Signal and Information Processing Laboratory

Prof. Dr. G.S. Moschytz (Director) / Prof. Dr. A. Lapidoth
Prof. Dr. F. Eggimann / Dr. K. Heutschi

ANNUAL REPORT

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Research Period   1999

Teaching Period   1998/1999

Foreword

This Annual Report for the year 1999 is a last of its kind in more ways than one. For one thing, it coincides with the turn of the century and millenium, and therefore may reflect in a subtle way the expectations and anxieties that seemed to grip most of mankind towards the end of the last year. This includes a preoccupation with the "Y2K bug" – which, for any future readers who may have no idea what this is, was the possibly exaggerated fear of a "computer-system melt down" and a resulting world-wide chaos due to the inextricably interconnected computers of our, and every other institute, not being able to differentiate between the 00 of the year 1900 and that of the year 2000. We shall never know for sure, but it seems reasonable to assume that the smooth and utterly uneventful transfer into the new millenium was fully due to our capable system manager, Dr. Max Duenki, who has carefully and expertly, if not to say lovingly, developed and nutured our complex system of over 50 workstations, servers and PCs into the outstandingly well-working and reliable backbone of the ISI Institute that it has become. This superbly working system is the envy of every visitor to our Institute, and responsible for much of the nostalgia that most ex-ISI members find so hard to shake off, once they encounter their new computer-based working place…

Another reason for the uniqueness of this Annual Report is that it documents the arrival of the new Professor for Information Theory, Professor Amos Lapidoth, who replaces Professor Jim Massey, (who retired in March 1998, see 1998 Annual Report). Dr. Lapidoth, who was previously at MIT, joined ISI in June of 1999, and is already thoroughly immersed in the teaching and research activities in the field of information theory at our Institute. He will undoubtedly continue the tradition of excellence that his predecessor began; we wish him much success and satisfaction in doing so.

Related to the changeover in the information-theory professorship, most of the remaining doctoral students of Prof. Massey completed their thesis and PhD requirements in the last year. These are Zsolt Kukorelly, Richard de Moliner, and Jossy Sayir. Dr. Kukorelly has begun a post-doc sojourn in the USA at the University of California, San Diego, Dr. de Moliner now works with Digital Copyright Technologies in Zurich as Head of Product Development, and Dr. Sayir is at the Technical University in Vienna. Doctoral students from the Signal-Processing group at ISI, who completed their thesis were Markus Helfenstein who has began work at Globespan in New Jersey, Martin Haenggi, who spent a post-doctoral year at Berkley University with Prof. Leon Chua, and Jürg Stettbacher, who is working freelance as a consultant in Zurich. Other ISI members to leave the Institute were Stefano Quarenghi, who had joined us with the intention of leaving within the year, and now has a position at Mandozzi in Ponte Capriasca (near Lugano), as well as Felix Frey and Patrick Schweizer, both of whom were permanent members of our electronic laboratory. Mr. Frey joined a small company in the field of laser plotting systems, and Mr. Schweizer wished to expand his knowledge of English and of life beyond Switzerland by spending an extended period of time in Australia. We wish all the parting ISI members all the best for their new endeavours, and hope that they will continue to have contact with us in the sense of an ex-ISI club. The new members at ISI are Stefan Moser, an ongoing

doctoral student in the field of information theory, and Ermanno Schinca, who will stay with us for the limited time he has before continuing his studies at the HSG in St. Gallen.

As usual we had a number of visitors from all over the world who invariably enrich the intellectual climate at the institute and collaborate in various phases of the ongoing research (see section 6.1).

On a personal note, this annual report will also be my last, since by the time of the next one, I will have retired from the ETH. It will have been over 29 years (including the first two in which I was a visitor) that I have spent at this Institute, which, when I joined it, was the Institute for Telecommunications (Fernmelde-technik). They have been wonderful years; this is due, in large part, to the outstanding graduate students and Institute members with whom I have had the privilege to work. It is also due to the very pleasant and cooperative attitude of my colleagues in the EE department and of those administrative persons of the ETH with whom I had personal contact. I shall remember and cherish these years for which I am very grateful, and hope, in the new challenges which, good health permitting, I hope to undertake, to be sustained and fortified by the fond memories that I have of this important period in my life.

April 2000   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. Amos Lapidoth
since 1.6.99

Professor for Information Technology:

Prof. Dr. Fritz Eggimann

Adjunct Lecturer:	**Dr. K. Heutschi**	
Secretaries:	**Mrs. Bernadette Röösli**	
	Mrs. Renate Agotai	
	Mrs. Heidi Schenkel	
Administr.Supervisor:	**Dr. Markus Helfenstein**	left on 31.12.99
Technical Supervisor	**Dr. Max Dünki**	

Teaching Assistants:	**Dieter Arnold**	Dipl.El.Eng.	
	Qun Gao	Dipl.El.Eng.	
	Markus Hofbauer	Dipl.El.Eng.	
	Zsolt Kukorelly	Dipl.Math.	left on 31.10.99
	Heinz Mathis	Dipl.El.Eng.	
	Ralf Kretzschmar	Dipl.Phys.	
Research Assistants:	**Richard De Moliner**	Dipl.El.Eng.	left on 31.1.99
	Markus Erne	Dipl.El.Eng.	
	Marcel Joho	Dipl.El.Eng.	
	Martin Hänggi	Dipl.El.Eng.	left on 30.9.99
	Dani Lippuner	Dipl.El.Eng.	
	Hans-Andrea Löliger	Dr.	
	Felix Lustenberger	Dipl.El.Eng.	
	Moser Stefan	Dipl.El.Eng.	since 19.10.99
	Quarenghi Stefano	Dipl.El.Eng.	1.4. – 31.12.99
	Jossy Sayir	Dipl.El.Eng.	left on 31.3.99
	Ermano Schinca	Dipl.El.Eng.	since 25.11.99
	Hanspeter Schmid	Dipl.El.Eng.	
	Jürg Stettbacher	Dipl.El.Eng.	left on 31.7.99
	Felix Tarköy	Dr.	left on 30.4.99
	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   left on 30.9.99
                         Thomas Schaerer
                         Patrick Schweizer   El.Eng.HTL   left on 31.7.99

Fellowship Recipients: (see 6.1 for report of activities)

A. Jurisic   University of Zagreb, Zagreb

Academic Guests: (see 6.1 for report of activities)

Tom Sennhauser	GlobeSpan Semiconductor Inc., New Jersey, USA	25.01.-27.01.99
Prof. Chris Toumazou	Imperial College London	03.05.-04.05.99
Prof. Hari Reddy	California State University, Long Beach, USA	28.06.-09.07.99
Prof. Y. Zeevi	Technion – Israel Institute of Technology, Haifa Israel	28.06.-04.07.99
Prof. Allen Lindgren	Univrsity of Rhode Island, Kingston, USA	01.07.-15.09.99
Aaron S. Cohen	MIT, Cambridge, USA	03.10.-31.12.99
Ibrahim Abou Faycal	MIT, Cambridge, USA	03.10.-31.12.99

2. Teaching

2.1 Lectures and Practica

Sem.	Instructors	Title	ETH-No.
5th	Dr. Loeliger	Zeitdiskrete Systeme & stochastische Signale	35-405
6th	Prof. Moschytz	Digitale Signalverarbeitung und Filterung	35-416
5/7th	Dr. Mittelholzer	Applied Digital Information Theory I	35-417
6th	Dr. Kraemer	Applied Digital Information Theory II	35-418
7th	Prof. Moschytz Prof. Eggimann	Adaptive Filter & neuronale Netzwerke	35-467
8th	Prof. Moschytz	Analoge Signalverarbeitung und Filterung	35-468
7th	Dr. Heutschi	Acoustics I	35-477
8th	Dr. Heutschi	Acoustics II	35-478
5/ 6th	Prof. Moschytz et al.	Laboratory for "Fundamentals in Electrical Engineering"	35-095/6
	Prof. Moschytz 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"	39-797
	Dr. Heutschi	Acoustics Colloquium	35-950

2.2 Semester Projects and Diploma Theses

During the winter semester 1998/99 and summer semester 1999, 10 Semester Projects (16 candidates) and 8 Diploma Theses (10 candidates) were carried out.

Candidates	Title	Supervisor

Semester Projects WS 98/99 (7th Semester)

Candidates	Title	Supervisor
Nussbaumer Marc Bachofner Patrick	Stabilisation einer Verstärkungsanlage für einen Hörsaal mittels adaptivem Equalizer und adaptivem Echokompensator	von Hoff Lippuner
Kaufmann Anton Regenscheit Simon	Implementation eines adaptiven Echokompensators	Lippuner von Hoff
Koster Alex	Sprechertrennung bei Videokonferenzen	Joho Mathis
Bühler Gerd Arpagaus Beat	Multitag-Separation für ein Hoch-frequenz-Identifikationssystem	Mathis Elektrobit
Curty Jari-Pasqual	Filterchip in SI-Technik	Dr. Helfenstein
Hofman Jan	Genetische Synthese von CMOS-Schaltungen	Schmid
Hösli Daniel Lanzetta Gennaro	Konvergenzverhalten des Summen-Produkt-Algorithmus bei turboartigen Codes	Arnold Lustenberger Endora Tech
Kalberer Gregor Spielmann Beat	Switchable Electronic Hybrid for Duplex High-speed Data Transmission II	Prof. Moschytz Lim, Dr. Muralt/ (Globespan)

Semester Projects SS 99 (8th Semester)

Candidates	Title	Supervisor
Roccioletti Ronald	Unifilare Turbo Codes	Arnold Lustenberger
Blunschi Reto Hauser Gabriel	Watermarking von Audiosignalen	Erne

Diploma Theses WS 98/99

Candidates	Title	Supervisor
Quarenghi Stefano	Genetische Algorithmen für Adaptive Filter	Lippuner Hänggi
Moser Stefan	JAVA basierter Simulator für iterativen Decoder	Lustenberger Arnold Endora Tech

Fromherz Georg Schinca Ermano	Interne Dynamik eines iterativen Decoders	Arnold Lustenberger Endora Tech
Stöckli Michael	Physical Modeling of Trombone and Player	Prof. Sullivan Schmid
Handlery Marc	Tailbiting BCJR Decoder with unbalanced Source Probabilities	Profos Anderson and Johannesson Dr. Mittelholzer

Diploma Theses SS 99

Berchtold Andreas	Optimale Bit-Allocation für einen Wavelet-basierenden Audiocoder	Erne
Faller Christof	Audio compression for broadcasting applications	Kroon/Lucent von Hoff
Gehring Alexander Huber Marcel	Regelung fuer einen Rauchmelder	Hofbauer Dr. Tuillard

Post-Diploma Thesis

Lippuner Daniel	Noice-Covariance Estimation in Model-Based Adaptive Algorithms	Moschytz
Joho Marcel	Connecting Partitioned Frequency-Domain Filters in Parallel or in Cascade	Moschytz
von Hoff Thomas	Transpose Properties in the Stability of the Classic Adaptive Algorithms for Blind Source Separation and Deconvolution	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:

- Watermarking for data protection
- Turbo codes and Low Density Parity Check codes
- Reduced-complexity receivers for InterSymbol Interference channels
- Magnetic recording

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

Decomposition of long-term intramuscular EMG signals using Wavelets

To analyse the causes of chronic muscle pain, intramuscular long-term measurements 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). The repetitive activation of several individual MUs results in a superposition of pulse trains, which constitute the EMG signal. In contrast to short-term recordings, the number of active MUs and the MUAP shapes change during long-term measurements. Therefore, the main goals of a decomposition algorithm are: the evaluation of the number of active MUs at any time, the determination of MUAP shapes, the detection of MUAP shape changes, and the complete decomposition of overlapping MUAPs.

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

Switched-Current Biquad Using Differential, Double-Sampling, Forward/Backward Difference Integrators

As proposed in many textbooks, the synthesis of sampled-data switched-current (SI) filters can be performed on the basis of simple analogies with continuous-time networks. Although this approach is appealing, it requires approximations which are valid only when the ratio between the sampling and the signal frequency is large, which in turn results in a large die area and increased power consumption. On the other hand, an exact design is possible when the synthesis of the discrete-time network is carried out in terms of discrete-time transfer functions. The former approach is often based on bilinearly transformed integrators, while the latter needs forward or backward difference integrators. However, due to the lack of double-sampling (DS) forward/backward difference integrators, exact designs were so far restricted to implementations without DS capabilities, thereby neglecting possible speed, power and SNR trade-offs.

This work introduces a new switched-current integrator which accomplishes forward and backward difference integration with double-sampling capabilities. This integrator promises significantly improved performance. Firstly, a single-ended version of a forward and backward DS integrator was derived. Subsequently, these integrators were merged into one differential DS integrator, which was then applied to the design of a low-sensitivity biquad. Transistor-level simulations demonstrated the potential of these integrators. It was shown that the pole sensitivity of the biquad is reduced by a factor of 2 and the coefficient spread by a factor of 3 when compared with a bilinearly transformed reference filter.

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 most difficult to build. Switched-capacitor and switched-current filters can be very precise, but the bandwith of time-continuous 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 coming 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 unprecise, 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 17-MHz lowpass filter with tunable pole frequency and two current amplifiers suitable for building 20-MHz filters, one with fixed, the other

with 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 phase of the project is the integration of a tunable 6th-order filter cascade, of a filter with both tunable pole Q and pole frequency, and of a charge-pump-controlled biquad with higher dynamic range.

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Supported by: ETH

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

Design of Analog VLSI Iterative Decoders (DAVID)

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

The main challenge of this project was to identify suitable computational primitives (elementary circuits) on the transistor level. This first goal was achieved in the first year of the project: a 'natural' mapping of the sum-product algorithm onto transistor circuits was found that applies, in particular, to turbo codes, to conventional trellis codes, and to low-density parity check codes. These circuits reveal an interesting connection between semiconductor physics and probability theory. The proof of concept was established by building a demonstration unit for a small binary trellis code using discrete transistors. Swiss and international patent applications have been filed.

To demonstrate the advantages of the new decoding approach, a first test chip for a binary (18,9,5) tailbiting trellis code was designed and fabricated in AMS 0.8um BiCMOS technology. Simulation results show the chip's robustness with regard to non-idealities such as transistor mismatch, finite output resistance of MOS transistors, and temperature effects. Furthermore, measurement results show that bit rates of over 100MBit/s can be achieved with a single 5V power supply and a power consumption of 50mW.

In a next step towards a full-sized decoding system, a test-chip for a more complex turbo-style code with digital interfaces was designed using the AMS 0.8um BiCMOS technology. Our decoders can be designed by construction using a C program to convert the parity-check matrix represenation of the code into a Verilog structural description. Subsequently, this file can be imported into the Cadence IC design environment, and digital place-and-route tools can be used to

generate the final layout of the decoder. This chip is now in fabrication and measurement results will be available shortly.

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

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

Fingerprint Recognition Using Cellular Neural Networks

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

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

Up to the present, it has been shown that CNNs are very well suited for image processing, pattern recognition and generation, nonlinear signal processing in general, and the solving of partial differential equations. In an extension of such tasks, the goal of this project is to develop robust CNN algorithms for fingerprint recognition.

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Keywords: fingerprint recognition, cellular neural networks

Fast Algorithms for Adaptive Beamforming

In acoustical applications related to hearing aids or teleconferencing, the signals received at a single microphone usually contain a mixture of several sound sources. Removing the disturbing noise sources is a difficult task, when only one microphone is used to pick up the sound, especially when the source signals overlap in their spectra. As the disturbing noise sources 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 the case of strong reverberation caused by acoustical reflections, standard beamforming techniques fail to work properly, because the assumption that each sound source impinges on to the microphone array from a single direction, is violated.

Blind source separation algorithms have demonstratet the capability of solving the multi-path problem in a simulation environment; they are therefore very promising for their use in real acoustical applications. Blind algorithms make only weak assumptions with regard to 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 converging behavior at the same time as 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. Various methods have been proposed on how to incorporate this additional information into a blind algorithm leading to a semi-blind algorithm.

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Supported by: ETH

Keywords: adaptive beamforming, blind source separation.

Blind Separation and Equalization for Wireless Communications

In wireless environments, channels often are frequency selective, making equalization mandatory. Equalization can be carried out either using training-sequence-based, decision-directed, or blind algorithms. The combination of both blind equalization of the transmission channels and blind separation of the sources, also referred to as multichannel blind deconvolution (MCBD) poses a particularly difficult problem. Besides, in mobile radio channels, the channel impulse response, and hence the mixing process of the sources, may change rapidly. Suitable equalization algorithms must therefore show fast convergence and good tracking capabilities.

We have investigated different algorithms suitable for blind separation of typical communication signals (PAM and QAM) and compared them in performance and complexity. A new, simple threshold nonlinearity has been derived and is shown to separate any mixture of sub-Gaussian signals. If an adaptive threshold is introduced, the proposed algorithm can even separate mixed-kurtosis signals.

Under the most general setup of sources and mixtures, higher-order statistics, either explicitly or implicitly, are required to solve the blind separation problem. Necessary conditions for separation are non-vanishing fourth-order cumulants. An extensive study of fourth-order cumulants of digitally modulated signals has revealed some insight into the process of blind signal separation and helped classify the suitability of different modulation formats for both blind separation and equalization.

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Keywords: blind source separation, blind equalization

Signal adaptive Audio Coding using Wavelets and Rate Optimization

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

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

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

Group 2: Adaptive Systems

Group Leader: Prof. Dr. A. Kälin

Adaptive Subband Signal Processing for Hearing Instruments

People with recruitment of loudness have a compressed dynamic range between the sound-pressure levels corresponding to threshold and discomfort. To compensate for this frequency-dependent compressed dynamic range, a nonlinear and time-varying filter has to be used. Furthermore, an echo canceler is needed to enable large gains in the hearing-loss compensating filter.

In realizing these signal processing tasks we have analyzed different computationally efficient adaptive subband schemes, especially the augmented modulated lapped transform (AMLT) and the DFT with the overlap-save technique.

The echo canceler is adapted using the available speech input signal only.

The concatenation of the nonlinear hearing-loss compensator and the adaptive echo canceler causes distortions in the loudspeaker signal. To reduce the distortions we developed a novel step-size control for the echo canceler and a closed-loop gain measure to supervise and control the stability of the system. Furthermore, a efficient prototype filter structure is used for the hearing-loss compensator.

The proposed algorithms were implemented and tested on a real-time DSP system using a dummy behind-the-ear hearing aid and a manikin. The test showed an increase of the hearing-loss gain of more than 20dB compared to systems without echo canceler.

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In Collaboration with: Phonak AG

Keywords: Recruitment of Loudness, Adaptive Feedback Cancellation, Multiband Compression, Subband Signal Processing, Digital Hearing Instrument

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

Over the last years the start-up company mechatronica has developed 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 using a floating 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 indicates the filling level. At very high temperatures (120 degrees C max.) the estimation of the

distance becomes very inaccurate because of the large temperature gradient along the tube causing a high variation in sound propagation speed, and the large sound attenuation.

Within the scope of this project, the mentioned problems have been investigated and suggestions for solutions have been 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 sound frequency used had to be reduced. Since most of the ultrasonic transducers are narrow-band devices, a new device operating on the desired frequency was developed by mechatronica. The impedance and the transient behavior of this ultrasonic transducer were investigated at our laboratory. Since the transducer serves as transmitter as well as receiver, the transient behavior is of great interest. However, it was stated that the impulse response was so long that the received echo was covered by the transient and could not be detected. This led to the conclusion that in a following project, an appropriate transducer would have to be found first. In a second step, an estimation algorithm for this new transducer needs to be developed. Even with noisy or distorted echos, such an algorithm should allow to make a precise and reliable estimation of the propagation time.

This interdisciplinary project with its complex problems required 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 providing the mechanical system to perform measurements.

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In Collaboration with: Mechatronica, TU Linz

Keywords: level gauge, ultrasonic, distance estimation

Convergence of Blind Algorithms

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

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

process introduces a temporal dependency in the sensor signals (Multichannel Blind Deconvolution (MCBD)).

Generally, blind algorithms are too slow to be able to track a changing mixing environment, particularly in the MCBD case. In order to improve the tracking behavior, a self-adjusting step size is proposed. The control mechanism increases the step size in case of a changing mixing environment such that the demixing system can track the new environment. As soon as the mixing system is invariant, the step size decreases towards zero enabling the demixing system to converge to the optimal solution.

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Professor: A. Kälin
Supported by: KTI (Noise reduction for a car mobile phone)
In Collaboration with: Siemens Schweiz AG

Keywords: blind source separation, multichannel blind deconvolution, step-size control

Adaptive Filters for Nonstationary Environments

Most adaptation algorithms converge rapidly 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. The estimation-error impairment may have two main reasons: either due to changes of the system parameters or due to the statistics of the measurement noise. It is important to distinguish between these two influences, because only in the former case the adaptive filter shall be readjusted.

Based on the Kalman filter theory several concepts for an improvement of the identification problem are being investigated. A first approach consists of the direct estimation of the process-noise and measurement-noise variances. In a second approach, the identification error is determined by observing the propagation of the own adaptive filter rather than the usually applied input-output signal correlation. Further approaches contain nonlinear Bayesian estimators resulting from a non-Gaussian model of the involved disturbances.

All algorithms are being designed to possibly adapt thousands of FIR filter coeffcents and to operate on a digital signal processor (DSP). It is intended to verify the performance of these algorithms in the application of acoustic echo cancellation.

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Professor: A. Kälin
Supported by: ETH

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

Group 3: Information Technology

Group Leader: Prof. Dr. F. Eggimann

Prediction of local winds with neural networks

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

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

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Professor: F. Eggimann
In Collaboration with: Dr. N.B. Karayiannis, The University of Houston, Houston, Texas, USA

Keywords: nonlinear signal processing, neural networks, wind prediction

Section 2:   Digital Information Theory

Section Leader: Prof. Dr. Amos Lapidoth

Channel Models and Codes for High Density Magnetic Recording

While communicators are concerned with maximizing the data rate (in bits per second) whereby digital information can be transmitted reliably from here to there, storage researchers are mainly concerned with maximizing the area density (in bits per square inch) for storing and retrieving reliably information from now to then. Hence, the magnetic recording channel possesses several characteristics that differentiate it from ordinary communication links. Especially at areal densities foreseen for the next generations of recording devices, the channel is expected to be highly nonlinear, the noise to be signal-dependent with the media noise overpowering the electronics noise by a ratio 9:1. The major sources of noise in high density magnetic recording are pulse jitter and partial signal erasure, which is a consequence of magnetization percolation of high linear densities. Error correcting codes designed for the high density magnetic recording cannel must therefore be resilient to these two noise manifestations.

As information theory provides the theoretical underpinnings for digital communications, it also serves as a foundation for understanding the fundamental

limits on reliable digital information recording as measured in terms of data rate and storage density. Based on a thorough understanding of these limits, the goal of this work is to create a new channel model that captures the essences of high density magnetic recording. In order to find good error correcting codes, the capacity of this new channel model has first to be investigated. Afterwards, new error correcting codes have to be found and their performance in terms of error correting capability must be assessed.

Algebraic Coding and Iterative Decoding

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

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

The goal is to build a bridge between algebraic and turbo codes: finding factor graph representations of known codes and designing new codes with focus on block lengths smaller than 1000 symbols. The construction of these codes will not only be based on ideas from algebra but also be in the spirit of Tanner's transform theory. We want to explore graph properties like cycles and connectivity; their impact on code parameters and especially on decoding performance are not well understood until now.

In the report period we designed a [44,22,8] binary linear block code for use in an analog-VLSI decoder; the actual hardware implementation was done by Felix Lustenberger. This code belongs to the category of low-density parity-check codes, i.e. it has a parity-check matrix which is sparsely filled with ones.

3.3 Completed Research Projects

DE MOLINER Richard

On the Statistical Testing of Block Ciphers

ETH-Diss. Nr. 13106 (Referee: Prof. Dr. J.L. Massey)

Tests that are capable of analyzing any practical block cipher, no matter what the internal structure of the block cipher may be, are the subject of this work. It is argued that such tests must be statistical.

A discrete memoryless source producing a fixed-length sequence of output digits from a finite alphabet is considered. The problem of deciding whether the single letter probability distribution of the discrete memoryless source is equal to a given probability distribution or not is analyzed in detail. For this problem of statistical hypothesis testing the Pearson statistic is used. What can validly be concluded from statistical hypothesis testing is carefully considered.

We show that if a cryptanalyst cannot solve at least one of two basic problems for a given block cipher, then he cannot "break" this block cipher. These two basic problems are (1) to find an algorithm that is *distinguishing* for the given block cipher and (2) to find an algorithm that is *key-subspace distinguishing* for the given block cipher and for a given decomposition of the key space.

An approach to finding an algorithm that is distinguishing for a given block cipher as well as an approach to finding an algorithm that is key-subspace distinguishing for a given block cipher and for a given decomposition of the key space are described. These two approaches form the framework for the statistical testing of block ciphers.

A family of tests called bit-dependency tests is presented. The aim of a bit-dependency test is to say as much as possible about the quality of a block cipher when only a given subset of bits of the plaintext blocks and a given subset of bits of the corresponding ciphertext blocks are observed.

Keywords: cryptography, cryptanalysis, block ciphers, bit-dependency tests, statistical hypothesis testing, statistical tests, Pearson statistic.

KUKORELLY Zsolt

On the Validity of Certain Hypotheses used in Linear Cryptanalysis

ETH-Diss. Nr. 13076 (Referee: Prof. Dr. J.L. Massey)

Linear cryptanalysis and its generalisations are possible ways to attack an iterated block cipher. Their success relies on a certain number of assumptions made by the attacker. In this thesis, the validity of some of these assumptions is investigated.

According to Matsui's Piling-up Lemma, the imbalance of a sum of independent binary random variables is equal to the product of the imbalances of these random

variables. One uses this fact in linear cryptanalysis to compute a lower bound on the probability of success of one's attack. It is shown that, on average, the imbalance of the sum is at least as large as the product of the imbalances and that for large sample spaces, both expressions are almost always approximately equal. It is deduced that, at least as an approximation, the Piling-up Lemma is applicable in a linear cryptanalysis attack to linked threefold sums even if they are not independent.

The validity of the hypothesis of fixed-key equivalence is investigated. The hypothesis asserts that for any effective input/output sum (I/O sum) virtually all key-dependent imbalances are approximately equal to their average, the average-key imbalance of the I/O sum. A counter-example is given. It is further proved that, for one round of encryption, the average and the variance of the key-dependent imbalances are approximately the same for virtually all I/O sums. Whether the key-dependent imbalances of an I/O sum can then be considered as "approximately equal" is subjective and therefore no conclusion about it is drawn. Finally, the average, over all I/O sums, of the average-key imbalances is computed for any number of rounds. Based on this result, a new quantitative definition of effective I/O sums is given.

The validity of the piling-up hypothesis is studied. This hypothesis is an m-ary analogue to Matsui's Piling-up Lemma. It says that (for certain imbalance measures) the imbalance of a product of independent m-ary random variables is in virtually all cases approximately equal to the product of the imbalances of these random variables. The family of all imbalance measures that are convex-$\cup$ on the set of m-ary probability distributions, equal to 1 for a constant random variable and equal to 0 for a uniformly distributed random variable, is considered. It is argued that they are all equally appropriate for measuring the goodness of an expression used in the group generalisation of linear cryptanalysis attack. For the measure I_2^2 which belongs to this family, it is shown that the imbalance of a product of two random variables is on average equal to the product of the imbalance s of the two random variables. It is inferred that the piling-up hypothesis holds for two random variables when m is large enough. By induction, it is shown that the hypothesis also holds for any number of random variables when m-ary is large enough. Finally, it is argued that I_2^2 is an appropriate imbalance measure to use in the group generalisation of linear cryptanalysis.

Keywords:Iterated block cipher, linear cryptanalysis, imbalance, linebreak Piling-up Lemma, hypothesis of fixed-key equivalence, piling-up hypothesis.

SAYIR Jossy

On Coding by Probability Transformation

ETH-Diss. Nr. 13099 (Referee: Prof. Dr. J.L. Massey)

An introduction to arithmetic source coding is given, showing how this technique transforms the output probability distribution of the source into an almost uniform probability distribution. A similar approach is investigated for block coding for

noisy channels, leading to the definition of the (N, K) block coding capacity of a discrete memoryless channel. Arithmetic coding is modified for data transmission over noisy channels and a metric is developed for a sequential decoder to be used in conjunction with an arithmetic encoder.

Universal arithmetic source coding is investigated for a class of sources whose output distributions lie within a polytope of probability distributions. The properties of the optimal coding distribution over a polytope of distributions are derived. Gallager's redundancy-capacity theorem is presented. An iterated version of the Arimoto-Blahut algorithm is fomulated to compute the optimal coding distribution for a polytope with many vertices, or, alternatively, to compute the capacity of a discrete memoryless channel with a large input alphabet. The optimal coding distribution is computed for the polytope of all monotone non-increasing probability distributions with a given expected value.

Universal source coding with a source transformation is described. Two source transformers are investigated: the recency-rank calculator (also called the move-to-front list) and the competitive list transformer. It is shown that the steady-state output distribution of these transformers is monotone non-increasing when the transformers are applied to the output of a discrete memoryless source. A context-tree algorithm is formulated which uses competitive list transformers followed by a universal arithmetic source encoder. The performance of this algorithm is compared to the performance of other universal source coding algorithms.

LIM Drahoslav

Implementation of a Programmable, Modularly Extendable Cellular Neural Network Signal Processor

ETH-Diss. Nr. 13219 (Referee: Prof. Dr. G.S. Moschytz)

A CNN is a non-linear array of locally interconnected analog processors (cells) which operate in parallel. In this thesis we present an architecture suitable for implementing a programmable general purpose cellular neural network (CNN) processor.

The architecture deals with some of the characteristics and restrictions inherent in CMOS VLSI technologies, and allows a multi-chip, mixed analog-digital hardware realization of a large, continuous-time CNN, with step-wise programmable template values, and a piece-wise linear output function. It is a direct implementation of the network described used in the theoretical literature. The input data is not inherently restricted to bipolar images.

In the first Chapter we give a brief outline of the CNN structure and its mathematical description and examples of a few simple CNN processing tasks.

Chapter 2 describes the design goals of this thesis, defines what is to be built, and translates general system specifications into more detailed, circuit-oriented specifications, on whose basis we can proceed with the design. In Chapter 3 we present the general architecture of the blocks which form the CNN processor and the reasoning and tradeoffs involved in choosing the configuration of each block.

Chapters 4 and 5 present the design of the analog and digital portions of the chip, respectively, together with related layout issues.

The chip was manufactured and samples were available from several wafers from two different manufacturing runs. The functions and performance of the chip is verified by static measurements and matching results in Chapter 6. The operation of the CNN processor chip is demonstrated in Chapter 7.

Finally, in Chapter 8 we give a summary of the work done, what was achieved, and what future work could be carried out.

KEYWORDS: cellular neural networks, artificial neural networks, programmable analog VLSI, mixed analog-digital design, master-slave tuning, auto-zero, offset nulling, image processing, 2-D signal processing, signal classification, pattern recognition.

HAENGGI Martin

Analysis, Design, and Optimization of Cellular Neural Networks

ETH-Diss. Nr. 13225 (Referee: Prof. Dr. G.S. Moschytz)

Cellular neural networks (CNNs) constitute a class of recurrent and locally coupled arrays of identical dynamical systems (cells). The underlying equation governing the dynamics of each cell is nonlinear 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.

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. By far not all mathematically possible CNN tasks can be carried out reliably on an analog chip, some of them are inherently too sensitive.

We define a robustness measure to quantify the degree of robustness and propose an exact and direct analytical design method for the synthesis of optimally robust network parameters. This method is restricted to the class of so-called locally regular templates, which is rigorously defined. It turns out that the complementary class, the locally irregular templates, constitute precisely the class of inherently sensitive templates, which makes the synthesis method generally applicable to all tasks that allow robust operation.

Processing speed is always crucial when discussing signal processing devices. In the case of the CNN, it is shown that the settling time of locally regular templates can be specified in closed analytical expressions, which permits, on the one hand, template optimization with respect to speed and, on the other hand, efficient numerical integration of CNNs. Interdependence between robustness and speed issues are also addressed

Another goal pursued is the unification of the theory of continuous-time and discrete-time CNNs. By means of a delta-operator approach, it is proven that basically the same templates can be used for both of these classes, even if their nonlinear output functions differ. However, not all tasks are feasible on all types of implementation. The common subset of templates that run on virtually any CNN chip built so far is, once again, the set of locally regular templates --- we conclude that local regularity in fact is a key concept in chip-oriented CNN theory.

More complex CNN optimization problems that cannot be solved analytically necessitate resorting to numerical methods. Among these, stochastic optimization techniques such as genetic algorithms prove their usefulness, for example in image classification problems.

3.1 Completed Dissertations

DE MOLINER Richard   On the Statistical Testing of Block Ciphers

> *ETH-Diss. Nr. 13106*
> Referee: Prof. emer. Dr. James L. Massey
> Co-referee: Dr. M. Dichtl, Siemens AG, München
>   Prof. Dr. U. Maurer

KUKORELLY Zsolt   On the Validity of Certain Hypotheses Used in Linear Cryptanalysis

> *ETH-Diss. Nr. 13076*
> Referee: Prof. emer. Dr. James L. Massey
> Co-referee: Prof. Dr. U. Maurer

SAYIR Jossy   On Coding by Probability Transformation

> *ETH-Diss. Nr. 13099*
> Referee: Prof. emer. Dr. James L. Massey
> Co-referee: Prof. Dr. F. Willems, Techn. Universität,
>   Eindhoven
>   Prof. Dr. F. Eggimann

LIM Drahoslav   Implementation of a Programmable, Modularly Extendable Cellular-Neural-N etwork Signal Processor

> *ETH-Diss. Nr. 13219*
> Referee: Prof. Dr. G.S. Moschytz
> Co-referee: Prof. Dr. Q. Huang

HAENGGI Martin   Analysis, Design, and Optimization of Cellular Neural Networks

> *ETH-Diss. Nr. 13225*
> Referee: Prof. Dr. G.S. Moschytz
> Co-referee: Prof. Dr. L. Chua, University of
>   California, Berkeley

13.01   Internal Reports

9901   Lippuner Daniel   Noice-Covariance Estimation in Model-Based Adaptive Algorithms

9902   von Hoff Thomas   Optimierung eines Füllstandsmesssystems
         Frey Felix   basierend auf Ultraschalltechnik

| 9903 | Joho Marcel | Connecting Partitioned Frequency-Domain Filters in Parallel or in Cascade |
| 9904 | von Hoff Thomas | Transpose Properties in the Stability of the Classic Adaptive Algorithms for Blind Source Separation and Deconvolution |

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).

President of the IEEE Circuits and Systems Society.

Organization of fourth ETHZ-EPFL Summer School on Identification of Dynamical Systems together with Prof. M. Hasler, CIRC EPFL and Prof. L.O. Chua, UC Berkeley.

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

Prof. Lapidoth

Organizer and Session Chairman for the session on Network Shannon Information Theory, 1999 IEEE Information Theory and Networking Workshop, Metsovo, Greece.

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 16.3.-29.3.99 and 12.7.-19.8.99.
Moschytz George S.	ISCAS'99, International Symposium on Circuits and Systems, Orlando, USA, 30.5.-2.6.99.
Moschytz George S.	IEEE CAS-COM Workshop'99, Princeton University, Princeton, NJ, 26.-28.7.99.
Moschytz George S.	41st Midwest Symposium on Circuits and Systems, Notre Dame, IN, USA, 9.-12.8.99.
Moschytz George S.	ECCTD'99, European Conference on Circuit Theory and Design, Stresa, Italy, 30.8.-2.9.99.
Moschytz George S.	ICECS'99, IEEE International Conference on Electronics, Circuits and Systems, Paphos, Cyprus, 5.-9.9.99.
Moschytz George S.	Procid-Meeting, Copenhagen, Denmark, 25.-27.11.99.
Moschytz George S.	Visit Imperial College, London, 28.11.-5.12.99.
Erne Markus	106-th AES-Convention, Munich, Germany, 8.-12.5.99.
Erne Markus	ICASSP 99, International Conference on Acoustics, Speech and Signal Processing, Phoenix, USA, 15.-19.5.99.
Erne Markus	48-th ISO-MPEG-meeting on MPEG4, Vancouver, Canada, 12.-16.7.99.
Erne Markus	AES 17-th International Conference on High Quality Audio Coding, Florence, Italy, 2.–5.9.99.
Erne Markus	107th AES-Convention, New York, USA, 24.–27.9.99.
Erne Markus	50-th ISO-MPEG-meeting on MPEG4, Maui, USA, 6.-10.12.99.
Gao Qun Hänggi Martin Schmid Hanspeter	ECCTD'99, Europen Conference on Circuit Theory and Design, Stresa, Italy, 29.8.-2.9.99.
Hänggi Martin Helfenstein Markus Lustenberger Felix Mathis Heinz Schmid Hanspeter	ISCAS'99, International Conference on Circuits and Systems, Orlando, Florida, USA, 30.5.-2.6.99.
Hänggi Martin Wellig Peter	ICECS'99, International Conference on Electronics, Circuits and Systems, Paphos, Cyprus, 5.-8.9.99.
Helfenstein Markus	MEDEA, Projekct Review Meeting, Bruxelles, Belgium, 26.-27.4.99.

Helfenstein Markus	Professional Development Conference, Dallas, USA, 4.-6.9.999.
Helfenstein Markus Schmid Hanspeter	International Workshop on Low Power RF Integrated Circuits, Lausanne, Switzerland, 19.-20.10.99.
Hofbauer Markus Mathis Heinz	ETHZ-EPFL Summer School, Lausanne, Switzerland, 5.-9.7.99.
Joho Marcel	International Conference on Independent Component Analysis and Blind Source Separation, Aussois, France, 11.-15.1.99.
Joho Marcel	ITG Fachgruppensitzung, ETH Zuerich, Switzerland, 5.3.99.
Kretzschmar Ralf	Research-Colloquium SMA-Meteo Schweiz, Zuerich, Switzerland, 11.5.99.
Kretzschmar Ralf	MAP'99, Mesoscale Alpine Programme Meeting, Appenzell, Switzerland, 9.-11.6.99.
Kretzschmar Ralf	Research stay at the University of Houston, Texas, USA, 9.8.-10.9.99.
Kretzschmar Ralf	179. SANW Jahresversammlung, Schweizerische Akademie der Naturwissenschaften, Lucerne, Switzerland, 13.-16.10.99.
Lustenberger Felix	AACD Workshop, Nice, France, 23.-25.3.99.
Lustenberger Felix	ESSCIRC'99, Duisburg, Germany, 21.-24.9.99.
Mathis Heinz	ICA'99, Independent Component Analysis and Blind Signal Separation, Aussois, France, 11.-15.1.99.
Schmid Hanspeter	Workshop on Advances in Analog Circuit Design, Opio, France, 23.-25.3.99.
Wellig Peter	21th Annual International conference on the IEEE Engineering in Medicine and Biology Society, Atlanta, USA, 13.-16.10.99.
Wellig Peter	Procid Meeting, Copenhagen, Denmark, 4.-5.2.99.
Wellig Peter	Procid Meeting, Stresa, Italy, 6.-8.5.99.
Wellig Peter	Procid Meeting, Göteborg, Sweden, 3.-4.9.99.
Wellig Peter	Symposium Meeting, Copenhagen, Denmark, 25.-27.11.99.

Group: Adaptive Systems

Lippuner Daniel von Hoff Thomas	ETHZ-EPFL Summer School, Lausanne, Switzerland, 5.-9.7.99
Lippuner Daniel von Hoff Thomas	ECCTD'99, Europen Conference on Circuit Theory and Design, Stresa, Italy, 29.8.-2.9.99.

Lippuner Daniel   IWAENC'99, International Workshop on Acoustic Echo
von Hoff Thomas   Cancellation and Noise Control, Pocono Manor, USA,
                           27.-30.9.99.

Von Hoff Thomas   ICA'99, Independent Component Analysis and Blind Signal
                           Separation, Aussois, France, 11.-15.1.99

Wyrsch Sigi   ICASSP 99, International Conference on Acoustics, Speech
                           and Signal Processing, Phoenix, USA, 15.-19.5.99

Wyrsch Sigi   ISCAS'99, International Conference on Circuits and Systems,
                           Orlando, Florida, USA, 30.5.-2.6.99.

Wyrsch Sigi   ICECS'99, International Conference on Electronics, Circuits
                           and Systems, Paphos, Cyprus, 5.-8.9.99.

Group: Applied Acoustics

Heutschi Kurt 2nd Convention of the European Acoustics Association,
                           Berlin, Germany, 14.-19.3.99.

Heutschi Kurt   NAFEMS Seminar Computational Acoustics, Wiesbaden,
                           Germany, 10.-11.11.99.

Group: Digital Information Theory

Lapidoth Amos   1999 IEEE Information Theory Workshop, Kruger National
                           Park, South Africa 20.-25.6.99.

Lapidoth Amos   1999 IEEE Information Theory and Networking Workshop,
                           Metsovo, Greece, 27.6.-17.99.

Lapidoth Amos   IBM Workshop on „Trends in Coding and Detection for
Arnold Dieter   Magnetic Recording", Rüschlikon, Switzerland, 11.-
                           12.11.99.

Arnold Dieter   IMA Workshop on Codes, Systems and Graphical, Institute
Vontober Pascal   for Mathematics and its Applications, University of
                           Minnesota, Minneapolis, USA, 2.-13.8.99.

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 of IEEE Circuits and Systems Society

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

Prof. Lapidoth

Member of IEEE New York

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'Ingénieurs de Fribourg, Switzerland, 6.5.99.
Erne Markus	"Perceptual and Near-Lossless Audio Coding based on a Signal-adaptive Waveletfilterbank", 106th AES-Convention, Munich, Germany, 8.5.99 – 1.5.99.
Erne Markus	"Best Wavelet-Packet Bases for Audio Coding using Perceptual and Rate-Distortion Criteria", ICASSP'99, Phoenix, Arizona, USA, 15.5.99 – 19.5.99.
Erne Markus	"Audio Coding Based on Rate-Distortion and Perceptual Optimization Techniques", 17th AES International Conference on High Quality Audio Coding, Florence, Italy, 2.9.99 – 5.9.99.
Erne Markus	"Wavelet basierte Audiocodierungsverfahren", ITG-Fachtagung, Institut für Rundfunktechnik, Munich, Germany, 26.11.99.
Hänggi Martin	"Unifying Results in CNN Theory Using Delta Operator", ISCAS'99, Orlando, Florida, USA, 1.6.99.
Hänggi Martin	Attacking the General Classification Problem with CNNs", ECCTD'99, Stresa, Italy, 1.9.99.
Hänggi Martin	"The CNN Sampling Theorem", Special Session on CNN Applications, ECCTD'99, Stresa, Italy, 2.9.99.
Hänggi Martin	"A Sampling Theorem in Numerical Integration", ICECS'99, Paphos, Cyprus, 6.9.99
Helfenstein Markus	"Decoding in analog VLSI," SOLID-STATE Chapter Foundation, ETH Zurich, Switzerland, March 1999.
Helfenstein Markus	"High-Speed Interfaces for Analogue, Iterative Decoders," Int. Symposium on Circuits and Systems, Orlando, USA, June 1999.
Helfenstein Markus	"Switched-Current Networks for Filtering and High-Speed Interface Applications," IBM, Zurich, Switzerland, July, 1999.
Helfenstein Markus	"Decoding in analog VLSI," Lucent Technologies, New Jersey, USA, Sept. 1999.
Helfenstein Markus	"Decoding in analog VLSI: Towards ADC-less Data Transmission?!," Int. Workshop on Low Power RF Integrated Circuits, EPFL Lausanne, Switzerland, Oct. 1999.
Joho Marcel	"Blinde Quellenseparation", ITG Fachgruppensitzung: Algorithmen der Signalverarbeitung, ETH Zürich, Switzerland. März 1999.

Joho Marcel	"Performance comparison of combined blind/non-blind source separation algorithms", International Conference on Independent Component Analysis and Blind Source Separation, Aussois, France, 1.12.1999.
Lippuner Daniel	"The Kalman Filter in the Context of Adaptive Signal Processing", 4th ETHZ-EPFL Summerschool, Lausanne, Switzerland, 5.7.1999.
Lippuner Daniel	"A Combined Process- and Measurement-Noise Covariance Estimation in Model-Based Adaptive FIR Filters", ECCTD'99, Stresa, Italy, 2.9.99.
Lippuner Daniel	International Workshop on Acoustic Echo Cancellation and Noise Control", IWAENC'99, Pocono Manor,Pennsylvania, USA, 29.9. 1999.
Lustenberger Felix	"Design of Analog VLSI Iterative Decoders (DAVID)", ITG Fachtagung, ETH Zuerich, Switzerland, 5.3.99.
Lustenberger Felix	"An Analog Decoding Technique for Digital Codes", ISCAS'99, Orlando, USA, 2.6.99.
Lustenberger Felix	"Design of Analog VLSI Iterative Decoders (DAVID)", Invited Talk at TU Dresden (Prof. Adolf Finger), Germany, 8.7.99.
Lustenberger Felix	"Design of Analog VLSI Iterative Decoders (DAVID)", Jahrgangstagung Diplomanden 1948, ETH Zurich, Switzerland, 10.11.99.
Mathis Heinz	An FFT-based Algorithm for Multichannel Blind Deconvolution", ISCAS'99, Orlando, USA, 1.6.99.
Mathis Heinz	"Blind Source Separation", Philips Semiconductors, Zurich, Switzerland, Switzerland,15.9.99.
Schmid Hanspeter	"A Video-Frequency, Low-Power, Single-Amplifier Biquadratic Filter in CMOS," ISCAS 1999, 30.5.-2.6.1999, Orlando, U.S.A.
Schmid Hanspeter	"Minimum-Sensitivity Single-Amplifier Biquadratic Filters," ECCTD 1999, 29.8.-2.9.1999, Stresa, Italiy.
Schmid Hanspeter	"A Continuously Adjustable Video-Frequency Current Amplifier for Filter Applications", ECCTD 1999, 29.8.-2.9.1999, Stresa, Italiy.
Schmid Hanspeter	"Single-amplifier Biquadratic MOSFET--C Video-Frequency Filters", Internarional Workshop on low-power RF integrated circuits, EPFL, Lausanne, Switzerland, 19.-20.10.1999,
Schmid Hanspeter	"Single-amplifier Biquadratic MOSFET-C Video-Frequency Filters", Colloquium of the Zurich Chapter, Solid State Circuits Society, IEEE, held in Zurich, Switzerland, 4.11.1999.
Wellig Peter	"Biomedical Signal Processing: Decomposition of Electromyograms", ITG Fachgruppensitzung, Zurich, Switzerland, 5.3.99.
Wellig Peter	"Decomposition of EMG Long-Term Recordings", ExCom Member Meeting, Zurich, Switzerland, 8.3.99.

Wellig Peter	"Wavelets in der Signalverarbeitung", Seminarvortrag am Institut für Hygiene und Arbeitsphysiologie, Zurich, Switzerland, 27.4.99.
Wellig Peter	"Impedance Changes of the Wire Electrodes and their Influences on the EMG Characteristics", PROCID, Göteborg, Sweden, 3.9.99.
Wellig Peter	"Electromyogram Decomposition using the Single-Linkage Clustering Algorithm and Wavelets", 6th IEEE International Conference on Electronics, Circuits and Systems, Paphos, Cyprus, 6.9.99.
Wellig Peter	"Classification of Time-Varying Signals using Time-Frequency atOms", 21th Annual International Conference on the IEEE Engineering in Medicine and Biology Society, Atlanta, USA, 15.10.99.
Wellig Peter	"Decomposition of electromyogram long-term recordings", 21th Annual International Conference on the IEEE Engineering in Medicine and Biology Society, Atlanta, USA, 16.10.99.
Wellig Peter	"EMG Long-Term Signal Decomposition", Symposium Meeting, Muscular Disorders in Computer Users: Mechanisms and Models, Copenhagen, Denmark, 25.11.99.

Group: Adaptive Systeme

Von Hoff Thomas	"Multichannel Blind Deconvolution", 4th ETHZ-EPFL Summer School, Lausanne, Switzerland, 9.7. 99.
Von Hoff Thomas	"Using Preprocessing in Blind Source Separation of Convolutive Mixtures to Accelarate Convergence", ECCTD'99: European Conference on Circuit Theory Design, Stresa, Italy, 2.9. 99.
Von Hoff Thomas	"Two-Stage Approach for Multichannel Blind Deconvolution", 6th IEEE International Workshop on Acoustic Echo and Noise Control, Pocono Manor, PA, USA, 29.9. 99.
Wyrsch Sigi	"Adaptive Feedback Cancelling in Subbands for Hearing Aids", IEEE International Conference on Acoustics, Speech & Signal Processing, Phoenix, Arizona, USA, 16.3.99.
Wyrsch Sigi	"Subband Signal Processing for Hearing Aids", IEEE International Symposium on Circuits and Systems, Orlando, Florida, USA, 31.5.99.
Wyrsch Sigi	"Performance Comparison of PBFDAF Algor ithms", IEEE International Conference on Electronics Circuits & Systems, Paphos, Cyprus, 7.9.99.

Group: Digital Information Theory

Lapidoth Amos	"Fading Channels: How Perfect need "Perfect Side Information" be?", 1999 IEEE Information Theory Workshop, Kruger National Park, South Africa, 20.-25.6.99.

4.5 Organization of Lectures, Seminars, and Colloquia

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

Invited by Prof. Moschytz:

25.01.99 **Tom Sennhauser,** Chief Operating Officer GlobeSpan
Semiconductor Inc., New Jersey, USA,
"Exciting Opportunities in High Speed Communication Technology
Development".

03.05.99 **Prof. Chris Toumazou,** Imperial College, London, UK,
"New Developments in High Frequency Current-Mode Analogue
Technology".

23.06.99 **Prof. David A. Johns,** Dept. of Elec. and Comp. Engineering,
University of Toronto, Canada,
"Integrated Filters for Equalization".

01.07.99 **Prof. Ch. Schlegel,** Dept. of Electrical Engineering, University of
Utah, USA,
"Iterative Implementations for Linear Multiuser Detectors".

28.06.99 **Drahoslav Lim,** Institut für Signal- und Informationsverarbeitung,
ETH Zürich, Switzerland,
"Implementation of a Programmable Modularly Extendable Cellular-
Neural-Network Signal Processor".

29.06.99 **Prof. Leon O. Chua,** University of California, Berkeley, USA,
"CNN: A Paradigm for Information Processing and Self-
Organization".

14.06.99 **Prof. R. Michael Tanner,** Dept. of Computer Science, University of
California, Santa Cruz, USA,
"Error-Correcting Codes, Graphs, and Iterative Algorithms: Progress
and Challenges".

30.06.99 **Prof. Yehoshua Y. Zeevi,** Faculty of Elec. Engineering, Technion-
Israel Institute of Technology, Technion, Haifa, Israel,
"Two-Dimensional Nonseparable Multiwavelets and a Wavelet-Type
Approach".

19.11.99 **Prof. Dr. Simon Haykin,** McMaster University, Communications
Research Laboratory, Hamilton, Ontario, Canada,
"Wireless Communications: A Challenging Environment for Adaptive
Signal Processing".

19.11.99   **Dr. Robert W. Lucky,** Corporate Vice President, Applied Research
Telcordia Technologies, Red Bank, NJ, USA,
"The Telephone meets the Internet – Or is it the other way Around?

Invited by Dr. Heutschi:

13.01.99   **Dr. Stanislav Pietrzko,** Abt. Akustik/Lärmbekämpfung, EMPA Dübendorf,
"Grundzüge der Statistischen Energieanalyse".

20.01.99   **Jorgen Kragh, M. Sc.,** DELTA Acoustics & Vibration, Denmark,
"New Nordic Sound Propagation Models – Development and First Results".

03.02.99   **PD Dr.sc.techn. Norbert Dillier,** Universitätsspital Zurich,
"Digitale Hörgeräte und implantierbare Hörprothesen für das Jahr 2000 – audiologische und technologische Entwicklungsperspektiven".

05.05.99   **Dr. Hansrudolf Graf,** Sulzer Innotec, Winterthur,
"Wo sind bei Lastwagen die Lärmquellen, Schallortung mit Arraytechnik".

26.05.99   **Dr. Eckhard Kahle** (Vortragender), **Russell Johnson,** Artec,
New York, USA,
"Der Konzertsaal im neuen Kultur- und Kongresszentrum Luzern, erste Erfahrungen im Betrieb".

30.06.99   **Dr. Robert Hofmann,** Leiter Abteilung Akustik und Lärmbekämpfung, EMPA Dübendorf,
"25 Jahre Lärmbekämpfung – eine Bilanz".

01.12.99   **Gary Levinson,** Levinson AG, Allschwil,
"Lautsprecherzeilen mit Digital Directivity Control".

5. Publications

Group: Analog and Digital Signal Processing

Moschytz George S.	"Realizability Constraints for Third-Order Impedance-Tapered Allpole Filters", IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing, vol. 46, no. 8, August 1999, pp. 1073-1077.
Moschytz George S.	"Low-Sensitivity, Low Power Active-RC Allpole Filters using Impedance Tapering", IEEE Transactions on Circuits and Systems II: Analog and Digital Signal Processing, vol. 46, no. 8, August 1999, pp. 1009-1026.
Erne Markus	"Perceptual and Near-Lossless Audio Coding based on a Signal-adaptive Waveletfilterbank", AES 106-th Convention, Munich, May1999, Reprint 4934 (12 pages).
Erne Markus	"Best Wavelet-Packet Bases for Audio Coding using Perceptual and Rate-Distortion Criteria", Proceedings of ICASSP 99, Phoenix, May 1999 (4 Pages).
Erne Markus	"Audio Coding Based on Rate-Distortion and Perceptual Optimization Techniques", Proc. of the AES 17th Intern. Conference on High Quality Audio Coding, Florence, September 1999, pp. 220-225.
Haenggi 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, pp. 304-311, Feb. 1999.
Haenggi Martin Moser Simon Pfaffhauser Eric Moschytz George S.	Simulation and Visualization of CNN Dynamics", International Journal of Bifurcation and Chaos, no. 9, pp. 1897-1899, Sept. 1999.
Haenggi Martin Moschytz George S.	"Optimization of CNN Template Robustness", IEICE Transactions on Fundamentals of Electronics, Communications and Computer Sciences, Vol. E82-A, no. 9, pp. 1897-1899, Sept. 1999.
Haenggi Martin Reddy Hari C. Moschytz George S.	"Unifying Results in CNN Theory Using Delta Operator", Proceedings of ISCAS'99, June 1999, vol. 5, pp. 547-550.
Haenggi Martin Moschytz George S.	"Attacking the General Classification Problem with CNNs", Proceedings of ECCTD'99, Sept. 1999, vol. 2, pp. 771-774.

Haenggi Martin Reddy Hari C. Moschytz George S.	"The CNN Sampling Theorem", Proceedings of ECCTD'99, Sept. 1999, vol. 2, pp. 995-998.
Haenggi Martin Reddy Hari C. Moschytz George S.	"A Sampling Theorem in Numerical Integration", Proceedings of ICECS'99, Sept. 1999, pp. 561-565.
Helfenstein Markus Lustenberger Felix Loeliger Hans-Andrea Tarköy Felix Moschytz George S.	"High-speed interfaces for analogue, iterative decoders," Proc. IEEE Int. Symposium on Circuits and Systems, Orlando, vol.2, pp. 428-431, Jun. 1999.
Helfenstein Markus Lustenberger Felix Loeliger Hans-Andrea Tarköy Felix Moschytz George S.	"Decoding in Analog VLSI: towards ADC-less Transmission?!," Proc. Int. Workshop on Low Power RF Integrated Circuits, Lausanne, Oct. 1999.
Helfenstein Markus Curty Jari-Pascal Moschytz George S.	"Switched-Current Biquad using Differential, Double-Sampling, Forward/Backward Difference Integrators," submitted to IEEE Int. Symposium on Circuits and Systems, Lausanne, Oct. 1999.
Helfenstein Markus Huang Q. Moschytz George S.	"90dB, 90MHz, 30mW OTA for a High Capacitive Load," Int. Journal of Circuit Theory and Applications, vol. 27, pp. 473-483, Oct. 1999.
M. Helfenstein Moschytz George S.	Book: "Circuits and Systems for Wireless Communications", Kluwer Academic Publishers, 1999.
Joho Marcel Mathis Heinz	"Performance Comparison of Combined Blind/Non-Blind Source Separation Algorithms", Proc. International Conference on Independent Component Analysis and Blind Source Separation, Aussois, January 11-15, 1999, pp. 139-142.
Joho Marcel Mathis Heinz	"An FFT-Based Algorithm for Multichannel Blind Deconvolution", Proc. IEEE International Symposium on Circuits and Systems, Orlando, May 30 - June 2, 1999, pp. III 203-206.
Joho Marcel Moschytz George S.	"On the Design of the Target-Signal Filter in Adaptive Beamforming", IEEE Transactions on Circuits and Systems-II, vol. 46, no. 7, pp. 963-966, July 1999.
Lippuner Daniel	"A Combined Process- and Measurement-Noise Covariance Estimation in Model-Based Adaptive FIR Filters", Proceedings of the ECCTD'99 vol.II, pp.1391-1394.
Lippuner Daniel	"Tracking Behavior of Model-Based Adaptive FIR Filters with Noise-Covariance Estimation", Proceedings of the IWAENC'99, pp.156-159.

Lustenberger Felix Loeliger Hans-Andrea Helfenstein Markus Tarköy Felix	"Decoding in Analog VLSI", in IEEE Communications Magazine, April 1999, pp. 99-101.
Lustenberger Felix Loeliger Hans-Andrea Helfenstein Markus Tarköy Felix Moschytz George S.	"An Analog Decoding Technique for Digital Codes", Proceedings of ISCAS '99, Orlando, Florida, May 30 - June 2, 1999, vol. II, pp. 428-431.
Lustenberger Felix Loeliger Hans-Andrea Helfenstein Markus Tarköy Felix Moschytz George S.	"All-Analog Decoder for a Binary (18,9,5) Tail-Biting Trellis Code", Proceedings of ESSCIRC'99, Duisburg, September 21-23, 1999, pp. 362-365.
Schmid Hanspeter Moschytz George S.	"A Continuously Adjustable Video-Frequency Current Amplifier for Filter Applications", Proceedings of the European Conference on Circuit Theory and Design, vol. 1, pages 305-309, Stresa, August 1999.
Schmid Hanspeter Moschytz George S.	"Minimum-Sensitivity Single-Amplifier Biquadratic Filters", Proceedings of the European Conference on Circuit Theory and Design, vol. 2, pages 1027-1030, Stresa, August 1999.
Schmid Hanspeter Moschytz George S.	"A Tunable, Video-Frequency, Low-Power, Single-Amplifier Biquadratic Filter in CMOS", Proceedings of the IEEE International Symposium on Circuits and Systems, vol. 2, pages 128-131, Orlando, Florida, June 1999.
Wellig Peter Moschytz George S.	"Electromyogram Decomposition using the Single-Linkage Clustering Algorithm and Wavelets", Proceedings of ICECS'99, Paphos, Sept. 1999, pp. 537-541.
Wellig Peter Moschytz George S. Läubli Thomas	"Decomposition of Electromyogram Long-Term Recordings", Proceedings of EMBS'99, Atlanta, Oct. 1999, p. 578.
Wellig Peter Moschytz George S.	"Classification of Time-Varying Signals using Time-Frequency Atoms", Proceedings of EMBS'99, Atlanta, Oct. 1999, p. 9.
Wellig Peter Moschytz George S.	"EMG Long-Term Signal Decomposition", Proceedings of the Symposium Meeting on Muscular Disorders in Computer Users, Copenhagen, Nov. 1999, pp. 67-72.
Schaerer Thomas	"DCDC-Hochspannungswandler mit niedriger Leerlaufverlustleistung und gutem Wirkungsgrad", (Teil 1), MEGALINK 13-14, 1999, Rubrik: Praxis, 23.8.1999, p. 46.

| Schaerer Thomas | "DCDC-Hochspannungswandler mit niedriger Leerlauf-verlustleistung und gutem Wirkungsgrad", (Teil 2), MEGALINK 16, 1999, Rubrik: Praxis, 20.9.1999, p. 26. |
| Schaerer Thomas | "Selbstgemachte 48VDC, Phantom-Speisung für Konden-satormikrofone", MEGALINK 21, 1999, 29.11.99, p. 14. |

Group: Adaptive Systems

von Hoff Thomas Kaelin August	"Using preprocessing in Blind Source Separation of Convolutive Mixtures to accelarate Convergence", European Conference on Circuit, Theory and Design, vol. 2, pp. 1395-1398, Stresa, August 1999.
von Hoff Thomas Kaelin August	"Two-stage Approach for Multichannel Blind Deconvolution", IEEE Workshop on Acoustic Echo and Noise Control, pp. 180-183, Pocono Manor, Pennsylvania, September 1999.
Wyrsch Sigi Kaelin August	"Adaptive Feedback Cancelling in Subbands for Hearing Aids", IEEE International Conference on Acoustics, Speech & Signal Processing, vol. 2, pages 921-924, March 1999.
Wyrsch Sigi Kaelin August	"Subband Signal Processing for Hearing Aids", IEEE International Symposium on Circuits and Systems, vol. 3, pages 29-32, May 30-June 2, 1999.
Wyrsch Sigi Kaelin August	"Performance Comparison of PBFDAF Algorithms", IEEE International Conference on Electronics Circuits & Systems, vol. II, pages 831-834, September 1999.

Group: Digital Information Theory

| Lapidoth Amos
Ziv Jacob | "On the Decoding of Convolutional Codes on an Unknown Channel", IEEE Transactions on Information Theory, vol. 45, pp. 2321-2332, Nov. 1999. |
| Cohen Aaron
Lapidoth Amos | "On the Gaussian Watermarking Game", LIDS Technical Report LIDS-P-2564, Oct. 1999. |

6.   Guests, Visitors

6.1 Activities of Academic Guests at the Institute

Guests of Prof. Moschytz:

Tom Sennhauser,	GlobeSpan Semiconductor Inc., New Jersey, USA. Discussion of prospects and opportunities for graduates and doctoral studens in communications industry.	25.01. – 27.01.99
Prof. Chris Toumazou,	Dept. of Electrical Engineering Imperial College, London,England. Exchange of research results and ideas with analog circuit design groups.	03.05. – 04.05.99
Prof. Hari Reddy,	California State University, Long Beach, USA. Ongoing continued research into aspects of signal processing related to the delta operator.	28.06. – 09.07.99
Prof. Y. Zeevi,	Technion – Israel Institute of Technology, Haifa, Israel. Ongoing exchange of ideas and programs with research groups at Technion and ISI.	28.06. – 04.07.99
Prof. Allen Lindgren,	University of Rhode Island, Kingston, USA. Collaboration with the Adaptive Filter Group.	01.07. – 15.09.99

Guests of Prof. Lapidoth:

Aaron S. Cohen	MIT, Cambridge, USA, studied and presented a talk on watermarking for data protection.	03.10. – 31.12.99
Ibrahim Abou Faycal	MIT, Cambridge,USA, studied and presented a talk on an	

information theoretic approach
to the design of reduced-
complexity receives for
ISI channels.   03.10. - 31.12.99

7. Honors and Awards

Moschytz George S.   Recipient of the CASS Golden Jubilee Medal,
IEEE Circuits & Systems Society.

Moschytz George S.   Recipient of the IEEE Millenium Medal,
IEEE Circuits & Systems Society.