EURASIP

NEWS LETTER

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European Association
for Signal, Speech,
and Image Processing

Newsletter, Volume 14, Number 3, September 2003

Contents

Nominations for the EURASIP Awards 2004

EURASIP gives on a biennial basis the following awards:

Best Paper Awards

These awards are given to the best papers published in the journals sponsored by EURASIP, that is,

Signal Processing (2 awards)
Image Communication (1 award)
Speech Communication (1 award)
Journal on Applied Signal Processing (2 awards).

The awards consist of a certificate and a check for the authors. The papers considered for the awards are preselected by the journal Editors according to the score of the reviews or the evaluations of members of the editorial boards including the guest editors of the special issues. Then a comparative evaluation is performed by specific subcommittees, one for each of the EURASIP journals, according to the following criteria:

- relevance of the topic dealt with,
- quality of the technical content,
- style of the presentation.

The AdCom takes the final decision on the awards on the basis of the evaluations of the award subcommittees.

EURASIP Technical Achievement Award

This award will be given at the rate, decided by the AdCom, of one or two per EUSIPCO, to individuals for major contributions in the field of signal, speech, and image processing. The candidate should

- be actively involved in an international context with impact on the European research activities,
- actively participate in the conferences sponsored by EURASIP, and in particular in EUSIPCOS's,
- publish on the EURASIP journals.

The award consists of a medal presented to such individuals. The nominations are submitted to the EURASIP Awards Chairman by members of the EURASIP AdCom and/or Advisory Committee by the end of December 2003. The final decision about the award will be taken unanimously by the AdCom in March/April 2004.

EURASIP Meritorious Service Award

This award is given at the maximum rate of one per EUSIPCO to individuals for distinguished services to the society. The award consists of a medal presented to such individuals. The nominations are submitted to the EURASIP Awards Chairman by members of the EURASIP AdCom and/or Advisory Committee by the end of December 2003. The final decision about the award will be taken unanimously by the AdCom in March/April 2004.

Annual European Group Technical Achievement Awards

These awards are reserved to the leaders of European research groups with relevant performances in the field of signal, speech, and image processing. The awards, given at the maximum rate of one per year, consist of a plaque presented to the leader of the group. The AdCom favoured requesting nominations from EURASIP members with the expectation that each AdCom member would make a minimum of two proposals, from outside his own country, each year. The deadline for the submission of the candidacies is the end of December 2003. EURASIP members are asked to email the EURASIP President, Prof. Ferran Marqués (ferran.marques@eurasip.org) or the EURASIP Awards Chairman, Prof. Giovanni Sicuranza (giovanni.sicuranza@eurasip.org), providing a name, affiliation, and 4–5 bullet points of achievement. The decision on these awards will be taken by the AdCom in March/April 2004.

All the above mentioned awards will be presented during EUSIPCO-2004 in Vienna.

Giovanni Sicuranza
EURASIP Awards Chairman
June 2003

New EURASIP Awards for the EURASIP Journal on Applied Signal Processing

As many of the EURASIP members may know, the publisher and editor of the EURASIP Journal on Applied Signal Processing (EURASIP JASP) have recently been changed. Hindawi is the new publisher and Prof. Ray Liu has been the Editor-in-Chief for the years 2001–2002, while Prof. Marc Moonen succeded him at the beginning of 2003. Through his relentless work for well over two years, Ray Liu completely transformed the journal into an extremely successful forum for the whole signal processing society. As a measure of the success of the new EURASIP JASP, 4 issues with about 30 papers have been published in 2001, while 12 issues, 133 papers, 1500 pages have been published in 2002. Moreover, EURASIP JASP was chosen to be covered in many ISI products only a few months after its launch. Given the quality of papers, authors, guest editors, and the number of submissions to EURASIP JASP, the EURASIP AdCom recently decided to give two Best Paper Awards, one for the years 2001–2002 and another one for 2003. The awards will be presented to the winners during EUSIPCO-2004 in Vienna.

The paper that will receive the award for 2001–2002 has been selected by the committee formed by Profs. Bastiaan Kleijn, Hideaki Sakai, and Phillip Regalia. This choice has been formally approved by the EURASIP AdCom. Title of the paper, authors, and motivation of the award are reported here below.

> **Design and DSP Implementation of Fixed-Point Systems**, Volume 2002, Number 9, September 2002, pp. 908–925.
>
> Martin Coors, Holger Keding, Olaf Luthje, and Heinrich Meyr are with the Institute for Integrated Signal Processing Systems, Aachen University of Technology, Germany.
>
> This paper gives a thorough and readable account of an important aspect of applied signal processing: algorithm implementation on dedicated DSP chips. Rather than focusing on one aspect of implementation, it provides a comprehensive treatment which captures the multifaceted tasks that confront applied signal processing.

Congratulations to the winners!
I look forward to meeting them at EUSIPCO-2004 in Vienna.

Giovanni Sicuranza
EURASIP Awards Chairman

2002 EURASIP Best Paper Award for Signal Processing

The paper that will receive the EURASIP Best Paper Award for 2002 has been chosen by the committee formed by Profs. Nicholas Kalouptsidis, Steve McLaughlin, and Jean-Yves Tourneret among eight papers preselected by the journal editor Prof. Murat Kunt. This choice has been formally approved by the EURASIP AdCom during the telephone conference of June 25, 2003. The award will be presented to the winners at EUSIPCO-2004 in Vienna. Title of the paper, authors, and motivation of the award are reported here below.

> **New Multiscale Transforms, Minimum Total Variation Synthesis: Application to Edge Preserving Image Reconstruction**, Volume 82, Number 11, November 2002, pp. 1519–1545.
>
> Emmanuel J. Candes and Franck Guo are with the Department of Applied and Computational Mathematics, California Institute of Technology, Pasadena, CA 91125, USA.
>
> This excellent paper addresses the problem of image reconstruction by using the ridgelet transform. It describes new multiscale transforms which combine multiscale analysis and geometry and suggests numerical implementation techniques. This paper fits the criteria for a prize paper in that it takes novel mathematical ideas and implements them in an efficient manner to ensure that they can be used for real world problems. The new image reconstruction methodology consists of minimizing the total variation norm of an appropriate criterion. The proposed strategy provides a very good reconstruction of edges, in comparison to the traditional wavelet and ridgelet transforms. Simulation results on synthetic and real images nicely illustrate the performance of this new method.

Congratulations to the winners!
I look forward to meeting them at EUSIPCO-2004 in Vienna.

Giovanni Sicuranza
EURASIP Awards Chairman

Calendar of Events

Year	Date	Event	Location	EURASIP Involvement	Chairperson/Information
2003	September 2–5	Advanced Concepts for Intelligent Systems (ACIVS-2003)	Ghent, Belgium	Cooperation	J. Blanc-Talon http://eltodo.rug.ac.be/acivs2003/
	September 8–11	6th Int. Conference on Digital Audio Effects	London, UK	Cooperation	Mark Sandler http://www.elec.qmul.ac.uk/dafx03
	September 18–20	3rd Int. Symposium on Image and Signal Processing and Analysis	Rome, Italy	Cooperation	A. Neri and H. Babic http://www.isispa.org
	September 18–19	International Workshop VLBV03	Madrid, Spain	Cooperation	Narciso Garcia http://vlbv03.upm.es
	September 22–23	IEE Colloquium on DSP Enabled Radio	Livingstone, Scotland, UK	Cooperation	Bob Stewart http://www.eee.strath.ac.uk/r.w.stewart/dsp_radio.pdf
	October 9–10	1st Int. Workshop on Interactive Rich Media Content Production (WS2003)	Lausanne, Switzerland	Cooperation	I. Pitas and D. Thalmann http://www.richmedia2003.org
	December 11–12	ISCA Workshop on Multimodal User Authentication	Santa Barbara, USA	Cooperation	Jean-Luc Dugelay authentication@research.panasonic.com
2004	June 23–25	17th Int. EURASIP Conf. BIOSIGNAL	Brno, Czech Republic	Co-sponsorship	Jiri Jan http://www.feec.vutbr.cz/UBMI/bs2004.html
	July 20–22	4th CSNDSP, Int. Symposium on Communication Systems, Networks and DSP	Newcastle, UK	Cooperation	T. Boukouvalas http://www.shu.ac.uk/ocr/csndsp/
	September 7–10	12th European Signal Processing Conference (EUSIPCO-2004)	Vienna, Austria	Sponsor	W. Mecklenbräuker http://www.nt.tuwien.ac.at/eusipco2004/

Sergios Theodoridis
Workshops/Confs Coordinator EURASIP

EURASIP (CO-)SPONSORED EVENTS

12th European Signal Processing Conference

EUSIPCO-2004

September 7–10, 2004, Vienna, Austria

Technische Universität Wien
Vienna University of Technology

CALL FOR PAPERS

The 2004 European Signal Processing Conference (EUSIPCO-2004) is the twelfth biennial conference promoted by EURASIP, the European Association for Signal, Speech, and Image Processing (www.eurasip.org), and organized by the Institute of Communications and Radio-Frequency Engineering at Vienna University of Technology (www.nt.tuwien.ac.at) and the ftw. Telecommunications Research Center Vienna (www.ftw.at). EUSIPCO-2004 aims to cover all aspects of signal processing theory and applications. Sessions will include invited presentations with review character in addition to presentations of new research results. An extensive technical exhibition will also be organized.

Camera-ready final papers describing original work are invited in any of the areas listed below. Accepted papers will be published in the Proceedings of EUSIPCO-2004. Acceptance will be based on quality, relevance, and originality. A contest for Best Paper Awards will be held and awards will be given at the banquet. Proposals for special sessions and tutorials are also invited.

AREAS OF INTEREST

Digital Signal Processing; Statistical Signal Processing; Sensor Array and Multichannel Processing; Signal Processing for Communications; Speech Processing; Image and Multidimensional Signal Processing; Video and Multimedia Signal Processing; Nonlinear Signal Processing; Wavelet and Time-Frequency Signal Processing; Audio and Electroacoustics; DSP Implementations and Embedded Systems; Rapid Prototyping and Tools for DSP Design; Signal Processing Applications; Signal Processing Education and Training; Emerging Technologies.

BEST PAPER AWARDS

Three Best Paper Awards, of 300 Euro each, will be given to young authors (under 30). The young author's name must appear first on the paper.

SUBMISSION

To submit a paper or a proposal for special sessions or tutorials, follow the electronic submission procedure on the website www.nt.tuwien.ac.at/eusipco2004. Submitted papers must be camera-ready, final, of length no more than four pages including figures and references, and conforming to the format specified on the EUSIPCO website. No more than four submissions are allowed per contributor, as an author or a coauthor.

IMPORTANT DATES

Proposal for Special Sessions and Tutorials	November 28, 2003
Submission of Camera-Ready Final Papers	January 16, 2004
Acceptance Notification	March 31, 2004
Authors' Registration	April 26, 2004

Please visit our website: www.nt.tuwien.ac.at/eusipco2004

Chair

Wolfgang Mecklenbräuker
w.mecklenbruker@tuwien.ac.at
Vienna University of Technology

Cochairs

Markus Rupp
mrupp@nt.tuwien.ac.at

Hans Weinrichter
jweinri@nt.tuwien.ac.at
Vienna University of Technology

Program Chairs

Franz Hlawatsch
f.hlawatsch@tuwien.ac.at

Gerald Matz
g.matz@tuwien.ac.at
Vienna University of Technology

Special Sessions/ Tutorials Chairs

Gernot Kubin
g.kubin@ieee.org
Graz University of Technology

Walter Kropatsch
krw@prip.tuwien.ac.at
Vienna University of Technology

Publicity/Exhibits Chairs

Markus Kommenda
kommenda@ftw.at

Judith Mair
mair@ftw.at
ftw. Telecommunications
Research Center Vienna

Publications Chairs

Markus Rupp
mrupp@nt.tuwien.ac.at

Bernhard Wistawel
b.wistawel@tuwien.ac.at
Vienna University of Technology

Finance Chair

Gerhard Doblinger
g.doblinger@tuwien.ac.at
Vienna University of Technology

Conference Secretariat/ Registration

Mondial Congress Vienna
congress@mondial.at
Phone: +43/1/588040
Fax: +43/1/5869185

Report on the IEEE-EURASIP Workshop on Nonlinear Signal and Image Processing, NSIP-03, June 8–11, 2003, Grado (Trieste), Italy

The sixth NSIP Workshop has been held at the Congress Palace in Grado on June 8–11, 2003. The main workshop topics have been centered on nonlinear theory and tools, nonlinear processing applications, and implementation of nonlinear systems. More than 100 papers selected from submissions received by researchers working in 24 countries have been presented in the following oral or poster sessions:

- Signal analysis and modeling
- Signal detection and estimation
- Nonlinear time-frequency methods
- Nonlinear methods
- Mathematical morphology
- Order statistics
- Volterra operators and HOS
- Neural and evolutionary systems
- Image and video processing
- Image enhancement
- Image and video coding
- Image analysis and representation
- Fuzzy systems
- Biomedical signal processing
- Communications
- Array and multichannel processing
- Circuits, systems, and architectures

In addition, two special sessions have been organized:

- Nonlinear methods and tools for adaptive learning analysis, organized by Joao Destro-Filho and cochaired by Gerard Favier and Simone Fiori
- Distant-talking speech processing and speech recognition, organized and chaired by Maurizio Omologo

Three invited talks have been presented during the plenary sessions by

- Leon Cohen on “Time-frequency analysis: the current issues”
- Touradj Ebrahimi on “Perceptual quality assessment of audiovisual information”
- Vito Cappellini on “Information technology trends for cultural heritage”

The participants in the workshop have been encouraged to expand their conference contributions into full papers for submission to the Special Issue on Nonlinear Signal and Image Processing to be published in early 2004 by the EURASIP Journal on Applied Signal Processing.

Complete information about the technical program and the workshop proceedings can be found at http://ipl.univ.trieste.it/nsip03/ or http://imagets10.univ.trieste.it/nsip03/.

The next NSIP Workshop will be organized by Prof. Pao-Ta Yu and Prof. Akira Taguchi in Sapporo (Japan) in 2005.

Giovanni L. Sicuranza
Conference Cochair

Report on the 13th International Packet Video Workshop

The well-known Packet Video Workshop has been held at the end of April 2003 in Ecole polytechnique University of Nantes, France. This year, the meeting was organized by the team Image and Video Communications of laboratory IRCCyN (the Nantes Institute of Communications and Cybernetics). The staff organizing the meeting for a whole year is to be warmly acknowledged: Drs. Nicolas Normand, Vincent Ricordel, Salima Hamma, Benoit Parrein, Pr. Dominique Barba, and Leila Neicibi. Packet Video was in conjunction with Picture Coding Symposium held in Saint Malo and organized by IRISA, Rennes and chaired by Drs. Christine Guillemot and Claude Labit.

Sixty papers were selected by the committee to be presented at the Packet Video Meeting (30 oral presentation and 30 posters). Almost all the speakers attended the meeting which was really difficult for Asian and American researchers at this period of time. Professor Bernd Girod deserves a special mention by presenting both a brilliant invited paper (available at our website) and the works of two American colleagues who could not attend the meeting because of visa reasons. The full audience was composed of more than 85 researchers, with all the European major teams represented.

The papers were of very good quality and spread in the different subareas focused by the Packet Video Meeting: Network and Video were equally served, the strong relationships between them always presented and questioned by the assistance. Two points have to be mentioned by their pregnancy through the presented papers: the use of joint source-channel coding versus multiple description in conjunction with Internet services and the implications of wireless networks uses and specificities. For the latter, the team of Dr. Thomas Stockhammer also deserves a special mention by animating a special implementation session presenting a wireless video platform.

Notice that the whole sixty papers of the conference will be available soon on our server: www.polytech.univ-nantes.fr/pv2003.

The Sunday morning prior to the meeting opening allows researchers to visit the old city of Nantes and enjoy the gothic cathedral, its old Middle Age streets, and its 19 century renewal harbour area with Jules Verne atmosphere. The workshop banquet was given on the river Erdre which was mentioned by king Francois 1st as "the most beautiful French river."

The Packet Video steering committee met during the meeting and decided two important points:

- The 14th International Packet Video will be held in California at Irvine University in December 2004. It will be chaired by Professor Magda El Zarki who is known in our community for her video-networks research
- To take care of the history of the International Packet Video Workshop, all the papers from the 13th previous workshops will be put on websites around the world. Such a

website, as in Nantes, will figure all the possibly downloaded papers (as it is the case for the last three meetings) to allow everyone of the scientific community to use this highly relevant fund of science.

Professor Jeanpierre Guédon
General Chair Packet Video 2003

Acoustic Echo Control

Eberhard Hänsler

Electrical Engineering and Information Technology, Darmstadt University of Technology, Merckstrasse 25, D-64283 Darmstadt, Germany
Email: Eberhard.Haensler@nt.tu-darmstadt.de

1. Introduction

Commercial telephone services started about 125 years ago. At this time users conducting a phone conversation had their two hands busy [1]. Each user had to hold a microphone close to his mouth and a loudspeaker close to one ear. It did not take long to get one hand free: microphone and loudspeaker were assembled in a handset. Desk telephones looking very similar to the ones we are using today were offered already before the turn of the century. However, more than 100 years later, starting a telephone call in the majority of cases still means "to grab the handset."

In the early years of the telephone, efficient electroacoustic transducers and amplifiers were not yet available. Therefore, positioning the loudspeaker and the microphone very close to the ear and the mouth, respectively, was the only way to achieve an—at that time—acceptable speech quality. Nowadays, however, it is difficult to understand why it should still be the same and that it takes all the signal processing capabilities available today to switch to hands-free operation.

Researchers and development engineers have been very busy during the past decades. A large number of papers on the topic have been published including bibliographies [2, 3, 4, 5] and reports and books on the state of the art [6, 7, 8, 9, 10, 11, 12]. Adaptive algorithms for acoustic echo cancellation gained special attention in [13].

This paper explains the problem of acoustic echoes and their cancellation. It will focus on the hands-free telephone as one of the applications most asked for. The considerations, however, hold for other applications such as hearing aids, voice input systems, and public address systems as well. The problem of noise reduction that is generally linked to hands-free conversations will not be discussed in this paper.

Acoustic echoes and the problem of their cancellation arise wherever a loudspeaker and a microphone are placed such that the microphone picks up the signal radiated by the loudspeaker and its reflections at the borders of the enclosure. As a result, the electro-acoustic circuit may become unstable and produce howling. In addition, telecommunication users are annoyed by listening to their own speech delayed by the round trip time of the system. These problems are avoided if the attenuation of the acoustic path between loudspeaker and microphone is sufficiently high.

In general, acoustic echo cancellation units are comprised of three subunits: a loss control circuit (LC), a filter parallel to the loudspeaker-enclosure-microphone system (LEMS), the echo cancellation filter (ECF), and a second filter, the residual echo suppressing filter

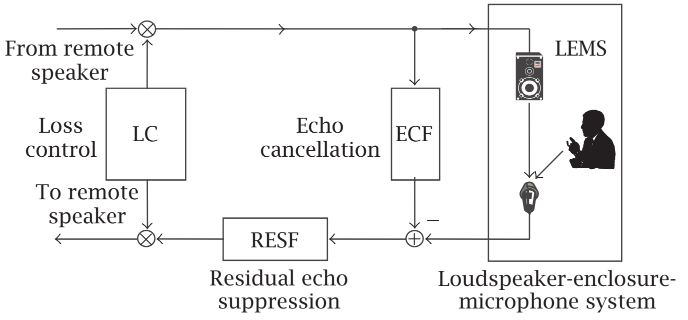

Figure 1: General structure of an acoustic echo cancellation system.

(RESF), which is within the path of the output signal (see Figure 1). The LC can attenuate the input and/or output signal such that the communication loop always remains stable. The ECF in parallel to the LEMS is able to cancel echoes picked up by the microphone according to the degree to which this filter is matched to the LEMS. The RESF within the path of the output signal can be used to suppress residual echoes and background noise.

Of these subunits, the LC has the longest history in hands-free communication systems. In its simplest form it reduces the full-duplex communication system to a half-duplex one by alternatively switching the input and output lines on and off. This can be done manually, as it is still done by astronauts, or by voice-controlled switches. The problem related to those switches is that they do not distinguish between speech and noise signals. Therefore, strong noise can mislead the switching circuit. Consequently, an active speaker may be switched off in favor of noise at the remote location. Thus, apart from preventing howling and suppressing echoes, any natural conversation is prevented too. A moderate form of switching off one direction of a connection consists of increasing its attenuation. In the case that no activity is detected on both circuits, the additional attenuation can be equally distributed between both directions. If the attenuation already provided by echo cancelling (and other) circuits can be estimated, only the lacking attenuation has to be inserted. In the case of a well-adapted ECF, this may only be a few decibels, which does not disturb both conversation partners. The LC, however, cannot be completely removed from a hands-free calling device. It has to serve as an "emergency brake" in cases where the echo attenuation achieved by the ECF and the RESF does not comply with ITU recommendations [14].

2. Loudspeaker-enclosure-microphone system (LEMS)

In an LEMS the loudspeaker and the microphone are connected by an acoustical path formed by a direct connection (if both can "see" each other) and a large number of additional paths caused by reflections at the boundaries of the enclosure. For low sound pressure and no overload of the converters, this system may be modeled as a linear system. The impulse response can be described by a sequence of delta impulses delayed proportionally to the geometrical length of the related path and the inverse of the sound velocity. As a first order approximation, one can assume that the impulse response decays exponentially. A measure for the degree of this decay is the so-called *reverberation time* T_{60} [15]. Depending on the application, it may be possible to design the boundaries of the enclosure such that

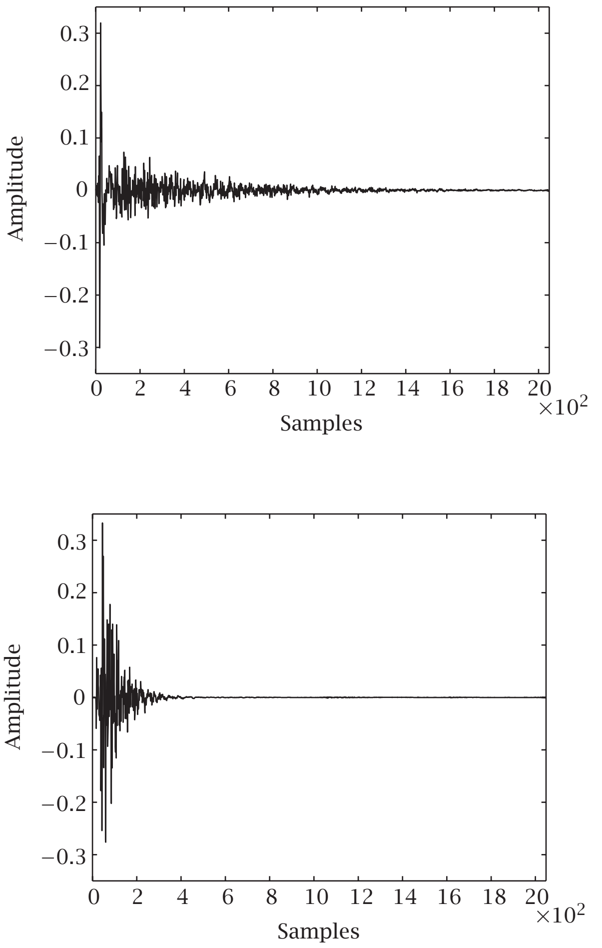

Figure 2: Impulse responses measured in an office (top) and in a car (bottom) (sampling frequency = 8 kHz).

the reverberation time is small, resulting in a short impulse response. Examples include telecommunication studios. For ordinary offices, the reverberation time T_{60} is typically in the order of a few hundred milliseconds. For the interior of a passenger car, this quantity measures a few tens of milliseconds. Figure 2 shows the impulse responses of LEMSs measured in an office (top) and in a passenger car (bottom). The microphone signals have been sampled at 8 kHz according to the standards for telephone signals.

The impulse response of a LEMS is highly sensitive to any changes caused, for example, by the movement of a person within it. This is explained by the fact that, assuming a sound velocity of 343 m/s and 8 kHz sampling frequency, the distance travelled by the sound wave between two sampling instants is 4.3 cm. Therefore, a 4.3 cm change in the length of an echo path, for example, the movement of a person by only a few centimeters, shifts the related impulse by one sampling interval. Thus, an *adaptive* ECF is required.

3. The echo cancelling filter (ECF)

In principle, acoustic echo cancelling constitutes a system identification problem. However, the system to be identified, the LEMS, is highly complex: its impulse response exhibits up to several thousand sample values with each one representing an echo path and, as explained above, it is time varying. The question of the optimal structure of the ECF has been discussed intensively in [16]. Since a long impulse response has to be modeled by the ECF, a recursive (IIR) filter seems best suited at first glance. At second glance, however, the impulse response exhibits a highly detailed and irregular shape. To achieve a sufficiently good match, the replica must offer a large number of adjustable parameters without the danger of becoming instable during adaptation. Therefore, in all applications known to the author, FIR filters are applied.

4. Speech signals

In the problem discussed, here one has to deal primarily with speech signals additively disturbed by other speech signals, if both communication partners talk simultaneously, and by noise. Performing signal processing with these types of signals turns out to be one of the most difficult tasks in digital signal processing.

Speech is characterized by nearly periodic segments, by noise-like segments, and by pauses. The signal envelope fluctuates greatly. In speech processing, it is widely accepted that parameters derived from a speech signal have to be updated after intervals of about 20 milliseconds. In the case of an 8 kHz sampling frequency, the standard for telephone systems, the normalized autocorrelation coefficient of neighboring samples $s_{xx}(1)/s_{xx}(0)$, assumes values in the range of 0.8 to 0.95. Because of periodicities and pauses, short-time autocorrelation matrices very often become singular. Thus, special precautions are necessary to prevent the instability of algorithms that use, directly or indirectly, the inverse of the autocorrelation matrix. Figure 3 shows a segment of a speech signal (top) and an estimate of its power spectral density (bottom).

The acoustic echo cancellers used in telephones have to comply with a number of standards issued by international standardization organizations like the International Telecommunication Union (ITU) and the European Telecommunications Standards Institute (ETSI). Especially important are requirements concerning delay and echo attenuation [14]. For ordinary telephones, the ITU allows only 2 milliseconds of additional delay for front-end processing. In the case of mobile telephones, 39 milliseconds are permitted. The maximum of 2 milliseconds prohibits the application of efficient frequency domain or block processing methods. Concerning the overall echo attenuation, a minimum of 45 dB is necessary in single talk situations. In the case of double talk, the attenuation can be reduced to 30 dB taking into consideration that the echo of the far-end signal is masked by the local speech signal. This high echo attenuation has to be provided at all times of connection and in all situations. Since adaptive filters (the ECF and the RESF) may not (yet) have converged to their optimal settings, the attenuation required by standards can only be guaranteed with the help of the loss control device.

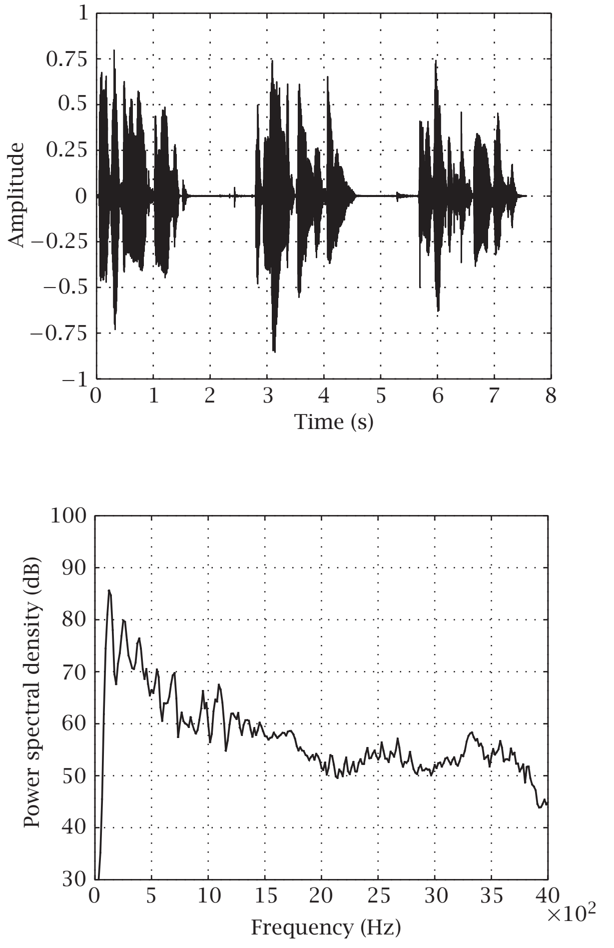

Figure 3: Section of a speech signal and an estimate of its power spectral density.

5. Echo cancellation

With the availability of powerful digital signal processors, an adaptive filter to cancel acoustic echoes (the ECF) and a second adaptive filter to suppress residual echoes (the RESF), (see Figure 1) became feasible.

The majority of implemented acoustic echo cancelling systems still use the *normalized least-mean-square* (NLMS) *algorithm* to update the ECF. This gradient type algorithm minimizes a mean-square error. Its update uses a normalization according to the energy of the short-term input speech signal that exhibits a high variance. Additional control by a step-size factor is necessary to prevent divergence in case of double talk and local noise.

The NLMS algorithm has no memory, that is, it uses only signals that are available at the time of the update. This is advantageous for tracking changes in the LEMS. The update is performed proportional to an input signal vector that is formed by the current input sample and the $N-1$ preceding ones, where N equals the number of filter coefficients to be updated. For speech signals, consecutive signal vectors are highly correlated. This is the reason for the low speed of convergence of the NLMS algorithm in case of speech excitation. Additional measures like decorrelation, that may be performed by low-order prediction error filters, are necessary to speed up convergence. The motivation for using the NLMS algorithm in the application discussed here is its robustness and low-computational complexity; it takes only one multiply-accumulate (MAC) operation to update a coefficient.

The *affine projection* (AP) *algorithm* overcomes the weakness of the NLMS algorithm concerning correlated input signals by updating the filter coefficients not just according to the current input signal vector but also using its $M - 1$ immediate predecessors. To accomplish this, an input signal matrix is formed and an error vector is calculated, collecting the errors for the current input vector and the $M - 1$ errors that would occur by applying the past input vectors to the ECF with the current coefficient setting. The price to be paid for the improved convergence is the increased computational complexity caused by the inversion of an $M \times M$ matrix required at each coefficient update. Fast versions of this algorithm, however, are available [17, 18]. For $M = 1$, the AP algorithm is equal to the NLMS algorithm. For speech input signals, even $M = 2$ leads to a considerably faster convergence of the filter coefficients. Suggested values for M are between two and five for the ECF update.

It should be noted, however, that faster convergence of the ECF coefficients also means faster divergence in the case of strong local signals. Therefore, faster control of the algorithm is required as well. Optimal control parameters have to be based upon estimated quantities. Since their reliability depends upon the lengths of the available data records, a too high speed of convergence may not always be desirable.

The *recursive least-squares* (RLS) *algorithm* minimizes the exponentially weighted sum of the squared errors. It calculates an estimate of the $N \times N$ autocorrelation matrix of the speech input vector and uses the inverse of this matrix to decorrelate the input for the update of the filter coefficients. In contrast to the AP algorithm, now an $N \times N$ matrix has to be inverted at each coefficient update and N may be in the order of 1000. The memory of this algorithm is controlled by a forgetting factor. A long memory, a forgetting factor very close to one, stabilizes the procedure. It degrades, however, its tracking capabilities. Fast versions of the RLS algorithm have been developed [19]. Special precautions, however, are necessary to ensure their stability.

Independent of the algorithm used, the computational load can be reduced by the introduction of *subband* or *block processing*. Cancelling echoes in subbands increases the degrees of freedom for optimizing the echo cancelling system since different algorithms and different filter lengths may be used in different bands. Combining signal samples into blocks and processing a block of data at a time reduces the demand for signal processing power. The efficiency of block processing methods increases with the block length. However, inevitably connected to both procedures is an increased storage demand and, even more important in real-time applications, an enlarged signal delay. Partitioning the impulse response of the LEMS provides a compromise between computational demands and delay [20, 21].

To summarize the remarks on algorithms which adapt the ECF, the NLMS algorithm plays the role of the workhorse being modest, stable, but slow. The AP algorithm has the potential to take over this role because it offers better performance at the price of a man-

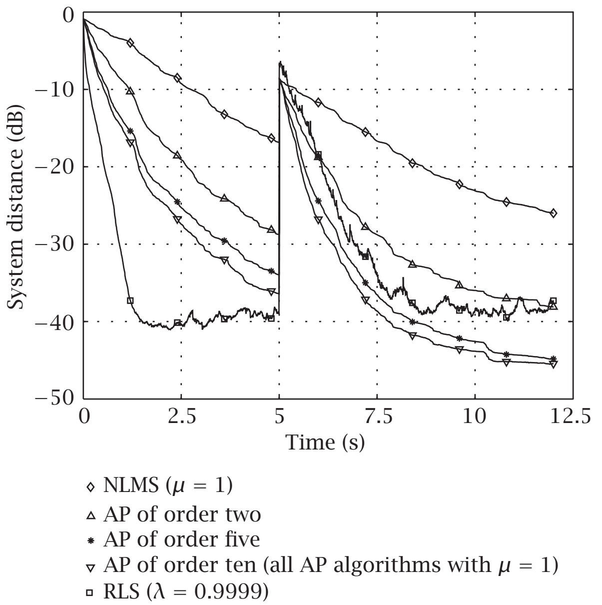

Figure 4: Convergence of the filter coefficients for different adaptation algorithms (filter length = 1024, sampling frequency = 8 kHz). "System distance" stands for the sum over the squared differences between the coefficients of the impulse responses of the LEMS and the ECF. The impulse response of the LEMS is changed at $t = 5$ s.

ageable increase in complexity. The RLS algorithm definitely performs like a race horse: it is fast; however, it behaves capriciously and needs a lot of attention, that is, signal processing power. This property makes its use for acoustical echo cancellation extremely difficult, if not prohibitive. Figure 4 compares the convergence achieved with the algorithms discussed above.

6. The residual echo suppressing filter (RESF)

The impact of the ECF on the acoustical echo is limited by at least two factors: only echoes due to the linear part of the transfer function of the LEMS can be cancelled, and the order of the ECF is typically much smaller than the order of the LEMS. Therefore, the RESF is introduced to reduce the echo further. In contrary to the ECF, the RECF lies in the path of the output signal (see Figure 1). Therefore, its influence on the locally generated speech signal cannot be neglected.

The RECF is basically a Wiener filter with some mainly heuristically justified modifications. The problem to be solved is to estimate the residual echo that is not directly measurable. It can be calculated from the input signal and the difference of the transfer functions of the LEMS and the ECF. As such, the estimation task is shifted to estimating the transfer function of the LEMS [22, 23].

The problem of residual echo suppression is often treated simultaneously with the problem of noise reduction [24, 25].

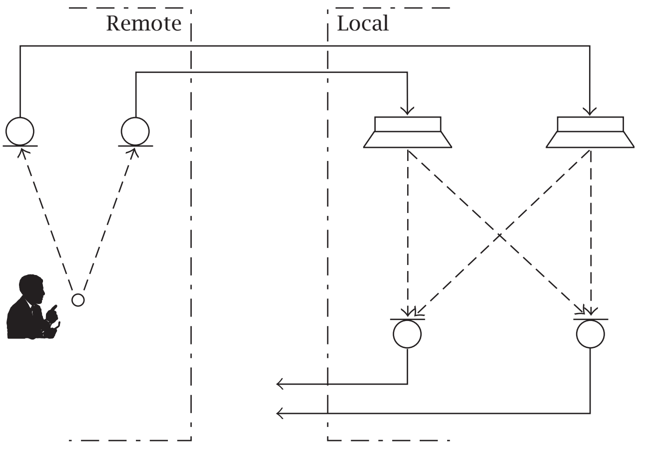

Figure 5: Signal paths in a stereophonic telecommunication system.

7. Echo cancellation for stereophonic systems

In *stereophonic telecommunication systems*, there are four ECFs (per location) necessary to model the four echo paths between left and right loudspeakers and left and right microphones (see Figure 5). In addition to providing the increased processing power, a problem specific to stereophonic systems has to be solved: the signals on the left and the right channels originate from the same signal source and differ only by the convolution with the impulse responses of the transmission paths between the signal source and the left and right microphones at the remote location. Typically, both impulse responses exhibit minimal phase components that are invertible. Therefore, the impulse responses of the ECFs are not uniquely determined and do not necessarily converge to the real impulse responses of the related echo paths. Preprocessing of the loudspeaker input signals is required in order to "separate" both signals. A number of methods to achieve this goal have been suggested. The methods suggested have the common property that they deliberately modify the audio signals such as adding signals that are nonlinear functions, adding half-wave rectifications of these signals [26], inserting of a periodically varying delay [27, 28], and adding noise such that it is masked by the speech signal [29].

8. Conclusions

In the opinion of the author, single-channel acoustic echo control is very well understood and single-channel solutions that provide acceptable speech quality are available. Further work, however, is still necessary. Multichannel systems are certainly a "hot topic." Solutions for stereophonic systems that do not need to disturb the speech signal are desirable. Microphone arrays (in monophonic transmission) can use spatial information to improve the ratio between "useful" local speech and echo and local noise. The mounting of more than one microphone in a single small housing breaks the argument of high wiring cost. In single-channel systems, nonlinear ECFs [30] can remove the demand for a linear LEMS and, as such, allow the use of less expensive loudspeakers and microphones.

The continuous progress in signal processing hardware allows highly sophisticated procedures to be implemented at an affordable cost. The author of this paper started to work on acoustic echo cancellation problems about twenty years ago. At this time a multiprocessor system running up to 16 signal processors, with top performance at that time, was necessary to implement some basic algorithms. Today, simple low-cost processors can do this job with a fraction of its capacity.

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SIGNAL PROCESSING

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Signal Processing is an interdisciplinary journal presenting the theory and practice of signal processing. Its primary objectives are the following:

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Editors-in-Chief

EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

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The overall aim of EURASIP Journal on Applied Signal Processing (EURASIP JASP) is to bring science and applications together with emphasis on practical aspects of signal processing in new and emerging technologies. It is directed as much at the practicing engineers as at the academic researchers. EURASIP JASP will highlight the diverse applications of signal processing and encourage a cross-fertilization of techniques. All papers should attempt to bring theory to life with practical simulations and examples. Tutorial articles on topics of interest are also welcomed. EURASIP JASP employs paperless, electronic review process to foster fast and speedy turnaround in review process.

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- Signal processing for communications and networking
- Statistical and adaptive signal processing
- Nonlinear signal processing techniques
- Signal processing design tools
- Signal processing for security, authentication, and cryptography
- Analog signal processing
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Editor-in-Chief

Marc Moonen, Department of Electrical Engineering, Katholieke Universiteit Leuven, ESAT-SISTA, Kasteelpark Arenberg 10, B-3001 Heverlee, Belgium

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Subjects

Subject areas include, but are by no means limited to: Ad-hoc networks; Channel modeling and propagation; Detection, estimation, and synchronization; Diversity and space-time techniques; End-to-end design techniques; Error control coding; Iterative techniques for joint optimization; Modulation techniques (CDMA, OFDM, multicarrier, spread-spectrum, etc.); Multiuser, MIMO channels, and multiple access schemes; Network performance, reliability, and quality of service; Resource allocation over wireless networks Security, authentication, and cryptography; Signal processing techniques and tools; Ultra wide-band systems; Wireless network services and medium access control.

Editor-in-Chief

EURASIP Journal on Applied Signal Processing, Volume 2003, No. 7, 1 June 2003

Best Paper Award 2001–2002

The EURASIP Advisory Committee has recently decided to install and support an *Annual Best Paper Award* for the EURASIP Journal on Applied Signal Processing, in recognition of the continued growth of the journal as well as the quality of the papers it publishes.

The first EURASIP JASP Best Paper Award covers the period 2001–2002, where over 160 papers were published. From 2003 on, the award will be given each year. Both the 2001–2002 and the 2003 awards will be presented at the EUSIPCO-2004 Awards Ceremony, September 2004.

The 2001–2002 Best Paper Award Committee was appointed by former Editor-in-Chief *Prof. K. J. Ray Liu* (University of Maryland, College Park, Md, USA) and consisted of EURASIP JASP Editorial Board Members *Prof. Bastiaan Kleijn*, Chair of the Committee (Royal Institute of Technology, Stockholm, Sweden), *Prof. Phillip Regalia* (Institut National des Télécommunications, Evry, France), and *Prof. Hideaki Sakai* (Kyoto University, Kyoto, Japan).

From a short list of five outstanding papers nominated by the members of the Editorial Board and/or Guest Editors of the special issues, the Award Committee unanimously decided to give the 2001–2002 Best Paper Award to the paper entitled *Design and DSP Implementation of Fixed-Point Systems* by *Martin Coors, Holger Keding, Olaf Lüthje, and Heinrich Meyr* which appeared in EURASIP JASP, vol. 2002, no. 9 (September 2002), pp. 908–925.

I sincerely congratulate the authors for this award, and at the same time I wish to thank all the other EURASIP JASP authors for submitting their fine papers to our journal. I hope this award will be a true stimulus for everyone to continue to view EURASIP JASP as a proper place to publish his/her research results.

I would also like to thank the EURASIP AdCom and especially EURASIP President Prof. Ferran Marqués for their support, Prof. Giovanni L. Sicuranza for his smooth coordination of the annual EURASIP Best Paper Award activities, and last but by no means least the EURASIP JASP Award Committee for their outstanding selection work.

Marc Moonen
Editor-in-Chief

Design and DSP Implementation of Fixed-Point Systems

Martin Coors, Holger Keding, Olaf Lüthje, and Heinrich Meyr

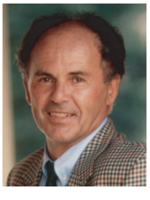
This article is an introduction to the FRIDGE design environment which supports the design and DSP implementation of fixed-point digital signal processing systems. We present the tool-supported transformation of signal processing algorithms coded in floating-point

ANSI C to a fixed-point representation in SystemC. We introduce the novel approach to control and data flow analysis, which is necessary for the transformation. The design environment enables fast bit-true simulation by mapping the fixed-point algorithm to integral data types of the host machine. A speedup by a factor of 20 to 400 can be achieved compared to C++-library-based bit-true simulation. FRIDGE also provides a direct link to DSP implementation by processor specific C code generation and advanced code optimization.

Martin Coors received the diploma in electrical engineering from Aachen University of Technology (RWTH), Aachen, Germany. In 1997, he joined the Institute for Integrated Signal Processing Systems (ISS) at RWTH Aachen as a research assistant. His research interests include DSP code optimization techniques, fixed-point design methodologies, and code generation for embedded processors.

Olaf Lüthje received the diploma in electrical engineering from Aachen University of Technology (RWTH), Aachen, Germany, and is currently working towards the Ph.D. degree in electrical engineering at the same institute. His research interests focus on fixed-point design methodology and data flow analysis.

Holger Keding received the diploma in electrical engineering from Aachen University of Technology (RWTH), Aachen, Germany. From 1996 to 2001 he was with ISS to work towards his Ph.D. thesis. Having finished his Ph.D., he joined the system level design group of Synopsys as a senior corporate application engineer. His research interests include fast bit-true simulation and fixed-point and system-level design methodology.



Heinrich Meyr received his M.S. and Ph.D. from ETH Zurich, Switzerland. He spent over 12 years in various research and management positions in industry before accepting a professorship in electrical engineering at Aachen University of Technology (RWTH Aachen) in 1977. He has worked extensively in the areas of communication theory, synchronization, and digital signal processing for the last thirty years. His research has been applied to the design of many industrial products. At RWTH Aachen he heads an institute involved in the analysis and design of complex signal processing systems for communication applications. He was a cofounder of CADIS GmbH (acquired 1993 by Synopsys, Mountain View, California), a company which commercialized the tool suite COSSAP extensively worldwide used in industry. He is a member of the Board of Directors of two companies in the communications industry. Dr. Meyr has published numerous IEEE papers. He is author together with Dr. G. Ascheid of the book "Synchronization in Digital Communications," Wiley 1990, and of the book "Digital Communication Receivers." He is also the author of "Synchronization, Channel Estimation, and Signal Processing" (together with Dr. M. Moeneclaey and Dr. S. Fechtel), Wiley, October 1997. He holds many patents. He served as a Vice President for International Affairs of the IEEE Communications Society and is a Fellow of the IEEE.

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Special Issue on Neuromorphic Signal Processing and Implementations

Shihab A. Shamma

Department of Electrical and Computer Engineering and Center for Auditory and Acoustic Research, Institute for Systems Research, University of Maryland, College Park, MD 20742, USA
Email: sas@eng.umd.edu

André van Schaik

School of Electrical and Information Engineering, University of Sydney, Sydney, NSW 2006, Australia
Email: andre@ee.usyd.edu.au

Neuromorphic engineering is a novel direction in Bioengineering that is based on the design and fabrication of artificial neural systems, such as vision chips, head-eye systems, auditory processors, and autonomous robots, whose physical architecture and design principles are based on those of biological nervous systems. The understanding of the brain and the application of that knowledge for health and technology will be one of the major research activities of the 21st century.

Neuromorphic engineering applies principles found in biological organisms to perform tasks that biological systems execute seemingly without effort, but which have been proven difficult to solve using traditional engineering techniques. These problems include visual navigation, auditory localization, olfaction, recognition, compliant limb control, and locomotion. The principles that biological organisms employ are still under investigation. For this reason, neuromorphic engineering is closely related to biological research, especially research in computational neuroscience. Neuromorphic engineering contributes to our understanding of biological systems by formulating and testing hypotheses of biological organization in fully functional synthetic systems. The aim of this research is to build a new generation of intelligent systems that interact with the real world much as animals do. The possible intellectual rewards and practical applications of this research are obviously very significant.

To some extent, "Bionics," popular in the 1960s, can be seen as a precursor to neuromorphic engineering. It emphasized the solutions that biology had found for a host of practical problems, and proposed to emulate those solutions. At the time, the focus was on biological materials, such as skin and muscles, rather than on trying to understand the detailed computational architecture and the algorithms used by the brain. Bionics disappeared from view, primarily due to a lack of detailed knowledge about biological systems and the lack of a suitable technology to implement biological strategies.

In the early 1980s, Carver Mead at Caltech, a pioneer of very large scale integrated (VLSI) circuit design, started to think about how integrated circuits could be used to emulate and understand neurobiology. What was different to the previous attempts was firstly, the tremendous growth in our knowledge of the nervous system and secondly, the existence of a mature electronics industry that could reliably and cheaply integrate a few million transistors and related structures onto a square centimeter of silicon. Indeed, the width of elementary features on a state-of-the-art very large scale integrated (VLSI) circuit is now entering the 100-nanometer domain, comparable to the average diameter of a cortical axon.

Although we are now able to integrate a few hundred million transistors on a single piece of silicon, our ideas of how to use these transistors have changed very little from the time when John von Neumann first proposed the architecture for the programmable serial computer. The serial machine was designed at a time when digital switching elements were large and fragile. Memory was also problematic and was stored by material unrelated to the computational devices. These constraints were consistent with a computer architecture based on a single active processor and a physically distant memory store. The constraints under which the serial machine was developed are no longer entirely relevant. On the contrary, the assumptions implicit in the traditional digital computational paradigm may now be limiting the computational power of integrated circuit technology.

A primary feature of the majority of integrated circuits is the representation of numbers as binary digits. Binary digits are useful because it is not difficult to standardize the performance of transistors, which are physical analog devices, to the extent that their state can be reliably determined to a single bit of accuracy. Analog computing is potentially more dense, because a single electrical node can represent multiple bits of information. Of course, analog computation is old news to engineers of the 1940s and 1950s. At that time, digital computers, where still too cumbersome to be used for many practical problems and engineers, resorted to analog computers that occupied entire rooms. However, once the digital computer became easy to reprogram and reasonably fast and small, it replaced analog technology. Today analog computers represent, for the main part, lab curiosities.

Analog computing is difficult because the physics of the material used to construct the machine plays an important role in the solution of the problem. It is difficult to control the physical properties of micrometer-sized devices such that their analog characteristics are well matched. The matching of analog device characteristics is the major difficulty facing an analog designer, and digital machines have an advantage over analog ones when high precision is required. Nevertheless, it is surprising that the high precision computation possible with modern computing is necessary to deal with real-world tasks in which the precision of the measurement of the data is often only a few bits. At the end of his life, von Neumann wrote a fascinating book, entitled *The Computer and the Brain*, in which he points out that the precision of the modern digital computer is entirely mismatched to the precision of the data, but it is necessary because errors in representation may multiply at each stage of the computation. In a digital computer, every bit of every number of the computation is fully restored and numbers are represented to many bits of accuracy to prevent the growth of error as the computation proceeds. The brain, in contrast, seems to use an analog representation with restoration at the action-potential output of the neuron. A typical active neuron firing rate is less than 100 spikes/second, so a neuron only has very few bits of precision. Nevertheless, they compute accurately enough for a wide range of computationally intensive sensorimotor tasks. One of the mysteries that neuromorphic engineering is trying to solve is how biological systems can compute so exactly using low precision components. The key appears to lie in the circuit architectures of neural systems, which aggregate information over a broad area and use feedback to provide an adaptation signal to all of the components of the system.

Although we do not fully understand the detailed circuits of neurobiological systems, their gross parallel architecture is clearly different from the serial computer architecture established by von Neumann. Serial computation remains the dominant form in digital computers because it executes tasks in a well-specified order and regularizes the problem of organization and communication. Parallel computers have been built, but have not gained widespread use due to the difficulty of programming them. Fine-grained parallel systems present nearly intractable problems for state-of-the-art engineering. Complex systems in which many processes interact are virtually designed using a trail-and-error method. For example, the boot sequence for a certain well-known modern aircraft is not a reproducible event; it is empirically determined that it will be complete sometime within fifteen minutes of initiation! Although they are not presently widely used, parallel systems have advantages over serial ones. Parallel systems have distributed local control and memory and can be faster and more fault tolerant than serial systems. Fault tolerance is important for integrated circuits because the number of transistors that can be integrated on a single silicon surface is limited by errors in manufacture that introduce flaws in the circuitry. Since digital computation demands perfect performance from every element in the system, chips with flaws cannot be used and wafer-scale integration, while physically achievable, is not practical for serial digital machines. Local memory and processing minimizes the amount of communication but requires that the task is to be organized in accordance with the machine architecture.

With the recognition that neurobiology has solved many difficult computational and sensorimotor control problems, it is believed that we can improve our technology by directly learning from biology. Yet, learning from biology brings problems of its own. In particular, the detailed forms of the biological solutions are difficult to analyze. An important reason for this is that the complexity of neuronal processing, particularly as it relates to system organization and function, is essentially nonlinear and so requires special methods of explanation that go beyond simple description and dissection. One successful method of explaining system function is to synthesize working models that integrate well-understood subelements into functional units. Such models attempt to characterize the operation of the brain at various levels, from synapses through behaving systems. Some of these models simply provide a compact ordering of our knowledge about a particular problem by detailed simulations. Others abstract the computational principles used by the neurons, and so are often framed within an engineering and physics paradigm.

This special issue of EURASIP JASP contains some examples of models representing the current state of neuromorphic signal processing. The issue starts with a low-level look at implementing neurons and synapses, and ends in a high-level application of classification of EEGs for brain-computer interfaces. In between we look at signal processing based on our current understanding of the auditory system and the visual system. Five papers in this issue concern the auditory system, starting at the cochlea, working its way up the auditory nerve, through the brainstem to the auditory cortex. The three vision papers present high fill-factor imagers, binocular perception of motion-in-depth, and color segmentation and pattern matching.

The guest editors would like to thank all the authors for their work in submitting and revising manuscripts. We also thank all the reviewers for their effort in writing reviews and their feedback to the authors.

Shihab A. Shamma
André van Schaik

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Contents and Abstracts

Analog VLSI Circuits for Short-Term Dynamic Synapses

Shih-Chii Liu

http://dx.doi.org/10.1155/S1110865703302094

Short-term dynamical synapses increase the computational power of neuronal networks. These synapses act as additional filters to the inputs of a neuron before the subsequent integration of these signals at its cell body. In this work, we describe a model of depressing and facilitating synapses derived from a hardware circuit implementation. This model is equivalent to theoretical models of short-term synaptic dynamics in network simulations. These circuits have been added to a network of leaky integrate-and-fire neurons. A cortical model of direction-selectivity that uses short-term dynamic synapses has been implemented with this network.

An FPGA-Based Electronic Cochlea

M. P. Leong, Craig T. Jin, and Philip H. W. Leong

http://dx.doi.org/10.1155/S1110865703303038

A module generator which can produce an FPGA-based implementation of an electronic cochlea filter with arbitrary precision is presented. Although hardware implementations of electronic cochlea models have traditionally used analog VLSI as the implementation medium due to their small area, high speed, and low power consumption, FPGA-based implementations offer shorter design times, improved dynamic range, higher accuracy, and a simpler computer interface. The tool presented takes filter coefficients as input and produces a synthesizable VHDL description of an application-optimized design as output. Furthermore, the tool can use simulation test vectors in order to determine the appropriate scaling of the fixed-point precision parameters for each filter. The resulting model can be used as an accelerator for research in audition or as the front-end for embedded auditory signal processing systems. The application of this module generator to a real-time cochleagram display is also presented.

An Analogue VLSI Implementation of the Meddis Inner Hair Cell Model

Alistair McEwan and André van Schaik

http://dx.doi.org/10.1155/S1110865703303026

The Meddis inner hair cell model is a widely accepted, but computationally intensive computer model of mammalian inner hair cell function. We have produced an analogue VLSI implementation of this model that operates in real time in the current domain by using translinear and log-domain circuits. The circuit has been fabricated on a chip and tested against the Meddis model for (a) rate level functions for onset and steady-state response,

(b) recovery after masking, (c) additivity, (d) two-component adaptation, (e) phase locking, (f) recovery of spontaneous activity, and (g) computational efficiency. The advantage of this circuit, over other electronic inner hair cell models, is its nearly exact implementation of the Meddis model which can be tuned to behave similarly to the biological inner hair cell. This has important implications on our ability to simulate the auditory system in real time. Furthermore, the technique of mapping a mathematical model of first-order differential equations to a circuit of log-domain filters allows us to implement real-time neuromorphic signal processors for a host of models using the same approach.

Analog VLSI Models of Range-Tuned Neurons in the Bat Echolocation System

Matthew Cheely and Timothy Horiuchi

http://dx.doi.org/10.1155/S1110865703302045

Bat echolocation is a fascinating topic of research for both neuroscientists and engineers, due to the complex and extremely time-constrained nature of the problem and its potential for application to engineered systems. In the bat's brainstem and midbrain exist neural circuits that are sensitive to the specific difference in time between the outgoing sonar vocalization and the returning echo. While some of the details of the neural mechanisms are known to be species-specific, a basic model of reafference-triggered, postinhibitory rebound timing is reasonably well supported by available data. We have designed low-power, analog VLSI circuits to mimic this mechanism and have demonstrated range-dependent outputs for use in a real-time sonar system. These circuits are being used to implement range-dependent vocalization amplitude, vocalization rate, and closest target isolation.

Sparse Spectrotemporal Coding of Sounds

David J. Klein, Peter König, and Konrad P. Körding

http://dx.doi.org/10.1155/S1110865703303051

Recent studies of biological auditory processing have revealed that sophisticated spectrotemporal analyses are performed by central auditory systems of various animals. The analysis is typically well matched with the statistics of relevant natural sounds, suggesting that it produces an optimal representation of the animal's acoustic biotope. We address this topic using simulated neurons that learn an optimal representation of a speech corpus. As input, the neurons receive a spectrographic representation of sound produced by a peripheral auditory model. The output representation is deemed optimal when the responses of the neurons are maximally sparse. Following optimization, the simulated neurons are similar to real neurons in many respects. Most notably, a given neuron only analyzes the input over a localized region of time and frequency. In addition, multiple subregions either excite or inhibit the neuron, together producing selectivity to spectral and temporal modulation patterns. This suggests that the brain's solution is particularly well suited for coding natural sound; therefore, it may prove useful in the design of new computational methods for processing speech.

Joint Acoustic and Modulation Frequency

Les Atlas and Shihab A. Shamma

http://dx.doi.org/10.1155/S1110865703305013

There is a considerable evidence that our perception of sound uses important features which are related to underlying signal modulations. This topic has been studied extensively via perceptual experiments, yet there are few, if any, well-developed signal processing methods which capitalize on or model these effects. We begin by summarizing evidence of the importance of modulation representations from psychophysical, physiological, and other sources. The concept of a two-dimensional joint acoustic and modulation frequency representation is proposed. A simple single sinusoidal amplitude modulator of a sinusoidal carrier is then used to illustrate properties of an unconstrained and ideal joint representation. Added constraints are required to remove or reduce undesired interference terms and to provide invertibility. It is then noted that the constraints would be also applied to more general and complex cases of broader modulation and carriers. Applications in single-channel speaker separation and in audio coding are used to illustrate the applicability of this joint representation. Other applications in signal analysis and filtering are suggested.

High Fill-Factor Imagers for Neuromorphic Processing Enabled by Floating-Gate Circuits

Paul Hasler, Abhishek Bandyopadhyay, and David V. Anderson

http://dx.doi.org/10.1155/S1110865703303105

In neuromorphic modeling of the retina, it would be very nice to have processing capabilities at the focal plane while retaining the density of typical active pixel sensor (APS) imager designs. Unfortunately, these two goals have been mostly incompatible. We introduce our transform imager technology and basic architecture that uses analog floating-gate devices to make it possible to have computational imagers with high pixel densities. This imager approach allows programmable focal-plane processing that can perform retinal and higher-level bioinspired computation. The processing is performed continuously on the image via programmable matrix operations that can operate on the entire image or blocks within the image. The resulting dataflow architecture can directly perform computation of spatial transforms, motion computations, and stereo computations. The core imager performs computations at the pixel plane, but still holds a fill factor greater than 40 percent—comparable to the high fill factors of APS imagers. Each pixel is composed of a photodiode sensor element and a multiplier. We present experimental results from several imager arrays built in 0.5 micrometer process (up to 128×128 in an area of 4 millimeter squared).

Phase-Based Binocular Perception of Motion in Depth: Cortical-Like Operators and Analog VLSI Architectures

Silvio P. Sabatini, Fabio Solari, Paolo Cavalleri, and Giacomo Mario Bisio

http://dx.doi.org/10.1155/S1110865703302033

We present a cortical-like strategy to obtain reliable estimates of the motions of objects in a scene toward/away from the observer (motion in depth), from local measurements of binocular parameters derived from direct comparison of the results of monocular

spatiotemporal filtering operations performed on stereo image pairs. This approach is suitable for a hardware implementation, in which such parameters can be gained via a feedforward computation (i.e., collection, comparison, and punctual operations) on the outputs of the nodes of recurrent VLSI lattice networks, performing local computations. These networks act as efficient computational structures for embedded analog filtering operations in smart vision sensors. Extensive simulations on both synthetic and real-world image sequences prove the validity of the approach that allows to gain high-level information about the 3D structure of the scene, directly from sensorial data, without resorting to explicit scene reconstruction.

A Vision Chip for Color Segmentation and Pattern Matching

Ralph Etienne-Cummings, Philippe Pouliquen, and M. Anthony Lewis

http://dx.doi.org/10.1155/S1110865703302021

A 128(H) $\times$ 64(V) $\times$ RGB CMOS imager is integrated with region-of-interest selection, RGB-to-HSI transformation, HSI-based pixel segmentation, (36bins $\times$ 12bits)-HSI histogramming, and sum-of-absolute-difference (SAD) template matching. Thirty-two learned color templates are stored and compared to each image. The chip captures the R, G, and B images using in-pixel storage before passing the pixel content to a multiplying digital-to-analog converter (DAC) for white balancing. The DAC can also be used to pipe in images for a PC. The color processing uses a biologically inspired color opponent representation and an analog lookup table to determine the Hue (H) of each pixel. Saturation (S) is computed using a loser-take-all circuit. Intensity (I) is given by the sum of the color components. A histogram of the segments of the image, constructed by counting the number of pixels falling into 36 Hue intervals of 10 degrees, is stored on a chip and compared against the histograms of new segments using SAD comparisons. We demonstrate color-based image segmentation and object recognition with this chip. Running at 30 fps, it uses 1 mW. To our knowledge, this is the first chip that integrates imaging, color segmentation, and color-based object recognition at the focal plane.

Joint Time-Frequency-Space Classification of EEG in a Brain-Computer Interface Application

Gary N. Garcia Molina, Touradj Ebrahimi, and Jean-Marc Vesin

http://dx.doi.org/10.1155/S1110865703302082

Brain-computer interface is a growing field of interest in human-computer interaction with diverse applications ranging from medicine to entertainment. In this paper, we present a system which allows for classification of mental tasks based on a joint time-frequency-space decorrelation, in which mental tasks are measured via electroencephalogram (EEG) signals. The efficiency of this approach was evaluated by means of real-time experimentations on two subjects performing three different mental tasks. To do so, a number of protocols for visualization, as well as training with and without feedback, were also developed. Obtained results show that it is possible to obtain good classification of simple mental tasks, in view of command and control, after a relatively small amount of training, with accuracies around 80%, and in real time.

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Foreword

David E. Goldberg

Department of General Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801, USA
Email: deg@uiuc.edu

I was delighted when I was asked to write a foreword to this special issue on genetic algorithms (GAs) and evolutionary computation (EC) in image and signal processing edited by Riccardo Poli and Stefano Cagnoni for two reasons. First, the special issue is another piece of the mounting evidence that GAs and EC are finding an important niche in the solution of difficult real-world problems. Second, in reviewing the contents of the special issue, I find it almost archetypal in its reflection of the GA/EC applications world of 2003. In the remainder of this discussion, I briefly review a number of reasons why genetic and evolutionary techniques are becoming more and more important in real problems and discuss some of the ways this issue used to both demonstrate effective GA/EC application and foreshadow more signal and image processing by evolutionary and genetic means.

There are a number of reasons why GAs and EC are becoming more prevalent in real applications. The first reason is what I call *the buzz*. Let us face it, GAs are cool. The very idea of doing a Darwinian survival of the fittest and genetics on a computer is neat. But *cool* and *neat*, while they may attract our *attention*, do not merit our *sustained* involvement.

Another reason for which GAs have become more popular is the *motivation from artificial systems*. Although decades, even centuries, of optimization and operations research leave us with an impressive toolkit, the contingency basis of the methodology leaves us somewhat cold. By this I mean that the selection of an optimization technique or OR is *contingent* on the type of problem you face. If you have a linear problem with linear constraints, you choose linear programming. If you have a stage decomposable problem, you choose dynamic programming. If you have a nonlinear problem with sufficiently pleasant constraints, you choose nonlinear programming, and so on. But the very nature of this list of methods that work in particular problems is part of the problem. One of the promises of biologically inspired techniques is a *framework* that does not vary and a larger class of problems that can be tackled within that framework.

This vision of greater *robustness* is now being realized, but it is tied to whether the solutions obtained using these techniques are both *tractable* and *practical*. Results about a decade ago showed that simple GAs in common practice had a kind of Dr. Jekyll and Mr. Hyde nature. Simple genetic and evolutionary algorithms work well (subquadratically) on straightforward problems, but they require exponential times on more complex ones. This is not the place to review these results in detail, and the interested reader can look elsewhere

(D. E. Goldberg, *The Design of Innovation: Lessons from and for Competent Genetic Algorithms*, Kluwer, Boston, 2002) but it suffices to say that work on *adaptive* and *self-adaptive* crossover and mutation operators is overcoming the tractability hurdle on real problems, resulting in what appears to be broadly scalable (subquadratic) or *competent* solvers.

Yet, theoretical tractability is of little solace to a practitioner who faces the daunting prospect of performing a million costly function evaluations on a 1000-variable problem. As a result, increasing theory, implementation, and application are showing the way toward *principled efficiency enhancement* using parallelization, time utilization, hybridization, and evaluation relaxation, and these methods are moving us from the realm of the competent (the tractable) to the realm of the practical.

These fundamental reasons—the buzz, the need, the tractability, and the practicality of modern genetic and evolutionary algorithms—are driving an ever-increasing interest in these methods, and this volume reflects that range of interest in terms of the application areas, operators, codings, and accoutrements on display.

In terms of application, the use of GAs and EC in this volume spans such disparate applications as filter tuning, sensor planning, system identification, object detection, bioinformatic image processing, 3D model interpretation, and speech recognition. The range of different applications here is a reflection of the breadth of application elsewhere, and the utility of the GA/EC toolkit across this landscape is empirical evidence of the robustness of these methods.

Looking under the hood, we see a wide range of codings and operators in evidence, from floating-point vectors to permutations to program codes, from fixed to adaptive operators, and from crossover to mutation with various competitive or clustered (or niched) selection mechanisms. Additionally, many of the papers here demonstrate an understanding of the importance of efficiency enhancement in real-world problems, and a number of them combine the best of genetic and evolutionary computation with local search to form useful and efficient hybrids that solve the problem. Too often, methods specialists are enamored with the method they helped invent or perfect, but in the real world, efficient solutions are obtained with an effective combine of global and local techniques.

In all, this special issue is a useful compendium for those interested in signal and image processing and the proper application of genetic and evolutionary methods to the unsolved problems of these domains. To the field of genetic and evolutionary computation, this special issue is a growing evidence of the importance of what that field does in areas of human endeavor that matter. To audience members in both camps, I recommend without reservation that you study this special issue, and absorb and apply its many lessons.

David E. Goldberg

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Special Issue on

Genetic and Evolutionary Computation for Signal Processing and Image Analysis

Riccardo Poli

Department of Computer Science, University of Essex, Wivenhoe Park, Colchester, CO4 3SQ, UK
Email: rpoli@essex.ac.uk

Stefano Cagnoni

Department of Computer Engineering, University of Parma, 43100 Parma, Italy
Email: cagnoni@ce.unipr.it

1. INTRODUCTION

Darwinian evolution is probably the most intriguing and powerful mechanism of nature mankind has ever discovered. Its power is evident in the impressive level of adaptation reached by all species of animals and plants in nature. It is intriguing because despite its simplicity and randomness, it produces incredible complexity in a way that appears to be very directed, almost purposeful. Like for other powerful natural phenomena, it is no surprise that several decades ago a few brilliant researchers in engineering and computer science started wondering whether they could steal the secrets behind Darwinian evolution and use them to solve problems of practical interest in a variety of application domains. Those people were pioneers of a new field which, after more than 30 years from its inception, is now big, well established, and goes under the name of *genetic and evolutionary computation* (GEC).

An almost endless number of results and applications of evolutionary algorithms have been reported in the literature, showing that the ideas of these pioneers were indeed right. Nowadays, evolutionary techniques can routinely solve problems in domains such as automatic design, optimisation, pattern recognition, control, and many others. Until recently, however, only very occasionally could one claim that GEC techniques approached the performance of human experts in these same domains, particularly in the case of large scale applications and complex engineering problems. This is why, initially, successful applications of GEC techniques to the fields of computer vision, image analysis, and signal processing were few and far in between. Towards the late 1990s, however, the research interest in these areas seemed to be rapidly growing, and the time seemed right for the creation of an infrastructure that could foster the interaction between researchers in this area. This is what led, in early 1998, the two editors of this special issue, together with people from various other European institutions, to create a working group of the European Network of Excellence in Evolutionary Computation, entirely devoted to the applications of evolutionary algorithms to image analysis and signal processing. The working group organises a regular meeting, the European Workshop on Evolutionary Computation in Image Analysis and Signal Processing (EvoIASP), which reached its fifth edition this year. This event gives European and non-European researchers, as well as people from industry, an opportunity to present their latest research, discuss current developments and applications, besides fostering closer interaction between members of the GEC, image analysis, and signal processing scientific communities. However, the event is, and intends to remain, a workshop. Therefore, the work presented there can never have the depth allowed by more substantial and mature archival publications such as this journal.

This special issue of EURASIP JASP on GEC for signal processing and image analysis, being the first of its kind, has offered computer scientists and engineers from around the world a unique opportunity to submit their best mature research for inclusion in this unified, high-quality "venue." The timing of this special issue could not have been better; well over thirty papers were submitted by contributors from around the world. The papers were reviewed by a pool of over thirty international expert reviewers. Only about one third passed our strict criteria for acceptance and are now in this volume.

The rest of this editorial is organised as follows. In Section 2, we will provide a gentle introduction to the basics of evolutionary computation. In Section 3, we describe each of the papers present in this special issue, briefly summarising, for each one, the problem considered and the evolutionary techniques adopted to tackle it. In Section 4, we provide our final remarks and acknowledgments, while in the appendix, we give a brief commented bibliography with suggested further reading.

2. EVOLUTIONARY COMPUTATION: THE BASICS

What were the main secrets behind Darwinian evolution, that the pioneers of GEC stole to make them the propelling fuel of evolutionary computation processes?

Inheritance

Individuals have a genetic representation (in nature, the chromosomes and the DNA) such that it is possible for the offspring of an individual to inherit some of the features of its parent.

Variation

The offsprings are not exact copies of the parents, but instead, reproduction involves mechanisms that create innovation as new generations are born.

Natural selection

Individuals best adapted to the environment have longer life and higher chances of mating and spreading their genetic makeup.

Clearly, there is a lot more to natural evolution than these main forces. However, like for many other nature-inspired techniques, not all the details are necessary to obtain working models of a natural system. The three ingredients listed above are in fact sufficient to obtain

Table 1: Nature-to-computer mapping at the basis of EAs.

Nature	*Computer*
Individual	Solution to a problem
Population	Set of solutions
Fitness	Quality of a solution
Chromosome	Representation for a solution (e.g., set of parameters)
Gene	Part of the representation of a solution (e.g., parameter or degree of freedom)
Crossover	Search operator
Mutation	Search operator
Natural selection	Promoting the reuse of good (sub-)solutions

artificial systems showing the main characteristic of natural evolution, *the ability to search for highly fit individuals*.

For all these ingredients (representation, variation, and selection), one can focus on different realisations. For example, in nature, variation is produced both through mutations of the genome and through the effect of sexually recombining the genetic material coming from the parents when obtaining the offspring chromosomes (*crossover*). This is why many different classes of evolutionary algorithms have been proposed over the years. So, depending on the structures undergoing evolution, on the reproduction strategies and the variation (or *genetic*) operators adopted, and so on, evolutionary algorithms can be grouped into several evolutionary paradigms: genetic algorithms (GAs) [1], genetic programming (GP) [2], evolution strategies (ESs) [3, 4], and so forth.

All the inventors of these different evolutionary algorithms (EAs) have to make choices as to which bits of nature have a corresponding component in their algorithms. These choices are summarised in the nature-to-computer mapping shown in Table 1. That is, the notion of individual in nature corresponds to a tentative solution to a problem of interest in an EA. The fitness (ability to reproduce and have fertile offsprings that reach the age of reproduction) of natural individuals corresponds to the objective function used to evaluate the quality of the tentative solutions in the computer. The genetic variation processes of mutation and recombination are seen as mechanisms (search operators) to generate new tentative solutions to the problem. Finally, natural selection is interpreted as a mechanism to promote the diffusion and mixing of the genetic material of individuals representing good quality solutions and therefore having the potential to create even fitter individuals (better solutions).

Despite their differences, most EAs have the following general form.

(1) Initialise population and evaluate the fitness of each population member.
(2) Repeat.
  (a) Select subpopulation for reproduction on the basis of fitness (*selection*).
  (b) Copy some of the selected individuals without change (*cloning* or *reproduction*).
  (c) Recombine the "genes" of selected parents (*recombination* or *crossover*).
  (d) Mutate the mated population stochastically (*mutation*).
  (e) Evaluate the fitness of the new population.
  (f) Select the survivors based on fitness.

Not all these steps are present in all EAs. For example, in modern GAs [5] and in GP phase, (a) is part of phases (b) and (c), while phase (f) is absent. This algorithm is said to be *generational* because there is no overlap between generations (i.e., the offspring population always replaces the parent population). In generational EAs, cloning is used to simulate the survival of parents for more than one generation.

In the following, we will analyse the various components of an EA in more detail, mainly concentrating on the GA, although most of what we will say also applies to other paradigms.

2.1. Representations

Traditionally, in GAs, solutions are encoded as binary strings. Typically, an adult individual (a solution for a problem) takes the form of a vector of numbers. These are often interpreted as parameters (for a plant, for a design, etc.), but in combinatorial optimisation problems, these numbers can actually represent configurations, choices, schedules, paths, and so on. Anything that can be represented on a digital computer can also be represented in a GA using a binary representation. This is why, at least in principle, GAs have a really broad applicability. However, other nonbinary representations are available, which may be more suitable, for example, for problems with real-valued parameters.

Because normally the user of a GA has no ideas as to what constitutes a good initial set of choices/parameters for adult individuals (tentative solutions to a problem), the chromosomes to be manipulated by the GA are normally initialised in an entirely random manner. That is, the initial population is a set of random binary strings or of random real-valued vectors.

2.2. Selection in GAs

Selection is the operation by which individuals (i.e., their chromosomes) are selected for mating or cloning. To emulate natural selection, individuals with a higher fitness should be selected with higher probability. There are many models of selection, some of which, despite fitting well the biologically inspired computational model and producing effective results, are not biologically plausible. We briefly describe them below.

Fitness proportionate selection, besides being the most direct translation into the computational model of the probabilistic principles of evolution, is probably the most widely used selection scheme. This works as follows. Let N be the population size, f_i the fitness of individual i, and $\bar{f} = (1/N)\sum_j f_j$ the average population fitness. Then, in fitness proportionate selection, individual i is selected for reproduction with a probability

$$p_i = \frac{f_i}{\sum_j f_j} = \frac{f_i}{\bar{f}N}. \tag{1}$$

In normal GAs, populations are not allowed to grow or shrink, so N individuals have to be selected for reproduction. Therefore, the *expected number* of selected copies of each individual is

$$N_i = p_i N = \frac{f_i}{\bar{f}}. \tag{2}$$

So, individuals with an above-average quality ($f_i > \bar{f}$) tend to be selected more than once for mating or cloning, while individuals below the average tend not to be used.

Tournament selection, instead, works as follows. To select an individual, first a group of T ($T \geq 2$) random individuals is created. Then the individual with the highest fitness in the group is selected, the others are discarded (tournament).

Another alternative is *rank selection* where individuals are first sorted (ranked) on the ground of their fitness so that if an individual i has fitness $f_i > f_j$, then its rank is $i < j$. Then each individual is assigned a probability of being selected p_i taken from a given distribution (typically a monotonic rapidly decreasing function), with the constraint that $\sum_i p_i = 1$.

2.3. Operators

EAs work well only if their genetic operators allow an efficient and effective search of the space of tentative solutions.

One desirable property of recombination operators is to guarantee that two parents sharing a useful common characteristic always transmit such a characteristic to their offspring. Another important property is to also guarantee that different characteristics distinguishing two parents may be all inherited by their offspring. For binary GAs, there are many crossover operators with these properties.

One-point crossover, for example, aligns the two parent chromosomes (bit strings), then cuts them at a randomly chosen common point and exchanges the right-hand side (or left-hand side) subchromosomes (see Figure 1a). In *two-point crossover*, chromosomes are cut at two randomly chosen crossover points and their ends are swapped (see Figure 1b). A more modern operator, *uniform crossover*, builds the offspring, one bit at a time, by randomly selecting one of the corresponding bits from the parents (see Figure 1c).

Normally, crossover is applied to the individuals of a population with a constant probability p_c (often $p_c \in [0.5, 0.8]$). Cloning is then applied with a probability $1 - p_c$ to keep the number of individuals in each generation constant.

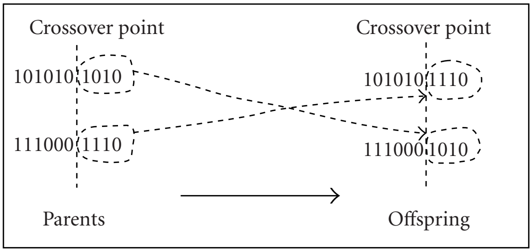
Mutation is the second main genetic operator used in GAs. A variety of mutation operators exist. Mutation typically consists of making (usually small) alterations to the values of one or more genes in a chromosome. Mutation is often applied to the individuals produced by crossover and cloning before they are added to the new population. In binary chromosomes, mutation often consists of inverting random bits of the genotypes (see Figure 2). The main goal with which mutation is applied is preservation of diversity, which helps GAs to explore as much of the search space as possible. However, due to its random nature, mutation may have disruptive effects onto evolution if it occurs too often. Therefore, in GAs, mutation is usually applied to genes with a very low probability.

In real-valued GAs, chromosomes have the form $x = \langle x_1, \ldots, x_\ell \rangle$ where each gene x_i is represented by a floating-point number. In these GAs, crossover is often seen as an interpolation process in a multidimensional Euclidean space. So, the components of the offspring o are calculated from the corresponding components of the parents p' and p'' as follows:

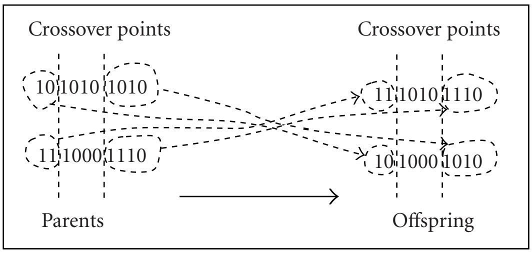
$$o_i = p'_i + r(p''_i - p'_i), \tag{3}$$

where r is a random number in the interval $[0, 1]$ (see Figure 3a). Alternatively, crossover can be seen as the exploration of a multidimensional hyperparallelepiped defined by the parents (see Figure 3b), that is, the components o_i are chosen uniformly at random within the intervals

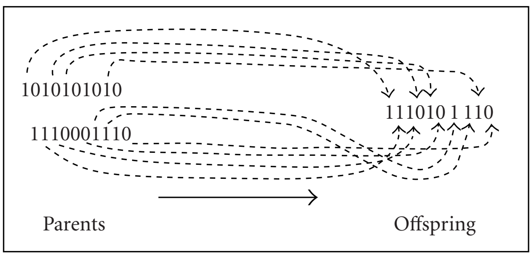
$$[\min(p'_i, p''_i), \max(p'_i, p''_i)]. \tag{4}$$



(a)



(b)



(c)

Figure 1: Three crossover operators for binary GAs: (a) one-point crossover, (b) two-point crossover, and (c) uniform crossover.

Mutation is often seen as the addition of a small random variation (e.g., Gaussian noise) to a point in a multidimensional space (see Figure 3c).

2.4. Other GEC paradigms

As mentioned before, the principles on which GAs are based are also shared by many other EAs. However, the use of different representations and operators has led to the development of a number of paradigms, each having its own peculiarities. With no pretence of being exhaustive, in the following, we will shortly mention those paradigms, other than GAs, that are used in the papers included in this special issue.

Genetic programming [2, 6] is a variant of GA in which the individuals being evolved are syntax trees, typically representing computer programs. The trees are created using

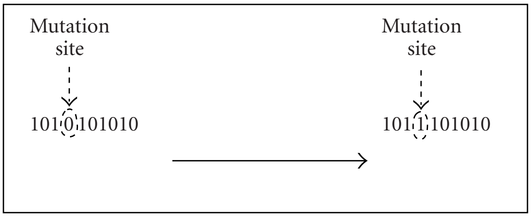


Figure 2: Bitwise mutation in binary GAs.

user-defined primitive sets, which typically include input variables, constants, and a variety of functions or instructions. The syntax trees are manipulated by specialised forms of crossover and mutation that guarantee the syntactic validity of the offspring. The fitness of the individual trees in the population is evaluated by running the corresponding programs (typically multiple times, for different values of their input variables).

Evolution strategies [3, 4] are real-valued EAs where mutation is the key variation operator (unlike GAs where crossover plays that role). Mutation typically consists of adding zero-mean Gaussian deviates to the individuals being optimised, with the mutation standard deviation being varied dynamically so as to maximise the performance of the algorithm.

Artificial immune systems (see [7, Part III, Chapters 10–13] or [8] for an extensive introduction) are distributed computational systems inspired by biological immune systems, which can recognise patterns and can remember previously seen patterns in an efficient and effective way. These systems are very close relatives of EAs (sometimes involving an evolutionary process in their inner mechanics) although they use a different biological metaphor.

3. THE PAPERS IN THIS SPECIAL ISSUE

In their paper entitled *Blind search for optimal Wiener equalizers using an artificial immune network model*, Attux et al. exploit recent advances in the field of artificial immune systems to obtain optimum equalisers for noisy communication channels, using a technology that does not require the availability of clean samples of the input signal. This approach is very successful in a variety of test equalisation problems. The approach is also compared with a more traditional EA, a GA with niching, showing superior performance.

The paper by Dunn and Olague, entitled *Evolutionary computation for sensor planning*, shows how well-designed evolutionary computation techniques can solve the problem of optimally specifying sensing tasks for a workcell provided with multiple manipulators and cameras. The problem is NP hard, effectively being a composition of a set partitioning problem and multiple traveling salesperson problems. Nonetheless, thanks to clever representations and the use of evolutionary search, this system is able to solve the problem, providing solutions of quality very close to that of the solutions obtained via exhaustive search, but in a tiny fraction of the time.

The paper entitled *An evolution approach for joint blind multichannel estimation and order detection* by Fangjiong et al. presents a method for the detection of the order and the estimation of the parameters of a single-input multiple-output channel. The method is based on a hybrid GA with specially designed operators. The method shows performances com-

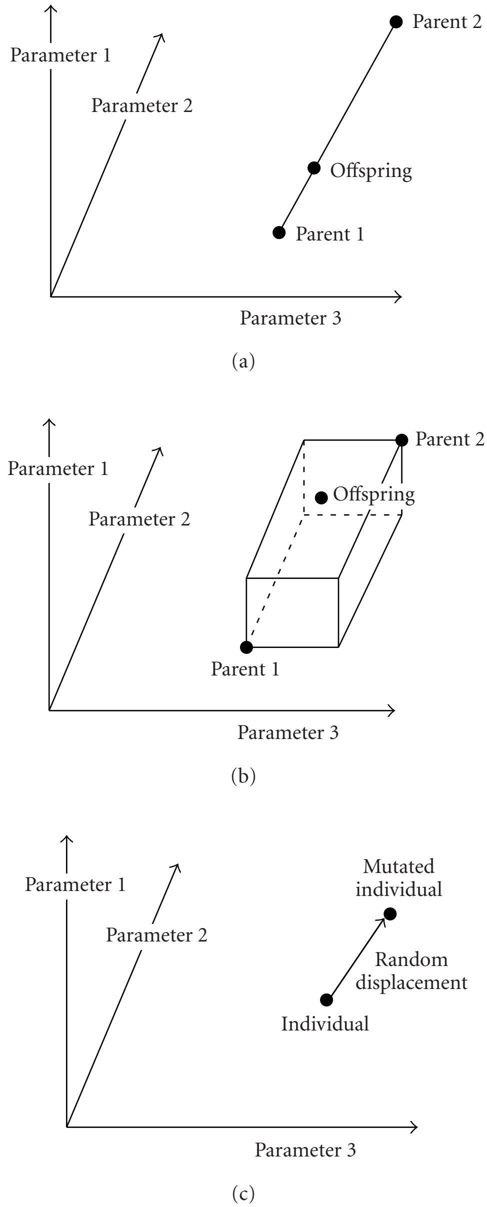


Figure 3: (a), (b) crossover operators and (c) mutation for real-valued GAs.

parable with existing closed-form approaches which, however, are much more restricted in that they either assume that the channel order is known or treat the problems of order estimation and parameter estimation separately.

In *Application of evolution strategies to the design of tracking filters with a large number of specifications*, Herrero et al. attack the problem of tracking civil aircrafts from radar information within the extremely tight performance constraints imposed by a civil aviation standard. They use interactive multiple mode filters optimised by using an ES and a multiobjective optimisation approach obtaining a high-performance aircraft tracker.

Making EAs more at hand and easy to apply for general practitioners by self-tuning their parameters is one of the main aims with which Pignalberi et al. developed GASE, a GA-based tool for range image segmentation. The system, along with some practical results, is described in the paper *Tuning range image segmentation by genetic algorithm*. A multiobjective fitness function is adopted to take into consideration problems that are typically encountered in range image segmentation.

The paper *Parameter estimation of a plucked string synthesis model using a genetic algorithm with perceptual fitness calculation* describes the use of GAs to estimate the control parameters for a widely used plucked string synthesis model. Using GAs, Riionheimo and Välimäki have been able to automate parameter extraction, which had been formerly achieved only through semiautomatic approaches, obtaining comparable results, both in quantitative and in qualitative terms. An interesting feature of the approach is the inclusion of knowledge about perceptual properties of the human hearing system into the fitness function.

Schell and Uhl compare results obtained with a GA-based approach to the near-best-basis (NBB) algorithm, a well-known suboptimal algorithm for wavelet packet decomposition. In their paper *Optimization and assessment of wavelet packet decompositions with evolutionary computation*, they highlight the problem of finding good cost functions in terms of correlation with actual image quality. They show that GAs provide lower-cost solutions that, however, provide lower-quality images than NBB.

In the paper entitled *On the use of evolutionary algorithms to improve the robustness of continuous speech recognition systems in adverse conditions*, Selouani and O'Shaughnessy show how a GA can tune a system based on state-of-the-art speech recognition technology so as to maximise its recognition accuracy in the presence of severe noise. This hybrid of evolution and conventional signal processing algorithms amply outperforms nonadaptive systems. The EA used is a GA with real-coded representation, rank selection, a heuristic type of crossover, and a nonuniform mutation operator.

The paper *Evolutionary techniques for image processing a large dataset of early Drosophila gene expression* by Spirov and Holloway describes an evolutionary approach to image processing to process confocal microscopy images of patterns of activity for genes governing early Drosophila development. The problem is approached using plain GAs, a simplex approach, and a hybrid between these two.

The use of GAs to track time-varying systems based on recursive models is tackled in *A comparison of evolutionary algorithms for tracking time-varying recursive systems*. The paper first compares a plain GA with a GA variant, called "random immigrant strategy," showing that the latter performs better in tracking time-varying systems even if it has problems with fast-varying systems. Finally, a hybrid combination of GAs and local search that is able to tackle even such hard tasks is proposed.

Zhang et al., in their paper *A domain-independent window approach to multiclass object detection using genetic programming*, propose an interesting approach in which GP is used to both detect and localise features of interest. The approach is compared with a neural network classifier, used as reference, showing that GP evolved programs can provide significantly lower false-alarm rates. Within the proposed approach, the choice of the primitive set is also discussed, comparing results obtained with two different sets: one comprises only the four basic arithmetical operators, and the other includes also transcendental functions. The results reported in the paper provide interesting clues to practitioners that would like to use GP to tackle image processing tasks.

4. CONCLUSIONS

The guest editors hope that the readership of the journal will enjoy reading the papers in this special issue as we did ourselves. We hope that the broadness of domains to which EAs can be applied, demonstrated by the contents of this issue, will convince other researchers in image analysis and signal processing to get acquainted with the exciting world of evolutionary computation and to apply its powerful techniques to solve important new and old problems in these areas.

APPENDIX

POINTERS TO FURTHER READING IN GEC

1. David E. Goldberg, *Genetic Algorithms in Search, Optimization, and Machine Learning*. Addison-Wesley, Reading, Massachusetts, 1989. A classic book on genetic algorithms and classifier systems.
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7. John R. Koza, *Genetic Programming: On the Programming of Computers by Means of Natural Selection*. MIT Press, 1992. The bible of genetic programming by the founder of the field. Followed by GP II (1994), GP III (1999), and GP IV (forthcoming).
8. Wolfgang Banzhaf, Peter Nordin, Robert E. Keller and Frank D. Francone, *Genetic Programming—An Introduction; On the Automatic Evolution of Computer Programs and its Applications*, Morgan Kaufmann, 1998. An excellent textbook on GP.
9. W. B. Langdon and Riccardo Poli, *Foundations of Genetic Programming*, Springer, February 2002. The only book entirely devoted to the theory of GP and its relations with the GA theory.
10. Proceedings of the International Conference on Genetic Algorithms (ICGA). ICGA is the oldest conference on EAs.

11. Proceedings of the Genetic Programming Conference. This was the first conference entirely devoted to GP.
12. Proceedings of the Genetic and Evolutionary Computation Conference (GECCO). Born in 1999 from the "recombination" of ICGA and the GP conference mentioned above, GECCO is the largest conference in the field.
13. Proceedings of the Foundations of Genetic Algorithms (FOGA) Workshop. FOGA is a biannual, small but very prestigious and highly selective workshop. It is mainly devoted to the theoretical foundations of EAs.
14. Proceedings of the Congress on Evolutionary Computation (CEC). CEC is a large conference under the patronage of IEEE.
15. Proceedings of Parallel Problem Solving from Nature (PPSN). This is a large biannual European conference, probably the oldest of its kind in Europe.
16. Proceedings of the European Workshop on Evolutionary Computation in Image Analysis and Signal Processing (EvoIASP). This is a small workshop, reaching its fifth edition in 2003. It is the only event worldwide uniquely devoted to the research topics covered by this special issue.
17. Proceedings of the European Conference on Genetic Programming. EuroGP was the first European event entirely devoted to GP. Run as a workshop in 1998 and 1999, it became a conference in 2000. It has now reached its sixth edition with EuroGP 2003 held at the University of Essex. Currently, this is the largest event worldwide solely devoted to GP.

ACKNOWLEDGMENTS

The guest editors would like to thank Professor David E. Goldberg for his insightful foreword, the former and present editors-in-chief of EURASIP JASP, Professor K. J. Ray Liu and Professor Marc Moonen, for their support in putting together this special issue, and all the reviewers who have generously devoted their time to help ensure the highest possible quality for the papers in this volume. All the authors of the manuscripts who have contributed to this special issue are also warmly thanked.

Riccardo Poli
Stefano Cagnoni

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Volume 2003, No. 8, 1 July 2003
Contents and Abstracts

Blind Search for Optimal Wiener Equalizers Using an Artificial Immune Network Model

Romis Ribeiro de Faissol Attux, Murilo Bellezoni Loiola, Ricardo Suyama, Leandro Nunes de Castro, Fernando José Von Zuben, and João Marcos Travassos Romano

http://dx.doi.org/10.1155/S1110865703303014

This work proposes a framework to determine the optimal Wiener equalizer by using an artificial immune network model together with the constant modulus (CM) cost function. This study was primarily motivated by recent theoretical results concerning the CM criterion and its relation to the Wiener approach. The proposed immune-based technique was tested under different channel models and filter orders, and benchmarked against a procedure using a genetic algorithm with niching. The results demonstrated that the proposed strategy has a clear superiority when compared with the more traditional technique. The proposed algorithm presents interesting features from the perspective of multimodal search, being capable of determining the optimal Wiener equalizer in most runs for all tested channels.

Evolutionary Computation for Sensor Planning: The Task Distribution Plan

Enrique Dunn and Gustavo Olague

http://dx.doi.org/10.1155/S1110865703303075

Autonomous sensor planning is a problem of interest to scientists in the fields of computer vision, robotics, and photogrammetry. In automated visual tasks, a sensing planner must make complex and critical decisions involving sensor placement and the sensing task specification. This paper addresses the problem of specifying sensing tasks for a multiple manipulator workcell given an optimal sensor placement configuration. The problem is conceptually divided in two different phases: *activity assignment* and *tour planning*. To solve such problems, an optimization methodology based on evolutionary computation is developed. Operational limitations originated from the workcell configuration are considered using specialized heuristics as well as a floating-point representation based on the random keys approach. Experiments and performance results are presented.

An Evolutionary Approach for Joint Blind Multichannel Estimation and Order Detection

Chen Fangjiong, Sam Kwong, and Wei Gang

http://dx.doi.org/10.1155/S1110865703304068

A joint blind order-detection and parameter-estimation algorithm for a single-input multiple-output (SIMO) channel is presented. Based on the subspace decomposition of the

channel output, an objective function including channel order and channel parameters is proposed. The problem is resolved by using a specifically designed genetic algorithm (GA). In the proposed GA, we encode both the channel order and parameters into a single chromosome, so they can be estimated simultaneously. Novel GA operators and convergence criteria are used to guarantee correct and high convergence speed. Simulation results show that the proposed GA achieves satisfactory convergence speed and performance.

Application of Evolution Strategies to the Design of Tracking Filters with a Large Number of Specifications

Jesús García Herrero, Juan A. Besada Portas, Antonio Berlanga de Jesús, José M. Molina López, Gonzalo de Miguel Vela, and José R. Casar Corredera

http://dx.doi.org/10.1155/S1110865703302057

This paper describes the application of evolution strategies to the design of interacting multiple model (IMM) tracking filters in order to fulfill a large table of performance specifications. These specifications define the desired filter performance in a thorough set of selected test scenarios, for different figures of merit and input conditions, imposing hundreds of performance goals. The design problem is stated as a numeric search in the filter parameters space to attain all specifications or at least minimize, in a compromise, the excess over some specifications as much as possible, applying global optimization techniques coming from evolutionary computation field. Besides, a new methodology is proposed to integrate specifications in a fitness function able to effectively guide the search to suitable solutions. The method has been applied to the design of an IMM tracker for a real-world civil air traffic control application: the accomplishment of specifications defined for the future European ARTAS system.

Tuning Range Image Segmentation by Genetic Algorithm

Gianluca Pignalberi, Rita Cucchiara, Luigi Cinque, and Stefano Levialdi

http://dx.doi.org/10.1155/S1110865703303087

Several range image segmentation algorithms have been proposed, each one to be tuned by a number of parameters in order to provide accurate results on a given class of images. Segmentation parameters are generally affected by the type of surfaces (e.g., planar versus curved) and the nature of the acquisition system (e.g., laser range finders or structured light scanners). It is impossible to answer the question, which is the best set of parameters given a range image within a class and a range segmentation algorithm? Systems proposing such a parameter optimization are often based either on careful selection or on solution space-partitioning methods. Their main drawback is that they have to limit their search to a subset of the solution space to provide an answer in acceptable time. In order to provide a different automated method to search a larger solution space, and possibly to answer more effectively the above question, we propose a tuning system based on genetic algorithms.

A complete set of tests was performed over a range of different images and with different segmentation algorithms. Our system provided a particularly high degree of effectiveness in terms of segmentation quality and search time.

Parameter Estimation of a Plucked String Synthesis Model Using a Genetic Algorithm with Perceptual Fitness Calculation

Janne Riionheimo and Vesa Välimäki

http://dx.doi.org/10.1155/S1110865703302100

We describe a technique for estimating control parameters for a plucked string synthesis model using a genetic algorithm. The model has been intensively used for sound synthesis of various string instruments but the fine tuning of the parameters has been carried out with a semiautomatic method that requires some hand adjustment with human listening. An automated method for extracting the parameters from recorded tones is described in this paper. The calculation of the fitness function utilizes knowledge of the properties of human hearing.

Optimization and Assessment of Wavelet Packet Decompositions with Evolutionary Computation

Thomas Schell and Andreas Uhl

http://dx.doi.org/10.1155/S111086570330407X

In image compression, the wavelet transformation is a state-of-the-art component. Recently, wavelet packet decomposition has received quite an interest. A popular approach for wavelet packet decomposition is the near-best-basis algorithm using nonadditive cost functions. In contrast to additive cost functions, the wavelet packet decomposition of the near-best-basis algorithm is only suboptimal. We apply methods from the field of evolutionary computation (EC) to test the quality of the near-best-basis results. We observe a phenomenon: the results of the near-best-basis algorithm are inferior in terms of cost-function optimization but are superior in terms of rate/distortion performance compared to EC methods.

On the Use of Evolutionary Algorithms to Improve the Robustness of Continuous Speech Recognition Systems in Adverse Conditions

Sid-Ahmed Selouani and Douglas O'Shaughnessy

http://dx.doi.org/10.1155/S1110865703302070

Limiting the decrease in performance due to acoustic environment changes remains a major challenge for continuous speech recognition (CSR) systems. We propose a novel approach which combines the Karhunen-Loève transform (KLT) in the mel-frequency domain with a genetic algorithm (GA) to enhance the data representing corrupted speech. The idea con-

sists of projecting noisy speech parameters onto the space generated by the genetically optimized principal axis issued from the KLT. The enhanced parameters increase the recognition rate for highly interfering noise environments. The proposed hybrid technique, when included in the front-end of an HTK-based CSR system, outperforms that of the conventional recognition process in severe interfering car noise environments for a wide range of signal-to-noise ratios (SNRs) varying from 16 dB to −4 dB. We also showed the effectiveness of the KLT-GA method in recognizing speech subject to telephone channel degradations.

Evolutionary Techniques for Image Processing a Large Dataset of Early *Drosophila* Gene Expression

Alexander Spirov and David M. Holloway

http://dx.doi.org/10.1155/S1110865703302069

Understanding how genetic networks act in embryonic development requires a detailed and statistically significant dataset integrating diverse observational results. The fruit fly (*Drosophila melanogaster*) is used as a model organism for studying developmental genetics. In recent years, several laboratories have systematically gathered confocal microscopy images of patterns of activity (expression) for genes governing early *Drosophila* development. Due to both the high variability between fruit fly embryos and diverse sources of observational errors, some new nontrivial procedures for processing and integrating the raw observations are required. Here we describe processing techniques based on genetic algorithms and discuss their efficacy in decreasing observational errors and illuminating the natural variability in gene expression patterns. The specific developmental problem studied is anteroposterior specification of the body plan.

A Comparison of Evolutionary Algorithms for Tracking Time-Varying Recursive Systems

Michael S. White and Stuart J. Flockton

http://dx.doi.org/10.1155/S1110865703303117

A comparison is made of the behaviour of some evolutionary algorithms in time-varying adaptive recursive filter systems. Simulations show that an algorithm including random immigrants outperforms a more conventional algorithm using the breeder genetic algorithm as the mutation operator when the time variation is discontinuous, but neither algorithms performs well when the time variation is rapid but smooth. To meet this deficit, a new hybrid algorithm which uses a hill climber as an additional genetic operator, applied for several steps at each generation, is introduced. A comparison is made of the effect of applying the hill climbing operator a few times to all members of the population or a larger number of times solely to the best individual; it is found that applying to the whole population yields the better results, substantially improved compared with those obtained using earlier methods.

A Domain-Independent Window Approach to Multiclass Object Detection Using Genetic Programming

Mengjie Zhang, Victor B. Ciesielski, and Peter Andreae

http://dx.doi.org/10.1155/S1110865703303063

This paper describes a domain-independent approach to the use of genetic programming for object detection problems in which the locations of small objects of multiple classes in large images must be found. The evolved program is scanned over the large images to locate the objects of interest. The paper develops three terminal sets based on domain-independent pixel statistics and considers two different function sets. The fitness function is based on the detection rate and the false alarm rate. We have tested the method on three object detection problems of increasing difficulty. This work not only extends genetic programming to multiclass-object detection problems, but also shows how to use a single evolved genetic program for both object classification and localisation. The object classification map developed in this approach can be used as a general classification strategy in genetic programming for multiple-class classification problems.

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Contents and Abstracts

MPEG-4 Authoring Tool Using Moving Object Segmentation and Tracking in Video Shots

Petros Daras, Ioannis Kompatsiaris, Ilias Grinias, Georgios Akrivas, Georgios Tziritas, Stefanos Kollias, and Michael G. Strintzis

http://dx.doi.org/10.1155/S1110865703301052

An Authoring tool for the MPEG-4 multimedia standard integrated with image sequence analysis algorithms is described. MPEG-4 offers numerous capabilities and is expected to be the future standard for multimedia applications. However, the implementation of these capabilities requires a complex authoring process, employing many different competencies from image sequence analysis and encoding of audio/visual/BIFS to the implementation of different delivery scenarios: local access on CD/DVD-ROM, Internet, or broadcast. However powerful the technologies underlying multimedia computing are, the success of these systems depends on their ease of authoring. In this paper, a novel Authoring tool fully exploiting the object-based coding and 3D synthetic functionalities of the MPEG-4 standard is described. It is based upon an open and modular architecture able to progress with MPEG-4 versions and it is easily adaptable to newly emerging better and higher-level authoring and image sequence analysis features.

Face Detection Using a First-Order RCE Classifier

Byeong Hwan Jeon, Kyoung Mu Lee, and Sang Uk Lee

http://dx.doi.org/10.1155/S1110865703304123

We present a new face detection algorithm based on a first-order reduced Coulomb energy (RCE) classifier. The algorithm locates frontal views of human faces at any degree of rotation and scale in complex scenes. The face candidates and their orientations are first determined by computing the Hausdorff distance between simple face abstraction models and binary test windows in an image pyramid. Then, after normalizing the energy, each face candidate is verified by two subsequent classifiers: a binary image classifier and the first-order RCE classifier. While the binary image classifier is employed as a preclassifier to discard nonfaces with minimum computational complexity, the first-order RCE classifier is used as the main face classifier for final verification. An optimal training method to construct the representative face model database is also presented. Experimental results show that the proposed algorithm yields a high detection ratio while yielding no false alarm.

An Efficient Feature Extraction Method with Pseudo-Zernike Moment in RBF Neural Network-Based Human Face Recognition System

Javad Haddadnia, Majid Ahmadi, and Karim Faez

http://dx.doi.org/10.1155/S1110865703305128

This paper introduces a novel method for the recognition of human faces in digital images using a new feature extraction method that combines the global and local information in frontal view of facial images. Radial basis function (RBF) neural network with a hybrid learning algorithm (HLA) has been used as a classifier. The proposed feature extraction method includes human face localization derived from the shape information. An efficient distance measure as facial candidate threshold (FCT) is defined to distinguish between face and nonface images. Pseudo-Zernike moment invariant (PZMI) with an efficient method for selecting moment order has been used. A newly defined parameter named axis correction ratio (ACR) of images for disregarding irrelevant information of face images is introduced. In this paper, the effect of these parameters in disregarding irrelevant information in recognition rate improvement is studied. Also we evaluate the effect of orders of PZMI in recognition rate of the proposed technique as well as RBF neural network learning speed. Simulation results on the face database of Olivetti Research Laboratory (ORL) indicate that the proposed method for human face recognition yielded a recognition rate of 99.3%.

Lapped Block Image Analysis via the Method of Legendre Moments

Hakim El Fadili, Khalid Zenkouar, and Hassan Qjidaa

http://dx.doi.org/10.1155/S1110865703305062

Research investigating the use of Legendre moments for pattern recognition has been performed in recent years. This field of research remains quite open. This paper proposes a new technique based on block-based reconstruction method (BBRM) using Legendre moments compared with the global reconstruction method (GRM). For alleviating the blocking artifact involved in the processing, we propose a new approach using lapped block-based reconstruction method (LBBRM). For the problem of selecting the optimal number of moment used to represent a given image, we propose the maximum entropy principle (MEP) method. The main motivation of the proposed approaches is to allow fast and efficient reconstruction algorithm, with improvement of the reconstructed images quality. A binary handwritten musical character and multi-gray-level "Lena" image are used to demonstrate the performance of our algorithm.

The PLSI Method of Stabilizing Two-Dimensional Nonsymmetric Half-Plane Recursive Digital Filters

N. Gangatharan and P. S. Reddy

http://dx.doi.org/10.1155/S1110865703303099

Two-dimensional (2D) recursive digital filters find application in image processing as in medical X-ray processing. Nonsymmetric half-plane (NSHP) filters have definitely positive

magnitude characteristics as opposed to quarter-plane (QP) filters. In this paper, we have provided methods for stabilizing the given 2D NSHP polynomial by the planar least squares inverse (PLSI) method. We have proved in this paper that if the given 2D unstable NSHP polynomial and its PLSI are of the same degree, the PLSI polynomial is always stable, irrespective of whether the coefficients of the given polynomial have relationship among its coefficients or not. Examples are given for 2D first-order and second-order cases to prove our results. The generalization is done for the Nth order polynomial.

Channel Effect Compensation in LSF Domain

An-Tze Yu and Hsiao-Chuan Wang

http://dx.doi.org/10.1155/S111086570330513X

This study addresses the problem of channel effect in the line spectrum frequency (LSF) domain. LSF parameters are the popular speech features encoded in the bit stream for low bit-rate speech transmission. A method of channel effect compensation in LSF domain is of interest for robust speech recognition on mobile communication and Internet systems. If the bit error rate in the transmission of digital encoded speech is negligibly low, the channel distortion comes mainly from the microphone or the handset. When the speech signal is represented in terms of the phase of inverse filter derived from LP analysis, this channel distortion can be expressed in terms of the channel phase. Further derivation shows that the mean subtraction performed on the phase of inverse filter can minimize the channel effect. Based on this finding, an iterative algorithm is proposed to remove the bias on LSFs due to channel effect. The experiments on the simulated channel distorted speech and the real telephone speech are conducted to show the effectiveness of our proposed method. The performance of the proposed method is comparable to that of cepstral mean normalization (CMN) in using cepstral coefficients.

Self-Tuning Blind Identification and Equalization of IIR Channels

Miloje Radenkovic, Tamal Bose, and Zhurun Zhang

http://dx.doi.org/10.1155/S111086570330410X

This paper considers self-tuning blind identification and equalization of fractionally spaced IIR channels. One recursive estimator is used to generate parameter estimates of the numerators of IIR systems, while the other estimates denominator of IIR channel. Equalizer parameters are calculated by solving Bezout type equation. It is shown that the numerator parameter estimates converge (a.s.) toward a scalar multiple of the true coefficients, while the second algorithm provides consistent denominator estimates. It is proved that the equalizer output converges (a.s.) to a scalar version of the actual symbol sequence.

Digital Audio for Multimedia Communications

Guest Editors: Gianpaolo Evangelista, Mark Kahrs, and Emmanuel Bacry

Publication Date: 3rd Quarter, 2003

Renewed interest in digital processing of audio signals has been stimulated by the introduction of multimedia communication via the Internet and digital broadcasting systems. These new applications demand high bandwidth and require innovative solutions to an old problem: how to achieve high quality at low bit rates. While modern psycho-acoustic theories are shedding new light on the assessment of audio quality—many issues are still open. Sophisticated time-frequency representations of signals have been introduced to increase flexibility and performance in coding. However, their use in audio coding is still at an early stage. The definition of suitable audio representations must be adapted to both models of sound and perception. Furthermore, the effect of quantization and the efficient quantization techniques must be carefully studied. Structured audio coding is another emerging paradigm for high quality music broadcasting. In structured coding, algorithms and synthesis parameters are transmitted rather than just encoded sound samples. While synthesis algorithms have been developed for many years, there is a great demand for procedures that automatically allocate synthesis resources and extract synthesis parameters, that is, fully automated structured coding. Other important issues include the development of new types of techniques for digital audio effects, realistic sound spatialization, sound restoration, and processing for multimedia electroacoustic transducers. Today's digital audio algorithms draw ideas from recent and past experience in synthesis and processing of music and speech signals, perceptual and physical models of sound and hearing, and transform coding and information theory. The key to success is to link several aspects of this multidisciplinary field together to develop new techniques or to improve existing methodologies. These new goals can strongly impact classical audio processing tools.

The aim of this special issue is to present state-of-the-art signal processing algorithms and methods for digital audio signal processing in the context of broadcasting, either via radio or the Internet.

Signal Processing for Acoustic Communication Systems

Guest Editors: Kees Janse, Walter Kellermann, Marc Moonen, and Piet Sommen

Publication Date: 3rd Quarter, 2003

The commercial application of advanced acoustic communication systems has become feasible in recent years due to the vast increase of available computational power. In new and future acoustic communication systems it is expected that people want to create virtual acoustic communication links that give conversation partners the impression of being in the same acoustic environment. Besides providing quality and robustness, these future acoustic communication systems should exploit the growing computer power to design more flexible systems in which an acoustic interface is built that, on the one hand, acquires speech and sound perfectly and yet allows people to move freely around without wearing or holding a microphone. On the other hand, sound that is reproduced at the human's ears should sound such that, ideally, the local acoustic environment is masked and remote or virtual environments can be created in the human's perception. The amount of signal processing involved in future acoustic communication systems grows exponentially due to the demand for more and more advanced systems. For this reason there is an increased interest in more sophisticated algorithms that can deal with multiple-source signals, multiple microphones, and multiple loudspeakers running in real time on one or more digital signal processing cores.

Target application areas include hands-free user interfaces for acoustic communication systems that are required for rooms, offices, cars, multimedia computers, hearing aids, voiced controlled machinery, and so on. Typical signal processing algorithms that are required for these applications are echo cancelling, (blind) signal separation, noise reduction, and dereverberation.

The aim of this special issue is to highlight innovative research in signal processing for acoustic communication systems, thus paving the way for future developments in the field.

Multimedia Human-Computer Interface

Guest Editors: Ryohei Nakasu, Richard Reilly, and Tsuhan Chen

Publication Date: 4th Quarter, 2003

Human-computer interfaces are an essential research topic for the realization of sophisticated human-computer communications and computer-mediated human communications. Until recently, research in this area treated only a single medium, speech or images. Typical research examples are speech recognition and gesture recognition. Recently, however, because researchers have recognized the importance of multimedia interfaces and also because technologies that treat a single medium have advanced a great deal, human-computer interfaces based on multimedia have become one of the centers of interest.

Research on multimedia human-computer interfaces is expected to realize the following:

(1) Natural communications between humans and computers.
(2) Integration of recognition technologies that utilize a single medium.
(3) Computer-human interfaces using multimedia to create new types of communications between computers and humans as well as between humans themselves.
(4) Multimedia human-computer interfaces to become breakthroughs for situations experiencing communications difficulty, for example, communications between handicapped people and computers by the introduction of multimedia human-computer technologies.

The aim of this special issue is to bring together contributions from the latest developments in the field of multimedia human-computer interfaces.

Biometric Signal Processing

Guest Editors: Herve Bourlard, Kenneth K. M. Lam, Ioannis Pitas, and Yue Wang

Publication Date: 4th Quarter, 2003

Biometric signal processing is an emerging technology that enables the authentication, identification, or verification of an individual based on physiological, behavioral, and molecular characteristics. With the advancement of computer vision and pattern recognition techniques, together with high-speed computers, research related to biometrics has developed rapidly in the last several decades and has led to various applications. Biometric techniques include recognizing faces, hands, voices, signatures, irises, fingerprints, DNA patterns, etc. These enabling technologies for biometrics will play an important role in security, smart card, and personalized eCommerce applications. The analysis of biometric information is a challenging task, and a wide range of signal processing techniques need to be applied. The success of the applications heavily relies on the efficiency, reliability, and accuracy of these biometric signal processing techniques.

The aim of this special issue is to bring together researchers working on biometric signal processing and its applications with a particular emphasis on person authentication and identification.

Genomic Signal Processing

Guest Editors: Xiaodong Wang, Yidong Chen, Edward R. Dougherty, and Carsten Peterson

Publication Date: 4th Quarter, 2003

Genomic signal processing is the science of processing genomic signals. It is fundamental to functional genomics, for which two critical goals are the discovery of gene combinations that explain specific cellular phenotypes (disease) on a mechanistic level, and the use of genomic signals to classify phenotypes on a molecular level.

Owing to the advent of microarray technology to simultaneously assess gene-expression levels from thousands of genes, there now exists the practical potential to apply signal processing methods to expression-based signaling within the genome. This potential has already been grasped by the bioinformatics community, where theoretical work aims at prediction and network modeling, and application-oriented studies aim at classification based on genotype expression measurements. Expression prediction and genetic regulatory networks provide understanding of multivariate gene relations and have the long-run potential to place expression profiling within the framework of dynamical systems, and early studies demonstrate that expression-based classification holds out the hope of gene-based diagnosis for a variety of diseases.

This special issue focuses on the application of signal processing methods to genomic signal processing and the development of new signal processing tools for the specific problems related to genomics. In addition to classification and prediction, we are particularly interested in applications involving dynamical systems, information theory, communications theory, network models such as Bayesian networks and probabilistic Boolean networks, and mathematical strategies to alter regulatory behavior.

Multimedia over IP and Wireless Networks

Guest Editors: Zixiang Xiong, Mihaela van der Schaar, Jie Chen, Eckehard Steinbach, and C.-C. Jay Kuo

Publication Date: 1st Quarter, 2004

Multimedia—an integrated and interactive presentation of speech, audio, video, graphics, and text—have become a major driving force behind today's information technology that merges practices of communication, computing, and information processing into an interdisciplinary field. Meanwhile, Internet protocol (IP) is becoming the common denominator for multimedia services and wireless access grows very rapidly recently. However, the intrinsic natures of the IP and wireless networks impose some necessary trade-off between QoS guarantee and resources utilization efficiency. Therefore, multimedia over IP and wireless networks face many challenges.

Two special issues will be published to address the challenges of how to deliver multimedia applications in a cost effective, ubiquitous, and QoS levels/classes adaptive manner. One special issue will focus on multimedia over IP networks, another one on multimedia over wireless networks.

MIMO Communications and Signal Processing

Guest Editors: H. Vincent Poor, Sergio Barbarossa, Constantinos Papadias, and Xiaowen Wang

Publication Date: 1st Quarter, 2004

Multiple-input multiple-output (MIMO) architectures enable powerful techniques for improving the capacity of wireless communication systems, especially in rich multipath environments. In particular, it is now well known that in such rich scattering environments, the attainable capacity of MIMO links increases approximately linearly with the minimum of the number of antennas at the transmitter and the receiver. These potentially significant capacity improvements of MIMO systems become even more desirable as the next generation wireless communication systems (such as 3G, wireless LANs, and beyond), create demands for higher data rates for increasing user populations. Although a considerable amount of progress has been made on MIMO techniques over the past decade, many technical challenges remain before the potential capacity gains can be realized in practical systems. These challenges include some of the most intriguing problems in wireless communications and signal processing, such as space-time coding and decoding, optimal multiple access techniques for systems equipped with multiple antennas, multiuser detection, and adaptive space-time beamforming. New challenges also emerge within some classical signal processing problems, such as MIMO channel characterization, estimation and equalization. Meanwhile, the associated practical implementation issues of MIMO systems, such as receiver complexity, power dissipation, cost, as well as robust operation, are critical in making MIMO transceivers viable. Further, system-level performance characterization and optimization of MIMO systems, taking into account undesired signal interference, remains a largely unexplored topic. Finally, the desired re-configurable operation of MIMO transceivers across a wide range of channel, antenna, and interference parameters remains yet another important challenge in designing MIMO systems.

The aim of this special issue is to showcase recent research in both MIMO communications and MIMO signal processing together in the same forum, in order to present a range of perspectives and innovative results with potential to enable practical MIMO systems.

Object-Based and Semantic Image and Video Analysis

Guest Editors: Kiyoharu Aizawa, Thomas Huang, Stefanos Kollias, Petros Maragos, and Ralf Schaefer

Publication Date: 1st Quarter, 2004

Recent progress and prospects in cognitive vision, multimedia, human-computer interaction, communications, and the Web call for and can benefit from applications of advanced image and video analysis technologies. Adaptive robust systems are required for analysis, indexing and summarisation of large amounts of audio-visual data. Advanced image analysis technologies are needed for next-generation description and browsing services characterised by structured, object-based representations, and personalised information access. Automatic extraction of semantic information from still or moving images and the analysis of their content is necessary for automatic annotation, indexing, and categorisation.

The aim of this special issue is to bring together contributions from the latest developments in the field of object-oriented and semantic image and video analysis applications.

Signal Processing for Broadband Access Systems: Techniques and Implementations

Guest Editors: An-Yeu (Andy) Wu, Ut-Va Koc, and Keshab K. Parhi

Publication Date: 2nd Quarter, 2004

With the rapid growth of Internet access and voice/data-centric communications, many access technologies have been developed to meet the stringent demand of high-speed data transmission and bridge the wide bandwidth gap between ever-increasing high-data-rate core networks and bandwidth-hungry end-user networks. To make efficient utilization of the limited bandwidth of existing access routes and cope with the adverse channel environment, many standards have been proposed for a variety of broadband access systems over different access environments (twisted pairs, coaxial cables, optical fibers, and fixed or mobile wireless). These access environments may create different channel impairments and dictate unique sets of signal processing algorithms and techniques to combat specific impairments. In the design and implementation domain of those systems, many research issues arise. Firstly, multistandards coexist in many access environments. In addition, multimode or rate-adaptive designs are encountered frequently even within one single standard. Hence, dynamically configurable platform or multimode transceiver architectures need to be investigated for ubiquitous access. Secondly, among all layers of the access systems, application-specific accelerator designs are desired to either enhance the computational power or lower the burden of general-purpose MCU and DSP processors. Finally, most DSP algorithms in broadband access systems are computationally intensive in nature. With currently available processing power of DSP processors and area/speed/power/fixed-wordlength limitation of ASIC designs, some of the originally developed DSP algorithms cannot be feasibly implemented, or need to suffer from performance degradation due to the implementational constraint. Consequently, novel cost-efficient DSP techniques at the implementational level need to be developed to either meet the cost/speed/power criteria of hardware design or help to reduce the MIPS count in running DSP processors. The design goal is to achieve the targeted transmission rate under feasible implementational vehicles (FPGA, SOPC, SOC, etc.) in attempt to maintain the desired SNR performance.

The aim of this special issue is to present state-of-the-art signal processing techniques and implementation issues for broadband access systems over different access channels.

Multicarrier Communications and Signal Processing

Guest Editors: Ye (Geoffrey) Li, Hamid R. Sadjadpour, and Dirk Dahlhaus

Publication Date: 2nd Quarter, 2004

Multicarrier (MC) transmission, especially orthogonal frequency division multiplexing (OFDM), has recently attracted considerable attention since it has been shown to be an effective technique to combat delay spread or frequency selective fading of wireless or wireline channels. This approach has been adopted as the standards in several outdoor and indoor high-speed wireless and wireline data applications, including wireless local area networks, digital audio and video broadcasting, and digital subscriber line modems. MC transmission requires no equalizers, which makes it possible to combine with many advanced techniques to improve the capacity and enhance the performance of transmission. At the same time, many issues in MC communications, such as time- and frequency-offset estimation and correction, channel estimation, and peak-to-average power ratio (PAPR) reduction, need to be solved. This special issue focuses on multicarrier communications and signal processing.

Multimedia Security and Rights Management

Guest Editors: Min Wu, Nasir Memon, Touradj Ebrahimi, and Ingemar Cox

Publication Date: 3rd Quarter, 2004

The digital information revolution has brought about profound changes in our society and our lives. New devices and powerful software have made it possible for consumers worldwide to create, manipulate, share, and enjoy the multimedia data. Internet and wireless networks offer ubiquitous channels to deliver and to exchange multimedia information for such purposes as remote collaboration, distant learning, and entertainment. With all these advances in multimedia coding and communication technologies over the past decade, the major hurdle for allowing much broader access of multimedia assets and deployment of multimedia services no longer lies with bandwidth-related issues, but with assuring that the content is used for its intended purpose by its intended recipients. The core issue now becomes the development of secure management of multimedia content usage and delivery across communication networks.

The aim of this special issue is to bring together the contributions from the latest research and development in multimedia security and rights management.

Particle Filtering in Signal Processing

Guest Editors: Petar M. Djuric, Simon J. Godsill, and Arnaud Doucet

Publication Date: 3rd Quarter, 2004

Particle filtering is a Monte Carlo methodology that is used for nonlinear and non-Gaussian sequential signal processing. Its beginning can be traced back to the late 1940s and early 1950s, which were followed in the last fifty years with sporadic outbreaks of intense activity.

In the past few years, particle filtering has again gained the attention of scientists, statisticians, and engineers; and as a result, many new contributions have been reported. Although its implementation is computationally intensive, the widespread availability of fast computers and the amenability of the particle filtering methods for parallel implementation make them very attractive for solving difficult signal processing problems. The objective of this special issue is to present original research results on particle filtering and bring its vast scope of applications closer to the signal processing community.

Advances in Smart Antennas

Guest Editors: Andreas Czylwik, Alex Gershman, Thomas Kaiser

Publication Date: 3rd Quarter, 2004

Smart antennas have emerged as a key technology for third and higher generations of wireless communication systems because they add a new spatial dimension to the currently used time, frequency, and code multiple access technologies. The recent past of 3G wireless systems licensing process in Europe has shown that spectral bandwidth may cost billions of euros to wireless system providers. In light of this fact, smart antennas offer an elegant and relatively inexpensive opportunity of increasing system capacity, number of users served, and quality of service. Today, developments, and progress in this strategic area are far away from cost efficient practical implementation, and a large amount of both theoretical and experimental study is of great demand to enable future successful applications of smart antennas.

The aim of this special issue is to present recent research in smart antennas from multiple points of view with focus on future applications in the area of wireless communications.

Improved CDMA Detection Techniques for Future Wireless Systems

Guest Editors: Geert Leus, Philippe Loubaton, Dirk Slock, and Michael D. Zoltowski

Publication Date: 3rd Quarter, 2004

The past few years have been marked by a worldwide standardization activity for third generation (3G) wireless systems, which are intended to deliver high data rates and are expected to handle multimedia applications in addition to voice. The key multiple access technique that has been chosen for 3G wireless systems is CDMA.

The performance of 3G wireless systems might not be sufficient to meet the needs of future high-performance multimedia applications such as full-motion video and teleconferencing. Hence, there will be a need for systems that extend the capabilities of 3G wireless systems, sometimes referred to as fourth generation (4G) wireless systems, whatever they might be. It is likely that these wireless systems will retain a CDMA component, but compared to 3G wireless systems, the changed operating conditions will present a new set of challenges in the development of CDMA detection techniques.

The aim of this special issue is to cover present research in the development of improved CDMA detection techniques for future wireless systems.

Model-Based Sound Synthesis

Guest Editors: Vesa Välimäki, Augusto Sarti, Matti Karjalainen, Rudolf Rabenstein, and Lauri Savioja

Publication Date: 4th Quarter, 2004

Model-based sound synthesis has become one of the most active research areas in musical signal processing. The earliest attempts in generating musical sound with a physical model were made three decades ago. The first products were seen only some 20 years later. Recently, many refinements to previous signal processing algorithms and several new ones have been introduced. We have learned that new signal processing methods can still be devised or old ones can be modified to advance the field. Today there exist efficient model-based synthesis algorithms for many sound sources, while there are still some for which we do not have a good model. Certain issues, such as parameter estimation and real-time control, require further work for many model-based approaches. Finally, the capabilities of human listeners to perceive details in synthetic sound should be accounted for in a way similar to that in perceptual audio coding in order to optimize the algorithms. The success and future of the model-based approach depends on researchers and the results of their work.

The aim of this special issue is to present recent research in model-based sound synthesis.

Cross Layer Design for Communications and Signal Processing Systems

Guest Editors: Antonio Ortega, Lang Tong, Haitao Zheng, and Michele Zorzi

Publication Date: 4th Quarter, 2004

An important aspect of wireless networks is a dynamic behavior. The conventional protocol structure is inflexible as various protocol layers can only communicate in a strict manner. In such a case, the layers are designed to operate under the worst conditions, rather than adapting to changing conditions. This leads to inefficient use of spectrum and energy.

Adaptation represents the ability of network protocols and applications to observe and respond to the channel variation. Central to adaptation is the concept of cross-layer design. In general, cross-layer design involves four key layers in the overall protocol stack (i.e., application layer, transport layer, network layer, and link-layer). The application can adjust its behavior, for example, its flow rate or the amount of overhead devoted to error resilience according to the changing network and channel conditions. The adaptation can also take place in the underlying layers such as TCP and UDP so that the application originally developed for different networks remains unchanged. Information derived from the application, such as its QoS requirements and the priorities of the packets it produces, could be used in coordinating the behavior of the lower layers for resource efficiency. For example, the persistence level of the link layer ARQ mechanism should be varied according to each application's latency and reliability requirements as well as the traffic load. Another essential factor to adaptation is each layer's ability to estimate the current and even predict the future network and channel conditions and exchange the information across different layers.

This special issue is devoted to the latest developments in the field of cross-layer design, where the emphasis is on interactions among different network layers so as to improve the performance of communication and signal processing systems.

Special Issue on

Nonlinear Signal and Image Processing

CALL FOR PAPERS

While the field of signal and image processing has matured within the framework of linear systems, novel areas of nonlinear signal processing continue to appear. This, due to the fact that the physics of image formation are inherently nonlinear and that the stochastic components of signals and images are inherently nonstationary and non-Gaussian, for which linear processing is not optimum.

The goal of this special issue is to bring together the latest advances in the areas of nonlinear signal and image processing as represented in the 2003 NSIP Workshop to be held in Grado, Italy. We encourage participants of the workshop to extend their conference paper contributions into full pages for submission to this special issue. Prospective manuscripts should be unpublished in journals and should present innovative contributions either from a methodological or applications point of view.

Areas of interest are targeted to advanced nonlinear signal and image processing (but not limited to):

- Lower and higher order statistics
- Median and order-statistics signal processing
- Mathematical morphology
- Multichannel and array nonlinear signal processing
- Nonlinear time-frequency methods
- Nonlinear methods for communications and networking
- Signal processing in bio-computing
- Polynomial signal processing structures
- Image and video signal processing applications
- Halftoning and digital printing
- Image, video, audio, and multimedia applications
- Signal processing for security, authentication, and cryptography

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GUEST EDITORS:

Gian Luca Foresti, Department of Computer Science, University of Udine, 33100 Udine, Italy; foresti@dimi.uniud.it

Gianni Ramponi, Signal and Image Processing, University of Trieste, 34127 Trieste, Italy; ramponi@univ.trieste.it

Carlo S. Regazzoni, Department of Biophysical and Electronic Engineering, University of Genoa, 16145 Genoa, Italy; carlo@dibe.unige.it

Giovanni Sicuranza, Signal and Image Processing, University of Trieste, 34127 Trieste, Italy; sicuranza@univ.trieste.it

Gianni Vernazza, Department of Biophysical and Electronic Engineering, University of Genoa, 16145 Genoa, Italy; vernazza@dibe.unige.it

Special Issue on Machine Perception on a Chip

CALL FOR PAPERS

Perception is where different functional modules—sensing, computing, information processing, and machine interfacing—merge in different technologies—MEMS, optics, and semiconductors, etc. It is one of the most active areas in both academia and industry. Perception is what differentiates a smart computer and system from a standard number crunching and storage machine. Perception is in its way to have more intelligence in future machines.

With the emergence of the "System-on-a-Chip" technology, perception systems have found a natural technological fit where a heterogeneous system—both in function and technology—can be implemented in a single chip. Although "perception on a chip" is not a reality, yet, several research groups have been working towards this goal.

The focus of the proposed special issue is to bring to the research and development community the latest research results and efforts at different levels: technologies, design paradigms, system integration, software-hardware codesign, high level architectures, sensors technologies, etc. The proposed issue will be a very useful resource for people who are just starting in this area, and will provide an update to those who have been working in the area. It will address some of the challenges these researchers are facing. Several examples of existing systems and prototypes will be included.

Many of these related issues have been the scope of a biannual workshop on "Computer Architectures for Machine Perception (CAMP)." The first CAMP workshop was held in Paris in 1991 (chair: Prof Zavidovique), the second one was held in New Orleans in 1993 (chairs: Profs Bayoumi and Larry Davis). It was held in Italy (twice) and Boston. CAMP '2003 took place in New Orleans, May 2003 (chair: Prof Guna Seetharaman). The special issue will also feature extended versions of papers presented at CAMP '2003.

Topics of interest include (but are not limited to):

- Smart sensing
- Image understanding
- Recognition
- Configurable and FPGA-based perception architecture
- Intelligent architecture

- Network of sensors
- Sensor fusion
- Internet imaging
- Motion and stereo vision
- Active vision
- Emerging technologies
- Others

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GUEST EDITORS:

Magdy A. Bayoumi, The Center for Advanced Computer Studies, Computer Science Department, University of Louisiana at Lafayette, P.O. Box 44330, Lafayette, LA 70504, USA; mbayoumi@cacs.louisiana.edu

Bertrand Zavidovique, Institut d'Électronique Fondamentale, Université Paris-Sud, Bât 220, 91405 Orsay Cedex, France

EDITORIAL BOARD REPRESENTATIVE:

Marc Moonen, Dept. of Electrical Engineering, Katholieke Univ. Leuven, ESAT-SISTA, Kasteelpark Arenberg 10, B-3001 Heverlee, Belgium; marc.moonen@esat.kuleuven.ac.be

Special Issue on Turbo Processing

CALL FOR PAPERS

Turbo codes first appeared in 1993. Although limited at the beginning to coding, the idea of exchanging soft information between SISO (soft-input/soft-output) modules has subsequently been applied to other elements of a digital communications receiver, leading to the nowadays famous turbo principle. Due to their excellent performance, turbo codes are already used in a number of standards, and have become a very active area of research.

The goal of this special issue is therefore to present recent research results on the theory and applications of the "turbo-principle."

Topics of interest include (but are not limited to):

- Design of turbo codes
- Performance and bounds of turbo codes
- Design of SISO algorithms
- Modelling of turbo/iterative processing
- Turbo detection/equalization for time dispersive channels
- Turbo joint detection for multiuser communications
- Turbo space-time coding
- Turbo reception for MIMO systems
- Turbo synchronization
- Turbo demodulation
- Joint source-channel decoding based on soft information
- Implementation issues
- Applications and standards

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GUEST EDITORS:

Alex M. Haimovich, Professor of Electrical and Computer Engineering, Director, New Jersey Center for Wireless Telecommunications, New Jersey Institute of Technology, University Heights, Newark, NJ 07102, USA; haimovic@njit.edu

Ramesh Pyndiah, Signal & Communications Department, ENST BRETAGNE Technopole Brest IROISE, BP832-29285 BREST Cedex, France; ramesh.pyndiah@enst-bretagne.fr

Luc Vandendorpe, Communications Laboratory, Université Catholique de Louvain, B-1348 Louvain la Neuve, Belgium; vandendorpe@tele.ucl.ac.be

EDITORIAL BOARD REPRESENTATIVE:

Marc Moonen, Department of Electrical Engineering, Katholieke Universiteit Leuven, ESAT-SISTA, Kasteelpark Arenberg 10, B-3001 Heverlee, Belgium; marc.moonen@esat.kuleuven.ac.be

Special Issue on UWB—State of the Art

CALL FOR PAPERS

Recent standardisation outcomes substantiate the potential impact of Ultra-Wideband (UWB) systems, and motivate the development of UWB products for the mass-market. In this light the moment seems appropriate for summarising the state of the art in UWB methods and technologies, and for setting an overview covering open issues in this broad research and application area. In particular, coexistence with other wireless standards plays a crucial role. Moreover, broadband antenna and receiver design is a real challenge, as well as the development of sophisticated algorithms for dense multipath environment, synchronization and several other topics such as channel and interference modelling. A low-complexity, low-power, and low-cost system solution may be far away.

The aim of this special issue is to present recent research in UWB systems and technology from multiple points of view with emphasis on future applications in the area of wireless communications. Prospective papers should be unpublished and present novel, fundamental research offering innovative contributions either from a methodological or an application perspective.

Topics of interest include (but are not limited to):

- Channel modeling and measurement
- Channel estimation and equalisation
- Synchronisation
- Modulation and multiple access
- Interference and coexistence issues
- Pulse shaping and filtering
- Antenna and receiver design
- Information theory
- Ad hoc networks and sensor networks
- Routing and MAC design
- Standardisation
- Applications
- UWBMIMO
- Multiband UWB

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GUEST EDITORS:

Gabriella di Benedetto, INFOCOM Department, Engineering School, University of Rome La Sapienza, Via Eudossiana 18, 00184 Rome, Italy; gaby@acts.ing.uniroma1.it

Thomas Kaiser, Department of Comm. Systems, Faculty of Engineering, University of Duisburg-Essen, Bismarckstr. 81, 47057 Duisburg, Germany; thomas.kaiser@uni-duisburg.de

Norbert Schmidt, IMST GmbH, Information and Communications Systems, Mobilfunksysteme, Carl-Friedrich-Gauß Str. 2, 47475 Kamp-Lintfort, Germany; schmidt@imst.de

Armin Wittneben, ETH, Swiss Federal Institute of Technology, Zurich, Communication Technology Laboratory, Sternwartstrasse 7, CH-8092 Zurich, Switzerland; wittneben@nari.ee.ethz.ch

EDITORIAL BOARD REPRESENTATIVE:

Mos Kaveh, Department of Electrical and Computer Engineering, University of Minnesota, 200 Union Street, SE Minneapolis, MN 55455, USA; kaveh@ece.umn.edu

Special Issue on
Anthropomorphic Processing of Audio and Speech

CALL FOR PAPERS

Anthropomorphic systems process signals "at the image of man." They are designed to solve a problem in signal processing by imitation of the processes that accomplish the same task in humans. In the area of audio and speech processing, remarkable successes have been obtained by anthropomorphic systems: perceptual audio coding even caused an MP3 hype.

At first sight, it could seem obvious that the performance of audio processing systems should benefit from taking into account the perceptual properties of human audition. For example, front-ends that extract perceptually meaningful features currently show the best results in speech recognizers. However, their features are typically used for a stochastic optimization that is itself not anthropomorphic at all. Thus, it is not obvious why they should perform best, and perhaps the truly optimal features have not yet been found because, after all, "airplanes do not flap their wings."

In general, we believe that there are several situations when an anthropomorphic approach may not be the best solution. First, its combination with nonanthropomorphic systems could result in a suboptimal overall performance (the quantization noise that was cleverly concealed by a perceptual audio coder could become unmasked by subsequent linear or nonlinear processing). Second, other than anthropomorphic approaches might be better adapted to the technology that is chosen for the implementation (airplanes do not flap their wings because it is technically much more efficient to use jet engines for propulsion). Nevertheless, a lot can be learned from imitating natural systems. As such anthropomorphic and, by extension, biomorphic systems can be considered to play an important role in the process of developing new technologies.

The aim of this special issue is to bring together papers from different areas of audio and speech processing that deal with aspects of anthropomorphic processing or in which an anthropomorphic or perceptual approach was taken. Papers with a research nature, review papers, and tutorial papers will be considered, provided that they are unpublished.

Topics of interest include (but are not limited to):

- Speech and audio coding
- Audio measurements and speech analysis
- Objective quality measures for audio and speech
- Speech synthesis (rule-based, articulatory, . . .)

- Audio virtual reality
- Content-based music search
- Music and instrument recognition
- Audio classification and retrieval
- Speech and speaker recognition

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GUEST EDITORS:

Werner Verhelst, Vrije Universiteit Brussel, Belgium; wverhels@etro.vub.ac.be
Jürgen Herre, Fraunhofer IIS-A, Germany; hrr@iis.fhg.de
Gernot Kubin, Technical University Graz, Austria; g.kubin@ieee.org
Hynek Hermansky, Oregon Health & Science University, USA; hynek@ece.ogi.edu

EDITORIAL BOARD REPRESENTATIVE:

Soeren Hold Jensen, Aalborg University, Fredrik Bajers Vej 7, A3, DK-9220 Aalborg Oest, Denmark; shj@cpk.auc.dk

Special Issue on Advances in Sensor Array Processing Technology

CALL FOR PAPERS

Sensor array processing has been a key technology in many diverse areas including radar, sonar, communications, astronomical observations, and microphone and seismic array applications. Arrays of sensors have been used extensively in most radar and sonar systems to improve target detection, Doppler estimation (i.e., platform velocity vector), interference suppression including jamming and clutter, and target angle of arrival estimation. Since the flourishing development of radar and sonar arrays, the same fundamental principles have been applied to different types of sensors with different classes of algorithms that adapt to the different signals and media of propagation, for example, microphone and seismic arrays. Antenna arrays have also been commonly applied to various communications systems to dramatically improve data throughput and combat performance degradation suffered by severe fading due to multipath. Presently, the advance in sensor array processing lends itself to many emerging applications such as next generation wireless communication systems, sensor networks, multimedia systems, bistatic radar, space-based radar, and communications, and many technical challenges remain before the advanced techniques can be realized in practical systems.

This special issue will gather the latest research and development of the sensor array processing area and address the most current issues and challenges. It will include the latest research results and efforts at different levels including novel algorithm designs, theoretical performance analysis, performance and capability demonstration via simulation and experimental results, robust algorithms under nonideal environmental and system conditions, and real-time implementation of advanced algorithms. Prospective papers should be unpublished and present novel, fundamental research offering innovative contributions either from a methodological or an application perspective.

Topics of interest include (but are not limited to):

- Source localization and angle of arrival estimation
- Adaptive beamforming, multichannel signal enhancement, interference cancellation, and source separation
- Multichannel detection and estimation
- Multichannel channel estimation, system identification, and calibration
- Space-time adaptive processing (STAP) and parameter estimation

- Theoretical bounds of multichannel signal processing algorithms
- Robust array signal processing methods in severe environmental and system conditions
- Practical implementation of multichannel signal processing algorithms

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Joe C. Chen, Space Communication Systems, Northrop Grumman Space Technology, One Space Park, O3/1633, Redondo Beach, CA 90266, USA; joe.chen@ngc.com

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Special Issue on

Advances in Intelligent Vision Systems: Methods and Applications

CALL FOR PAPERS

Computer vision has proved especially successful in well-constrained industrial environments (for instance when illumination, objects types, and orientations are known). However, in many practical applications such as airborne or remote sensing, medical imaging, face recognition, outdoor robotics, and surveillance applications, the environment can scarcely be controlled.

These challenging applications require a more sophisticated approach. The resulting intelligent computer vision systems usually integrate several image and video processing algorithms, ranging from low-level preprocessing and medium-level algorithms to high-level recognition techniques. These solutions usually involve a specific adaptation of generic image processing techniques to the application.

This special issue will be dedicated to original contributions on state-of-the art components at any of the above-mentioned three levels of an intelligent vision system and on their interconnection. We also welcome submissions detailing complete vision systems or specific applications.

Topics of interest include (but are not limited to):

- Low level processing: filtering, segmentation, edge detection, and image transforms;
- Intermediate level analysis: clustering, object tracking, and data fusion;
- High level analysis: pattern matching, classification, and recognition;
- Empirical and algorithmic evaluation of techniques.

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Acceptance Notification	May 1, 2004
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Special Issue on

Innovative Signal Transmission and Detection Techniques for Next Generation Cellular CDMA System

CALL FOR PAPERS

Code division multiple access (CDMA) has been identified as one of the major techniques for next generation communications system. In CDMA system, in addition to intersymbol interference (ISI) caused by multipath propagation, simultaneous transmission also introduces multiuser interfernce (MUI). The receiver, therefore, is required to separate and recover the information signal of the desired user(s). Compared to the conventional single user detectors where interfering users are modeled as noise, significant improvement can be obtained with multiuser detectors where MUI is explicitly part of the signal model.

In literature, if the spreading sequences are periodic and repeat every information symbol, the system is refered to as short CDMA, and if the spreading sequences are aperiodic or essentially pseudorandom, we call it long CDMA. Mainly, due to the time-varying nature of long code systems, researches on multiuser detection have been largely limited to short CDMA. On the other hand, long codes are widely used in virtually all operational and commercially proposed CDMA systems. To bridge up the gap, researchers have proposed code-hopping scheme for short code systems and have also been targeting on simplified multiuser detectors for long CDMA systems. Moreover, multirate design has been proposed to support multimedia services with high data rate and variable quality of service. Meanwhile, novel techniques on spreading sequences design, time-frequency analysis, multiple transmit and receive antennas, space-time coding, multicarrier CDMA, and other related topics have continuously been explored to improve the performance and communication security of CDMA systems.

This special issue aims to cover the present research on the development of signal transmission and detection techniques for next generation cellular CDMA systems. Prospective papers should present original and fundamental research offering innovative contributions to the wireless communications community.

Topics of interest include (but are not limited to):

- Spreading Sequences Design
- Multiple access design

- Multirate CDMA systems
- Code-hopping CDMA
- Long CDMA systems
- Multicarrier CDMA and OFDM
- Time-varying channel tracking for CDMA
- Space-time coding for CDMA
- Communication security of CDMA

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GUEST EDITORS:

Jitendra K. Tugnait, Department of Electrical and Computer Engineering, Auburn University, 200 Broun Hall, Auburn, AL 36849, USA; tugnait@eng.auburn.edu

Hui Liu, Department of Electrical Engineering, University of Washington, 307P EE/CSE Building, Campus Box 352500, Seattle, WA 98195-2500, USA; hliu@ee.washington.edu

Guang Gong, Department of Electrical and Computer Engineering, University of Waterloo, Waterloo, Ontario, Canada N2L 3G1; ggong@calliope.uwaterloo.ca

Tongtong Li, Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824-1226, USA; tongli@egr.msu.edu

Special Issue on
Multiuser MIMO Networks

CALL FOR PAPERS

The potential benefits of using multiple transmit and receive antennas on both ends of a single-user wireless communications link are by now well understood. Depending on the structure of the resulting multiple-input multiple-output (MIMO) channel, dramatic gains in capacity can result, and the additional spatial degrees of freedom can be used to enhance the diversity and robustness of the communications link. Numerous space-time coding schemes have recently been developed in an attempt to exploit the available capacity and spatial diversity. Most of the work on this point has focused on single point-to-point communications, where each of the transmitter and receiver has arrays, and the presence of other cochannel users is not considered. Such a work ignores the larger system-level issues that are critical to the successful operation of a network of users that potentially all possess multiple antennas.

The goal of this special issue is to emphasize research that addresses how a network of multiantenna nodes can be coordinated to achieve the competing objectives of high total network throughput, a minimum quality-of-service level for all users, and low multiuser interference.

Topics of interest include (but are not limited to):

- Multiple access in MIMO networks
- Multiuser detection in MIMO systems
- Vector broadcast channels
- Downlink beamforming
- Uplink and downlink duality
- Linear and nonlinear precoding for multiuser systems
- “Dirty paper” techniques
- Obtaining and exploiting channel knowledge
- Multiuser interference mitigation
- Ad hoc networks of multiple antenna nodes
- Space-time routing/scheduling algorithms
- Space-time network resource allocation Network

- Capacity Network performance and reliability
- Related quality-of-service objectives

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A. Lee Swindlehurst, Department of Electrical and Computer Engineering, Brigham Young University, Provo, UT 84602, USA; swindle@ee.byu.edu

Brian Sadler, Army Research Laboratory, AMSRL-IS-TA, 2800 Powder Mill Road, Adelphi, MD 20783-1197, USA; bsadler@arl.mil

Robert Fischer, Lehrstuhl für Informationsübertragung, Friedrich-Alexander-Universität Erlangen-Nürnberg, Causerstraße 7/NT, D-91058 Erlangen, Germany; fischer@LNT.de

Special Issue on

Optical Wireless Communications

CALL FOR PAPERS

EURASIP Journal on Wireless Communications and Networking invites submissions to a special issue on the topic of Optical Wireless Communications. Submissions to this issue should follow the guidelines and submission procedure described below. When technologies penetrate and reach through many products our home, they become pervasive without many of us realizing it, such technologies are very successful. In many ways, optical wireless links have succeeded in permeating our homes and in this vision, and little credit has been given to this important technology.

Products ranging from the TV remote control unit to IrDA ports with a worldwide installed base on products of over 200 million units and growing at 40% annually, optical wireless is widely available on personal computers, peripherals, embedded systems, and devices of all types.

Optical wireless has an impact on other outdoor applications such as bringing broadband to the home, (The last mile), Optical Wireless LANs, (802.11), and considered in intersatellite link applications.

This special issue seeks to highlight the exciting of ongoing research in this important area. Original manuscripts are solicited on issues of Optical Wireless Techniques as they are related to indoor applications.

Topics of interest include (but are not limited to):

- Indoor and outdoor links
- Optical LANs
- Propagation effects
- Indoor and outdoor channel modeling and ambient noise
- Modulation and coding techniques
- Capacity limits
- Transceiver design
- Link layer, network and transport protocols and modeling
- High speed system design
- Demonstrators (cameras, wrist watches, phones, printers, LAN extensions, home network control, etc.)
- Applications (financial, aircraft, automobile, etc.)

- Standards and eye safety issues, (802.11, IrDA, IEC 825-1, etc.)
- ATM and wireless multimedia
- Coexistence with radio technology (Bluetooth)
- Future market and technology prospects

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Final Manuscript Due	August 31, 2004
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GUEST EDITOR:

A. Boucouvalas, School of Design, Engineering and Computing, Bournemouth University, Fern Barrow, Poole Dorset BH12 5 BB, UK; tboucouv@bournemouth.ac.uk

EURASIP JOURNAL ON

WIRELESS COMMUNICATIONS AND NETWORKING

The overall aim of EURASIP Journal on Wireless Communications and Networking (EURASIP JWCN) is to bring science and applications together on wireless communications and networking technologies with emphasis on signal processing techniques and tools. It is directed at both practicing engineers and academic researchers. EURASIP JWCN will highlight the continued growth and new challenges in wireless technology, both for application development and basic research. Papers should emphasize original results relating to the theory and/or applications of wireless communications and networking. Tutorial papers, especially those emphasizing multidisciplinary views of communications and networking, are also welcome. EURASIP JWCN employs a paperless, electronic submission, and evaluation system to promote a rapid turnaround in the peer review process.

The journal publishes two types of issues: regular issues and special issues. Regular issues publish collections of papers without special solicitation. The special issues feature specifically aimed and targeted topics of interest contributed by authors responding to a particular Call-for-Papers or by invitation, edited by invited guest editor(s). Regular papers can be submitted at any time, while special issue papers can be submitted only based on planned schedules and submission guidelines of the Call-for-Papers. Proposals for special issues can be submitted directly to the Editor-in-Chief.

Subject areas include, but are by no means limited to: Ad-hoc networks; Channel modeling and propagation; Detection, estimation, and synchronization; Diversity and space-time techniques; End-to-end design techniques; Error control coding; Iterative techniques for joint optimization; Modulation techniques (CDMA, OFDM, multicarrier, spread-spectrum, etc.); Multiuser, MIMO channels, and multiple access schemes; Network performance, reliability, and quality of service; Resource allocation over wireless networks; Security, authentication, and cryptography; Signal Processing Techniques and Tools; Ultra wide-band systems; Wireless network services and medium access control.

http://wcn.hindawi.com

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The Editor reserves the right to edit these to fit into the available space as appropriate for the next newsletter publication.