaural conditions. To decide what kind of experiments are needed a model of cognitive hearing has been established.

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In Collaboration with: ORL, Universitätsspital Zürich.

Keywords: binaural hearing, sound localization, audiology, audiometry, DSP, digital signal processing, parallel computing

Section 2: Digital Information Theory

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

Directed Information for the Discrete Memoryless Network

The Discrete Memoryless Network is a general communications model for channels with many users and noisy feedback. For example, it includes the two-way channel, the multiple-access channel and the broadcast channel as special cases. The capacity region of these channels can be expressed by using causal conditioning and directed information. The capacity expressions are applied to the multple-accesschannel with noiseless feedback for which improvements of the best existing rate regions are obtained. The coding technique to achieve these rates involves adaptive codewords, i.e., codewords that adapt with the feedback. Further investigations into the construction of codes with adaptive codewords are being done. The goal is to understand how best to use feedback to improve codes.

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Keywords: causal conditioning, directed information, feedback, capacity

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.

The success of the linear cryptanalysis attack relies on a certain number of hypotheses. Although the ignorance as to their validity will not keep anyone back from applying the attack, they are what we are working on presently. The most important ones are the hypothesis of wrong-key randomization and the hypothesis of fixed-key equivalence. Also, it is studied whether it is possible to use Matsui's Piling-up Lemma for dependent random variables if the equality is replaced by an approximation.

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Keywords: Cryptograpghy, Iterated Block Ciphers, Linear Cryptanalysis

Statistical Testing of 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, 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 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

Arithmetic Coding for Noisy Channels

Arithmetic Coding is a well-known technique to encode the output of a source efficiently for transmission over a noiseless channel. The technique functions by covering the unit interval with intervals representing the possible output sequences of the source. By introducing gaps between the those intervals, the technique is modified to provide a coding method for noisy channels.

In collaboration with Claudio Weidmann of the EPF Lausanne, the properties and the performance of arithmetic encoders for noisy channels were investigated and a decoding procedure was devised to be used in conjunction with an arithmetic channel encoder.

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Keywords: channel coding, sequential decoding, arithmetic coding

3.3 Completed Research Projects

OBERLE Stefan

"Detection and Estimation of Acoustical SignalsUsing Hidden Markov Models"

ETH-Diss. Nr. 12936 (Referee: Prof. Dr. A. Kälin)

The recognition of acoustical signals and the enhancement of speech both play an important role in many communication and signal processing systems. With the increasing portability of these systems, solutions which provide reliable operation in various acoustical environments are of particular interest.

In the present thesis, hidden Markov models (HMM) are shown to be a powerful tool for dealing with the areas mentioned above, approaching them as statistical detection and estimation problems, and putting them on a solid mathematical foundation. Based on this, methods and applications of HMM-based signal detection and estimation for block processing systems are presented.

In the area of acoustical signal recognition, automatic speech recognition systems are the most common application. Since the performance of such systems can decrease significantly in noisy environments, robustness against background noise plays an important role. For single word recognition, this thesis gives a contribution in the form of a robust algorithm for signal preprocessing. It includes an endpoint detection scheme for the determination of word boundaries which allows reliable operation also in noisy environments by using a background noise estimation scheme.

As an application example, an automatic command recognition system for operating speech-controlled devices is described. In addition to background noise disturbances, this application also addresses another problem: Since the command recognition system should always be active, it has to reliably detect and extract the relevant command words out of irrelevant speech data (for example conversations in the room). For this "word-spotting" problem a solution based on word duration models is presented. The speech recognition system is followed by a validation stage which uses a suitable probability normalization to distinguish the command words from irrelevant signals.

Moreover, this thesis shows that the use of hidden Markov models is not limited to the modeling of speech signals. Using an alarm signal recognition scheme for the profoundly deaf as an example, the HMM-based modeling and recognition of environmental sounds is considered. It is shown that compared to other schemes the use of HMMs gives the best recognition rates.

In the area of speech enhancement, the focus of this thesis lies on single-channel methods for reducing background noise in speech signals. Special emphasis is placed on the HMM-based MMSE estimator, where the statistical properties of the speech and noise signal are described by a speech-HMM and a noise-HMM. These two models can be combined into a composite-HMM which serves as the model for the noisy speech signal and can be used to determine the optimal filter for every noisy speech segment.

It is shown that this scheme can be further improved if additional a priori knowledge of the speech signal is employed. In voiced speech segments, an improvement of the noise reduction can be achieved by using the pitch period which allows the disturbing frequency components between the harmonics of the speech signal to be sufficiently removed. Moreover, it is shown that speech segments with low energy have to be processed separately because the HMM-based scheme does not give a reliable estimation for these segments.

The improvement in the HMM-based scheme obtained through the extensions mentioned is demonstrated using a paired comparison test for speech quality judgement.

KRAEMER Gerhard

"Directed Information for Channels with Feedback"

ETH-Diss. Nr. 12656 (Referee: Prof. Dr. J.L. Massey)

The capacity regions of channels with feedback are investigated. The corresponding information rates are simplified by using the conditional independence of random variables. To establish conditional independence, use is made of d-separation in functional dependence graphs. A weaker condition called fd-separation is introduced and also shown to establish conditional independence in functional dependence graphs. Causally conditioned uncertainty and causally conditioned directed information are defined and used to express the capacity region of the two-way channel and the multiple-access channel with feedback. For both of these channels, rate regions whose points are approachable with error probability approaching zero are developed, including generalizations of Han's rate region for the two-way channel and generalizations of Cover and Leung's rate region for the multiple-access channel with feedback. Finally, feedback strategies are designed for the class of multiple-access channels for which one of the channel inputs is determined by the second channel input and the channel output. These strategies approach all rate points in the capacity region of these channels.

LOHER Urs

Information-Theoretic and Genie-Aided Analyses of Random-Access Algorithms

ETH-Diss. Nr. 12627 (Referee: Prof. Dr. J.L. Massey)

The random access problem is one of organizing a population of users so that they may efficiently share the resources of a single communications channel.

In this thesis, the identification of information which must be exchanged within a set of infinitely many, uncoordinated users possibly attempting to access a common collision-type channel, is explicitly addressed. Particularly, the essential information being gathered in a collision resolution algorithm to successfully transmit the messages of the users, is specified. This is the first time in the literature where pure information-theoretic arguments were used to describe random-accessing systems and are applied to calculate the performance of specific protocols. Moreover, information-theoretic bounds are applied to ternary feedback to yield an upper bound on the maximal stable throughput of a random access broadcast channel; multiplicity feedback is discussed as well. Nonetheless, this concept still offers interesting unanswered aspects to find the capacity of the collision channel with feedback.

The problem of resolving collisions is then viewed as a generic search problem with N independent random variables, e.g., active users, where the concept of a *probabilistic* genie is introduced and the capacity for the first non-trivial case, namely $N = 3$ collided packets, where N is known to all users, is given. For $N = 4$ new lower and upper bounds on the effciency are derived. Unfortunately, they do not meet each other such that the capacity for $N \geq 4$ users is still an open problem.

Furthermore, various facets of selected problems applying different degrees of feedback are considered. For binary feedback, in particular success/no success feedback, a collision resolution protocol is introduced whose throughput is shown to be 0:372, the highest reported to date. Using a genie argument, a new upper bound on First-Come First-Served Algorithms is derived. This upper bound of 0:4906 is only slightly higher than the effciency of the best algorithm presently known which achieves a maximum stable throughput of 0:4878. Finally, a new strategy achieving capacity 1 for the collision channel with multiplicity feedback is presented. A virtue of this protocol is its simplicity.

Some remarks on future work and a brief summary of the results then conclude the thesis.

STETTBACHER Jürg

Audiometry to Evaluate Binaural Hearing

ETH-Diss. Nr. 12723 (Referee: Prof. Dr. G.S. Moschytz)

Normal binaural hearing provides comprehensive information on our acoustic environment. This includes clues on the location and movement of sound sources, and on the multiplicity of reflections and reverberation signals that characterize every room and space. The ability of a listener to interpret these elements can be very important for personal safety in traffic or workplace situations. Another aspect of binaural hearing is the capability to discriminate interfering sound sources and to focus on one particular source. In so-called Cocktail-Party-Situations this leads to a unique enhancement of the speech intelligibility which is of considerable social importance.

When the sense of hearing is impaired, the binaural capacities are affected most seriously. Unfortunately the benefits from hearing and from hearing aids are usually only measured under very artificial conditions. For this reason the conclusions of traditional audiometry may not coincide with practical experience in everyday life. Further improvements in evaluating the overall usefulness of hearing will be possible when audiometry includes testing of binaural effectiveness in real situations. As a contribution in this direction this report presents

- a concept for binaural testing and evaluation,
- a flexible hardware solution which enables binaural audiometry,
- a limited set of stimuli, including a classification system,
- and a collection of algorithms to reproduce static and dynamic virtual sound sources.

Together these elements form a complete system for binaural audiometry. The concept is derived from a model of the cognitive hearing process, which is established for this purpose. It distinguishes the four processes (1) perception of sound and sound quality, (2) localisation and perception of spaciousness, (3) identification of sound sources, and (4) perception of semantic contents of sounds. The concept presumes that all of the four hearing processes contribute to the usefulness of hearing and therefore should be included in the audiometric evaluation.

The measurement device incorporates an array of loudspeakers and serves to reproduce acoustic signals in a controlled free field setting. Real situations with several sound sources can be simulated. By measuring in the free field, all individual hearing properties of the particular test person are automatically included in the result, for example the direction-dependent transfer characteristics of the auricle. Various stimuli are examined and classified according to their psychoacoustic properties in binaural measurements. The main aspects are (1) the apparent spacial distribution of a sound source or the sharpness of the perceived source location, and (2) the minimum audible angle of displacement. The selection of stimuli includes the signals commonly used in audiometry as well as natural sounds like animal voices, traffic noise, etc.

Different algorithms to generate virtual sound sources by using the array of fixed loudspeakers are compared. Experiments reveal that the published and commonly used algorithms are not satisfactory for audiometric use, especially in the case of dynamic virtual sources. Therefore the quality of the most promising approach - which is a stereophonic one - is enhanced by physical and psychoacoustic models. Finally an audiometric procedure is proposed. It includes various experiments covering the different aspects of natural and everyday hearing. A short example illustrates the interpretation of the measurement results.

3.4 Completed Dissertations

OBERLE Stefan	Detection and Estimation of Acoustical Signals using Hidden Markov Models *ETH Diss. Nr 12936* Referee: Prof. Dr. A. Kälin Co-referee: Prof. Dr. G.S. Moschytz PD Dr. habil. H. Reiniger, Johann Wolfgang Goethe-Universität, Frankfurt a/Main
KRAEMER Gerhard	Directed Information for Channels with Feedback *ETH-Diss. Nr. 12656* Referee: Prof. Dr. J.L. Massey Co-referee: Prof. Dr. A.J. Han Vinck, Universität Essen
LOHER Urs	Information-Theoretic and Genie Aided Analyses of Random-Access Algorithms *ETH-Diss. Nr. 12627* Referee: Prof. Dr. J.L. Massey Co-referee: Dr. P. Humblet, Institut Eurécom Antipolis
STETTBACHER Jürg	Beitrag zur Audiometrie für binaurales Hören *ETH-Diss. Nr.12723* Referee: Prof. Dr. G.S. Moschytz Co-referee: Prof. Dr. E. Rathe

3.5. Internal Reports

9801	Schmid Hanspeter	The Ideal Amplifier Myth, and a complete Amplifier Classification
9801/2	Schmid Hanspeter	Understanding Integrated Amplifiers
9802	Hänggi Martin	An Analysis of CNN Settling Time
9803	Lustenberger Felix Löliger H.A. Helfenstein Markus Tarköy Felix	Probability Propagation in Analog VLSI

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 third ETHZ-EPFL Summer School on Introduction to Wavelet Theory and Applications for Signal Processing together with Prof. M. Hasler, CIRC EPFL and Prof. L.O. Chua, UC Berkeley.

Program Committee and Steering Committee of of ICECS.

Organization of 1st IEEE – CAS Workshop on Wireless-Communication Circuits and Systems in Lucerne.

Prof. Massey

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

Co-Organizer and Session Chair, 1998 Workshop on Cryptographic Protocols, Monte Verità, Switzerland.

Session Chairman, 1998 Symposium on Communications and Coding, Lancaster, England.

Organizer and Session Chairman for the Session on Cryptography, 1998 IEEE Information Theory Workshop, Killarney, Ireland.

Session Chairman, 1998 IEEE International Symposium on Information Theory, Cambridge, USA.

Rump Session Chairman, EUROCRYPT'98, Helsinki, Finland.

4.2 Participation in Congresses and Meetings

Prof. Moschytz Group:	**Analog and Digital Signal Processing**
Moschytz George S.	Research Stay with Globespan Technologies, New Jersey, USA, 9.2.-11.3.1998 and 13.7.-31.8.1998.
Moschytz George S.	ECBS'98 Conference, Israel, 26.3.-2.4.1998
Moschytz George S.	IEEE Ex-Com Meeting, Chicago, USA 1.-2.3.1998.
Moschytz George S.	Midwest Symposium on Circuits and Systems, Notre Dame, IN, USA, 9.-12.8.1998.
Moschytz George S.	Visit to Imperial College, London, England, 30.10.-3.11.1998.
Moschytz George S.	IEEE Meeting, San José and New Jersey, USA, 5.-15.11.1998.
Moschytz George S.	IEEE – CAS ACE, Advanced Continued Education Courses, Bangkok, Thailand, 7.-15.12.1998.
Erne Markus	42nd ISO-MPEG Committee Meeting, San José, USA, 2.-6.21998.
Erne Markus	DAGA 98, ETH Zurich, Switzerland, 23.-26.3.1998.
Erne Markus	104th AES-Convention, Amsterdam, Netherlands, 16.-19.5.1998.
Erne Markus	45th ISO-MPEG Committee Meeting, Atlantic City, USA, 12.-16.10.1998.
Hänggi Martin Lustenberger Felix	IZS'98, International Zurich Seminar on Broadband Communications, Zurich, Switzerland, 17.-19.2.1998.
Hänggi Martin Mirzai Bahram	CNNA'98 International Workshop on Cellular Neural Networks and their Applications, London, England, 14.-17.4.1998.
Hänggi Martin	ICEC'98, International Conference on Evolutionary Computation, Anchorage, Alaska, 4.-9.5.1998.
Hänggi Martin	ICASSP'98, International Conference on Acoustics, Speech and Signal Processing, Seattle, USA, 11.-15.5.1998.
Hänggi Martin	Research Stay at University of California, Berkeley, USA, 19.-27.5.1998.
Hänggi Martin	NOLTA'98, International Symposium on Nonlinear Theory and its Applications, Crans-Montana, Switzerland, 14.-17.9.1998.

Moschytz George S. Hänggi Martin Helfenstein Markus Joho Marcel Mirzai Bahram Lim Drahoslav Schmid Hanspeter	ISCAS'98, International Conference on Circuits and Systems, Monterey, USA, 31.5.-3.6.1998.
Helfenstein Markus Schmid Hanspeter Lim Drahoslav Lustenberger Felix	1st IEEE-CAS Workshop on Wireless-Communication Circuits and Systems, Lucerne, Switzerland, 22.-24.6.1998.
Lim Drahoslav Mirzai Bahram	ESANN'98, European Symposium on Artificial Neural Networks, Bruges, Belgium, 22.-24.4.98.
Schmid Hanspeter Helfenstein Markus Lim Drahoslav	Workshop on Advances in Analog Circuit Design, Copenhagen, Denmark, 28.-30.4.1998.
Hofbauer Markus	ETHZ-EPFL Summer School: Introduction to Wavelet Theory and Applications for Signal Processing, Zurich, Switzerland, 6.-10.7.1998.
Moschytz George S. Helfenstein Markus Wellig Peter	ICECS'98, International Conference on Electronics, Circuits and Systems, Lisbon, Portugal, 7.-10.9.1998.
Wellig Peter	PROCID Meeting Göteborg, Sweden, 25.-26.5.1998.
Wellig Peter	PROCID Meeting, Helsinki, Finland, 20.9.1998.
Wellig Peter	3rd International Scientific Conference on Prevention of Work-Related Musculoskeletal Disorders, Helsinki, Finland, 21.-23.9.1998.
Hänggi Martin	Research Stay at Southeast University, Nanjing, China, 4.-18.10.1998.
Wellig Peter	20th Annual International Conference on the IEEE Engineering in Medicine and Biology Society, Hong Kong, 29.10.-1.11.1998.
Wellig Peter	PROCID Meeting, Winterthur, Switzerland, 5.-6.12.1998.

Group: Adaptive Systems

Steiner Rolf	INTERNOISE'98, Christchurch, Neuseeland, 15.-18.11.98.

Group: Applied Acoustics

Heutschi Kurt Stettbacher Jürg	DAGA'98, Deutsche Jahrestagung für Akustik, Zurich, Switzerland, 23.-27.3.1998.
Heutschi Kurt	ICA'98, 16th International Congress on Acoustics, Seattle, Washington, USA, 20.-26.6.1998.

Group: Digital Information Theory

Massey James L.	Symposium on Communications and Coding, Lancaster, England, 25.-28.1.98.
Massey James L.	1998 IEEE Information Theory Workshop, San Diego, USA, 8.-11.2.98.
Massey James L.	International Zurich Seminar on Wideband Communications, Zurich, Switzerland, 17.-19.2.98.
Massey James L.	Journée SISR-EIG, Geneva, Switzerland, 5.3.98.
Massey James L.	Monte Verità Workshop on Cryptographic Protocols for Distributed Systems, Monte Verità, Switzerland, 8.-13.3.98.
Massey James L.	Colloquium Lecture, Univ. of Essen, Germany, 18.3.98.
Massey James L.	IPPI, Moscow, Russia, 23.-30.3.98.
Massey James L.	Netherlands Acad. Of Sciences Colloquium, Amsterdam, The Netherlands, 17.6.98.
Massey James L.	IEEE Information Theory Workshop, Killarney, Ireland, 22.-26.6.98.
Massey James L. Krämer Gerhard Kukorelly Zsolt Sayir Jossy	IEEE International Symposium on Information Theory, Cambridge, USA, 16.-21.8.98.
Krämer Gerhard	Lucent Bell Labs, Murray Hill, USA, 13.8.98.
Krämer Gerhard	EPFL Lausanne, Switzerland, 12.11.98.
Kukorelly Zsolt	Winter School in Coding and Information Theory, Ebeltoft, Denmark, 13.-16.12.98.
Sayir Jossy	Data Compression Conference 1998, Snowbird, USA, 30.3.-1.4.98.
Sayir Jossy	EIDMA Mini-Course on Universal Compression, Eindhoven, The Netherlands, 11.-15.5.98.
Sayir Jossy	Workshop on Convolutional Coding and Applications, Kamp-Lintfort, Germany, 5.-6.11.98.

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 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

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

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

President-Elect of IEEE Circuits and Systems Society

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, Journal of Information and Optimization Sciences

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

Fellow of the IEEE New York

Member, Swiss Academy of Engineering Sciences

Member, Swiss Electrical Engineering 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, ComSoc Awards Board for 1996-98

Member, Scientific Advisory Board, Cylink Corporation

Member, Election Committee, Professorships in Communications, EPFL

Member, Evaluation Committee, Professorship in Informatics, University of Bergen, Norway

Sayir Jossy

Chairperson of the IEEE Student Branch, ETH Zurich,

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

Erne Markus	Invited Tutorial on "Codage audionumérique", Ecole d'Ingenieurs de Fribourg, Switzerland, 7.5.98.
Erne Markus	"Watermarking of Audio Signals", 104th Convention of the Audio Engineering Society, Amsterdam, Netherlands, 17.5.98.
Erne Markus	"Embedded Audio Compression based on Wavelets and Improved Psychoacoustic Models", ETHZ-EPFL Summerschool on Introduction to Wavelet Theory and Applications to Signal Processing, ETH Zurich, Switzerland, 6.-10.7.98.
Erne Markus	"Embedded Audio Compression based on Wavelets and Improved Psychoacoustic Models", Wavelets and Applications Workshop, Monte Verita, Ascona, Switzerland, 28.9.98.
Erne Markus	"Tutorial on Audio Coding", COST-G6 Workshop on Digital Audio Effects, Barcelona, Spain, 19.-21.11.98.
Hänggi Martin	"Stochastic and Hybrid Methods Towards Robust CNN Templates", CNNA'98, London, England, 15.4.98.
Hänggi Martin	"Genetic Optimization of Cellular Networks", ICEC'98, Anchorage, Alaska, 6.5.98.
Hänggi Martin	"CNN Settling Time and an Analytical Approach for the Design of Optimally Robust Templates", University of California, Berkeley, USA, 20.5.98.
Hänggi Martin	"The Cellular Neural Network: A Paradigm for Analog Supercomputing", NASA Ames, Moffett Field, USA, 28.5.98.
Hänggi Martin	"An Analysis of CNN Settling Time", ISCAS'98, Monterey, USA, 3.6.98.
Hänggi Martin	"Making CNN Templates Optimally Robust", NOLTA'98, Crans-Montana, Switzerland, 17.9.98.
Hänggi Martin	"Cellular Neural Networks: Template Design and Applications in Nonlinear Signal Processing", Southeast University, Nanjing, China, 14.10.98.
Helfenstein Markus	"Decoding in Analog VLSI", Lucent Technologies, New Jersey, USA, July 1998.

Helfenstein Markus	"Analysis and Design of Switched-Current Networks", Lucent Technologies, New Jersey, USA, July 1998.
Helfenstein Markus	"Exact Design of Multirate Switched-Current FIR Filters with improved Phase Linearity", IEEE Intern. Conference on Electronics, Circuits and Systems, Lisbon, Portugal, Sept. 98.
Joho Marcel	"On the Design of the Target-Signal Filter in Adaptive Beamforming", IEEE Intern.Symposium on Circuits & Systems ISCAS, Monterey, CA, USA, 2.6.98.
Lim Drahoslav	"On the Robust Design of Uncoupled CNNs", European Symposium on Artificial Neural Networks, Bruges, Belgium, 7.4.98.
Lim Drahoslav	"A Programmable gm-C CNN Implementation", International Workshop on Cellular Neural Networks and their Applications, London, 15.4.98.
Lim Drahoslav	"A Modular gm-C Programmable Implementation", IEEE International Symposium on Circuits and Systems, Monterey, Calif., USA, 6.6.98.
Lustenberger Felix	"Design on Analog VLSI Iterative Decoders (DAVID)", Fachtagung ProTelecom, ETH Zurich, Switzerland, 30.10.1998.
Wellig Peter	"Long-Term Electromyogram Decomposition under Semi-Dynamic Conditions", PROCID, Göteborg, 26.5.98.
Schmid Hanspeter	"Fundamental Frequency Limitations in Current-Mode Sallen-Key Filters", IEEE International Symposium on Circuits and Systems ISCAS, Monterey, USA, 1.6.98.
Wellig Peter	"Wavelets in Biomedical Signal Processing", ETHZ-EPFL Summer School on Linear and Nonlinear Adaptive signal Processing, Zurich, Switzerland, 10.7.98.
Wellig Peter	"Analysis of Wavelet Features for Myoelectric Signal Classification", 5th IEEE International Conference on Electronics, Circuits and Systems, Lisbon, Portugal, 9.9.98.
Wellig Peter	"Wavelet Features for MUAP Classification", PROCID, Helsinki, Finland, 20.9.98.
Wellig Peter	"Decomposition of EMG Signals using Time-Frequency Features", 20th Annual International Conference on the IEEE Engineering in Medicine and Biology Society, Hong Kong, 31.10.98.

Wellig Peter	"Electromyogram Data Compression using Single-tree and modified Zero-tree Wavelet Encoding", 20th Annual International Conference on the IEEE Engineering in Medicine and Biology Society, Hong Kong, 31.10.98.

Group: Applied Acoustics

Stettbacher Jürg	"Audiometrie zur Beurteilung des binauralen Gehoers", DAGA, 23.-26. März 1998, Zurich.

Group: Digital Information Theory

Massey James L.	“Orthogonal, Antiorthogonal and Self-Orthogonal Matrices and Their Codes“, Symposium on Communications and Coding, Lancaster, England, 25.-28.1.98.
Massey James L.	“The Discrete Fourier Transform in Coding and Cryptography“, Plenary Speaker, IEEE Information Theory Workshop, San Diego, USA, 8.-18.2.98.
Massey James L.	“Block Ciphers and the IDEA Algorithm“, Journée SISR-EIG “La Cryptographie“, Geneva, Switzerland, 5.3.98.
Massey James L.	“On the Design of Block Ciphers“, Colloquium Lecture, University of Essen, Germany, 18.3.98.
Massey James L.	“Cryptography: A Theoretical Mess“, Netherlands Academy of Sciences Colloquium, Amsterdam, The Netherlands, 17.-19.6.98.
Massey James L.	“On Antiorthogonal Matrices and Their Codes“, IEEE International Symposium on Informationa Theory, Cambridge, USA, 17.8.98.
Krämer Gerhard	“Directed information for the discrete memoryless network“, Lucent Bell Labs, Murray Hill, USA, 13.8.98.
Krämer Gerhard	“Causal conditioning, directed information and the multiple-access channel with feedback“, ISIT’98, Cambridge, USA, 18.8.98.
Krämer Gerhard	“Feedback strategies for a class of two-user multiple-access channels“, ISIT’98, Cambridge, USA, 20.8.98.
Kukorelly Zsolt	“The Hypothesis of Fixed-Key Equivalence for One Round of Encryption“, ISIT’98, Cambridge, USA, 17.8.98.
Kukorelly Zsolt	“Various Hypotheses in Relation with Linear Cryptanalysis“, Winter School in Coding and Information Theory, Ebeltoft, Denmark, 15.12.98.

Sayir Jossy	"Universal Source Coding with Competitive Lists", DCC 98, Snowbird, USA, 30.3.-1.4.98.
Sayir Jossy	"Arithmetic Channel Coding", Workshop on Convolutional Coding and Applications, Kamp.Lintfort, Germany, 5.-6.11.98.

4.5 Organization of Lectures, Seminars, and Colloquia

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

Invited by Prof. Moschytz:

12.01.98 **Dr. Andreas Poncet**, ABB Corporate Research Center, Baden-Dätwil "Data-Based Modeling for Prediction and Classification".

10.07.98 **Prof. Dr. Leon Chua,** University of California, Berkeley, USA "Visions and Trend of Wavelet Technology".

24.08.98 **Prof. Arie Arbel,** Technion – Institute of Technology, Haifa "Differential Signal Processing: Review, some novel ideas & wishful thinking".

Invited by Prof. Massey:

26.02.98 **Prof. Dr. Israel Bar-David**, Technion, Haifa,Israel: “An incremental freqency, amplitude and phase tracking algorithm for coherent demodulation over fast flat fading channels“.

26.02.98 **Prod. Dr. Leonid Bassalygo**, IPPI, Moscow, Russia: “The effect of unsynchronized symbols and channel errors on the capacity of a multiple-access channel“.

28.04.98 **Prof. Dr. A.J. Vinck**, University of Essen, Germany: “State Recovery for Convolutional Encoders“.

Invited by Dr. Heutschi:

14.01.98 **Dr. Jürg Stettbacher**, Institut für Signal- und Informationsverarbeitung, ETH, Zürich, "Audiometrie zur Beurteilung des binauralen Gehörs".

21.01.98 **Prof. Dr. Eric Rathe,** Russikon, Switzerland "37 Jahre Akustik".

04.02.98 **Dr. Daniel von Grünigen,** Ingenieurschule Burgdorf "Auslöschung von Lärm durch Gegenlärm in einem Rohr".

06.05.98 **Frau Dr. Sigrun Hirsekorn,** Fraunhofer Institut für zerstörungsfreie Prüfverfahren, Saarbrücken "Materialcharakterisierung durch Ultraschallabbildungsverfahren".

5. Publications

Group: Analog and Digital Signal Processing

Moschytz George S.	"Low-Sensitivity, Low-Power Active-RC Allpole Filters Using Impedance Tapering", IEEE Transactions on Circuits and Systems, accepted for publication in 1999.
Moschytz George S.	"Realizability Constraints for Third-Order Impedance-Tapered Allpole Filters", IEEE Transactions on Circuits and Systems, accepted for publication in 1999.
Erne Markus	"Tutorial on Audio Coding", Proceeding of the COST-G6 Workshop on Digital Audio Effects, Barcelona, November 98, pp. 151-158.
Erne Markus	"Embedded Audio Compression using Wavelets", Proceedings of the COST-G6 Workshop on Digital Audio Effects, Barcelona, November 98, pp. 147-150.
Hänggi Martin Moschytz George S.	"Stochastic and Hybrid Methods towards Robust CNN Templates", Proceedings of CNNA'98, April 1998, pp. 366-371.
Hänggi Martin Moschytz George S.	"Genetic Optimization of Cellular Neural Networks", Proceedings of ICEC'98, May 1998, pp. 381-386.
Hänggi Martin Moschytz George S.	"An Analysis of CNN Settling Time", Proceedings of ISCAS'98, June 1998, Vol. 3, pp. 155-158.
Hänggi Martin Moschytz George S.	"Making CNN Templates Optimally Robust", Proceedings of NOLTA'98, Sept. 1998, pp. 935-938.
Hänggi Martin Moschytz George S.	"An Exact and Direct Analytical Method for the Design of Optimally Robust Templates", IEEE Transactions on Circuits and Systems I, vol. 46, no. 2, Feb., 1998.
Helfenstein Markus Moschytz George S.	"Distortion Analysis of Switched-Current Circuits", IEEE Proceedings International Symposium on Circuits and Systems, Monterey, May 1998, vol. 1, pp. 29-32.
Helfenstein Markus Muralt Arnold Moschytz George S.	"Direct Analysis and Synthesis of Multiphase Switched-Current Networks using Signal-Flow Graphs", International Journal on Circuit Theory and Applications, vol. 26, pp. 253-280.
Helfenstein Markus Moschytz George S.	"Improved Two Step Compensation Technique for Switched-Currents", IEEE Transactions on Circuits and Systems, vol. 45, no. 6, pp. 739-743, June 1998.
Helfenstein Markus Moschytz George S.	"Exact Design of Multirate Switched-Current FIR Filters with Improved Phase Linearity", IEEE International Conference on Electronics, Circuits and Systems, Lisbon, Sept. 1998.

Joho Marcel Moschytz George S.	"On the Design of the Target-Signal Filter in Adaptive Beamforming", Proceedings of ISCAS, Monterey, USA, vol. 5, pp. 166-169, June 1998.
Loeliger Hans-Andrea Helfenstein Markus Lustenberger Felix Tarkoey Felix	"Probability Propagation and Decoding in Analog VLSI", International Symposium on Information Theory, Cambridge, USA, Aug. 98, p. 146.
Mathis Heinz	"Differential Detection of GMSK Signals with Low BT using the SOVA", IEEE Transactions on Communications, vol. 46, no. 4, pp. 428-430, April 1998.
Lim Drahoslav Moschytz George S.	"A Progammable gm-C CNN Implementation", IEEE International Workshop on Cellular Neural Networks and their Applications, V. Tavsanoglu, E., London, April 1998, pp. 294-299.
Lim Drahoslav Moschytz George S.	"A Modular gm-C Programmable Implementation", IEEE International Symposium on Circuits and Systems, Monterey, Calif., USA, June 1998, vol. 3, pp. 139-142.
Mirzai Bahram Moschytz George S.	"The Influence of the Boundary Conditions on the Robustness of a CNN", IEEE Transactions on Circuits and Systems, I: Fundamental Theory and Applications, April 1998, vol. 45, no. 4, pp. 511-515.
Mirzai Baharm Lim Drahoslav Moschytz George S.	"Applications of CNN Processing by Template Composition", IEEE International Workshop on Cellular Neural Networks and their Applications.,V. Tavsanoglu, Ed., London, April 1998, pp. 379-384.
Mirzai Bahram Lim Drahoslav	"On the Robust Design of Uncoupled CNNs", European Symposium on Artificial Neural Networks, Bruges, Belgium, April 1998, pp. 297-302.
Mirzai Bahram Moschytz George S Chen Zhenlan.	"Learning Algorithms for Cellular Neural Networks", ISCAS'98, Proceedings of the 1998 IEEE International Symposium on Circuits and Systems, vol. 3, pp. 159-162.
Schaerer Thomas	"Kostengünstiges Netzteil – Kondensator statt Trafo", MEGALINK, no. 21, pp. 2, 30.11.98.
Schaerer Thomas	"Kondensator statt Trafo (Teil II)", MEGALINK, no. 22, pp. 3, 14.12.98.
Schmid Hanspeter Moschytz George S.	"Fundamental Frequency Limitations in Current-Mode Sallen-Key Filters", Proceedings of ISCAS, Monterey, USA, vol. 1, pp. 57-60, 1998.
Wellig Peter Moschytz George S. Läubli Thomas	"Decomposition of EMG Signals using Time-Frequency Features", Proceedings of EMBS'98, Hong Kong, vol. 20, part 3, pp. 1497-1500.

Wellig Peter Cheng Zhenlan Semling Michael Moschytz George S.	"Electromyogram Data Compression using Single-tree and modified Zero-tree Wavelet Encoding", Proceeding of EMBS'98, Hong Kong, vol. 20, part 3, pp. 1303-1306, 1998.
Wellig Peter Moschytz George S.	"Analysis of Wavelet Features for Myoelectric Signal Classification", Proceedings of ICECS'98, Lisbon, Portugal, vol. 3, pp. 109-112, 1998.

Group: Adaptive Systems

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", International Journal on Signal Processing, vol. 64, no. 1, pp. 71.-85, February 1998.
Steiner Rolf Kälin August	"Sound Field Reconstruction of Moving Noise Sources by means of Acoustical Holography", International Congress on Noise Control Engineering, Christchurch, New Zealand, 1998.
Kälin August von Grünigen Daniel	"On the Use of a Priori Knowledge in Adaptive Inverse Control", IEEE Transactions on Circuits and Systems, Part I, submitted for publication.
Steiner Rolf Kälin August	"Localization of Moving Noise Sources by Means of Acoustical Holography", The Journal of the Acoustical Society of America, submitted for publication.
Estermann Pius Kälin August Lindgren Allen	"Analysis of Partitioned Frequency-Domain LMS Adaptive Algorithm with Application to a Hands-Free Telephone System Echo Canceler", Internationl Journal on Signal Processing, EURASIP, submitted for publication.

Group: Applied Acoustics

Stettbacher Jürg	"Beitrag zur Audiometrie fuer binaurales Hoeren", Hartung-Gore Verlag, Konstanz, 1998, p. 160.

Group: Digital Information Theory

Massey James L.	“Orthogonal, Antiorthogonal and Self-Orthogonal Matrices and Their Codes“, in Communications and Coding (Ed. M.Darnell and B. Honary). Taunton, England and New York: Reseach Studies Press and Wiley, 1998, pp. 3-9.

Massey James L.	“The Discrete Fourier Transform in Coding and Cryptography“, Proc. IEEE Information Theory Workshop, San Diego, USA,8.-11.Feb. 1998, pp. 1-2.
Massey James L.	"Codes and Ciphers: Fourier and Blahut", in Codes, Curves and Signals: Common Threads in Communication, (Ed. A. Vardy). Kluwer, 1998, pp. 105-119.
Massey James L.	"On Antiorthogonal Matrices and Their Codes", Proc. ISIT'98, p. 64.
Massey James L.	"Rudi at the Board", (poem in honor of Prof. Rudi Ahlswede) IEEE Information Theory Society Newsletter, December 1998, pp. 29-30.
Massey James L.	"Feedback strategies for a class of two-user multiple-access channels", Proc. ISIT'98, p. 407.
Krämer Gerhard Gastpar Michael	“A spectral criterion for feedback coding“, Proc. ISIT’98, p.26.
Krämer Gerhard	“Causal conditioning, directed information and the multiple-access channel with feedback", Proc. ISIT’98, p. 189.
Krämer Gerhard	“Feedback strategies for a class of two-user multiple-access channels“, Proc. ISIT’98, p. 407.

6. Guests, Visitors

6.1 Activities of Academic Guests at the Institute

Guests of Prof. Moschytz:

Prof. Arie Feuer	Technion – Israel Institute of Technology, Haifa, Israel held a lecture and research activities	19.04. – 27.04.98
Prof. Leon Chua	University of California, Berkeley, USA held a talk on "Visions and Trend of Wavelet Technology"	15.05. – 15.08.98
Prof. Allen Lindgren	University of Rhode Island, Kingston, USA Collaboration with the Adaptive Filter Group	01.07. – 31.08.98
Prof. Arie Arbel	Technion – Israel Institute of Technology, Haifa, Israel held two talks on "Differential Amplifiers, floating current sources, biquadratic Filters and A/D converters"	24.08. – 25.08.98
Prof. Cederbaum	Technion – Israel Institute of Technology, Haifa, Israel research activities	23.09. – 24.09.98
Prof. Dejan V. Tosic	University of Belgrade, Belgrade, Serbia lectures and research activities	15.12. – 18.12.98
Dr. M.D. Lutovac	Telecommunicaitons & Electronics Institute IRITEL Belgrade, Serbia lectures and research activities	15.12. – 18.12.98

Guests of Prof. Massey:

Dr. Shirlei Serconek	Brazilian Defense Ministry Goiania, Brazil,	

	worked with Prof. Massey on generalizing the Discrete Fourier Transform to sequences of arbitrary period	01.01. - 28.02.98
Prof. L. Bassalygo	IPPI, Moscow, Russia Institutional partnership and cooperation between the information theory researchers at the ETHZ and their counterparts in the Dobrushin Mathematical Lab., Supported by the Swiss National Science Foundation	22.02. - 08.03.98
	Honors and Awards	

7. Honors and Awards

Moschytz George S.	President-Elect of the IEEE CAS Circuits and Systems Society.
Massey James L.	Degree of Professor, honoris causa, Institute for Problems of Information Transmission, Russian Academy of Sciences, Moscow, Russia.
Massey James L.	Election as Distinguished Israel Pollak Lecturer Technion, Haifa, Israel.
Massey James L.	Recipient of the Golden Jubilee Paper Award, IEEE Int. Symposium on Information Theory 1998, Cambridge, USA.
Hänggi Martin	Winner of the ETH-SEU award for excellent young scientists.