EURASIP

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European Association
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and Image Processing

Newsletter, Volume 15, Number 3, September 2004

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President's Message: EUSIPCO 2004

This Newsletter is being distributed to all current EURASIP members and to all EUSIPCO 2004 registrants. As you may know, all EUSIPCO 2004 registrants will become EURASIP members after the conference and until the end of year 2005. Thus, if you have not come across EURASIP before, we welcome you to the association. The renewal of those members not attending EUSIPCO 2004 and the handling of all the journal subscription process will be managed through the new EURASIP website that will be launched in September 2004. You can find more details of this new service in this Newsletter or directly visit our web page http://www.eurasip.org.

On Wednesday, 8 September 2004, the EURASIP General Assembly Meeting will take place. All our members are invited to attend and participate in this meeting, although only current members have the right to vote for the new Administrative Committee members. To help you in your decision, this Newsletter includes the information about the six candidates that were already presented in the previous Newsletter.

Finally, on Thursday evening, the EUSIPCO 2004 banquet will take place. During this event, the EURASIP achievement awards and the EURASIP best paper awards will be presented. It is our pleasure to include in this Newsletter the details of those individuals who have received such awards.

Looking forward to welcoming you all at EUSIPCO 2004 in Vienna this September.

Ferran Marqués
President

EURASIP Secretary-Treasurer's Report: 1st January 2003–30th June 2004

Opening balance of **155,820.92€** was held as follows on 1st January 2003:

1CHF=0,64104 EURO

CHF cheque account . . . . . (CHF 114,615.80) . . . . . 73,473.31€
Euro cheque account . . . . . 13,593.81€
Euro money market account . . . . . (CHF 75,216.00) . . . . . 48,216.46€
Postal account . . . . . (CHF 8,638.05) . . . . . 5,537.34€
Loans (to EUSIPCO-2002) . . . . . 15,000.00€

The EURASIP main account movements during the financial period from 1st January 2003 until 30th June 2004 are documented in the table below in Euros (€).

Incoming balance of 1st January 2003			**155,820.92€**
Income:	Membership fees incl. journal subscriptions	53,546.22€	
	Bosch Awards	1,500.00€	
Total:		55,046.22€	
Expenses:	Elsevier (various concepts)	60,440.66€	
	Hindawi (various concepts, incl. Newsletter)	30,414.19€	
	ACCO (Newsletter)	2,722.25€	
	Administrative expenses	10,294.46€	
	Taxes, bank costs, interests	−4171.62€	
Total:		99,699.95€	
Loans	To EUSIPCO-04	15,006.54€	
	To EUSIPCO-05	15,009.81€	
	To NSIP-03	1,343.24€	
Outgoing balance on 30th June 2004 = **Incoming balance** on 1st July 2004			**111,167.20€**

Total balances on 30th June 2004 in EURO:

CHF cheque account	(CHF 1,461.35)	938.78€
Euro cheque account		203.27€
Euro money market account	(CHF 82,491.00)	52,880.03€
Postal account	(CHF 16,560.88)	10,616.19€
Dollar account	(USD 210.31)	171.34€
Loans (to EUSIPCO-2002, -2004, -2005, NSIP-03)		46,359.59€
Total in EURO		**111,167,20€**

2004 AdCom Elections

Deadline for Mail voting: 20 August 2004 (for return address, see newsletter inside cover)

Deadline for Direct voting: 8 September 2004 (General Assembly Meeting at EUSIPO 2004)

Please cross 4 of the candidates and mail it to the AdCom's President Professor Ferran Marqués (for the address, see newsletter inside cover)

Jonathon Chambers	☐	Paulo Correia	☐	Fulvio Gini	☐
Søren Holdt	☐	Marc Moonen	☐	Markus Rupp	☐

2004 AdCom Elections: Candidates Biographies

Jonathon Chambers is a Cardiff Professorial Fellow of Digital Signal Processing within the Centre for Signal Processing, Cardiff School of Engineering, Cardiff, Wales, UK. He leads a team of researchers involved in the analysis, design and evaluation of novel algorithms for digital signal processing with application in acoustics, biomedicine, and wireless communications.

Jonathon served in the Royal Navy as an Artificer Apprentice in Action, Data and Control until 1982, after which he studied for his first degree in electronic engineering at the Polytechnic of Central London. He graduated in 1985 with a first-class honours degree and was awarded the Robert Mitchell Medal as the top graduate of the Polytechnic. He was immediately appointed to a lectureship in VLSI Signal Processing before moving to Cambridge University, Peterhouse, for postgraduate research. He attained his Ph.D. in adaptive signal processing in 1990. He has since held academic and industrial positions at Bath University, Imperial College London, King's College London, and Schlumberger Cambridge Research, Cambridge, UK.

Jonathon's research contributions have been in adaptive and blind signal processing. He has authored/coauthored more than 180 conference and journal publications, and supervised 20 Ph.D. graduates. He has served as an Associate Editor for IEEE Transaction Signal Processing and Circuits & Systems, and is a Past Chairman of the IEE Professional Group E5, Signal Processing. He is currently serving as an Associate Editor for IEEE Signal Processing Letters and as the University Liaison on the EURASIP AdCom Committee. He believes that there is a great opportunity for further collaborative growth within the European Signal Processing community, particularly given the recent expansion of the EU, and desires to play a major role in this process.

Paulo Correia is currently an Assistant Professor at the Department of Electrical Engineering and Computer Engineering, Superior Institute Technician (IST), Lisbon, Portugal. He is also a Researcher at the Image Group of the Telecommunications Institute (IT), Lisbon. He has lectured several Telecommunication related subjects (fundamentals, systems, mobile communications, computer networks). In 2004–005 he is also responsible for an M.S. course on Advanced Digital Image Processing.

He holds Ph.D. degree on electrical and computers engineering, on the field of video analysis and image communications, and an M.S. degree on the field of mobile communication systems simulation, both from IST. In 1989 he won the Marconi and Alcatel awards for the best telecommunications student graduating from IST.

Paulo Correia participates in several international projects in the field of video analysis, coding and description, notably the IST Network of Excellence VISNET (NETworked audioVISual media technologies). He is the Portuguese representative in the Management Committee of the COST 211 European project (redundancy reduction techniques and content analysis for multimedia services). He has also participated in international projects related to audiovisual processing (e.g. ACTS DICEMAN, ACTS MoMuSys), mobile communications systems (e.g. RACE MBS), and communication systems simulation (e.g. COST 228).

He acts as reviewer for several journals: IEEE Transactions on Image Processing, IEEE Transactions of Circuits and Systems for Video Technology, IEEE Transactions on Pattern Analysis and Machine Intelligence, Elsevier Signal Processing: Image Communication, EURASIP Journal on Applied Signal Processing, SPIE Optical Engineering.

He is also a Member of the technical committees of international conferences: International Workshop on Image Analysis for Multimedia Interactive Services (WIAMIS), International Conference on Image Processing (ICIP). In 2004 he was the General Cochairman of WIAMIS.

He has evaluated research proposals for the European Commission, namely for the INCO-DC, INCO-COPERNICUS, and Information Society Technologies (IST) programmes.

He cooperates with EURASIP (European Association for Signal, Speech and Image Processing) since 2003 in the role of "Internal Coordinator Officer."

His current research interests are in the areas of video analysis (notably segmentation and objective segmentation evaluation), video description, video coding, and multimedia communications.

Fulvio Gini received the Doctor Engineer (cum laude) and the Research Doctor degrees in electronic engineering from the University of Pisa, Italy, in 1990 and 1995, respectively. In 1993 he joined the Departmento "Ingegneria dell'Informazione", Università di Pasa, where he is an Associate Professor since October 2000. From July 1996 through January 1997, he was a Visiting Researcher at the Department of Electrical Engineering, University of Virginia, Charlottesville. He has been Session Chairman and member of the technical committee for various international conferences. He is a coauthor of two tutorials entitled *Coherent detection and fusion in high resolution radar systems* presented at the International Conference on Radar (Brest, May 1999) and *Advanced Radar Detection and Fusion* presented at the International Radar Conference (Washington D.C., May 2000). In February 2002, he was the Faculty Opponent at the Chalmers University of Technology, for a Ph.D. thesis defense. He is an IEEE Member since 1992 and Senior Member since January 1, 2000. He is an Associate Editor for the IEEE Transactions on Signal Processing (term starting August 1st, 2000) and a Member of the EURASIP JASP Editorial Board (term starting July 1st, 2003). He was corecipient of the 2001 IEEE Aerospace and Electronic Systems Society's Barry Carlton Award for Best Paper. He was a recipient of the 2003 IEE Achievement Award for outstanding contribution in signal processing and of the 2003 IEEE Aerospace and Electronic Systems Society Nathanson Award for the Young Engineer of the Year. He is a Member of the Signal Processing Theory and Methods (SPTM) Technical Committee (TC) of the IEEE Signal Processing Society since January 1st, 2003. His general interests are in the areas of statistical signal processing, estimation, and detection theory. In particular, his research interests include modeling and statistical analysis of recorded live sea and ground radar clutter data, non-Gaussian signal detection and estimation, parameter estimation and data extraction from multichannel interferometric SAR data, cyclostationary signal analysis, and estimation of nonstationary signals, with applications to radar signal processing. He authored or coauthored more than 60 journal papers and about 60 conference papers.

Søren Holdt Jensen was born in Denmark in 1964. He received the M.S. degree in electrical engineering from Aalborg University, Aalborg, Denmark, and the Ph.D. degree from the Technical University of Denmark, Lyngby, Denmark. Currently, he is an Associate Professor at Aalborg University. Before joining the Department of Communication Technology of Aalborg University, he was with the Telecommunications Laboratory of Telecom Denmark, the Electronics Institute of the Technical University of Denmark, the Scientific Computing Group of the Danish Computing Center for Research and Education (UNI-C), the Electrical Engineering Department of Katholieke Universiteit Leuven, Belgium, and the Center for PersonKommunikation (CPK) of Aalborg University. His research activities are in digital signal processing, digital communications, and speech and audio processing. Dr. Jensen is a Member of the editorial board of the Journal on Applied Signal Processing, a former Chairman of the IEEE Denmark Section, and founder and first Chairman of the IEEE Denmark Section - Signal Processing Chapter.

Marc Moonen received the electrical engineering degree and the Ph.D. degree in applied sciences from the Katholieke Universiteit Leuven, Leuven, Belgium, in 1986 and 1990, respectively. Since 2000 he has been an Associate Professor at the Electrical Engineering Department of Katholieke Universiteit Leuven, where he is currently heading a research team of 16 Ph.D. candidates and postdocs, working in the area of signal processing for digital communications, wireless communications, DSL, and audio signal processing. He received the 1994 K.U. Leuven Research Council Award, the 1997 Alcatel Bell (Belgium) Award (with Piet Vandaele), and was a 1997 "Laureate of the Belgium Royal Academy of Science." He was Chairman of the IEEE Benelux Signal Processing Chapter (1998–2002), and is currently a EURASIP AdCom Member (European Association for Signal, Speech and Image Processing, 2000–·). He has been a member of the editorial board of IEEE Transactions on Circuits and Systems II (2002–2003). He is currently Editor-in-Chief for the EURASIP Journal on Applied Signal Processing (2003–·) and a Member of the editorial board of Integration, the VLSI Journal, EURASIP Journal on Wireless Communications and Networking, and IEEE Signal Processing Magazine.

Prof. Dr.-Ing. Markus Rupp is presently a Full Professor for digital signal processing in mobile communications at the Technical University of Vienna. He graduated in February 1993 with a Ph.D. degree in electrical engineering from Technische Universitaet Darmstadt, Germany, where he worked with Eberhardt Haensler on designing new algorithms for acoustical and electrical echo compensation. From November 1993 until July 1995, he had a postdoctoral position at the University of Santa Barbara, California with Sanjit Mitra where he worked with Ali H. Sayed on a robustness description of adaptive filters with impacts on neural networks and active noise control.

From October 1995 until August 2001, he was a member of the technical staff in the Wireless Technology Research Department of Bell Labs. It is located in the Crawford Hill facility of the Bell Laboratories, now part of Lucent Technologies

During his Bell Labs time, he spent 15 months in the Netherlands and 13 months in Germany on international assignments. During this time he was working on various topics related to adaptive equalization and rapid implementation. He designed equalizers and symbol timing recovery algorithms for mobile telephone sets in the cellular and PCS band. He developed a rapid prototyping modem in a high speed Advanced Wireless Access project. He worked on rapid prototyping methodologies for Hyperlan II and UMTS.

During his last year at Bell Labs he was leading two teams: one to build a BLAST-on-UMTS prototype that allows to support four-transmit and four-receive antennas. The second team focused on the 4th Generation Mobile Communications and defined new methods to improve spectral efficiency and quality of services in future mobile networks.

In July 2002 he received a Research Grant for a Christian Doppler Laboratory on design methodology for digital signal processing algorithms in collaboration with Infineon Technologies.

Since February 2003 he has been an associate editor of IEEE Transactions on Signal Processing and since January 2004 an Associate Editor of EURASIP JASP (Journal on Applied Signal Processing).

He is a Publication Chair and Cochair of EUSIPCO 2004 held in Vienna, Austria and a member of the steering committee for URSI-ISSSE04 held in Linz, Austria.

He authored and coauthored more than 100 papers and patents on adaptive filtering, wireless communications, and rapid prototyping.

Minutes of Previous EURASIP General Assembly

Toulouse, Thursday 5th September 2002 at 18.10 hrs

1. The president, Peter Grant, opened the meeting and introduced the Adcom.
2. The president then introduced the six election candidates and the ballot took place.
3. The president gave a summary of EURASIP events organised over the last two years.
4. The treasurer, Ferran Marqués, presented the budgets for the last two years with an increased bank balance of 172,000 EURO. The treasurer explained that the relatively large balance usually reduces substantially when we receive the publisher's bill for our members' journal subscriptions. It was explained that this was really a small balance for a society when running large expensive conferences such as the EUSIPCO series. The treasurer's report was formally accepted after proposal by Ulrich Heute and seconding by Wolfgang Mecklenbrauker.
5. The president then opened a discussion on what EURASIP members expect from the newsletter and how it could be improved. The resulting discussion focused on printing of the call for papers for special issues in our journals, and the editorials (and contents lists) from recently published special issues to alert the members to these. The concept of a new low-cost EURASIP book series was encouraged by the members, and on EU/FP6 issues the members felt that there was little impact that EURASIP as a whole could have on the commission activities, except for encouraging special interest groups within DSP, and for supplying the commission with names and contact details of leading groups who could be expected to assist with proposal review. Finally, there was a request for providing EURASIP email addresses to the members.
6. The results of the Adcom elections were then announced as

   Sergios Theodoridis: 46 votes
   Ferran Marqués: 38 votes
   Bob Stewart: 23 votes
   Francis Castanie: 21 votes
   Helmut Bolcskei: 15 votes
   Jonathon Chambers: 9 votes.

   The first three candidates were thus elected to the Adcom from the 51 voting papers including 13 received by mail in advance of the General Assembly. The other candidates will be invited to assist with Adcom in officer positions.
7. The president thanked all for attending and closed the assembly at 18.50 hrs.

Peter Grant and Ferran Marqués

Awards at EUSIPCO 2004

EURASIP Achievement Awards

EURASIP Meritorious Service Award

The EURASIP Meritorious Service Award for 2004 is awarded to Ray Liu for his activities in relaunching the EURASIP Journal on Applied Signal Processing as an extremely successful forum for the whole signal processing society and promoting as Editor-in-Chief the new "EURASIP Book Series on Signal Processing and Communications."

K. J. Ray Liu received his B.S. degree from the National Taiwan University in 1983, and the Ph.D. degree from UCLA in 1990, both in electrical engineering. He is a Professor at the Electrical and Computer Engineering Department and Institute for Systems Research of the University of Maryland, College Park. His research interests span broad aspects of signal processing algorithms architectures, multimedia communications and signal processing, wireless communications and networking, information security, and bioinformatics, in which he has published over 250 refereed papers. Dr. Liu is the recipient of numerous awards including the 1994 National Science Foundation Young Investigator Award, the IEEE Signal Processing Society's 1993 Senior Award (Best Paper Award), IEEE 50th Vehicular Technology Conference Best Paper Award, Amsterdam, 1999. He also received the George Corcoran Award in 1994 for outstanding contributions to electrical engineering education and the Outstanding Systems Engineering Faculty Award in 1996 in recognition of outstanding contributions in interdisciplinary research, both from the University of Maryland. Dr. Liu is a Fellow of IEEE. He has served as an Associate Editor of IEEE Transactions on Signal Processing, a Guest Editor of the special issues on Multimedia Signal Processing of Proceedings of the IEEE, a Guest Editor of the special issue on Signal Processing for Wireless Communications of IEEE Journal of Selected Areas in Communications, a Guest Editor of the special issue on Multimedia Communications over Networks of IEEE Signal Processing Magazine, a Guest Editor of the special issue on Multimedia over IP of IEEE Trans. on Multimedia, and an Editor of the Journal of VLSI Signal Processing Systems. Dr. Liu has served as a Chairman of Multimedia Signal Processing Technical Committee of IEEE Signal Processing Society. He is a coauthor of Design of Digital Video Encoder: A Complete Compressed Domain Approach, Marcel Dekker, 2001, and a coeditor of High Performance VLSI Signal Processing I: System Design and Methodology; II: Algorithms, Architectures, and Applications, IEEE Press, 1998.

Ray Liu was the relaunching Editor-in-Chief of EURASIP JASP. Under his leadership the journal achieved the following:

1. Number of submissions grew from about 60 manuscripts per year in pre 2001 years to 275 in 2001, 361 in 2002, and 679 in 2003.
2. Number of published papers increased from about 40 papers per year in pre 2001 to about 140 papers in 2003 and this number is expected to exceed 200 in 2004.
3. Number of subscribers for EURASIP JASP increased from about 60 to over 200 in the last 3 years (consequently EURASIP JASP has an increased visibility in the signal processing community).

4. Ray Liu is currently the Senior Advisory Editor of EURASIP JASP offering his valuable advice in managing, running, and further promoting the journal.

Ray Liu is also the Editor-in-Chief of the new "EURASIP Book Series on Signal Processing and Communications" where in a very short time a number of book proposals were reviewed and approved by the editorial board of the series including both edited volumes and monographs. The first book in the series is expected to be published during year 2004.

Annual European Group Technical Achievement Awards

The European Group Technical Achievement Award for the year 2003 is awarded to John McWhirter for his fundamental contributions, as leader of the Advanced Signal Processing Group at QinetiQ, to applications in defence electronic systems for radar, sonar and communications, and development of algorithms and signal processor architectures for airborne early warning radars, phased-array radars, and sonar systems.

The Advanced Signal Processing group at QinetiQ Ltd is a world-leading group specialising in signal processing and data analysis for multisensor systems. Until recently it was part of the UK government's defence evaluation and research agency (DERA) in Malvern, UK. Much of its work has therefore been directed towards applications in defence electronic systems for radar, sonar, and communications. It has developed algorithms and signal processor architectures for airborne early warning radars, phased-array radars such as MESAR, and sonar systems, specialising in fast adaptive beamforming, interference cancellation, and the detection and recovery of signals in noise and interference. The group became part of QinetiQ Ltd in 2002, and is continuing to make important contributions in many areas, including both the theory of blind signal separation methods, and their application, for example to noninvasive fetal ECG and artefact suppression in EEG.

The group has been led technically by Professor John McWhirter, BSc, PhD, FRS, FREng, FIEE, FIMA, CMath, who has spent all his working life at Malvern. His personal achievements include the development of methods for applying two-dimensional (parallel) arrays of VLSI signal processors (so-called systolic arrays [IEE Proc. F, April 89]) to the implementation of high-throughput DSP systems. He designed the first VLSI implementation of minimum variance distortionless response (MVDR) beam-forming processors [IEE Proc. F, p. 241, August 91], based on triangularised QR decomposition. These were later implemented commercially as the STL NODE chip set, and the ideas developed by the group were also adopted by Charles Rader at MIT for wafer scale integration. This work led to a beamformer capability which is not exceeded in any other overseas laboratory or company.

The Group contains several internationally recognised researchers. Professor McWhirter is a world leading authority in the field of advanced signal processing. He worked extensively with John McCanny of Queen's University Belfast, where McWhirter was appointed a Visiting Professor in 1986. They received the Northern Ireland Information Technology Award from the British Computer Society in 1987 for work on "bit level systolic arrays for digital filter design." In 1990, Professor McWhirter received the JJ Thomson Paper Premium from the Institution of Electrical Engineers (IEE), London, and in 1994 he was the first scientist within DERA to receive the IEE JJ Thomson medal in recognition of his "outstanding work on signal processing using parallel electronic architectures, which gives the UK science community its prominent innovative position in this field." In 1999 his contributions were further recognised by the award of a Fellowship of Royal Society, the UK Academy of Sciences, to which only 40 are invited annually in recognition of their outstanding scientific achievements and leadership, which is a rare honour for a UK

engineering mathematician. He is also a fellow of the Royal Academy of Engineering (FREng). Finally, he is the current president of the Institute of Mathematics and Its Applications in the UK. In January 2002, Professor Proudler was the group's second recipient of the IEE JJ Thomson medal, for his work on orthogonalised lattice algorithms.

Other achievements of the group include the work of Professor Proudler on the design of efficient algorithms and the design of high-performance magnetic sensors, using adaptive signal processing and inexpensive magnetic sensors to build sensitive magnetometers, greatly reducing cost. Ira Clarke's work on IMP, an iterative beamforming algorithm, and the BLISS algorithm for blind signal separation, which are of major importance. Dr. John Mather is well recognised for his work on detection in sensor array signal processing in clutter, reverberation, jamming and interference, and, with Dr. Steve Hayward, for space-time adaptive processing (STAP) in nonstationary signal environments. Dr. Hayward's work on adaptive beamforming for phased-array radar seekers contributed to QinetiQ's world first with a successful "closed loop" test of its phased-array radar seeker sensor concept, announced recently. He has also developed patented algorithms which greatly increase the sensitivity of interferometric porous silicon chemical sensors. Dr. Richard Walke did novel work on Givens rotations for implementing RLS algorithms, and Professor David Broomhead pioneered the use of nonlinear radial basis function networks for pattern recognition, for which the Group hold various international patents. Dr. David Rees devised a radically new concept for phased-array airborne radars using element-level digitisation, and the group developed the penalty function method for interference-nulling beamformer design, now used in the MESAR phased-array radar. Dr. Geoff de Villiers and Dr. Fabienne Marchand have published papers on both phased-array antenna design and inverse problems on antenna theory. Dr. Malcolm Macleod has widely published in signal processing journals and conferences for over twenty years and has recently patented new techniques for direction finding. Dr. Ed Warner's publications cover adaptive multichannel equalisation of multipath, most recently applied to HF radio systems, and interference rejection.

The European Group Technical Achievement Award for the year 2004 is awarded to Ulrich Heute for his effective leadership in speech-signal processing, coding, and recognition. Ulrich Heute, born in 1944 in Magdeburg, Germany, studied electrical engineering at Stuttgart, with a Dipl.-Ing. degree in 1970. He worked on digital filters with H.W. Schuessler at Erlangen, receiving his Dr.-Ing. degree in 1975. His "Habilitation," in 1982, dealt with DFT / FFT realization and application. He expanded his research to spectral analysis and speech-signal processing. In 1987, he became a professor for digital signal processing at Ruhr-Universität, Bochum, Germany. Here, speech enhancement, instrumental speech-quality measurement, and speech-model signals became a central theme in his research. The same topics remained his focus when accepting a chair at Kiel in 1993.

Now, he holds a Chair for Circuit and System Theory within the Faculty of Engineering of Kiel University where he teaches signals and systems, system theory, digital signal processing and speech-signal processing. He has been active within EURASIP as an AdCom member since 1983, a Workshop Organizer in 1985 and 1989, Secretary-Treasurer from 1988 until 1994, President from 1994 until 1998, Past President until 2000, and member of the Advisory Board since then.

He coauthored a book on speech-signal processing (with P. Vary and W. Hess), is a (co-) author of several book chapters, and 25 journal plus 50 conference-proceedings contributions, and he is proud to see his group publish another 85 papers, without including him as author, plus 26 dissertations or theses.

The group's speech-enhancement work led to very good results with classical methods, more remarkably, however, to new insights and success by nonclassical approaches, such as artificial bandwidth enlargement as well as spectral subtraction in generalized spectral domains or with adaptive bandwidths. The group research on instrumental speech-quality measures has been restarted recently; after several general proposals, a particularly notable contribution is the work on algorithms for detecting certain coding and error-concealment algorithms in nonintrusive telephone-quality monitoring. The search for speech-signal models led to a participation in an ITU recommendation. The early digital-filter activities are still alive, and the group's expertise on filter and filter-bank design and their fixed-point realization makes it a well-accepted industry partner.

Especially, however, the group became one of few German centers of excellence in medium-to-low rate speech coding, contributing a hardware solution for the German pre-GSM test (no. 2 in the competition), hard- and software components in the GSM half-rate development phase within a EUREKA project, a competitive proposal for the enhanced full-rate system as well as hardware optimizations for MPEG-4 coders, both with strong industrial cooperation. More recently, activities in combined source-and-channel coding have produced exciting results.

EURASIP Technical Achievement Awards

The first Individual Technical Achievement Award for 2004 is awarded to Wolfgang Mecklenbräuker for his pioneering contributions in digital signal processing and his seminal work on reinterpreting the two-dimensional Wigner Distribution, from its use as a quantum-mechanical function, into an electronic time-frequency signal analysis technique.

Wolfgang F. G. Mecklenbruker, Dipl.-Ing, Dr.-Ing, is Professor at the Technical University of Vienna, Austria. He personally initiated much of the early work on adaptive filters, which have such importance for reducing distortions and noise in communication systems. Further, he made groundbreaking advances in our understanding of the limitations of quantisation and overflow errors from finite word-length arithmetic in digital filter hardware. His most significant technical achievement was the seminal three-part paper summarising his work on reinterpreting the two-dimensional Wigner Distribution, from its 1932 use as a quantum-mechanical function, into an electronic time-frequency signal analysis technique. This spawned a completely new area of research, culminating in the later discovery of the "wavelet" transform, which forms a vital part of today's image and video compression standards. He was an elected Fellow of the IEEE (USA) almost 2 decades ago, a testament to his considerable early research achievements. He has received the NTG Preis, IEEE Microwave & Communications Societies Paper Prize Awards. He was the recipient of a Fulbright Award and has been elected to both the Austrian and the New York Academies of Sciences.

As a Visiting Scholar at MIT (USA), he stimulated his interests in two-dimensional signal processing by researching with Mersereau, a leading authority in multidimensional signal processing. This work resulted in a number of papers defining design methods for these complicated filters.

At Philips Research Laboratories, he was a pioneer in two research areas. With Claasen, he contributed to the early work on digital and adaptive filters. The latter have major importance for reducing distortions and noise in communication systems, and are now vital components in the modem which lets you access the internet from your home computer. Further research, with Claasen and Peek, advanced our understanding of the limitations

of finite word-length arithmetic in hardware realisations of fixed-point digital filters and this was a groundbreaking achievement at the time. Not being content with these advances, in 1980, he and Claasen published the seminal three-part paper reinterpreting the two-dimensional Wigner Distribution from its original 1932 use as a quantum-mechanical function into an electronic time-frequency signal analysis technique. This analysis technique simultaneously measures a signals instantaneous power and spectral density and it spawned a completely new area of research in the 1980s and 1990s, culminating subsequently in the discovery of the "wavelet" transform, which is so important in today's photographic still and video image compression systems and is embodied in many standards. Time-frequency techniques are particularly important or relevant for analysing time-varying signals such as radar transmissions and bat sonar emissions, as well as in speech processing. Mecklenbruker summarised the research in this field in his seminal 1997 Elsevier textbook.

He moved in 1981 to the Technical University Vienna and he led from 1991 the "Nachrichtentecknik und Hochfrequenztechnik" Institute. Since 1981 he has grown his signal processing research group within the institute into one of the leading European research groups in time-frequency, speech, and signal processing techniques. He continues to publish on linear time invariant DSP system design.

The second Individual Technical Achievement Award for 2004 is awarded to Bernd Girod for his fundamental contributions to networked media systems, video signal compression and coding, and 3D image analysis and synthesis.

Bernd Girod is Professor of Electrical Engineering in the Information Systems Laboratory of Stanford University, California. He also holds a courtesy appointment with the Stanford Department of Computer Science and he serves as Director of the Image Systems Engineering Program at Stanford. His research interests include networked media systems, video signal compression and coding, and 3D image analysis and synthesis.

He received his M.S. degree in electrical engineering from Georgia Institute of Technology, in 1980, and his Doctoral degree "with highest honours" from the University of Hannover, Germany, in 1987. Until 1987 he was a Member of the research staff at the Institut für Theoretische Nachrichtentechnik und Informationsverarbeitung, University of Hannover, working on moving image coding, human visual perception, and information theory. In 1988, he joined Massachusetts Institute of Technology, Cambridge, Mass, USA, first as a Visiting Scientist with the Research Laboratory of Electronics, then as an Assistant Professor of Media Technology at the Media Laboratory. From 1990 to 1993, he was Professor of computer graphics and Technical Director of the Academy of Media Arts in Cologne, Germany, jointly appointed with the Computer Science Section, Cologne University. He was a Visiting Adjunct Professor with the Digital Signal Processing Group at Georgia Institute of Technology, Atlanta, Ga, USA, in 1993. From 1993 until 1999, he was Chaired Professor of electrical engineering/telecommunications at the University of Erlangen-Nuremberg, Germany, and the Head of the Telecommunications Institute I, codirecting the Telecommunications Laboratory. He has served as the Chairman of the Electrical Engineering Department from 1995 to 1997, and as a Director of the Center of Excellence "3D Image Analysis and Synthesis" from 1995 to 1999.

As an entrepreneur, Professor Girod has worked successfully with several start-up ventures as a founder, investor, director, or advisor. Most notably, he has been a cofounder and Chief Scientist of Vivo Software, Inc., Waltham, Mass (1993–1998); after Vivo's acquisition, 1998–2002, he was the Chief Scientist of RealNetworks, Inc. (Nasdaq: RNWK); and, since 1996, he has been an outside Director of 8×8, Inc. (Nasdaq: EGHT).

Professor Girod has authored or coauthored one major textbook, two monographs, and over 250 book chapters, journal articles and conference papers in his field, and he holds about 20 international patents. He has served as a member on the Editorial Board or as an Associate Editor for several journals in his field, and is currently Area Editor for Speech, Image, Video & Signal Processing of the "IEEE Transactions on Communications." He has served on numerous conference committees, e.g., as a Tutorial Chair of ICASSP-97 in Munich and ICIP-2000 in Vancouver, as a General Chair of the 1998 IEEE Image and Multidimensional Signal Processing Workshop in Alpbach, Austria, and as a General Chair of the Visual Communication and Image Processing Conference (VCIP) in San Jose, Calif, in 2001.

Professor Girod has been a Member of the IEEE Image and Multidimensional Signal Processing Committee from 1989 to 1997 and was elected Fellow of the IEEE in 1998 "for his contributions to the theory and practice of video communications." He has been named "Distinguished Lecturer" for the year 2002 by the IEEE Signal Processing Society. Together with J. Eggers, he was the recipient of the 2001 EURASIP Best Paper Award for Signal Processing.

His Image, Video, and Multimedia Systems Group forms part of the Information Systems Laboratory in the Stanford Department of Electrical Engineering, carrying out fundamental and applied research on various aspects of video compression, coding, and networked real-time media systems.

Best Paper Awards for EURASIP Journals

Signal Processing 2002 (sponsored by Robert Bosch GMBH)

"New multiscale transforms, minimum total variation synthesis: application to edge preserving image reconstruction," vol. 82, no. 11, November 2002, pp. 1519–1545.

Emmanuel J. Candes and Franck Guo, Department of Applied and Computational Mathematics, California Institute of Technology, Pasadena, Calif, 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.

Signal Processing 2003 (sponsored by Robert Bosch GMBH)

"Design of far-field and near-field broadband beamformers using eigenfilters," vol. 83, no. 12, December 2003, pp. 2641–2673.

Simon Doclo and Marc Moonen, Katholieke Universiteit, Leuven, Belgium.

This superb paper deals with a difficult practical problem, as that of broadband beamforming by using eigenfilters, and presents a comprehensive study of it. The analysis is very

elegant and it connects well with the practical issues that motivated the study. The paper has an important tutorial value since it reviews the main beamforming strategies proposed in the literature. It also studies new noniterative beamforming procedures based on eigenfilters. The performance of these new procedures is close to that obtained with a nonlinear cost function but exhibits lower computational complexity. All these methods are studied under near-field and far-field assumptions, which make the paper of major interest.

Members of the SP award subcommittee:

Nicholas Kalouptsidis, Steve Mc Laughlin, and Jean-Yves Tourneret

Image Communication 2002-2003

"A shape-based image retrieval method using salient edges," vol. 18, no. 2, February 2003, pp. 141–156.

Jun Wei Han and Lei Guo, Department of Automatic Control, Northwestern Polytechnical University, Xi'an, China.

This paper addresses an important and relevant research topic. Content-based image retrieval (CBIR) applications require improved methods in order to satisfy customer needs. In this paper a novel method is proposed and tested. It has theoretical background and it shows some improvement in CBIR performance compared to other systems in literature. The award subcommittee also believes that this paper will serve as one basic reference in the future scientific literature in this particular field.

Members of the IC award subcommittee:

Pauli Kuosmanen, Ioannis Pitas, and Carlo Regazzoni

Speech Communication 2001-2002

"Comparison of discriminative training criteria and optimization methods for speech recognition," vol. 34, pp. 287–310, 2001.

Ralf Schl uter, Wolfgang Macherey, Boris Müller, and Hermann Ney, Aachen University of Technology, Germany.

This paper builds up a framework for a class of discrimative training criteria and optimization methods for continuous speech recognition. New particular criteria are derived therefrom for specific applications. Analytical and experimental results are presented. The contribution covers important aspects of digital speech-signal processing in a unified presentation, provides the reader with essential insights, and is very well written.

Members of the SC award subcommittee:

Ulrich Heute (coordinator), Hynek Hermansky, Baastian Klein, and Luis Oliveira

EURASIP Journal on Applied Signal Processing 2001-2002

"Design and DSP implementation of fixed-point systems," vol. 2002, no. 9, September 2002, pp. 908–925.

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

Members of the EURASIP JASP award subcommittee:

Bastiaan Kleijn, Hideaki Sakai, and Philippe Regalia

EURASIP Journal on Applied Signal Processing 2003

"A vision chip for color segmentation and pattern matching," vol. 2003, no. 7, June 2003, pp. 703–712.

Ralph Etienne-Cummings, Philippe Pouliquen, M. Anthony Lewis, Iguana Robotics, Urbana, Ill, USA.

The paper by Etienne-Cummings, Pouliquen, and Lewis describes a 128×64 CMOS imager which integrates color segmentation and color-based object recognition. The chip uses a biologically inspired color opponent representation with an analogue lookup table the determine the hue, while saturation is obtained from a loser-take-all circuit, in addition to intensity obtained as the sum of color components. The claimed power consumption is but 1 mW at 30 frames per second.

Members of the EURASIP JASP award subcommittee:

Phil Regalia, Hideaki Sakai, and Jaakko Astola

Award

Ulrich Heute, born 1944 in Magdeburg, Germany, studied electrical engineering at Stuttgart, with a Dipl.-Ing. degree in 1970. He worked on digital filters with H.W. Schuessler at Erlangen, receiving his Dr.-Ing. degree in 1975. His "habilitation," in 1982, dealt with DFT/FFT realization and application. He expanded his research to spectral analysis and speech-signal processing. In 1987, he became a professor for digital signal processing at Ruhr-Universität, Bochum, Germany. Here, speech enhancement, instrumental speech-quality measurement, and speech-model signals became a central theme in his research. The same topics remained his focus when accepting a chair at Kiel in 1993.

Now, he holds a Chair for Circuit and System Theory within the Faculty of Engineering of Kiel University where he teaches signals and systems, system theory, digital signal processing, and speech-signal processing. He has been active within EURASIP as an AdCom Member since 1983, a Workshop Organizer in 1985 and 1989, Secretary-Treasurer from 1988 until 1994, President from 1994 until 1998, Past President until 2000, and Member of the Advisory Board since then.

He coauthored a book on speech-signal processing (with P. Vary and W. Hess), is a (co-)author of several book chapters, and 25 journal plus 50 conference-proceedings contributions, and he is proud to see his group publish another 85 papers, without including him as author, plus 26 dissertations or theses.

The group's speech-enhancement work led to very good results with classical methods, more remarkably, however, to new insights and success by nonclassical approaches, such as artificial bandwidth enlargement as well as spectral subtraction in generalized spectral domains or with adaptive bandwidths. The group research on instrumental speech-quality measures has been restarted recently; after several general proposals, a particularly notable contribution is the work on algorithms for detecting certain coding and error-concealment algorithms in nonintrusive telephone-quality monitoring. The search for speech-signal models led to a participation in an ITU recommendation. The early digital-filter activities are still alive, and the group's expertise on filter and filter-bank design and their fixed-point realization makes it a well accepted industry partner

Especially, however, the group became one of few German centers of excellence in medium-to-low rate speech coding, contributing a hardware solution for the German pre-GSM test (no. 2 in the competition), hard- and software components in the GSM half-rate development phase within a EUREKA project, a competitive proposal for the enhanced full-rate system, as well as hardware optimizations for MPEG-4 coders, both with strong industrial cooperation. More recently, activities in combined source-and-channel coding have produced exciting results.

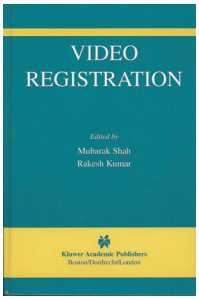
Calendar of Events

Year	Date	Event	Location	EURASIP Involvement	Chairperson/Information
2004	September 7–10	12th European Signal Processing Conference (EUSIPCO 2004)	Vienna, Austria	Sponsor	W. Mecklenbräuker http://www.nt.tuwien.ac.at/eusipco2004/
	September 13–15	11th Int. Workshop on Systems, Signals and Image Ambient Multimedia Processing (IWSSIP 2004)	Poznan, Poland	Cooperation	M. Domanski http://iwssip2004.et.put.poznan.pl
	October 5–8	7th Int. Conference on Dgital Audio Effects (DAFx 04)	Naples, Italy	Cooperation	G. Evangelista http://dafx04.na.infn.it
	December 14–16	6th Int. Conference on Mathematics in Signal Processing	Cirencester, UK	Cooperation	J. G. McWirter and I. K. Proudler http://www.ima.org.uk/mathematics/conferences.htm
2005	April 13–15	6th International Workshop on Image Analysis for Multimedia Interactive Services (WIAMIS 2005)	Montreux, Switzerland	Cooperation	F. Dufaux, T. Ebrahimi, and M. Strintzis http://amalia.img.lx.it.pt/wiamis2004/
	April 18–20	Int. Syposium on Mathematical Morphology (ISMM 05)	Paris, France	Cooperation	C. Ronsé, L. Najman, and E. Decenciére Ferrandiéres http://ismm05.esiee.fr/
	May 18–20	Nonlinear Signal and Image Processing (NSIP 2005)	Sapporo, Japan	Cooperation	Y. Miyama and P.-T. Yu http://www.ice.eng.hokudai.ac.jp/nsip/
	June 5–8	Int. Workshop on Geomic SP and Statistics	Rhode Island, USA	Cooperation	R. Liu and J. Astola http://www.newport.hyatt.com/property/index.jhtml
	September 11–14	Intl. Conference on Image Processing (ICIP-05)	Genova, Italy	Cooperation	G. Vernazza and G. Sicuranza http://www.icip05.org/

Sergios Theodoridis
Workshops/Confs Coordinator EURASIP

Video Registration

Edited by Mubarak Shah and Rakesh Kumar
Kluwer international series in video computing
Kluwer Academic Publisher, ISBN 1-4020-7460-3



Video registration in its broadest sense refers to models and techniques to align video frames with each other or with reference images and models. Video registration techniques play a key role in several video processing and management applications: video frame alignment can be used to stabilize video sequences or to create panoramic mosaics; background/foreground separation can be used for compression and indexing; computation of displacement fields between video frames can be used for 3D recovery; alignment of video frames with 3D models can be used for augmented reality, georegistration, and site and activity monitoring.

Organized into nine chapters, this book comprises a collection of studies from an IEEE workshop on video registration held in Vancouver in 2001. International experts document a well-balanced investigation on key issues including camera motion and 3D structure estimation, correlation-based and model-based image-to-image registration, construction of 3D models from aerial videos, registration of aerial images to 3D models, and registration of aerial video to reference imagery.

The opening and concluding chapters present a comprehensive perspective on key challenges and potential impacts of video registration on computer vision research. In Chapter 2, accurate estimation of camera motion and 3D structure from uncalibrated image sequences is addressed. In Chapter 3, a method to register images with subpixel accuracy based on phase correlation is presented. Image registration is also addressed in Chapter 5, where a feature-based approach to registration of images of urban scenes is described. Construction of 3D models through stereo mosaics based on aerial videos is addressed in Chapter 4. Finally, Chapters 6, 7, and 8 deal with registration of aerial video to 3D data. In Chapter 6, video is registered to 3D data in the form of site models for activity monitoring purposes. Differently, in Chapters 7 and 8 video data is aligned with reference imagery. Each chapter provides the reader with background and state of the art information about design and modelling principles that are relevant to the chapter's topic.

This book is a valid reference for academics, engineers, researchers, and graduates undertaking or planning research on video registration.

Pietro Pala

Pietro Pala is Assistant Professor of Computer Engineering at the University of Firenze, Italy, where he teaches database management systems and image and video processing. He is also Contract Professor of Multimedia Systems on internet engineering and content design of master's theses in Multimedia at the University of Firenze. His main scientific interests include pattern recognition, image and video databases, and content-based multimedia information retrieval. Pietro Pala is author of over 50 publications in the most distinguished international journals and conference proceedings and has served on the program committee of several international conferences and workshops.

New EURASIP Website to be Launched: http://www.EURASIP.org

EURASIP will be launching a new website in September 2004. The address remains the same at http://www.eurasip.org but new content and upgraded services make this a significant investment for our Association. Over the last few years EURASIP has effectively used their website to inform members about events, conferences, and special interest groups, to provide electronic access to Newsletters, and more recently to provide a resource to link the Ph.D. theses on signal and image processing that are available on the web.

The new website will provide the same information and services as the old site, but it also has a number of new features. First, EURASIP aims to move all membership subscription services and payment procedures to the web. Therefore we now have a new membership server, where one can join EURASIP for the first time, or exisiting members can renew their membership, change their profile, order new journals, and so on. EURASIP also plans to operate an e-mail-alert service (this is of course optional and members can opt not to receive email!) where information on forthcoming events will be sent to members. Finally a "news service" is planned where latest signal and image processing information from our members and the community at large will be presented.

The AdCom are pleased to again welcome Beatrice Pesquet to her appointment as a EURASIP Web Officer. She is currently closely involved with the transition to the new site. Beatrice will be coordinating the content and updates to the new website to ensure that it is current, offering useful content, and serving our members.

We look forward to your visit to the new website in September 2004, and hope that it would provide a more efficient and vibrant service for our members.

Bob Stewart
AdCom

Recent European Ph.D. Theses

Title: Error resilience for object-based video coding
Institution: Instituto Superior Técnico/Instituto de Telecomunicacoes, Portugal
Author: Luis Ducla Soares
Advisor: Fernando Pereira
Date: 20th April 2004
Link: www.img.lx.it.pt/publications/Theses/LuisDuclaSoares_PhD.pdf

Please send details of recent Ph.D. theses in the above format for publication in future Newsletters to Jonathon Chambers, e-mail: chambersj@cf.ac.uk.

Jonathon Chambers
EURASIP AdCom Academic Coordinator

Postdoc Openings

University of Edinburgh, School of Informatics
One Research Post - Computer Vision

Applications are invited for one post-doctoral researcher to work in the School of Informatics on an EPSRC-funded project entitled "BEHAVE: Computed-assisted prescreening of video streams for unusual activities." Informatics at Edinburgh is one of the top-ranked departments in Europe.

The researchers will investigate topics related to appearance-based recognition of human behaviour, such as might be applied to human safety and harm prevention. These include feature extraction, feature grouping, spatial and temporal attention, representing and recognising objects, contexts and situations, reactive integration, and control architectures.

See: http://homepages.inf.ed.ac.uk/rbf/BEHAVE/details.htm for more details.

Applicants for the post must have a Ph.D. in an appropriate area, such as computer vision or image processing, and should have experience with the MATLAB, C or C++ or JAVA programming languages.

The post is on the AR1A scale (18893-28279 pounds/annum). Placement for the post is according to experience and qualifications. The post is available from October 1, 2004 and will last until September 30, 2007.

Further particulars are available from, and applications are sent to:

Irene Madison (ipab-sec@informatics.ed.ac.uk)
School of Informatics
University of Edinburgh
Room 2107E, James Clerk Maxwell Building
The King's Buildings
Mayfield Road
Edinburgh EH9 3JZ
UK

Please quote reference number 3002012.

Closing date for applications is June 30, 2004.

The application form can be found soon at www.jobs.ed.ac.uk, which also allows online application.

Informal inquiries may be made to Bob Fisher: rbf@inf.ed.ac.uk.

2nd Call for Participation
2nd Summer School 'Cognitive Vision'

This is supported by the European Network for Research in Cognitive Computer Vision Systems (ECVision) and funded by the EC IST programme.

The summer school is at Hotel Activotel, Bonn, Germany (http://www.activotel.de/)

Dates: August 16–20, 2004

The goal of the summer school is to provide an intensive and challenging introduction to the area of cognitive computer vision. The summer school modules will be given by acknowledged experts in each key area:

Bob Fisher, UK
Martin Giese, Germany
Ales Leonardis, Slovenia
Markus Vincze, Austria
Monique Thonnat, France

The objective is to provide postgraduate students with a comprehensive introduction to all of the constituent areas of cognitive vision. This will help create a new generation of researchers in the area and will help maximize the impact of the ECVision network in the long run. In addition, it will provide practising researchers with an opportunity to learn about areas outside their main speciality and, hence, foster the cross-fertilization of ideas that is essential for real progress in the area.

We kindly invite you to participate. The deadline for application with a reduced fee is extended until July 15th, 2004.

For more information please visit our home page at http://www.ipb.uni-bonn.de/events/summerschool04/summerschool04.html

Wolfgang Förstner
Institut für Photogrammetrie
Nussallee 15
D-53121 Bonn
Germany

3 Research Fellow Posts in Computer Vision at The University of Reading
AVITRACK & SAFEE Projects
School of Systems Engineering

2 Research Fellows (AVITRACK project) Ref R0419
RA1A up to 24,951 per annum
Full-time, fixed-term to 31 January 2006

1 Research Fellow (SAFEE project) Ref R0420
RA1A up to 21,010 per annum
Full-time, fixed-term for three years

As a result of funding under the Space and Aeronautics Priority of the European Commission's 6th Framework Programme, we are seeking three researchers to work on two high-profile international research and development projects. The projects will employ computational vision techniques for object (people and vehicle) tracking, event detection, and behavioural analysis. Both projects aim to research, develop, and evaluate a multicamera vision system in an aviation context. Both projects are strongly commercially driven.

You will have a strong scientific background, ideally in computer science, experience with programming in C/C++ and with Linux. You will be knowledgeable of techniques in image processing and you will have either a Ph.D. degree, or have similar research experience in a relevant area. You will be enthusiastic about state-of-the-art research, development and technology translation towards commercial vision systems and working alongside academic and commercial partners to achieve goals within tight time constraints.

For more information please contact Dr. James Ferryman

E-mail: j.m.ferryman@reading.ac.uk, Tel: +44(0) 118 378 6697.

More details on the projects can be found respectively at:

http://europa.eu.int/comm/research/aeronautics/info/news/news26_en.html
http://europa.eu.int/comm/research/aeronautics/info/news/news24_en.html

Application forms are available from

Personnel Office
The University of Reading
Whiteknights
P.O. Box 217
Reading, RG6 6AH
Tel: 0118 378 6771 (voicemail)
E-mail: personnel@reading.ac.uk giving full name and address.

Application forms are also available at www.reading.ac.uk/Jobs.

Closing date for applications is 16 July 2004.

It is hoped that the chosen candidates will be able to start as soon as possible. Please quote appropriate reference number.

Short Course Report: DSP for FPGA Short Course, May 2004

EURASIP cosponsored the DSP for FPGAs short course which was presented in Livingston, Scotland, UK, in May 2004. The course was presented in association with the Institute of System Level Integration (ISLI, http://www.sli-institute.ac.uk) and more than 20 engineers from industry attended. The course leader was Professor Bob Stewart (r.stewart@eee.strath.ac.uk) of the University of Strathclyde, with technical presentation and laboratory support from Dr. Daniel Garcia-Alis, Dr. Garrey Rice, Kenneth Macpherson, Iain Stirling, Steven Alexander, and Graham Freeland.

The aim of the four-day course was to educate electronic engineers from varying backgrounds in the design and implementation of DSP algorithms on FPGAs with a particular emphasis on digital communications. The course was presented as around 50% lecture and 50% hand-on experience designing real time DSP algorithms for FPGA implementation. Attendees were from a wide range of backgrounds including DSP engineers, FPGA engineers, Satellite designers, and ASIC designers, from a wide variety of companies including EADS-Astrium, Elixent, Epson, IMRA Europe, Hewlett Packard, KOP, National Semiconductor, Pentland Systems, Philips, QinetiQ, and Wolfson Microelectronics.

The general syllabus topics that were presented in the course are given below:

- Introduction to DSP Hardware Technologies
- Linear Systems DSP Algorithm Review
- FPGA Technology
- DSP Arithmetic Fundamentals
- FPGA Elements for DSP Algorithms
- Signal Flow Graph (SFG) Techniques
- "Strategic" Digital Filtering for FPGAs
- Channel Coding and Decoding
- Adaptive DSP Algorithms and Applications
- DSP-Enabled Communications using FPGAs
- Timing and Sychronisation Issues
- Digital Mobile and Wireless Communication Case Studies

All attendees received a five-volume set of class notes with more than 2000 slides, and an educational copy of HDL Design Studio running on SystemView DSP design software.

In the next five years we can anticipate more communication standards, more pervasive computing, and more data available on the move—anytime and anywhere. The requirements for processing speeds of the order 10's or 100's of billions of operations per second, rapid prototyping, and software definable architectures will further the penetration of FPGAs into the DSP communication market. Traditionally FPGAs have provided an integrated device for sequential and combinational logic operations in many applications. However more recently the increased flexibility, programmability, and capability of FPGAs for fast and customised multiply-accumulate has meant that, in many DSP applications, they are the first choice for hardware verification, rapid prototyping, and in many cases, final product and system design. Therefore the demand for high speech DSP on FPGAs and similar technologies such as structured ASICs is only expected to grow in the next few years.

The course is planned to run again in the autumn from 11th–14th October 2004. More details from sian.williams@sli-institute.ac.uk or visit the ISLI website or EURASIP websites.

SIGNAL PROCESSING

Editorial Policy

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

- dissemination of research results and of engineering developments to all signal processing groups and individuals;
- presentation of practical solutions to current signal processing problems in engineering and science.

The editorial policy and the technical content of the journal are the responsibility of the Editor-in-Chief and the Editorial Board. The journal is self-supporting from the subscription income and contains a minimum amount of advertisements. Advertisements are subject to the prior approval of the Editor-in-Chief. The journal welcomes contributions from every country in the world.

Scope

Signal Processing incorporates all aspects of the theory and practice of signal processing (analogue and digital). It features original research work, tutorial and review articles, and accounts of practical developments. It is intended for a rapid dissemination of knowledge and experience to engineers and scientists working on signal processing research, development, or practical application.

Subjects

Subject areas covered by the journal include: Signal Theory; Stochastic Processes; Detection and Estimation; Spectral Analysis; Filtering; Communication Signal Processing; Biomedical Signal Processing; Geophysical and Astrophysical Signal Processing; Earth Resources Signal Processing; Acoustic and Vibration Signal Processing; Signal Processing Systems; Software Developments; Image Processing; Pattern Recognition; Optical Signal Processing; Multidimensional Signal Processing; Data Processing; Remote Sensing; Signal Processing Technology; Speech Processing; Radar Signal Processing; Sonar Signal Processing; Special Signal Processing; Industrial Applications; New Applications.

Editor-in-Chief

SIGNAL PROCESSING: IMAGE COMMUNICATION

Editorial Policy

Signal Processing: Image Communication is an international journal for the development of the theory and practice of image communication. Its primary objectives are the following:

- to present a forum for the advancement of the theory and practice of image communication;
- to simulate cross fertilization between areas similar in nature which have traditionally been separated, for example, various aspects of visual communications and information systems;
- to contribute to a rapid information exchange between the industrial and academic environments.

The editorial policy and the technical content of the journal are the responsibility of the Editor-in-Chief and the Editorial Board. The journal is self-supporting from the subscription income and contains a minimum amount of advertisements.

Advertisements are subject to the prior approval of the Editor-in-Chief. The journal welcomes contributions from every country in the world.

Scope

Signal Processing: Image Communication publishes articles relating to aspects of design, implementation, and use of image communication systems. Signal Processing: Image Communication features original research work, tutorial and review articles, and accounts of practical developments.

Subjects

Subject areas covered by the journal include: TV, HDTV, and 3DTV systems; Visual Science; Image; TV and Advanced TV; Broadcasting; Image Storage and Retrieval; Graphic Arts; Electronic Printing; Image Transmission; Interactive Image Coding Communication; Imaging Technology; Display Technology; VLSI Processors for Image Communications.

Editor-in-Chief

SPEECH COMMUNICATION

Editorial Policy

The journal's primary objectives are the following:

- to present a forum for the advancement of human and human-machine speech communication science;
- to stimulate cross fertilization between different fields of this domain;
- to contribute towards the rapid and wide diffusion of scientifically sound contributions in this domain.

Speech Communication is an interdisciplinary journal whose primary objective is to fulfill the need for the rapid dissemination and thorough discussion of basic and applied research results. In order to establish frameworks of inter-relate results from the various areas of the field, emphasis will be placed on viewpoints and topics of a transdisciplinary nature. The editorial policy and the technical content of the journal are the responsibility of the Editors and the Institutional Representatives. The Institutional Representatives assist the Editors in the definition and the control of editorial policy as well as in maintaining connections with scientific associations, international congresses, and regional events. The Editorial Board contributes towards the gathering of material for publication and assists the Editors in the editorial process.

Scope

Speech Communication is an interdisciplinary journal for the development and dissemination of all basic and applied aspects of speech communication processes. Speech Communication features original research work, tutorial and review articles dealing with the theoretical, empirical, and practical aspects of this scientific field.

Subject Coverage

Subject areas covered in this journal include:

- Basics of oral communication and dialogue: modelling of production and perception processes; phonetics and phonology; syntax; semantics of speech communication; cognitive aspects.
- Models and tools for language learning: functional organisation and developmental models of human language capabilities; acquisition and rehabilitation of spoken language; speech and hearing defects and aids.
- Speech signal processing: analysis; coding; transmission; enhancement, robustness to noise.
- Models for automatic speech communication: speech recognition; language identification; speaker recognition; speech synthesis; oral dialogue.

- Development and evaluation tools: monolingual and multilingual databases; assessment methodologies; specialised hardware and software packages; field experiments; market development.
- Multimodal human-computer interface: using speech I/O in combination with modalities, for example, gesture and handwriting.

EURASIP JOURNAL ON APPLIED SIGNAL PROCESSING

Scope

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.

There are two different issues: regular issues and special issues. The regular issues publish collections of papers without special solicitation. The special issues have 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.

Subjects

Subject areas include, but are by no means limited to:

- Signal processing theory, algorithm, architecture, design, and implementation
- Speech processing, coding, compression, and recognition
- Audio signal processing, coding, and compression
- Image/video processing, coding, compression, restoration, analysis and understanding, and communications
- Multimedia signal processing and technology
- 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
- Signal processing for smart sensor and systems

Application areas include, but not limited to: communications; networking; sensors and actuators; radar and sonar; medical imaging; biomedical applications; remote sensing; consumer electronics; computer vision; pattern recognition; robotics; fiber optic sensing/transducers; industrial automation; transportation; stock market and financial analysis; seismography; avionics.

Indexed/Abstracted In

The articles of the EURASIP JASP are reviewed/indexed in Acoustics Abstracts; Computer and Communications Security Abstracts (CCSA); Current Contents Engineering, Computing & Technology; INSPEC; Science Citation Index Expanded; and Technology and Management (TEMA).

Editor-in-Chief

EURASIP JOURNAL ON WIRELESS COMMUNICATIONS AND NETWORKING

Scope

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 highlights 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 welcomed. 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.

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 wideband systems; Wireless network services and medium access control.

Editor-in-Chief

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Special Issue on
Object-Based and Semantic Image and Video Analysis

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

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. Ten papers have been selected following the reviewing process and appear in this issue, which are briefly described below.

In the first paper, Cavallaro and Ebrahimi tackle semantic video object extraction by interacting between color change detection and region-based processing, achieving high spatial accuracy and temporal coherence. In the second paper, H.-Y. Wang and Ma propose a video object segmentation approach, involving image segmentation and motion estimation; the approach is based on spatial-constrained motion mask generation and motion-constrained spatial region merging.

Video object segmentation is also the topic of the third paper by Porikli and Y. Wang. The authors perform a spatio-temporal decomposition of the data, defining simple homogeneous, in terms of low-level visual descriptors, components; the latter, called volumes, are then expanded and grouped into objects, using hierarchical clustering. In the next paper, Li et al. use a Markov random field model to obtain object-based semantic image segmentation, focusing on remote sensing applications; their approach includes a Wold model decomposition of the original image generating both stochastic and structural texture image components.

The next two papers deal with technologies used in semantic image and video object analysis. In the first paper, Tsechpenakis et al. propose a model-based snake approach for object tracking, using a priori shape knowledge; a probabilistic rule-based approach is thus derived that copes with objects in cluttered and partially occluded scenes. In the second paper, Caldelli et al. analyze how estimation of objects' motion parameters can effectively be obtained, using appropriate MRF modeling and simple motion models.

The following two papers deal with content-based image retrieval. They both start with unsupervised image segmentation. In the first, R. Zhang and Z. Zhang use color object analysis and compute fuzzy color, texture, and shape parameters of the objects of the images. They also use clustering to obtain efficiency in the retrieval. In the second, Mezaris et al. extract similar low-level descriptors, forming a simple object ontology, which is used next for defining semantic objects. Relevance feedback is used here in the retrieval process.

The last two papers deal with specific applications of image and video analysis. In the first, Maragos et al. present an integrated system for the estimation of the bioecological quality of soils from analysis of soil section images, focusing on efficient extraction of multiscale geometric features from the data and object-oriented image analysis and using a neurofuzzy inference procedure. In the second paper, M. Kampmann proposes a maximum a posteriori algorithm for efficient chin and cheek contours estimation in video sequences, exploiting a priori knowledge about the shape and position of the contours.

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Volume 2004, No. 6, 1 June 2004

Contents and Abstracts

Interaction between High-Level and Low-Level Image Analysis for Semantic Video Object Extraction

Andrea Cavallaro and Touradj Ebrahimi

http://dx.doi.org/10.1155/S1110865704402157

The task of extracting a semantic video object is split into two subproblems, namely, object segmentation and region segmentation. Object segmentation relies on *a priori* assumptions, whereas region segmentation is data-driven and can be solved in an automatic manner. These two subproblems are not mutually independent, and they can benefit from interactions with each other. In this paper, a framework for such interaction is formulated. This representation scheme based on region segmentation and semantic segmentation is compatible with the view that image analysis and scene understanding problems can be decomposed into low-level and high-level tasks. Low-level tasks pertain to region-oriented processing, whereas the high-level tasks are closely related to object-level processing. This approach emulates the human visual system: what one "sees" in a scene depends on the scene itself (region segmentation) as well as on the cognitive task (semantic segmentation) at hand. The higher-level segmentation results in a partition corresponding to semantic video objects. Semantic video objects do not usually have invariant physical properties and the definition depends on the application. Hence, the definition incorporates complex domain-specific knowledge and is not easy to generalize. For the specific implementation used in this paper, motion is used as a clue to semantic information. In this framework, an automatic algorithm is presented for computing the semantic partition based on color change detection. The change detection strategy is designed to be immune to the sensor noise and local illumination variations. The lower-level segmentation identifies the partition corresponding to perceptually uniform regions. These regions are derived by clustering in an N-dimensional feature space, composed of static as well as dynamic image attributes. We propose an interaction mechanism between the semantic and the region partitions which allows to cope with multiple simultaneous objects. Experimental results show that the proposed method extracts semantic video objects with high spatial accuracy and temporal coherence.

Spatio-Temporal Video Object Segmentation via Scale-Adaptive 3D Structure Tensor

Hai-Yun Wang and Kai-Kuang Ma

http://dx.doi.org/10.1155/S111086570440122X

To address multiple motions and deformable objects' motions encountered in existing region-based approaches, an automatic video object (VO) segmentation methodology is proposed in this paper by exploiting the duality of image segmentation and motion estimation such that spatial and temporal information could assist each other to jointly yield much improved segmentation results. The key novelties of our method are (1) *scale-adaptive* tensor computation, (2) *spatial-constrained* motion mask generation *without* invoking dense

motion-field computation, (3) *rigidity analysis*, (4) motion mask generation and selection, and (5) *motion-constrained* spatial region merging. Experimental results demonstrate that these novelties jointly contribute much more accurate VO segmentation both in spatial and temporal domains.

Automatic Video Object Segmentation Using Volume Growing and Hierarchical Clustering

Fatih Porikli and Yao Wang

http://dx.doi.org/10.1155/S1110865704401152

We introduce an automatic segmentation framework that blends the advantages of color-, texture-, shape-, and motion-based segmentation methods in a computationally feasible way. A spatiotemporal data structure is first constructed for each group of video frames, in which each pixel is assigned a feature vector based on low-level visual information. Then, the smallest homogeneous components, so-called volumes, are expanded from selected marker points using an adaptive, three-dimensional, centroid-linkage method. Self descriptors that characterize each volume and relational descriptors that capture the mutual properties between pairs of volumes are determined by evaluating the boundary, trajectory, and motion of the volumes. These descriptors are used to measure the similarity between volumes based on which volumes are further grouped into objects. A fine-to-coarse clustering algorithm yields a multiresolution object tree representation as an output of the segmentation.

Object-Based and Semantic Image Segmentation Using MRF

Feng Li, Jiaxiong Peng, and Xiaojun Zheng

http://dx.doi.org/10.1155/S1110865704402182

The problem that the Markov random field (MRF) model captures the structural as well as the stochastic textures for remote sensing image segmentation is considered. As the one-point clique, namely, the external field, reflects the priori knowledge of the relative likelihood of the different region types which is often unknown, one would like to consider only two-pairwise clique in the texture. To this end, the MRF model cannot satisfactorily capture the structural component of the texture. In order to capture the structural texture, in this paper, a reference image is used as the external field. This reference image is obtained by Wold model decomposition which produces a purely random texture image and structural texture image from the original image. The structural component depicts the periodicity and directionality characteristics of the texture, while the former describes the stochastic. Furthermore, in order to achieve a good result of segmentation, such as improving smoothness of the texture edge, the proportion between the external and internal fields should be estimated by regarding it as a parameter of the MRF model. Due to periodicity of the structural texture, a useful by-product is that some long-range interaction is also taken into account. In addition, in order to reduce computation, a modified version of parameter estimation method is presented. Experimental results on remote sensing image demonstrating the performance of the algorithm are presented.

Rule-Driven Object Tracking in Clutter and Partial Occlusion with Model-Based Snakes

Gabriel Tsechpenakis, Konstantinos Rapantzikos, Nicolas Tsapatsoulis, and Stefanos Kollias

http://dx.doi.org/10.1155/S1110865704401103

In the last few years it has been made clear to the research community that further improvements in classic approaches for solving low-level computer vision and image/video understanding tasks are difficult to obtain. New approaches started evolving, employing knowledge-based processing, though transforming a priori knowledge to low-level models and rules are far from being straightforward. In this paper, we examine one of the most popular active contour models, snakes, and propose a snake model, modifying terms and introducing a model-based one that eliminates basic problems through the usage of prior shape knowledge in the model. A probabilistic rule-driven utilization of the proposed model follows, being able to handle (or cope with) objects of different shapes, contour complexities and motions; different environments, indoor and outdoor; cluttered sequences; and cases where background is complex (not smooth) and when moving objects get partially occluded. The proposed method has been tested in a variety of sequences and the experimental results verify its efficiency.

An Algorithm for Motion Parameter Direct Estimate

Roberto Caldelli, Franco Bartolini, and Vittorio Romagnoli

http://dx.doi.org/10.1155/S1110865704401012

Motion estimation in image sequences is undoubtedly one of the most studied research fields, given that motion estimation is a basic tool for disparate applications, ranging from video coding to pattern recognition. In this paper a new methodology which, by minimizing a specific potential function, directly determines for each image pixel the motion parameters of the object the pixel belongs to is presented. The approach is based on Markov random fields modelling, acting on a first-order neighborhood of each point and on a simple motion model that accounts for rotations and translations. Experimental results both on synthetic (noiseless and noisy) and real world sequences have been carried out and they demonstrate the good performance of the adopted technique. Furthermore a quantitative and qualitative comparison with other well-known approaches has confirmed the goodness of the proposed methodology.

A Robust Color Object Analysis Approach to Efficient Image Retrieval

Ruofei Zhang and Zhongfei (Mark) Zhang

http://dx.doi.org/10.1155/S111086570431214X

We describe a novel indexing and retrieval methodology integrating color, texture, and shape information for content-based image retrieval in image databases. This methodology, we call CLEAR, applies unsupervised image segmentation to partition an image into a

set of objects. Fuzzy color histogram, fuzzy texture, and fuzzy shape properties of each object are then calculated to be its signature. The fuzzification procedures effectively resolve the recognition uncertainty stemming from color quantization and human perception of colors. At the same time, the fuzzy scheme incorporates segmentation-related uncertainties into the retrieval algorithm. An adaptive and effective measure for the overall similarity between images is developed by integrating properties of all the objects in every image. In an effort to further improve the retrieval efficiency, a secondary clustering technique is developed and employed, which significantly saves query processing time without compromising retrieval precision. A prototypical system of CLEAR, we developed, demonstrated the promising retrieval performance and robustness in color variations and segmentation-related uncertainties for a test database containing 10 000 general-purpose color images, as compared with its peer systems in the literature.

Region-Based Image Retrieval Using an Object Ontology and Relevance Feedback

Vasileios Mezaris, Ioannis Kompatsiaris, and Michael G. Strintzis

http://dx.doi.org/10.1155/S1110865704401188

An image retrieval methodology suited for search in large collections of heterogeneous images is presented. The proposed approach employs a fully unsupervised segmentation algorithm to divide images into regions and endow the indexing and retrieval system with content-based functionalities. Low-level descriptors for the color, position, size, and shape of each region are subsequently extracted. These arithmetic descriptors are automatically associated with appropriate qualitative intermediate-level descriptors, which form a simple vocabulary termed *object ontology*. The object ontology is used to allow the qualitative definition of the high-level concepts the user queries for (*semantic objects*, each represented by a *keyword*) and their relations in a human-centered fashion. When querying for a specific semantic object (or objects), the intermediate-level descriptor values associated with both the semantic object and all image regions in the collection are initially compared, resulting in the rejection of most image regions as irrelevant. Following that, a relevance feedback mechanism, based on support vector machines and using the low-level descriptors, is invoked to rank the remaining potentially relevant image regions and produce the final query results. Experimental results and comparisons demonstrate, in practice, the effectiveness of our approach.

Image Analysis of Soil Micromorphology: Feature Extraction, Segmentation, and Quality Inference

Petros Maragos, Anastasia Sofou, Giorgos B. Stamou, Vassilis Tzouvaras, Efimia Papatheodorou, and George P. Stamou

http://dx.doi.org/10.1155/S1110865704402054

We present an automated system that we have developed for estimation of the bioecological quality of soils using various image analysis methodologies. Its goal is to analyze soilsection images, extract features related to their micromorphology, and relate the visual features to various degrees of soil fertility inferred from biochemical characteristics of the soil. The

image methodologies used range from low-level image processing tasks, such as nonlinear enhancement, multiscale analysis, geometric feature detection, and size distributions, to object-oriented analysis, such as segmentation, region texture, and shape analysis.

MAP Estimation of Chin and Cheek Contours in Video Sequences

Markus Kampmann

http://dx.doi.org/10.1155/S111086570440208X

An algorithm for the estimation of chin and cheek contours in video sequences is proposed. This algorithm exploits a priori knowledge about shape and position of chin and cheek contours in images. Exploiting knowledge about the shape, a parametric 2D model representing chin and cheek contours is introduced. Exploiting knowledge about the position, a MAP estimator is developed taking into account the observed luminance gradient as well as a priori probabilities of chin and cheek contours positions. The proposed algorithm was tested with head and shoulder video sequences (image resolution CIF). In nearly 70% of all investigated video frames, a subjectively error free estimation could be achieved. The 2D estimate error is measured as on average between 2.4 and 2.9 pel.

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Special Issue on Model-Based Sound Synthesis

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Model-based sound synthesis has become one of the most active research topics in musical signal processing and in musical acoustics. The earliest attempts in generating musical sound with a physical model were made over three decades ago. The first commercial products were seen only some twenty 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 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 as 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 roots of this special issue are in a European project called ALMA (Algorithms for the Modelling of Acoustic Interactions, IST-2001-33059, see http://www-dsp.elet.polimi.it/alma/) where the guest editors and their research teams collaborated in the period from 2001 to 2004. The goal of the ALMA project was to develop an elegant, general, and unifying strategy for a blockwise design of physical models for sound synthesis. A "divide-and-conquer" approach was taken, in which the elements of the structure are individually modeled and discretized, while their interaction topology is separately designed and implemented in a dynamical and physically sound fashion. As a result, several high-quality demonstrations of virtual musical instruments played in a virtual environment were developed. During the ALMA project, the guest editors realized that this special issue could be created, since the field was very active but there had not been a special issue devoted to it for a long time.

This EURASIP JASP special issue presents ten examples of recent research in model-based sound synthesis. The first two papers are related to keyboard instruments. First Giordano and Jiang discuss physical modeling synthesis of the piano using the finite-difference approach. Then Välimäki et al. show how to synthesize the sound of the harpsichord based on measurements of a real instrument. An efficient implementation using a visual software synthesis package is given for real-time synthesis.

In the third paper, Trautmann and Rabenstein present a multirate implementation of a vibrating string model that is based on the functional transformation method. In the next paper, Testa et al. investigate the modeling of stiff string behavior. The dispersive wave phenomenon, perceivable as inharmonicity in many string instrument sounds, is studied by deriving different physically inspired models.

In the fourth paper, Karjalainen and Erkut propose a very interesting and general solution to the problem of how to build composite models from digital waveguides and finite-difference time-domain blocks. The next contribution is from Guillemain, who proposes a real-time synthesis model of double-reed wind instruments based on a nonlinear physical model.

The paper by Howard and Rimell provides a viewpoint quite different from the others in this special issue. It deals with the design and implementation of user interfaces for model-based synthesis. An important aspect is the incorporation of tactile feedback into the interface.

Arroabarren and Carlosena have studied the modeling and analysis of human voice production, particularly the vibrato used in the singing voice. Source-filter modeling and sinusoidal modeling are compared to gain a deeper insight in these phenomena. Bensa et al. bring the discussion back to the physical modeling of musical instruments, with particular reference to the piano. They propose a source/resonator model of hammer-string interaction aimed at a realistic production of piano sound. Finally, Glass and Fukuodome incorporate a plucked-string model into an audio coder for audio compression and instrument synthesis.

The guest editors would like to thank all the authors for their contributions. We would also like to express our deep gratitude to the reviewers for their diligent efforts in evaluating all submitted manuscripts. We hope that this special issue will stimulate further research work on model-based sound synthesis.

Vesa Välimäki
Augusto Sarti
Matti Karjalainen
Rudolf Rabenstein
Lauri Savioja

Volume 2004, No. 7, 15 June 2004

Contents and Abstracts

Physical Modeling of the Piano

N. Giordano and M. Jiang

http://dx.doi.org/10.1155/S111086570440105X

A project aimed at constructing a physical model of the piano is described. Our goal is to calculate the sound produced by the instrument entirely from Newton's laws. The structure of the model is described along with experiments that augment and test the model calculations. The state of the model and what can be learned from it are discussed.

Sound Synthesis of the Harpsichord Using a Computationally Efficient Physical Model

Vesa Välimäki, Henri Penttinen, Jonte Knif, Mikael Laurson, and Cumhur Erkut

http://dx.doi.org/10.1155/S111086570440211X

A sound synthesis algorithm for the harpsichord has been developed by applying the principles of digital waveguide modeling. A modification to the loss filter of the string model is introduced that allows more flexible control of decay rates of partials than is possible with a one-pole digital filter, which is a usual choice for the loss filter. A version of the commuted waveguide synthesis approach is used, where each tone is generated with a parallel combination of the string model and a second-order resonator that are excited with a common excitation signal. The second-order resonator, previously proposed for this purpose, approximately simulates the beating effect appearing in many harpsichord tones. The characteristic key-release thump terminating harpsichord tones is reproduced by triggering a sample that has been extracted from a recording. A digital filter model for the soundboard has been designed based on recorded bridge impulse responses of the harpsichord. The output of the string models is injected in the soundboard filter that imitates the reverberant nature of the soundbox and, particularly, the ringing of the short parts of the strings behind the bridge.

Multirate Simulations of String Vibrations Including Nonlinear Fret-String Interactions Using the Functional Transformation Method

L. Trautmann and R. Rabenstein

http://dx.doi.org/10.1155/S1110865704312059

The functional transformation method (FTM) is a well-established mathematical method for accurate simulations of multidimensional physical systems from various fields of science, including optics, heat and mass transfer, electrical engineering, and acoustics. This paper applies the FTM to real-time simulations of transversal vibrating strings. First, a

physical model of a transversal vibrating lossy and dispersive string is derived. Afterwards, this model is solved with the FTM for two cases: the ideally linearly vibrating string and the string interacting nonlinearly with the frets. It is shown that accurate and stable simulations can be achieved with the discretization of the continuous solution at audio rate. Both simulations can also be performed with a multirate approach with only minor degradations of the simulation accuracy but with preservation of stability. This saves almost 80% of the computational cost for the simulation of a six-string guitar and therefore it is in the range of the computational cost for digital waveguide simulations.

Physically Inspired Models for the Synthesis of Stiff Strings with Dispersive Waveguides

I. Testa, G. Evangelista, and S. Cavaliere

http://dx.doi.org/10.1155/S1110865704402200

We review the derivation and design of digital waveguides from physical models of stiff systems, useful for the synthesis of sounds from strings, rods, and similar objects. A transform method approach is proposed to solve the classic fourth-order equations of stiff systems in order to reduce it to two second-order equations. By introducing scattering boundary matrices, the eigenfrequencies are determined and their n^2 dependency is discussed for the clamped, hinged, and intermediate cases. On the basis of the frequency-domain physical model, the numerical discretization is carried out, showing how the insertion of an all-pass delay line generalizes the Karplus-Strong algorithm for the synthesis of ideally flexible vibrating strings. Knowing the physical parameters, the synthesis can proceed using the generalized structure. Another point of view is offered by Laguerre expansions and frequency warping, which are introduced in order to show that a stiff system can be treated as a nonstiff one, provided that the solutions are warped. A method to compute the all-pass chain coefficients and the optimum warping curves from sound samples is discussed. Once the optimum warping characteristic is found, the length of the dispersive delay line to be employed in the simulation is simply determined from the requirement of matching the desired fundamental frequency. The regularization of the dispersion curves by means of optimum unwarping is experimentally evaluated.

Digital Waveguides versus Finite Difference Structures: Equivalence and Mixed Modeling

Matti Karjalainen and Cumhur Erkut

http://dx.doi.org/10.1155/S1110865704401176

Digital waveguides and finite difference time domain schemes have been used in physical modeling of spatially distributed systems. Both of them are known to provide exact modeling of ideal one-dimensional (1D) band-limited wave propagation, and both of them can be composed to approximate two-dimensional (2D) and three-dimensional (3D) mesh structures. Their equal capabilities in physical modeling have been shown for special cases and have been assumed to cover generalized cases as well. The ability to form mixed models by joining substructures of both classes through converter elements has been proposed recently. In this paper, we formulate a general digital signal processing (DSP)-oriented

framework where the functional equivalence of these two approaches is systematically elaborated and the conditions of building mixed models are studied. An example of mixed modeling of a 2D waveguide is presented.

A Digital Synthesis Model of Double-Reed Wind Instruments

Ph. Guillemain

http://dx.doi.org/10.1155/S1110865704402194

We present a real-time synthesis model for double-reed wind instruments based on a nonlinear physical model. One specificity of double-reed instruments, namely, the presence of a confined air jet in the embouchure, for which a physical model has been proposed recently, is included in the synthesis model. The synthesis procedure involves the use of the physical variables via a digital scheme giving the impedance relationship between pressure and flow in the time domain. Comparisons are made between the behavior of the model with and without the confined air jet in the case of a simple cylindrical bore and that of a more realistic bore, the geometry of which is an approximation of an oboe bore.

Real-Time Gesture-Controlled Physical Modelling Music Synthesis with Tactile Feedback

David M. Howard and Stuart Rimell

http://dx.doi.org/10.1155/S1110865704311182

Electronic sound synthesis continues to offer huge potential possibilities for the creation of new musical instruments. The traditional approach is, however, seriously limited in that it incorporates only auditory feedback and it will typically make use of a sound synthesis model (e.g., additive, subtractive, wavetable, and sampling) that is inherently limited and very often nonintuitive to the musician. In a direct attempt to challenge these issues, this paper describes a system that provides tactile as well as acoustic feedback, with real-time synthesis that invokes a more intuitive response from players since it is based upon mass-spring physical modelling. Virtual instruments are set up via a graphical user interface in terms of the physical properties of basic well-understood sounding objects such as strings, membranes, and solids. These can be interconnected to form complex integrated structures. Acoustic excitation can be applied at any point mass via virtual bowing, plucking, striking, specified waveform, or from any external sound source. Virtual microphones can be placed at any point masses to deliver the acoustic output. These aspects of the instrument are described along with the nature of the resulting acoustic output.

Vibrato in Singing Voice: The Link between Source-Filter and Sinusoidal Models

Ixone Arroabarren and Alfonso Carlosena

http://dx.doi.org/10.1155/S1110865704401127

The application of inverse filtering techniques for high-quality singing voice analysis/synthesis is discussed. In the context of source-filter models, inverse filtering provides

a noninvasive method to extract the voice source, and thus to study voice quality. Although this approach is widely used in speech synthesis, this is not the case in singing voice. Several studies have proved that inverse filtering techniques fail in the case of singing voice, the reasons being unclear. In order to shed light on this problem, we will consider here an additional feature of singing voice, not present in speech: the *vibrato*. Vibrato has been traditionally studied by sinusoidal modeling. As an alternative, we will introduce here a novel noninteractive source filter model that incorporates the mechanisms of vibrato generation. This model will also allow the comparison of the results produced by inverse filtering techniques and by sinusoidal modeling, as they apply to singing voice and not to speech. In this way, the limitations of these conventional techniques, described in previous literature, will be explained. Both synthetic signals and singer recordings are used to validate and compare the techniques presented in the paper.

A Hybrid Resynthesis Model for Hammer-String Interaction of Piano Tones

Julien Bensa, Kristoffer Jensen, and Richard Kronland-Martinet

http://dx.doi.org/10.1155/S1110865704402121

This paper presents a source/resonator model of hammer-string interaction that produces realistic piano sound. The source is generated using a subtractive signal model. Digital waveguides are used to simulate the propagation of waves in the resonator. This hybrid model allows resynthesis of the vibration measured on an experimental setup. In particular, the nonlinear behavior of the hammer-string interaction is taken into account in the source model and is well reproduced. The behavior of the model parameters (the resonant part and the excitation part) is studied with respect to the velocities and the notes played. This model exhibits physically and perceptually related parameters, allowing easy control of the sound produced. This research is an essential step in the design of a complete piano model.

Warped Linear Prediction of Physical Model Excitations with Applications in Audio Compression and Instrument Synthesis

Alexis Glass and Kimitoshi Fukudome

http://dx.doi.org/10.1155/S1110865704402078

A sound recording of a plucked string instrument is encoded and resynthesized using two stages of prediction. In the first stage of prediction, a simple physical model of a plucked string is estimated and the instrument excitation is obtained. The second stage of prediction compensates for the simplicity of the model in the first stage by encoding either the instrument excitation or the model error using warped linear prediction. These two methods of compensation are compared with each other, and to the case of single-stage warped linear prediction, adjustments are introduced, and their applications to instrument synthesis and MPEG4's audio compression within the structured audio format are discussed.

EURASIP Journal on Applied Signal Processing, Volume 2004, No. 8, 1 July 2004

Message from the Editor-in-Chief

EURASIP Meritorious Service Award

It is a pleasure to announce that the EURASIP Advisory Committee has decided to give the EURASIP 2004 Meritorious Service Award to Prof. K. J. Ray Liu (University of Maryland, College Park, USA) for his outstanding work as the Editor-in-Chief of the EURASIP Journal on Applied Signal Processing (2001-2002) and as the Editor-in-Chief of the new EURASIP Book Series on Signal Processing and Communications.

I sincerely congratulate Ray on this Award, and wish to thank him for his continued interest and efforts for EURASIP JASP as a Senior Advisory Editor.

The Award will be presented at the EUSIPCO 2004 Awards Ceremony, Vienna, September 2004.

Best Paper Award 2003

Last year, the EURASIP Advisory Committee decided to install 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 covered the period 2001-2002, while this second Award covers the year 2003, where over 130 papers were published.

The 2003 Best Paper Award Committee consisted of EURASIP JASP Associate Editors *Prof. Phillip Regalia* (Chair of the Committee, Institut National des Telecommunications, Evry, France), *Prof. Jaakko Astola* (Tampere University of Technology, Tampere, Finland), and *Prof. Hideaki Sakai* (Kyoto University, Kyoto, Japan).

From a shortlist of six papers nominated by the members of the Editorial Board and/or Guest Editors of Special Issues, the Award Committee decided to give the 2003 Best Paper Award to the paper entitled "A vision chip for color segmentation and pattern matching" by Ralph Etienne-Cummings, Philippe Pouliquen, and M. Anthony Lewis which appeared in EURASIP JASP Vol. 2003, No. 7 (June).

I sincerely congratulate the authors on this Award. I also would like to thank the EURASIP JASP Award Committee, for their outstanding selection work.

Both the 2001-2002 and the 2003 Awards will be presented at the EUSIPCO 2004 Awards Ceremony, Vienna, September 2004.

Editorial Board

On July 1st, ten new Associate Editors have agreed to join the EURASIP JASP Editorial Board: Kostas Berberidis (University of Patras, Patras, Greece); Joe Chen (Northrop Grumman Space Technology, Redondo Beach, USA); Frank Ehlers (Federal Armed Forces

Underwater Acoustics and Marine Geophysics Research Institute, Kiel, Germany); Walter Kellermann (University Erlangen-Nuremberg, Erlangen, Germany); Alex Kot (Nanyang Technological University, Singapore); Geert Leus (Delft University of Technology, Delft, The Netherlands); Vincent Poor (Princeton University, Princeton, USA); Yuan-Pei Lin (National Chiao Tung University, Hsinchu, Taiwan); Dimitrios Tzovaras (Informatics and Telematics Institiut, Thessaloniki, Greece); Jar-Ferr Yang (National Cheng Kung University, Tainan, Taiwan). I welcome them on Board and look forward to working with them.

I would like to express my heartfelt thanks to the Editorial Board Members currently leaving the Editorial Board: Kiyoharu Aizawa,Tony Constantinides, Sadaoki Furui, Mos Kaveh, Kyoung Mu Lee, Geoffrey Li, Ioannis Pitas, Wan-Chi Siu, Michael Strintzis, and Andy Wu.

Marc Moonen
Editor-in-Chief

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Contents and Abstracts

A Noise Reduction Preprocessor for Mobile Voice Communication

Rainer Martin, David Malah, Richard V. Cox, and Anthony J. Accardi

http://dx.doi.org/10.1155/S1110865704312138

We describe a speech enhancement algorithm which leads to significant quality and intelligibility improvements when used as a preprocessor to a low bit rate speech coder. This algorithm was developed in conjunction with the mixed excitation linear prediction (MELP) coder which, by itself, is highly susceptible to environmental noise. The paper presents novel as well as known speech and noise estimation techniques and combines them into a highly effective speech enhancement system. The algorithm is based on short-time spectral amplitude estimation, soft-decision gain modification, tracking of the a priori probability of speech absence, and minimum statistics noise power estimation. Special emphasis is placed on enhancing the performance of the preprocessor in nonstationary noise environments.

Estimation of Road Vehicle Speed Using Two Omnidirectional Microphones: A Maximum Likelihood Approach

Roberto López-Valcarce, Carlos Mosquera, and Fernando Pérez-González

http://dx.doi.org/10.1155/S1110865704311133

We address the problem of estimating the speed of a road vehicle from its acoustic signature, recorded by a pair of omnidirectional microphones located next to the road. This choice of sensors is motivated by their nonintrusive nature as well as low installation and maintenance costs. A novel estimation technique is proposed, which is based on the maximum likelihood principle. It directly estimates car speed without any assumptions on the acoustic signal emitted by the vehicle. This has the advantages of bypassing troublesome intermediate delay estimation steps as well as eliminating the need for an accurate yet general enough acoustic traffic model. An analysis of the estimate for narrowband and broadband sources is provided and verified with computer simulations. The estimation algorithm uses a bank of modified crosscorrelators and therefore it is well suited to DSP implementation, performing well with preliminary field data.

On the Determination of Optimal Model Order for GMM-Based Text-Independent Speaker Identification

M. F. Abu El-Yazeed, M. A. El Gamal, and M. M. H. El Ayadi

http://dx.doi.org/10.1155/S1110865704312205

Gaussian mixture models (GMMs) are recently employed to provide a robust technique for speaker identification. The determination of the appropriate number of Gaussian components in a model for adequate speaker representation is a crucial but difficult problem. This number is in fact speaker dependent. Therefore, assuming a fixed number of Gaussian

components for all speakers is not justified. In this paper, we develop a procedure for roughly estimating the maximum possible model order above which the estimation of model parameters becomes unreliable. In addition, a theoretical measure, namely, a goodness of fit (GOF) measure is derived and utilized in estimating the number of Gaussian components needed to characterize different speakers. The estimation is carried out by exploiting the distribution of the training data for each speaker. Experimental results indicate that the proposed technique provides results comparable to other well-known model selection criteria like the minimum description length (MDL) and the Akaike information criterion (AIC).

Fast Watermarking of MPEG-1/2 Streams Using Compressed-Domain Perceptual Embedding and a Generalized Correlator Detector

Dimitrios Simitopoulos, Sotirios A. Tsaftaris, Nikolaos V. Boulgouris, Alexia Briassouli, and Michael G. Strintzis

http://dx.doi.org/10.1155/S1110865704402029

A novel technique is proposed for watermarking of MPEG-1 and MPEG-2 compressed video streams. The proposed scheme is applied directly in the domain of MPEG-1 system streams and MPEG-2 program streams (multiplexed streams). Perceptual models are used during the embedding process in order to avoid degradation of the video quality. The watermark is detected without the use of the original video sequence. A modified correlation-based detector is introduced that applies nonlinear preprocessing before correlation. Experimental evaluation demonstrates that the proposed scheme is able to withstand several common attacks. The resulting watermarking system is very fast and therefore suitable for copyright protection of compressed video.

A Fast LSF Search Algorithm Based on Interframe Correlation in G.723.1

Sameer A. Kibey, Jaydeep P. Kulkarni, and Piyush D. Sarode

http://dx.doi.org/10.1155/S1110865704312011

We explain a time complexity reduction algorithm that improves the line spectral frequencies (LSF) search procedure on the unit circle for low bit rate speech codecs. The algorithm is based on strong interframe correlation exhibited by LSFs. The fixed point C code of ITU-T Recommendation G.723.1, which uses the "real root algorithm" was modified and the results were verified on ARM-7TDMI general purpose RISC processor. The algorithm works for all test vectors provided by International Telecommunications Union-Telecommunication (ITU-T) as well as real speech. The average time reduction in the search computation was found to be approximately 20%.

Estimating Intrinsic Camera Parameters from the Fundamental Matrix Using an Evolutionary Approach

Anthony Whitehead and Gerhard Roth

http://dx.doi.org/10.1155/S1110865704401024

Calibration is the process of computing the intrinsic (internal) camera parameters from a series of images. Normally calibration is done by placing predefined targets in the scene or by having special camera motions, such as rotations. If these two restrictions do not hold, then this calibration process is called autocalibration because it is done automatically, without user intervention. Using autocalibration, it is possible to create 3D reconstructions from a sequence of uncalibrated images without having to rely on a formal camera calibration process. The fundamental matrix describes the epipolar geometry between a pair of images, and it can be calculated directly from 2D image correspondences. We show that autocalibration from a set of fundamental matrices can simply be transformed into a global minimization problem utilizing a cost function. We use a stochastic optimization approach taken from the field of evolutionary computing to solve this problem. A number of experiments are performed on published and standardized data sets that show the effectiveness of the approach. The basic assumption of this method is that the internal (intrinsic) camera parameters remain constant throughout the image sequence, that is, the images are taken from the same camera without varying such quantities as the focal length. We show that for the autocalibration of the focal length and aspect ratio, the evolutionary method achieves results comparable to published methods but is simpler to implement and is efficient enough to handle larger image sequences.

Optimizing Statistical Character Recognition Using Evolutionary Strategies to Recognize Aircraft Tail Numbers

Antonio Berlanga, Juan A. Besada, Jesús García Herrero, José M. Molina, Javier I. Portillo, and José R. Casar

http://dx.doi.org/10.1155/S1110865704312084

The design of statistical classification systems for optical character recognition (OCR) is a cumbersome task. This paper proposes a method using evolutionary strategies (ES) to evolve and upgrade the set of parameters in an OCR system. This OCR is applied to identify the tail number of aircrafts moving on the airport. The proposed approach is discussed and some results are obtained using a benchmark data set. This research demonstrates the successful application of ES to a difficult, noisy, and real-world problem.

An Improved Way to Make Large-Scale SVR Learning Practical

Quan Yong, Yang Jie, Yao Lixiu, and Ye Chenzhou

http://dx.doi.org/10.1155/S1110865704312096

We first put forward a new algorithm of reduced support vector regression (RSVR) and adopt a new approach to make a similar mathematical form as that of support vector classification. Then we describe a fast training algorithm for simplified support vector regression,

sequential minimal optimization (SMO) which was used to train SVM before. Experiments prove that this new method converges considerably faster than other methods that require the presence of a substantial amount of the data in memory.

Correction of Misclassifications Using a Proximity-Based Estimation Method

Antti Niemistö, Ilya Shmulevich, Vladimir V. Lukin, Alexander N. Dolia, and Olli Yli-Harja

http://dx.doi.org/10.1155/S1110865704402145

An estimation method for correcting misclassifications in signal and image processing is presented. The method is based on the use of context-based (temporal or spatial) information in a sliding-window fashion. The classes can be purely nominal, that is, an ordering of the classes is not required. The method employs nonlinear operations based on class proximities defined by a proximity matrix. Two case studies are presented. In the first, the proposed method is applied to one-dimensional signals for processing data that are obtained by a musical key-finding algorithm. In the second, the estimation method is applied to two-dimensional signals for correction of misclassifications in images. In the first case study, the proximity matrix employed by the estimation method follows directly from music perception studies, whereas in the second case study, the optimal proximity matrix is obtained with genetic algorithms as the learning rule in a training-based optimization framework. Simulation results are presented in both case studies and the degree of improvement in classification accuracy that is obtained by the proposed method is assessed statistically using Kappa analysis.

Detecting Impulses in Mechanical Signals by Wavelets

W.-X. Yang and X.-M. Ren

http://dx.doi.org/10.1155/S1110865704311091

The presence of periodical or nonperiodical impulses in vibration signals often indicates the occurrence of machine faults. This knowledge is applied to the fault diagnosis of such machines as engines, gearboxes, rolling element bearings, and so on. The development of an effective impulse detection technique is necessary and significant for evaluating the working condition of these machines, diagnosing their malfunctions, and keeping them running normally over prolong periods. With the aid of wavelet transforms, a wavelet-based envelope analysis method is proposed. In order to suppress any undesired information and highlight the features of interest, an improved soft threshold method has been designed so that the inspected signal is analyzed in a more exact way. Furthermore, an impulse detection technique is developed based on the aforementioned methods. The effectiveness of the proposed technique on the extraction of impulsive features of mechanical signals has been proved by both simulated and practical experiments.

Filter-Bank-Based Narrowband Interference Detection and Suppression in Spread Spectrum Systems

Tobias Hidalgo Stitz and Markku Renfors

http://dx.doi.org/10.1155/S1110865704312102

A filter-bank-based narrowband interference detection and suppression method is developed and its performance is studied in a spread spectrum system. The use of an efficient, complex, critically decimated perfect reconstruction filter bank with a highly selective subband filter prototype, in combination with a newly developed excision algorithm, offers a solution with efficient implementation and performance close to the theoretical limit derived as a function of the filter bank stopband attenuation. Also methods to cope with the transient effects in case of frequency hopping interference are developed and the resulting performance shows only minor degradation in comparison to the stationary case.

A New Method for Estimating the Number of Harmonic Components in Noise with Application in High Resolution Radar

Emanuel Radoi and André Quinquis

http://dx.doi.org/10.1155/S1110865704401097

In order to operate properly, the superresolution methods based on orthogonal subspace decomposition, such as multiple signal classification (MUSIC) or estimation of signal parameters by rotational invariance techniques (ESPRIT), need accurate estimation of the signal subspace dimension, that is, of the number of harmonic components that are superimposed and corrupted by noise. This estimation is particularly difficult when the S/N ratio is low and the statistical properties of the noise are unknown. Moreover, in some applications such as radar imagery, it is very important to avoid underestimation of the number of harmonic components which are associated to the target scattering centers. In this paper, we propose an effective method for the estimation of the signal subspace dimension which is able to operate against colored noise with performances superior to those exhibited by the classical information theoretic criteria of Akaike and Rissanen. The capabilities of the new method are demonstrated through computer simulations and it is proved that compared to three other methods it carries out the best trade-off from four points of view, S/N ratio in white noise, frequency band of colored noise, dynamic range of the harmonic component amplitudes, and computing time.

EURASIP JASP FORTHCOMING SPECIAL ISSUES

Multicarrier Communications and Signal Processing

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

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.

Nonlinear Signal and Image Processing

Guest Editors: Gian Luca Foresti, Gianni Ramponi, Carlo S. Regazzoni, Giovanni Sicuranza, and Gianni Vernazza

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 is 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 an application point of view.

Machine Perception on a Chip

Guest Editors: Magdy A. Bayoumi and Bertrand Zavidovique

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.

Multimedia Security and Rights Management

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

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. Djurić, Simon J. Godsill, and Arnaud Doucet

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, and Thomas Kaiser

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

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.

Cross-Layer Design for Communications and Signal Processing Systems

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

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, can 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 of 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.

Turbo Processing

Guest Editors: Alex M. Haimovich, Ramesh Pyndiah, and Luc Vandendorpe

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."

UWB—State of the Art

Guest Editors: Gabriella di Benedetto, Thomas Kaiser, Norbert Schmidt, and Armin Wittneben

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 are. 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.

Anthropomorphic Processing of Audio and Speech

Guest Editors: Werner Verhelst, Jürgen Herre, Gernot Kubin, and Hynek Hermansky

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.

Advances in Intelligent Vision Systems: Methods and Applications

Guest Editors: Jacques Blanc-Talon, Wilfried Philips, and Dan Popescu

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.

Advances in Sensor Array Processing Technology

Guest Editors: Joe C. Chen and Amin G. Jaffer

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.

DSP-Enabled Radio

Guest Editors: Robert W. Stewart, Stephan Weiss, and Michael W. Hoffman

Digital signal processing has experienced a tremendous growth in order to bring applications such as 2G and 3G mobile communications and wireless LAN to mass markets. The advance in DAC and ADC technology to sampling rates of around 100 MHz at high bit resolution has allowed DSP to be employed for versatile transmission and receiver tasks, which are most pronounced in programmable software radios characterised by their reconfigurability for multiband and/or multimode operations in potentially mobile devices.

In the past five years, this research area has witnessed a substantial increase in activity, with a number of events and special issues dedicated to the fast-expanding topic of software-defined radios. However, as the foundations advance, with sampling rates reaching the GHz range with good multibit resolution over the next few years and smart antennas being incorporated into transceiver systems, the challenges for both the design and implementation of DSP algorithms in programmable radio systems advance as well.

Therefore, this special issue aims to present an overview of current research into DSP design, algorithms, and methods that may shape the development of future radio systems and wireless networks.

Trends in Brain-Computer Interfaces

Guest Editors: Jean-Marc Vesin and Touradj Ebrahimi

Brain-computer interfaces (BCI), an emerging domain in the field of man-machine interaction, have attracted increasing attention in the last few years. Among reasons, one may cite the expansion of neurosciences, the development of powerful information processing and machine learning techniques and, last but not least, the mere fascination exerted by a direct control of human intellect upon the material world.

The goal of this special issue is to present a broad overview of state-of-the-art approaches to brain-computer communication with emphasis on signal processing issues.

System-Integration-Oriented Transceiver Designs for Wireless Networks Beyond 3G

Guest Editors: Sofiène Affes, Jacob Benesty, David Gesbert, Laurence Mailaender, and Mamoru Sawahashi

Current research on wireless will provide enhanced transceiver technologies that will enable future upgrade of wireless networks beyond 3G. Yet the prospective innovative solutions that are most likely to make their shortest way to integration in a future real-world wireless system are those that take into account interaction with other subsystem components, any source of imperfection such as estimation and modeling errors, implementation feasibility and costs, software/hardware codesign issues, and so forth to the proof-of-concept.

This special issue is seeking original research contributions in the design of new transceiver solutions for wireless networks beyond 3G with a development and assessment approach oriented towards implementation and integration in a real-world wireless system, that is, the methodology ranging from (i) realistic link/system-level software simulation to (ii) off-line verification and validation over channel measurements, (iii) real-time prototyping and validation, and (iv) on-air demonstration and field trials.

Signal Analysis Tools for Optical Information Processing

Guest Editors: Christi K. Madsen, Daniela Dragoman, José Azaña

The application of traditional signal analysis tools (e.g., Fourier transforms) to a wide variety of optical problems (e.g., diffraction, spatial filtering, holography, dispersion, etc.) has led to a new and deeper understanding of these optical problems. Novel analysis and synthesis methods for different photonics devices (e.g., fiber gratings, ring resonators, etc.) have also been developed based on well-known signal processing tools. The use of different photonic technologies for processing spatial or temporal information in the optical domain is also a field of growing importance, with a strong potential for interesting applications in fields such diverse as optical telecommunications, image processing, and optical computing, to name only a few. Advantages of processing the information in the optical domain include the tremendous available bandwidth and the parallelism intrinsic to the optical approach, which translate into ultrahigh processing speeds, which otherwise are not possible.

This special issue is seeking original research contributions regarding (i) the application of signal analysis tools to optical problems and (ii) the proposal and demonstration of innovative technologies, devices, and architectures for all-optical information processing in the spatial or temporal domains.

DSP in Hearing Aids and Cochlear Implants

Søren Holdt Jensen, Simon Doclo, Philippe Pango, Søren Riis, and Jan Wouters

Digital signal processing for *hearing aids* was initiated as a topic of research in the mid-late 1980s. However, it was not until 1995 that technology was matured to a level where size and power consumption made a market introduction of hearing aids with full digital signal processing possible. Today more and more hearing aids are turning digital even in the low-price segments. Current technology enables hearing aids that fit completely in the ear canal, and the introduction of truly programmable platforms has allowed the development of advanced digital signal processing algorithms that provide a natural sound picture with increased speech intelligibility and comfort to the hearing-impaired user.

Such signal processing technology is now also being adopted in *cochlear implants*. A cochlear implant needs, in addition, a sound processing strategy that converts the acoustical signal into electrical signals to be applied to the electrodes placed in the cochlea. The design of such sound processing strategies poses additional signal processing challenges, but at the same time builds on knowledge acquired through physiological and psychophysical studies.

The goal of this special issue is to present research in signal processing methods and algorithms for hearing aids and cochlear implants.

Applications of Signal Processing in Astrophysics and Cosmology

Ercan E. Kuruoglu and Carlo Baccigalupi

Recent satellite missions such as WMAP (Wilkinson anisotropy probe) have provided scientists with vast amounts of data which need to be analysed to extract vital information about the universe. In particular, scientists are interested in separating various sources in the radiation maps such as the cosmic microwave background radiation which provides a picture of the early universe shortly after the big bang and information about the future evolution of the universe. Many astrophysics problems, as in this specific example, require dealing with prohibitive amounts of the data which are nonstationary, non-Gaussian, and are corrupted severely by noise and nonlinearities in the measurement process. These challenges, which cannot be met by classical data analysis methods, have required the utilisation of the state-of-the-art signal processing techniques and, in the lack of suitable methods, have fuelled research into the development of new ones such as in the case of nonlinear spectral estimation. Similarly, techniques such as wavelet transforms and advanced signal separation techniques have been translated into the astrophysics field and have demonstrated promising results. Data mining and classification techniques coupled with the advances in the computational power have enabled the processing of data of big dimensions almost in real time which allowed the focusing of the astrophysics and the cosmology community on previously untractable problems in anticipation of new measurements to arrive from the Planck satellite.

This new and active research field is producing a wealth of scientific papers and conference proceedings. On the other hand, up to now, most literature have been published in astrophysics and cosmology journals and therefore have not attracted much attention in the signal processing field delaying crucial input from signal processing experts. In this special issue, we would like to create a forum in which the signal processing community would be introduced to the real problems in the astrophysics field as well as drawing the attention of the astrophysics community to the availability of signal processing tools for the solution of the problems, hence aiming at a cross fertilisation of ideas.

Advances in Interferometric Synthetic Aperture Radar Processing

Gianfranco Fornaro, Fabrizio Lombardini, Roland Romeiser, and Shane Cloude

Interferometric methods have successfully expanded in the last two decades the remote-sensing capabilities of high-resolution synthetic aperture radar (SAR), providing efficient operational topographic mapping and displacement monitoring tools for land and ice applications, and promising techniques for surface ocean velocity sensing and forest parameters estimation. The need for advanced signal processing techniques within the interferometric SAR processing field is continuously increasing, for improving existing functionalities, producing novel parameter extraction capabilities, and fully exploiting the potentials originated by new complex experimented and planned interferometric SAR sensor systems.

This special issue is seeking original research contributions in the development and assessment of advanced models and new signal processing algorithms in the interferometric SAR field, with an approach oriented towards the exploitation of statistical methods and of (baseline, time, frequency, or polarization) acquisition diversity, to face the challenges of an accurate, reliable, and fully capable interferometric radar remote sensing and to deal with increasingly various and difficult scenarios.

In particular, papers are concerned with the fertilization and application of methods and concepts from areas such as filtering, parameter estimation, detection, spectral estimation, array processing, model inversion, data fusion, and phenomenological-or physical-based statistical modeling.

Special Issue on

Information Mining from Multimedia Databases

CALL FOR PAPERS

The main focus of this special issue is on information mining techniques for the extraction and interpretation of semantic contents in multimedia databases. Due to the spatiotemporal nature of most multimedia data streams, an important requirement for this information mining process is the accurate extraction and characterization of salient events from the original signal-based representation, and the discovery of possible relationships between these events in the form of high-level association rules. The availability of these high-level representations will play an important role in applications such as content-based multimedia information retrieval, surveillance, and automatic image/video annotation. For this problem, the main challenges are in the design and analysis of mapping techniques between the signal-level and semantic-level representations, and the adaptive characterization of the notion of saliency for multimedia events in view of its dependence on the preferences of individual users and specific contexts. In other words, the eventual objective is to bridge the gap between the low-level feature representation and the high-level interpretation of multimedia contents.

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

- Mapping techniques between low-level features and high-level representations for bridging the semantic gap.
- The application of machine learning techniques (e.g. symbol-based inductive learning, neural networks, and evolutionary computation) for multimedia information mining.
- Detection, characterization, and representation of salient events in multimedia data streams.
- Automatic discovery of high-level association rules in image and video mining.
- Automatic image/video annotation, classification, and indexing.
- Multimedia information mining based on the integration of multiple modalities such as text, image, video, and audio.
- Adaptive characterization of users' preference in the interpretation of semantic information.
- Multimedia data mining within the MPEG-4 and MPEG-7 frameworks.

- Applications of multimedia data mining in areas such as video scene analysis, content-based retrieval, multimedia content summarization, surveillance, scientific visualization, and medical imaging.

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Special Issue on

Frames and Overcomplete Representations in Signal Processing, Communications, and Information Theory

CALL FOR PAPERS

Many problems in signal processing, communications, and information theory deal with linear signal expansions. The corresponding basis functions usually constitute a nonredundant set. It is well known that the use of redundancy in engineering systems improves robustness and numerical stability. Motivated by this observation, the use of redundant linear signal expansions (a.k.a. "frames" or "overcomplete representations") has found widespread use in many different engineering disciplines. Recent examples include sampling theory, A/D conversion, oversampled filter banks, multiple description source coding, error correcting codes, wavelet- and frame-based denoising, quantum detection and estimation, and space-time coding for wireless communications.

This special issue aims to present survey papers on frame theory and its applications and to bring together original contributions from the different areas mentioned above, containing original applications of frame theory. Prospective papers should be unpublished and present novel contributions, either in terms of fundamental research or from an applications perspective, or should be of survey nature.

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

- Sampling theory, A/D conversion
- Source coding, in particular, multiple description source coding
- Oversampled filter bank theory and design
- Error correcting codes
- Wavelet- and frame-based denoising
- Quantum detection and estimation
- Space-time coding and modulation
- Sigma-delta modulation
- OFDM systems
- Linear time-varying system theory
- Sparse representations

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Special Issue on

Design Methods for DSP Systems

CALL FOR PAPERS

Industrial implementations of DSP systems today require extreme complexity. Examples are wireless systems satisfying standards like WLAN or 3GPP, hearing aids components or transceivers for home building automation. At the same time, often harsh constraints like low power requirements burden the designer even more. Conventional methods for ASIC design are not sufficient any more to guarantee a fast conversion from initial concept to final product. In industry, the problem has been addressed by the wording design crisis or design gap. While this design gap exists in a complexity gap, that is, a difference between existing, available, and demanded complexity, there is also a productivity gap, that is, the difference between available complexity and how much we are able to efficiently convert into gate level representations. This special issue intends to present recent solutions to such gaps addressing algorithmic design methods, algorithms for floating-to-fixed-point conversion, automatic DSP coding strategies, architectural exploration methods, hardware/software partitioning, virtual and rapid prototyping, as well as automatic testing and verification.

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

- Automatic DSP coding strategies
- Floating-to-fixed-point conversion algorithms
- Automatic HW/SW partitioning
- Architectural exploartion
- Virtual prototyping
- Rapid prototyping
- Automatic testing and verification
- Complex SoC design experience

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Special Issue on

Radar Space-Time Adaptive Processing

CALL FOR PAPERS

Space-time adaptive processing (STAP) is a technique originally developped for detecting slow-moving targets from airborne radars. Although the main principles of STAP have been known for many years, the field has experienced a regain of interest in the early 1990s as a result of the significant increase in computational power.

Much of the 1990s focused on monostatic STAP configurations (where the transmitter and receiver are collocated) and on computationally efficient partially adaptive and beamspace techniques. More recently, much of the attention has shifted to the much more challenging case of bistatic configurations (where the transmitter and receiver are located on distinct, independently moving platforms).

Another major challenge to STAP systems is operation in strong heterogeneous environments that preclude conventional covariance estimation techniques based on a wide-sense stationarity assumption. Knowledge-aided methods have recently emerged as a potential solution to this problem. In addition, we are currently seeing STAP techniques moving into new areas such as sonar and communications.

The goal of this special issue is to discuss the state of the art in radar STAP techniques (suboptimal, bistatic, etc.) and to explain why STAP techniques are also proving useful in domains that were probably not initially anticipated.

Papers should emphasize advanced signal processing techniques, applications to real data, systems issues, and new concepts and applications.

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

- Operational environments: airborne, space-based, UAV
- Bistatic and multistatic STAP
- STAP and SAR
- Tracking with STAP radars
- STAP architectures (e.g., suboptimum processors)
- Novel concepts, systems, techniques, and algorithms
- Special hardware for real-time STAP
- Nonlinear and/or nonuniform antenna arrays
- Estimation of radar configuration parameters
- Range-dependence compensation

- Handling of nonstationary, heterogeneous environments, and knowledge-aided STAP
- Polarimetric STAP
- Simulation of realistic STAP data
- Test of STAP techniques on real-life data
- New application areas: sonar, communications (e.g., MIMO), navigation, seismics, and so forth.

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Special Issue on

Super-Resolution Imaging: Analysis, Algorithms, and Applications

CALL FOR PAPERS

The recent increase in the wide use of digital imaging technologies in consumer (e.g., digital video) and other markets (e.g., security and military) has brought with it a simultaneous demand for higher-resolution images. The demand for such high-resolution (HR) images can be met by algorithmic advances in super-resolution (SR) technology in place ofor in tandem withhardware development. Such HR images not only give the viewer a more pleasing picture but also offer additional details that are important for subsequent analysis in many applications.

The current approach to obtaining HR images mainly relies on sensor manufacturing technology that attempts to increase the number of pixels per unit area by reducing the pixel size. However, the cost for high-precision optics and sensors may be prohibitive for general purpose commercial applications, and there is a limitation to pixel size reduction due to shot noise encountered in the sensor itself. Therefore, a resolution enhancement (super-resolution) approach using computational, mathematical, and statistical techniques has received a great deal of attention recently. The relevant signal processing technology for this SR approach to high-quality imaging is the topic of this special issue.

The scope of techniques intended to overcome the above limitations that will be covered in this special issue will include: enhancement in spatial resolution for both gray-scale and color images and video, suppression of signal dependent noise, and various other associated artifacts.

Because of the recent emergence of many key relevant computational, mathematical, and statistical techniques, and the increasing importance of digital imaging technology, a special issue of the EURASIP JASP dedicated to the topic of SR imaging is quite timely.

A more detailed list of SR imaging topics of interest include (but are not limited to):

- Multiframe/multichannel direct and blind deconvolution in SR
- Subpixel motion estimation
- SR in time and dynamic range, etc.
- Artifact analysis of sensors and optics
- Video-to-video SR imaging

- Multiframe demosaicing and SR imaging
- Wavelet-based methods for SR imaging
- PDE-based methods for SR imaging
- Locally adaptive image interpolation
- SR in medical, astronomical, security/surveillance, and other applications

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Special Issue on

Implementation Aspects and Testbeds for MIMO Systems

CALL FOR PAPERS

MIMO (multiple input multiple output) systems have emerged as a key technology for wireless local area networks (WLAN), wireless metropolitan area networks (WMAN), and cellular mobile communication systems (3G, 4G) because they promise greater coverage, higher data rates, and improved link robustness by adding a spatial dimension to the time, frequency, and code dimensions. Recent progress in standardization and first MIMO prototype chipsets force manufacturers worldwide to pay more attention to MIMO implementation aspects. Moreover, MIMO testbeds become more and more attractive to universities and research institutes as has been observed in the past few years. The aim of this special issue is to reflect the current state of the art of MIMO testbeds and to point out the numerous MIMO implementation challenges for current and future wireless communication standards.

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

- Antenna arrays (topology, coupling, size limitations, cross-polarization, multiband capability)
- Multichannel analog front ends (transceiver architectures, power consumption, low complexity, high-scale integration, enabling technologies)
- Multichannel digital front ends (RF impairment cancelation, AD conversion, synchronization)
- Baseband signal processing (implementation, suboptimum algorithms, soft/hardware partitioning)
- Cross-layer issues (scheduling, power allocation, quality of service, protocol design, channel aware optimization, power conscious optimization)
- System level aspects (capacity enhancement, coverage extension, diversity/multiplex trade-offs)
- Overall complexity versus performance evaluations

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Special Issue on

Advanced Signal Processing Techniques for Digital Subscriber Lines

CALL FOR PAPERS

The recent deployment of digital subscriber line (DSL) technology around the world is rapidly making broadband access for the mass consumer market a reality. DSL allows telephone operators to get maximum leverage out of their existing infrastructure by delivering broadband access over existing twisted-pair telephone lines. At the heart of DSL lies a plethora of advanced signal processing techniques which enable such high-speed transmission to be achieved over a medium originally designed with only voice-band transmission in mind. As DSL networks are deployed, customer demand for ever higher data rates is growing. This has been fueled by the increasing popularity of applications like peer-to-peer (P2P) file-sharing networks, video streaming, and HDTV.

Achieving such high data rates will require the development of new, advanced signal processing techniques to address many issues that still exist in DSL networks such as crosstalk, impulse noise, high peak-to-average power ratios (PAPR), intersymbol/intercarrier interference (ISI/ICI), and radio frequency interference (RFI). The goal of this special issue is to discuss the state of the art in signal processing techniques for DSL.

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

- Dynamic spectrum management
- Vectoring, bonding, and phantom-mode transmission
- Alien crosstalk cancelation
- Other multiuser techniques
- Turbo/LDPC codes for DSL
- Ethernet in the first mile (EFM)
- Advanced modulation techniques for DSL
- PAPR reduction
- Windowing and RFI cancelation
- Equalization and echo cancelation
- Impulse noise mitigation
- Synchronization
- Wavelets and filterbanks

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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 welcomed. 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

EURASIP Journal on on Wireless Communications and Networking, Volume 2004, No. 1, 1 August 2004

Special Issue on

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

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This editorial reflects the emerging system design and signal detection methods for next-generation digital cellular CDMA system. In CDMA system, in addition to intersymbol interference (ISI) caused by multipath propagation, simultaneous transmission also introduces multiuser interference (MUI). The receiver is, therefore, 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 referred to as short-code CDMA, and if the spreading sequences are aperiodic or essentially pseudo-random, it is known as long-code CDMA. Since multiuser detection relies on the cyclostationarity of the received signal, which is significantly complicated

by the time-varying modeling of the long-code system, research on blind multiuser detection has largely been limited to short-code CDMA. On the other hand, long-code is widely used in virtually all operational and commercially proposed CDMA systems due to its performance stability in frequency fading environment and better information security. More recently, researchers have been targeting on effective and efficient multiuser detectors for long-code CDMA systems as well. Furthermore, combination of CDMA and OFDM is attracting more and more attention in order to take the advantages of both schemes.

In this special issue, novel techniques on spreading sequences design, space diversity (multiple transmit and receive antennas), time diversity (channel coding, interleaving), and combination of CDMA and OFDM, in conjunction with new channel estimation and signal extraction approaches, are intensively investigated to achieve good system performance while improving system capacity for broadband multimedia wireless communications.

This special issue contains the following five topics.

Spreading sequence design

Spreading sequence design is essential in synchronization, channel estimation, effective MUI suppression, and communication security. On this topic, (i) Cotae addresses the problem for an overloaded synchronous DS-CDMA system in a multicell environment. A promising algorithm has been derived to design orthogonal generalized WBE sequence sets for any processing gain. (ii) Ren proposes an efficient and flexible approach to construct pseudo-random sequences with long period, large complexity, balance statistics, and low correlation properties from addition of M-sequences with pairwise-prime linear spans. (iii) Fan gives a nice survey on the recent trends and results on generalized orthogonal and quasiorthogonal sequences design and theoretical limits.

Space-time signal processing

As a relatively new member in space-time signal processing, transmit antenna diversity is gaining increasing popularity in communication system design and signal extraction. On this topic, (i) Dai, Mailaender, and Poor study the algorithm choice in CDMA cellular downlink transmission with transmit antenna arrays over multipath fading channels. They conclude that, in general, maximum SNR beamforming is the best choice for circuit-switched systems, whereas for packet-switched systems, maximum SINR beamforming is the best choice. (ii) W. Li and Gulliver introduce a novel successive interference cancellation (SIC) technique for DS-CDMA systems employing space-time block codes (STBC) at the transmit side. Both hard- and soft-decision-based cancellation schemes are analyzed and simulated.

Multicarrier CDMA

Due to its strong capability in combating frequency-selective fading and tracking the time-varying channels, multicarrier CDMA (MC-CDMA), which is the combination of CDMA and OFDM, is a promising candidate for broadband communication systems. On this topic, (i) Rahman, Sesay, and Hefnawi consider a two-stage ML-based detector for a multitone CDMA system. In the first stage, the channel is estimated using a given symbol, while in the second stage, the estimated channel is used to detect the next symbol. The theoretical model

is validated with simulation results for Rayleigh fading environments. (ii) Preequalization techniques are derived by Silva and Gameiro for downlink TDD MC-CDMA system using space-frequency algorithms. The approaches effectively reduce multiple access interference at the base station, enabling low-cost terminal designs without sacrificing the system performance. (iii) Raulefs, Dammann, Sand, and Kaiser present an innovative rotated Walsh-Hadamard-based spreading scheme for MCCDMA applications. The rotated spread gain, stemmed from signal-space diversity, increases the system performance by almost 1 dB in a fading environment.

Channel estimation and signal detection

Accurate channel estimation is the guarantee for effective signal detection. On this topic, novel blind channel estimation and multiuser detection approaches are investigated for long-code CDMA and MC-CDMA systems. More specifically, (i) Sirbu and Koivunen address the problem of propagation delay estimation in asynchronous long-code DS-CDMA multiuser systems. By modeling the users' propagation delays in the MIMO channel matrix, delay estimates are obtained as a by-product of the channel estimation. (ii) P. Liu and Xu carry out a joint performance study of channel estimation and multiuser detection for long-code uplink CDMA systems using perturbation theory. Simulation and analytical results show good agreement. (iii) Dang and van der Veen present a joint multiuser source-channel estimation approach for long-code CDMA, which is the combination of the blind (decorrelating) RAKE receiver with an iterative symbol/channel estimation algorithm. The algorithm shows a significant improvement over the decorrelating RAKE receiver and the conventional RAKE receiver. (iv) Gelli, Paura, and Verde propose a novel two-stage blind multiuser detector for quasisynchronous MC-CDMA systems. The receive filter is factored into the production of two parts: $\mathbf{f} = \mathcal{F}\mathbf{u}$, and each part is optimized accordingly. $\mathbf{u}$ is calculated based on the constant modulus criterion, and $\mathcal{F}$ serves as the constraint so that the system will extract the desired user.

System design and signal processing

On this topic, the researchers explore innovative transmitter and receiver design for CDMA systems. (i) Hou, Yi, and Lee propose an intriguing multilevel coding scheme based on LDPC to facilitate multimedia applications in future-generation wireless networks. By offering one low-rate channel and two high-rate channels, the new method allows simultaneous transmission of voice and greater than 1 Mbps high-speed data with minimum error and latency. (ii) Madhukumar, Chin, Liang, and Yang propose a single-carrier cyclic prefix-assisted CDMA system with frequency domain equalization. The proposed system has the advantages of conventional MC-CDMA system, but does not suffer from the high peak-to-average ratio and sensitivity to frequency offset and phase noise. (iii) Vanhaverbeke and Moeneclaey consider to improve the performance of overloaded CDMA systems. The main idea is to introduce time shifts between users so that the overall MUI power is minimized. Simulation results demonstrate the effectiveness of the proposed approach. (iv) Park, Lim, and Gelfand present a performance study showing that with a low-complexity MMSE multiuser detector, superior performance can be obtained through coding across multicodes and time.

We thank all the authors who submitted their articles to the special issue. Special thanks go to all the reviewers for their hard work, on which the quality of the special issue relies. Last but not least, thanks to the Editor-in-Chief and the Editorial Board of EURASIP JWCN for their support and thanks to Hindawi Publishing Corporation for making the publication of this special issue possible.

Jitendra K. Tugnait
Hui Liu
Guang Gong
Tongtong Li

Volume 2004, No. 1, 1 August 2004

Contents and Abstracts

Spreading Sequence Design for Multiple-Cell Synchronous DS-CDMA Systems under Total Weighted Squared Correlation Criterion

Paul Cotae

http://dx.doi.org/10.1155/S1687147204405027

An algorithm for designing spreading sequences for an overloaded multicellular synchronous DS-CDMA system on uplink is introduced. The criterion used to measure the optimality of the design is the total weighted square correlation (TWSC) assuming the channel state information known perfectly at both transmitter and receiver. By using this algorithm it is possible to obtain orthogonal generalized WBE sequences sets for any processing gain. The bandwidth of initial generalized WBE signals of each cell is preserved in the extended signal space associated to multicellular system. Mathematical formalism is illustrated by selected numerical examples.

Design of Long Period Pseudo-Random Sequences from the Addition of m-Sequences over $\mathbb{F}_p$

Jian Ren

http://dx.doi.org/10.1155/S1687147204405052

Pseudo-random sequence with good correlation property and large linear span is widely used in code division multiple access (CDMA) communication systems and cryptology for reliable and secure information transmission. In this paper, sequences with long period, large complexity, balance statistics, and low cross-correlation property are constructed from the addition of m-sequences with pairwise-prime linear spans (AMPLS). Using m-sequences as building blocks, the proposed method proved to be an efficient and flexible approach to construct long period pseudo-random sequences with desirable properties from short period sequences. Applying the proposed method to $\mathbb{F}_2$, a signal set $((2^n-1)(2^m-1), (2^n+1)(2^m+1), (2^{(n+1)/2}+1)(2^{(m+1)/2}+1))$ is constructed.

Spreading Sequence Design and Theoretical Limits for Quasisynchronous CDMA Systems

Pingzhi Fan

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For various quasisynchronous (QS) CDMA systems such as LAS-CDMA system which emerged recently, in order to reduce or eliminate the multiple access interference and multipath interference, it is required to design a set of spreading sequences which are mutually orthogonal within a designed shift zone, called orthogonal zone. For traditional orthogonal

sequences, such as Walsh sequences and orthogonal Gold sequences, the orthogonality can only be achieved at the inphase point; in other words, the orthogonality is destroyed whenever there is a relative shift between the sequences, that is, their orthogonal zone is 0. In this paper, new concepts of generalized orthogonality (GO) and generalized quasiorthogonality (GQO) for spreading sequence design in both direct sequence (DS) QS-CDMA systems and time/frequency hopping (TH/FH) QS-CDMA systems are presented. Besides, selected GO/GQO sequence designs and general theoretical periodic and aperiodic limits, together with several applications in QS-CDMA systems, are also reviewed and analyzed.

CDMA Downlink Transmission with Transmit Antenna Arrays and Power Control in Multipath Fading Channels

Huaiyu Dai, Laurence Mailaender, and H. Vincent Poor

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Wireless code-division multiple-access (CDMA) cellular downlink communications with transmit antenna arrays in multipath fading channels is studied. Various array signal processing techniques at the transmit end are investigated and compared under various settings, in conjunction with power control. No instant downlink channel information is assumed; however, the obtained results are also compared with results assuming ideal feedback. The study is carried out for both circuit-switched and packet-switched systems, where different goals are pursued and different conclusions are drawn. In particular, it is found that the traffic type impacts the algorithm choice in downlink transmission, and that there is no need to seek optimum power control/allocation schemes, which are either too complex or infeasible in practice. Another interesting conclusion is that, even though feedback does not help much for packet-switched systems, it does help for circuit-switched systems, the gain of which increases with the number of antennas.

Successive Interference Cancellation for DS-CDMA Systems with Transmit Diversity

Wei Li and T. Aaron Gulliver

http://dx.doi.org/10.1155/S1687147204404022

We introduce a new successive interference cancellation (SIC) technique for direct sequence code division multiple access (DS-CDMA) systems with transmit diversity. The transmit diversity is achieved with a space-time block code (STBC). In our work we first consider hard decision SIC with an STBC, and then investigate the performance of soft decision SIC with an STBC. System performance over a Rayleigh fading channel is investigated and the analysis is confirmed by simulation.

Two-Stage Maximum Likelihood Estimation (TSMLE) for MT-CDMA Signals in the Indoor Environment

Quazi Mehbubar Rahman, Abu B. Sesay, and Mostafa Hefnawi

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This paper proposes a two-stage maximum likelihood estimation (TSMLE) technique suited for multitone code division multiple access (MT-CDMA) system. Here, an analytical framework is presented in the indoor environment for determining the average bit error rate (BER) of the system, over Rayleigh and Ricean fading channels. The analytical model is derived for quadrature phase shift keying (QPSK) modulation technique by taking into account the number of tones, signal bandwidth (BW), bit rate, and transmission power. Numerical results are presented to validate the analysis, and to justify the approximations made therein. Moreover, these results are shown to agree completely with those obtained by simulation.

Downlink Space-Frequency Preequalization Techniques for TDD MC-CDMA Mobile Radio Systems

Adão Silva and Atílio Gameiro

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The paper considers downlink space-frequency preequalizations techniques for time division duplex (TDD) MC-CDMA. We consider the use of antenna arrays at the base station (BS) and analytically derive different preequalization schemes for two different receiver configurations at the mobile terminal: a simple despread receiver without channel equalization and an equal-gain combiner (EGC) conventional receiver. We show that the space-frequency preequalization approach proposed allows to format the transmitted signals so that the multiple access interference at mobile terminals is reduced allowing to transfer the most computational complexity from mobile terminal to the BS. Simulation results are carried out to demonstrate the effectiveness of the proposed preequalization schemes.

Rotated Walsh-Hadamard Spreading with Robust Channel Estimation for a Coded MC-CDMA System

Ronald Raulefs, Armin Dammann, Stephan Sand, Stefan Kaiser, and Gunther Auer

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We investigate rotated Walsh-Hadamard spreading matrices for a broadband MC-CDMA system with robust channel estimation in the synchronous downlink. The similarities between rotated spreading and signal space diversity are outlined. In a multiuser MC-CDMA system, possible performance improvements are based on the chosen detector, the channel code, and its Hamming distance. By applying rotated spreading in comparison to a standard Walsh-Hadamard spreading code, a higher throughput can be achieved. As combining the channel code and the spreading code forms a concatenated code, the overall minimum Hamming distance of the concatenated code increases. This asymptotically results in an improvement of the bit error rate for high signal-to-noise ratio. Higher convolutional channel code rates are mostly generated by puncturing good low-rate channel codes. The overall

Hamming distance decreases significantly for the punctured channel codes. Higher channel code rates are favorable for MC-CDMA, as MC-CDMA utilizes diversity more efficiently compared to pure OFDMA. The application of rotated spreading in an MC-CDMA system allows exploiting diversity even further. We demonstrate that the rotated spreading gain is still present for a robust pilot-aided channel estimator. In a well-designed system, rotated spreading extends the performance by using a maximum likelihood detector with robust channel estimation at the receiver by about 1 dB.

Delay Estimation in Long-Code Asynchronous DS-CDMA Systems Using Multiple Antennas

Marius Sirbu and Visa Koivunen

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The problem of propagation delay estimation in asynchronous long-code DS-CDMA multiuser systems is addressed. Almost all the methods proposed so far in the literature for propagation delay estimation are derived for short codes and the knowledge of the codes is exploited by the estimators. In long-code CDMA, the spreading code is aperiodic and the methods developed for short codes may not be used or may increase the complexity significantly. For example, in the subspace-based estimators, the aperiodic nature of the code may require subspace tracking. In this paper we propose a novel method for simultaneous estimation of the propagation delays of several active users. A specific multiple-input multiple-output (MIMO) system model is constructed in a multiuser scenario. In such model the channel matrix contains information about both the users propagation delays and channel impulse responses. Consequently, estimates of the delays are obtained as a by-product of the channel estimation task. The channel matrix has a special structure that is exploited in estimating the delays. The proposed delay estimation method lends itself to an adaptive implementation. Thus, it may be applied to joint channel and delay estimation in uplink DS-CDMA analogously to the method presented by the authors in 2003. The performance of the proposed method is studied in simulation using realistic time-varying channel model and different SNR levels in the face of near-far effects, and using low spreading factor (high data rates).

Joint Performance Study of Channel Estimation and Multiuser Detection for Uplink Long-Code CDMA Systems

Ping Liu and Zhengyuan Xu

http://dx.doi.org/10.1155/S1687147204404058

Although numerous channel estimation and multiuser detection approaches have appeared for long-code uplink CDMA systems, joint performance study of channel estimators and symbol detectors remains largely open. In this paper, we construct three typical symbol-level linear receivers upon existing channel estimation method, known as zero-forcing (ZF), minimum mean-square-error (MMSE), and RAKE receivers, for symbol detection. Since the channel estimation error is rippled to the linear receivers, performance of all receivers is thus jointly analyzed with the channel estimator from a perturbation perspective. Extensive simulation examples involving different communication environments demonstrate high consistency between our analysis and experimental results.

A Low-Complexity Blind Multiuser Receiver for Long-Code CDMA

Quang Hieu Dang and Alle-Jan van der Veen

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Receivers for long-code systems are for computational reasons usually based on simple matched-filter techniques, and hence suffer from multiaccess interference. Decorrelating RAKE and MMSE receivers do not have this problem but have not been widely studied due to the apparent complexity of the inversion of a large code matrix. Tong, van der Veen Dewilde, and Sung (IEEE Tr. Signal Proc., 2003) derived a blind decorrelating RAKE receiver (DRR) and channel estimation algorithm for long-code CDMA systems, and showed how it can be efficiently implemented. In this paper, we continue on that work. We propose both single-user and multiuser blind source-channel estimation algorithms by making use of an iterative estimation scheme initialized by the DRR. Simulation results show significant improvement, even in heavily loaded systems. Moreover, with an implementation based on time-varying system theory, the proposed algorithm can be implemented efficiently at a cost similar to the RAKE.

Blind Direct Multiuser Detection for Uplink MC-CDMA: Performance Analysis and Robust Implementation

Giacinto Gelli, Luigi Paura, and Francesco Verde

http://dx.doi.org/10.1155/S1687147204405039

We consider the problem of blind (i.e., without training sequences) linear mitigation of multiple-access interference in the uplink of quasi-synchronous multicarrier code-division multiple-access (MC-CDMA) systems. In the first part of the paper, we present the analytical performance assessment of the recently proposed blind two-stage multiuser detector, whose synthesis requires only the knowledge of the spreading code of the desired user. The analysis allows one to evaluate the actual performance when the receiver's parameters are estimated by resorting to a finite data record. Based on this analysis, in the second part of the paper, we propose to improve the performance of the two-stage detector by adding a quadratic constraint in the first stage synthesis, which exploits the knowledge of the spreading codes of the active users within the cell of interest. It is shown analytically that incorporation of such a quadratic constraint improves the receiver robustness against errors in the estimated statistics of the received data, although it slightly reduces the interference suppression capabilities of the two-stage detector. The effectiveness of the proposed receiver is further corroborated by computer simulation results.

Multilevel LDPC Codes Design for Multimedia Communication CDMA System

Jia Hou, Yu Yi, and Moon Ho Lee

http://dx.doi.org/10.1155/S168714720440406X

We design multilevel coding (MLC) with a semi-bit interleaved coded modulation (BICM) scheme based on low density parity check (LDPC) codes. Different from the traditional de-

signs, we joined the MLC and BICM together by using the Gray mapping, which is suitable to transmit the data over several equivalent channels with different code rates. To perform well at signal-to-noise ratio (SNR) to be very close to the capacity of the additive white Gaussian noise (AWGN) channel, random regular LDPC code and a simple semialgebra LDPC (SA-LDPC) code are discussed in MLC with parallel independent decoding (PID). The numerical results demonstrate that the proposed scheme could achieve both power and bandwidth efficiency.

Single-Carrier Cyclic Prefix-Assisted CDMA System with Frequency Domain Equalization for High Data Rate Transmission

A. S. Madhukumar, Francois Chin, Ying-Chang Liang, and Kai Yang

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Multiple-access interference and interfinger interference limit the capacity of conventional single-carrier DS-CDMA systems. Even though multicarrier CDMA posses the advantages of conventional CDMA and OFDM, it suffers from two major implementation difficulties such as peak-to-average power ratio and high sensitivity to frequency offset and RF phase noise. A novel approach based on single-carrier cyclic prefix-assisted CDMA has been proposed to overcome the disadvantages of single-carrier CDMA and multicarrier modulation. The usefulness of the proposed approach for high-speed packet access with simplified channel estimation procedures are investigated in this paper. The paper also proposes a data-dependent pilot structure for the downlink transmission of the proposed system for enhancing pilot-assisted channel estimation in frequency domain. The performance of the proposed pilot structure is compared against the data-independent common pilot structure. The proposed system is extensively simulated for different channel parameters with different channel estimation and equalization methods and the results are compared against conventional multicarrier CDMA systems with identical system specifications.

Overloaded CDMA Systems with Displaced Binary Signatures

Frederik Vanhaverbeke and Marc Moeneclaey

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We extend three types of overloaded CDMA systems, by displacing in time the binary signature sequences of these systems: (1) random spreading (PN), (2) multiple-OCDMA (MO), and (3) PN/OCDMA (PN/O). For each of these systems, we determine the time shifts that minimize the overall multiuser interference power. The achievable channel load with coded and uncoded data is evaluated for the conventional (without displacement) and improved (with displacement) systems, as well as for systems based on quasi-Welch-bound-equality (QWBE) sequences, by means of several types of turbo detectors. For each system, the best performing turbo detector is selected in order to compare the performance of these systems. It is found that the improved systems substantially outperform their original counterparts. With uncoded data, (improved) PN/O yields the highest acceptable channel load. For coded data, MO allows for the highest acceptable channel load over all considered systems, both for the conventional and the improved systems. In the latter case, channel loads of about 280% are achievable with a low degradation as compared to a single user system.

Coding Across Multicodes and Time in CDMA Systems Employing MMSE Multiuser Detector

Jeongsoon Park, Jong-Han Lim, and Saul B. Gelfand

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When combining a multicode CDMA system with convolutional coding, two methods have been considered in the literature. In one method, coding is across time in each multicode channel while in the other the coding is across both multicodes and time. In this paper, a performance/complexity analysis of decoding metrics and trellis structures for the two schemes is carried out. It is shown that the latter scheme can exploit the multicode diversity inherent in convolutionally coded direct sequence code division multiple access (DS-CDMA) systems which employ minimum mean squared error (MMSE) multiuser detectors. In particular, when the MMSE detector provides sufficiently different signal-to-interference ratios (SIRs) for the multicode channels, coding across multicodes and time can obtain significant performance gain over coding across time, with nearly the same decoding complexity.

Performance Evaluation of Public Key-Based Authentication in Future Mobile Communication Systems

Georgios Kambourakis, Angelos Rouskas, and Stefanos Gritzalis

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While mobile hosts are evolving into full-IP enabled devices, there is a greater demand to provide a more flexible, reconfigurable, and scalable security mechanism in mobile communication systems beyond 3G (B3G). Work has already begun on such an "all-IP" end-to-end solution, commonly referred to as 4G systems. Fully fledged integration between heterogeneous networks, such as 2.5G, UMTS, WLAN, Bluetooth, and the Internet, demands fully compatible, time-tested, and reliable mechanisms to depend on. SSL protocol has proved its effectiveness in the wired Internet and it will probably be the most promising candidate for future wireless environments. In this paper, we discuss existing problems related to authentication and key agreement (AKA) procedures, such as compromised authentication vectors attacks, as they appear in current 2/2.5G/3G mobile communication systems, and propose how SSL, combined with public key infrastructure (PKI) elements, can be used to overcome these vulnerabilities. In this B3G environment, we perceive authentication as a service, which has to be performed at the higher protocol layers irrespective of the underlying network technology. Furthermore, we analyze the effectiveness of such a solution, based on measurements of a "prototype" implementation. Performance measurements indicate that SSL-based authentication can be possible in terms of service time in future wireless systems, while it can simultaneously provide both the necessary flexibility to network operators and a high level of confidence to end users.

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

Guest Editors: Jitendra K. Tugnait, Hui Liu, Guang Gong, and Tongtong Li

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.

Multiuser MIMO Networks

Guest Editors: A. Lee Swindlehurst, Brian Sadler, Robert Fischer

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 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.

Optical Wireless Communications

Guest Editor: A. Boucouvalas

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, 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.

Advanced Signal Processing Algorithms for Wireless Communications

Guest Editors: Erdal Panayırcı, Costas Georghiades, Xiadong Wang, and Hakan A. Çırpan

Traditional wireless technologies are confronted with new challenges in meeting the ubiquity and mobility requirements of cellular systems. Hostile channel characteristics and limited bandwidths in wireless applications provide key barriers that future generation systems must cope with. Advanced signal processing methods, such as the expectation-maximization algorithm and related techniques including SAGE, MCEM, HMM, Baum-Welch, and sequential Monte Carlo methods, in collaboration with inexpensive and rapid computing power, provide a promising avenue for overcoming the limitations of current technologies. Although such methods have been successfully applied in a variety of communication contexts, many technical challenges remain in emerging applications, whose solutions will provide the bridge between the theoretical potential of such techniques and their practical utility.

Ad Hoc Networks: Cross-Layer Issues

Guest Editors: Sergio Palazzo, Leandros Tassiulas, and Lang Tong

Ad Hoc Networks, due to their intended support of "no-limit" infrastructure-less communication, pose many significant new challenges with respect to traditional wireless networks. The main particularities of ad hoc networks, which typically require new solutions for distributed signal processing and control, can be summarized as follows: the autonomous and spontaneous nature of nodes which leads to dynamic unpredictable topology; node

mobility which may cause link failures and network partitions; battery limitations which imply constraints on transmission power and network connectivity; and the need for cooperative and/or opportunistic behaviour in spite of the natural energy conservative selfish attitude of nodes.

Traditional layered protocol architectures are not well suited to deal with these multifaceted issues, because they do not exploit the potential improvement in performance that can be obtained through cross-layer design. Typical examples of transversal objectives which deserve joint interaction of algorithms and techniques that span multiple layers are energy efficiency, quality-of-service support, reliability, and network scalability. This special issue sollicits research papers which shed new light on the potential benefits gained by applying a cross-layer design perspective to ad hoc networks. We seek original and unpublished contributions addressing novel architectures, algorithms, and/or protocols, where evidence of the performance gain obtained is shown by either theoretical analysis, simulation, or experimental results.

Special Issue on

Reconfigurable Radio for Future Generation Wireless Systems

CALL FOR PAPERS

Future generation wireless systems aim to support a wide variety of services over a wide variety of networks in a way transparent to the user. To deliver the optimal quality-of-service (QoS) for many different applications over many different communication environments, flexibility and adaptivity are key ingredients of these future generation wireless systems. Rather than relying on the traditional horizontal communication model consisting of a single wireless access system, these future 4G systems will employ a vertical communication model, which integrates different existing and evolving wireless access systems on a common IP-based platform, to complement each other for different service requirements and radio environments. To enable seamless and transparent interworking between these different wireless access systems, or communication modes, through horizontal (intrasystem) and vertical (intersystem) handovers, multimode terminals that support different existing and newly emerging air interfaces are needed.

As a deep penetration of the multimode terminal is aimed at in the telecommunication market, new challenges appear in terms of minimizing the terminal cost, size, and power consumption while, at the same time, maximizing its flexibility with respect to communication standards as well as its adaptivity with respect to varying user requirements and changing communication conditions. The conventional approach to the design of a multimode terminal is the provision of a custom baseband processor for every communication mode. However, with the growing number of standards and communication modes, this approach is becoming increasingly infeasible and economically unacceptable. A more efficient approach towards this design is to adopt a reconfigurable (as opposed to fixed) radio concept, such that the terminal can adapt to the best-suited communication mode under the control of a QoS manager. A high degree of flexibility is not only required for the digital baseband processing but also for the analog radio frequency (RF) front-end, which should accept a large range of carrier frequencies, possess a flexible bandwidth, and deal with a wide variety of operational conditions. Likewise, the same high degree of flexibility is not only called for at the physical layer but also at the medium access control (MAC) (and possibly higher) layer(s), to be compatible with the protocols of different standards.

This special issue aims to cover the present research on reconfigurable radio for future generation wireless systems. Prospective papers should present original and innovative contributions to the wireless communications community. Fundamental research results as well as practical implementations and demonstrators are solicited for.

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

- Flexible digital baseband signal processing
- Adaptive modulation and coding
- Flexible multiple access schemes
- Flexible MIMO signalling
- Channel quality information prediction
- Adaptive transmission schemes
- Reconfigurable transmission techniques
- Reconfigurable receiver algorithms
- Reconfigurable detection and equalization
- Reconfigurable channel estimation
- Reconfigurable synchronization
- Software-defined radio
- Flexible analog RF front-ends
- Multifrequency RF front-ends
- Multiband RF front-ends
- Flexible MAC protocols
- QoS management
- Reconfiguration algorithms
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- Link adaptation
- Dynamic resource allocation
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Special Issue on Wireless Sensor Networks

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Recent advances in integrated circuit and digital wireless communication technologies have enabled the design of wireless sensor networks to facilitate the joint processing of spatially and temporally distributed information. Such networks greatly enhance our ability to understand and evaluate complex systems and environments. Using wireless connectivity for sensor networks offers increased flexibility in the deployment and reconfiguration of the network and reduces the infrastructure cost. These advantages will enable sensor networks to monitor complex environments for applications ranging from battlefield surveillance to environment monitoring and telemedicine control.

Enormous challenges in the understanding of sensor networks presently impede deployment of many of the envisaged applications. In particular, for wireless sensor networks that employ in situ unattended sensors, physical constraints integrating power, bandwidth, and cost have presented significant challenges as well as research opportunities in the field. One of the major concerns is maintaining connectivity: the geographical disperseness of the sensor nodes and the ad hoc network structure, coupled with the above-mentioned resource constraints, make this a unique challenge. Maintaining efficient network operation is further exacerbated by the volume of data generated by the sensors, which is disproportionally large compared with the network capacity. This special issue is intended to provide a venue for the dissemination of high-quality research addressing these challenges for wireless sensor networks. We solicit original contributions that have direct connection to or impact on the communications and networking design of wireless sensor networks.

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

- Integrated sensing, processing, communications, and networking
- Channel-aware and distributed sensing and processing
- Joint-source channel coding for wireless sensor networks
- Fundamental performance limits and information-theoretic study of wireless sensor networks
- Fading and diversity in wireless sensor networks
- Distributed data and sensor management

- Network protocols for wireless sensor networks
- Cross-layer design techniques
- Energy management
- Distributed self-organization
- Support for quality of service (QoS) in wireless sensor networks

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Edited by: Edward R. Dougherty, Ilya Shmulevich, Jie Chen, and Z. Jane Wang

ISBN: 977-5945-07-0

Recent advances in genomic studies have stimulated synergetic research and development in many cross-disciplinary areas. Genomic data, especially the recent large-scale microarray gene expression data, represents enormous challenges for signal processing and statistics in processing these vast data to reveal the complex biological functionality. This perspective naturally leads to a new field, genomic signal processing (GSP), which studies the processing of genomic signals by integrating the theory of signal processing and statistics. Written by an international, interdisciplinary team of authors, this invaluable edited volume is accessible to students just entering this emergent field, and to researchers, both in academia and industry, in the fields of molecular biology, engineering, statistics, and signal processing. The book provides tutorial-level overviews and addresses the specific needs of genomic signal processing students and researchers as a reference book.

The book aims to address current genomic challenges by exploiting potential synergies between genomics, signal processing, and statistics, with special emphasis on signal processing and statistical tools for structural and functional understanding of genomic data. The book is partitioned into three parts. In part I, a brief history of genomic research and a background introduction from both biological and signal-processing/statistical perspectives are provided so that readers can easily follow the material presented in the rest of the book. In part II, overviews of state-of-the-art techniques are provided. We start with a chapter on sequence analysis, and follow with chapters on feature selection, clustering, and classification of microarray data. The next three chapters discuss the modeling, analysis, and simulation of biological regulatory networks, especially gene regulatory networks based on Boolean and Bayesian approaches. The next two chapters treat visualization and compression of gene data, and supercomputer implementation of genomic signal processing systems. Part II concludes with two chapters on systems biology and medical implications of genomic research. Finally, part III discusses the future trends in genomic signal processing and statistics research.

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