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Contents

EURASIP JASP ABSTRACTS

President's Message: Elections 2004

As you may know, the elected members of the EURASIP Administration Committee are partially renewed every two years. This year, four AdCom members will be elected for a four-year period. As President of the AdCom, I am pleased to present six candidates for the Elections 2004 in this Newsletter. Nominations are coming from the AdCom and EURASIP members, identifying outstanding members, some already serving as elected members or officers with different tasks. As always, biographies and photos assist you in your selection. The ballot allows every member to give four votes. The ballots can be mailed to me (the address in the Newsletter inside cover) before 20 August, 2004 or delivered at Vienna at the General Assembly Meeting along EUSIPCO 2004 scheduled to take place on Wednesday, 8 September, 2004, 6.30 p.m.

In addition to the renewal of the elected members, the AdCom is constantly analyzing the EURASIP management needs. As a result of this analysis, the structure of the AdCom is reorganized; that is, current member tasks are redefined and new members join the AdCom. Recently, two new officers have been nominated to take care of specific tasks in the organization: Paulo Correia (since October 2003) as *Internal Coordinator Officer* and Beatrice Pesquet-Popescu (since March 2004) as *WWW assistant*. I would like to publicly welcome Paulo and Beatrice to the AdCom and thank them for their commitment to EURASIP.

I look forward to welcoming you all at EUSIPCO 2004 in Vienna during the second week in September.

Frarran Marqués
President

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 Correira	☐	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.

EURASIP Awards: Awards Description and Policy

Introduction

The goal of the EURASIP Awards program is to pay tribute to those persons making outstanding technical contributions and achievements or to those institutions enabling the development of remarkable research and dissemination, leading to the advancement of the signal processing domain.

EURASIP Awards fall into several categories:

- Best Paper Awards
- Technical Achievements Award
- Meritorious Service Award
- European Group Technical Achievement Award

The nominations for awards can be initiated by EURASIP members and by the public. These nominations are then carefully reviewed by a panel of experts of the particular field to which the award relates. Their recommendations are submitted to the EURASIP Awards officer prior to their final approval by the EURASIP AdCom.

EURASIP Awards program promotes excellence in the different areas related to the signal processing community: education, research, industry, and services.

The EURASIP Awards attribution ceremony takes place during the annual EUSIPCO conference. The list of awards is published in the EURASIP Newsletter that is under distribution at that moment.

Best Paper Awards

Best paper awards are attributed to the more relevant contributions published in the journals sponsored by the EURASIP, that is:

- Signal Processing (1 award annually)
- Image Communication (1 award every 2 years)
- Speech Communication (1 award every 2 years)
- Journal of Applied Signal Processing (1 award annually)

These awards are attributed to each of the authors of the selected papers by the EURASIP AdCom at the annual EUSIPCO conference.

These awards consist in a certificate, one for each author, and a monetary compensation of 500.00€.

The papers considered for the awards are preselected by the journal Editors according to the score of the reviews or the evaluations of members of the Editorial Boards including the Guest Editors of the special issues. Then a comparative evaluation is performed by specific subcommittees, one for each of the EURASIP journals, according to the following criteria.

- *Relevance of the topic.* The degree of contribution of the paper to the advancement of the state of the art is evaluated. Under this topic the evaluation committee will assess the value of the paper in terms of the significance of the problem addressed in the paper for the signal processing community.
- *Quality of the technical content.* The degree of excellence of the work described is evaluated. Under this topic the evaluation committee will assess the techniques and skills used to achieve the reported results.
- *Style of the presentation.* The degree to which the paper message is clearly and effectively expressed is evaluated. Under this topic the evaluation committee will assess the quality of the writing of the paper, namely, completeness, conciseness, clarity, efficacy or balancing between the various sections of the paper.
- *Originality.* The degree to which the paper is original is evaluated. Under this topic the evaluation committee will assess the novelty of the technical material presented, namely, in terms of addressing new problems, presenting new results, or using new methodologies.

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

Technical Achievements Awards

The technical achievements awards are attributed annually to individuals responsible for major contributions in the field of signal, speech, and image processing.

One technical achievements award is attributed by the EURASIP AdCom at the annual EUSIPCO conference.

The nominations are submitted to the EURASIP Awards Chairman by members of the EURASIP AdCom and/or Advisory Committee, or by the general public by the end of December of the preceding year. The final decision about the award will be taken by the AdCom in March/April of the year of the award attribution.

The candidates should

- be actively involved with the international signal processing community, having a recognised leadership in their field of action and a significant impact on the European research activities;
- have a significant list of achievements, reflected by their *curriculum vitae*, namely, in terms of relevant publications, original contributions, or inventive value (patents);
- publish regularly in EURASIP journals;
- participate regularly in conferences sponsored by the EURASIP and, in particular, in EUSIPCO conferences.

The nominations should be well supported, namely, in terms of

- providing the complete name of the candidate, education degrees, business titles, contact information, professional curriculum, and EURASIP involvement;
- documenting the major technical achievements of the candidate, namely, by referencing books, papers, reports, standards contributions, patents, development of products, systems, facilities or services, technical presentations and courses developed;
- justification why the candidate is worthy to receive this award, namely, in terms of innovation, originality, creativity, and relevance of contributions;

- including support endorsement letters.

Nomination and endorsements should be as specific, accurate, and complete as possible.

The technical achievements awards consist of a gold medal presented to the selected individuals.

EURASIP Meritorious Service Award

The EURASIP meritorious service award is attributed annually to individuals for distinguished services to the Society.

The nominations are submitted to the EURASIP Awards Chairman by members of the EURASIP AdCom and/or Advisory Committee by the end of December of the preceding year. The final decision about the award will be taken by the AdCom in March/April of the year of the award attribution.

The candidates should

- have provided significant contributions to the development of EURASIP;
- have furnished distinguished service to the advancement and pursuit of the technical objectives of EURASIP;
- cooperate actively in actions supported or sponsored by EURASIP.

This award, in the form of a gold medal, is attributed by the EURASIP AdCom at the annual EUSIPCO conference.

The nominations should be well supported, namely, in terms of

- providing, complete name of the candidate, education degrees, business titles, contact information, professional curriculum, and EURASIP involvement;
- justification why the candidate is worthy to receive this award, namely, in terms of the major contributions to the development of the EURASIP.

Nomination should be as specific, accurate, and complete as possible.

European Group Technical Achievement Award

The European group technical achievement award is attributed annually to individuals who are leaders of European research groups with relevant performances in the field of signal, speech, and image processing.

The nominations can be proposed by the general public. The candidate research groups should

- be actively involved with the international signal processing community, having a recognised leadership in their field of action and a significant impact on the European research activities;
- have a significant list of achievements, namely, in terms of relevant publications, original contributions, or inventive value (patents);
- publish regularly in EURASIP journals;
- participate regularly in conferences sponsored by the EURASIP and, in particular, in EUSIPCO conferences.

These awards, in the form of a plaque presented to the leader of the selected research group, are attributed by the EURASIP AdCom at the annual EUSIPCO conference.

The nominations should be well supported, namely, in terms of

- providing the complete name of the candidate who represents the research group, his/her education degrees, business titles, contact information, professional curriculum, and EURASIP involvement;
- documenting the major technical achievements of the research group, namely, by referencing books, papers, reports, standards contributions, patents, development of products, systems, facilities or services, and technical presentations and courses developed;
- justification why the candidate research group is worthy to receive this award, namely, in terms of innovation, originality, creativity, and relevance of contributions;
- including support endorsement letters.

Nomination and endorsements should be as specific, accurate, and complete as possible.

Calendar of Events

Year	Date	Event	Location	EURASIP Involvement	Chairperson/Information
2004	June 23–25	17th Int. EURASIP Conf. BIOSIGNAL	Brno, Czech Republic	Cosponsorship	Jiri Jan http://www.feec.vutbr.cz/UBMI/bs2004.html
	July 20–22	4th Int. Symposium on Communication Systems, Networks and DSP (CSNDSP)	Newcastle, UK	Cooperation	T. Boukouvalas http://www.shu.ac.uk/ocr/csndsp/
	September 7–10	12th European Signal Processing Conference (EUSIPCO 2004)	Vienna, Austria	Sponsor	W. Mecklenbräuker http://www.nt.tuwien.ac.at/eusipco2004/
	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 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

Sergios Theodoridis
Workshops/Confs Coordinator EURASIP

Report on the Fifth International Workshop on Image Analysis for Multimedia Interactive Services (WIAMIS '2004)

The Fifth International Workshop on Image Analysis for Multimedia Interactive Services has been organized by Instituto de Telecomunicações/Instituto Superior Técnico on April 21–23, 2004, in Lisbon, Portugal. The workshop was chaired by Prof. Fernando Pereira and Prof. Paulo Correia from Instituto de Telecomunicações/Instituto Superior Técnico.

Around 175 papers from 35 countries have been submitted. All papers were peer-reviewed by a panel of experts selected by the WIAMIS '2004 technical committee, which accepted 135 regular papers and 8 additional papers for a special session. There were also 12 panel position statements and 3 invited papers. The conference counted 180 participants.

One key feature of WIAMIS is the single track organization of the technical sessions. This has once again proved to be a fruitful approach, allowing more in-depth discussions among all the participants, since they spend more time together. This was also confirmed by the high level of attendance during the three days of the workshop.

The technical program was organized to start each day with an invited speaker, followed by oral and poster sessions until lunch time. The afternoon started with an oral session, followed by a panel, and concluded with a poster session.

The first invited speaker, Prof. Tsuhan Chen from Carnegie Mellon University, talked about the advances on facial recognition with a presentation entitled "Face Recognition by the Human and by the Computer: Two sides of the same Coin, or Not?." In the afternoon, the first panel, chaired by Prof. Joern Ostermann from the Technical University of Hannover, addressed also facial analysis, focusing on tools and applications. The other members of the panel were Prof. Tsuhan Chen, Prof. Robert Forchheimer from the University of Linköping, and Prof. Luis Torres from the Technical University of Catalonia. The other sessions in the first day were about facial analysis and recognition, facial analysis tools, error resilience and rate control, watermarking, data hiding and protection and analysis for surveillance.

The second invited speaker, Dr. Ton Kalker from Philips Research Eindhoven, talked about the analysis of content protection with a presentation entitled "The Sense and Nonsense of Watermarking." In the afternoon, the second panel, chaired by Prof. Moncef Gabbouj from the Tampere University of Technology, analyzed the current status of segmentation and indexing tools and applications. The other members of the panel were Dr. Henri Sanson from France Telecom R&D, Prof. Thomas Sikora from the Technical University of Berlin, and Prof. Murat Tekalp from the Kok University. The other sessions in the second day were about segmentation, semantic-based multimedia analysis (a special session organized by the European project SCHEMA), indexing and retrieval, quality evaluation, detection and tracking and extraction, structuring and classification.

The third invited speaker, Prof. Jens-Rainer Ohm from Aachen University of Technology, talked about the recent advances on video coding with a presentation entitled "Wavelet

Video Coding." In the afternoon, the third panel, chaired by Prof. Touradj Ebrahimi from EPFL, discussed the trends and challenges for image and video analysis. The other members of the panel were Prof. Ed Delp from Purdue University, Prof. Aggelos Katsaggelos from Northwestern University, and Prof. Riccardo Leonardi from the University of Brescia. The other sessions in the third day were about content adaptation, scalability, transconding and transmoding, image and video coding, object detection and tracking, personalization and applications.

The WIAMIS Technical Program Committee met on the 22nd of April at lunch time. It was agreed to keep the annual periodicity of WIAMIS, with WIAMIS '2005 being organized in Montreux, Switzerland, chaired by Prof. Touradj Ebrahimi. The possibility to organize WIAMIS outside Europe was left open for 2006, with Chicago, USA, or Korea being possible future locations.

The best Workshop papers will be selected for publication in a WIAMIS '2004 special issue of the IEE Proceedings on Vision, Image and Signal Processing. The best papers selection will be done by the WIAMIS '2004 session chairs and the conference chairs. After WIAMIS '2004, the authors of these papers will be invited to submit an extended version of their paper which will be reviewed using the normal IEE reviewing process. Also, authors of papers addressing relevant issues are invited to submit an extended version of their paper to a special issue of IEEE Transactions on Circuits and Systems for Video Technology (CSVT) on the topic of "Analysis and Understanding for Video Adaptation" (see call). These papers will be reviewed using the normal IEEE CSVT reviewing process.

Fernando Pereira and Paulo Lobato Correia
WIAMIS '2004 Organising Committee

EURASIP Newsletter Guidelines for Book Reviews

Introduction

The EURASIP Newsletter is distributed to all EURASIP members, being a privileged means to communicate important topics for the signal processing community. In particular, EURASIP Newsletters include a book review section devoted to the analysis of a book publication covering topics of significant relevance and broad interest for this community. Book reviews are under the responsibility of the EURASIP Newsletter Editor.

This document includes relevant information for those individuals willing to submit a book review, as well as for book authors or publishers willing to provide sample book copies for eventual book review.

Book Review Structure

A book review for EURASIP Newsletter consists of a detailed comment of the text under analysis. The book review must be written in English, about 500 words long, and must contain at least the following elements.

- *Book identification.* Include all relevant information about the book, namely, title, author(s), series (if applicable), publisher, location of publisher, publication year, ISBN number, number of pages, cover type (paperback or hardcover), and price (if known).
- *Review title.* The reviewer should suggest a title for the review, which may eventually be changed by the Newsletter editors.
- *Overview of the book.* The reviewer describes the general scope of the book, along with a short description of the content for each chapter. A first discussion on the merits of the book can be included.
- *Appropriate audience for the book.* The reviewer explains to whom the book may be directed, grouping the users in general categories such as students (at which level introductory, undergraduate, graduate), professionals, academic, and researchers. What background should the reader have, what readers may expect to learn from the reading.
- *Review body.* The reviewer provides a description of the book subject with a technical depth ranging from a general and disseminative approach to a deep technical insight. The goal here is to give the reader an idea of the book, assessing the book's strengths and weaknesses.
- *Review summary.* The reviewer may conclude with a recommendation on the book merits to help the reader in deciding whether to buy the book or not.
- *Reviewer Signature.* It is placed at the end of the review, containing name, affiliation, current position, and up to 50 words biographic outline. The reviewer can include his/her webpage and e-mail address.

Book reviews should be returned to the Newsletter Editor by e-mail, by the deadline indicated in the invitation letter, in the one of the following encoding and file formats:

- ASCII Text (also in the e-mail body)
- Microsoft Word
- RTF
- PDF

Information for Reviewers

Reviews can be done by invitation from the Newsletter Editor, or they can be proposed by the general public. In the later case, the person interested in submitting a book review should contact the Newsletter Editor, indicating the following.

- The complete book title, author(s), publisher, publication year, and ISBN number.
- *The reviewer identification.* The complete contact information of the reviewer should be provided: name, affiliation, address (to which the review copy should be mailed), phone, fax, and email address.
- *The reviewer background.* A short summary of the reviewer background in the book's subject matter should be provided, together with the reviewer's motivation to write the review.

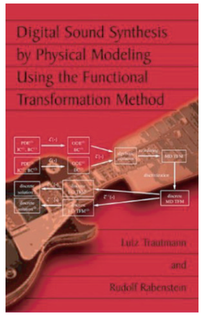
The proposed reviewer will receive an answer to the submitted request, indicating the adequacy of the proposal and the availability of the requested book title. All reviews are subject to approval and editing before publication.

Information for Authors/Publishers

Authors or publishers willing to have a review of their book published in EURASIP Newsletter are encouraged to contact the Newsletter editor, providing all relevant information about the book, namely, title, author(s), series (if applicable), publisher, location of publisher, publication year, ISBN number, number of pages, cover type (paperback or hardcover), and price (if known).

BOOK REVIEWS

Digital Sound Synthesis by Physical Modeling Using the Functional Transformation Method



Lutz Trautmann and Rudolf Rabenstein
Kluwer Academic/Plenum Publishers,
250 pp, hardcover, ISBN 0-306-47875-7

Physical modeling sound synthesis has been an active research topic for the last twenty years and is used in commercial applications for more than ten years. The authors of "Digital Sound Synthesis by Physical Modeling Using the Functional Transformation Method" have published more than 20 papers on the topic and this book provides a complete overview of their results.

Chapter 2 gives an overview of traditional sound-based synthesis techniques, including processing of recorded sounds using wavetable and granular synthesis, and spectral methods such as additive, substractive, and frequency modulation synthesis. Several examples are given, together with their range of applications and key references to the relevant literature.

Chapter 3 discusses the derivation of a physical model, starting from the observation of a musical instrument. The physical meaning of the partial differential equations (PDEs) governing the model is clarified by numerous examples and figures. The authors explain the subdivision of the physical model into its various parts before detailing three example structures: the vibrating string, the drum membrane, and the resonant body. While the descriptions of those structures are intentionally kept general, they are illustrated with examples from the slap-bass guitar and the bowed strings to the kettle drum and the piano hammer.

Chapter 4 reviews classical sound synthesis methods based on physical modeling: PDEs, digital waveguides, and modal synthesis. The finite difference method, which rely on the discretization of both temporal and spectral variable, is applied to solve those models. Many examples are given, along with equations and typical value for their physical parameters. Applications and computational load for each method are discussed. Modal synthesis theory is described with a special attention, as it will serve as the foundation of the next chapters.

Chapter 5, the key articulation of the book, is an in-depth introduction to the functional transformation method (FTM), a new physical modeling method developed by the authors. Section 1 states the fundamental principles of the FTM and presents its application to the various types of PDEs describing physical structures: scalar PDEs, vector PDEs, PDEs with nonlinear excitation functions, and PDEs with solution-dependant coefficients. The use of the Sturm-Liouville transformation, the underlying principle of the FTM, is clearly explained, together its analogy with the Laplace transformation. In section 2 the FTM is applied to vibrating strings. Two examples are treated: a piano string and a slapped bass guitar string. Section 3 treats the applications to vibrating membranes with a similar depth, using a rectangular reverberation plate and a circular drum head as examples. Section 4 discusses the applications of the FTM to the resonant bodies. Each of the examples is illustrated with typical values for their physical quantities, and the computational cost of their implementations is also considered.

Chapter 6 compares the functional transformation method to the different techniques presented in Chapter 4. Four main aspects are considered: their ability to solve problems at initial boundary values, the different structures implementation they can describe, their accuracy in simulations, and their computational complexity.

The authors conclude that a wider range of different structures can be represented using their functional transformation method, and that simulation accuracy is improved. The higher computational complexity is likely to become less and less a problem in the near future; this can already be balanced by combining the method with other synthesis approaches, such as digital waveguides.

In his preface, Vesa Välimäki describes the functional transformation method as a "truly novel" and promising approach. The book gives a comprehensive introduction to sound synthesis by physical modeling. The various methods are accompanied by many figures and examples to aid the explanations, and extensive references are provided to original research papers. A complete list of symbols used in the text will be useful for the reader to have as a reference. This title will provide an excellent introduction to physical models and the authors' own functional transformation method, suitable for graduate students and researchers interested in physical modeling of musical instruments and modern sound synthesis techniques.

Mark Sandler
Department of Electronic Engineering
Queen Mary University of London

The H.264/MPEG-4 AVC Video Coding Standard

Cristina Gomila

Thomson Inc., Corporate Research 2 Independence Way, Princeton, NJ 08540, USA
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1. Introduction

This tutorial is intended to provide a brief overview of the H.264/MPEG-4 AVC video coding standard, focusing on its new features and applications. Outperforming all previous standards over a wide range of bitrates, H.264 is expected to flood the market in a large number of applications ranging from real-time conversational services to TV broadcasting or Internet streaming. At the basis of its high performance, we can find a layered structure improving the network friendliness; the adoption of clean and simple solutions enabling efficient implementations; the capability of allowing flexible delays for a variety of services; in addition to a powerful set of tools maximizing the coding efficiency. A fidelity range extension (FRExt) is currently under development for the encoding of different bit depths and chroma formats.

2. Historical remark

For the last two decades, different video coding standards have been developed to ensure a seamless flow of data along the entire chain that covers the production, distribution, and reception of video content. Video coding standards are developed to provide the minimum set of tools that reaches the previous requirement at the lowest cost. Worldwide, two organizations dominate the video coding standardization processes, namely, the ITU-T Video Coding Experts Group (VCEG) and the ISO/IEC Moving Picture Experts Group (MPEG). VCEG has traditionally focussed on low bitrate video coding applications, where there is a need for high compression rates and error resilience tools. MPEG groups a larger community targeting higher bitrates for entertainment-quality broadcasting applications.

Both organizations have produced very successful standards in their respective domains. Figure 1 presents them in chronological order. H.261, developed by the ITU-T, introduced the hybrid video coding design currently in use. A few years later, MPEG-1 improved the performance of H.261 at higher bitrates. In a joint effort, the MPEG-2/H.262 [1] standard provided support for the first time to interlaced material and became the most commonly used video coding standard. Following MPEG-2, in 1995, H.263 [2] seemed to approach the best performance achievable with the classical hybrid coding scheme. For that reason, the ISO/IEC broadened its objectives from the pure compression gains. Along these lines, the MPEG-4 standard [3] focused on the development of new coding functionalities, MPEG-7 extended to the development of new description tools, while MPEG-21 aimed at defining a

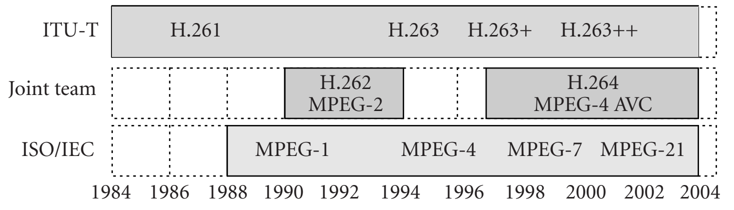

FIGURE 1: Historical evolution of the video coding standards.

TABLE 1: Slice coding types.

Slice coding types	Allowed MB prediction types
Intra (I)	Intra
Predictive (P)	Intra, Predictive
Bipredictive (B)	Intra, Predictive, Bipredictive
S-Intra (SI)	S-Intra, Intra
S-Predictive (SP)	Intra, S-Predictive

new multimedia framework. The ITU-T group continued the extension of the H.263 standard, standardizing two new versions, H.263+ in 1997 and H.263++ in 2000, focusing on the compression efficiency.

In parallel with these efforts, in 1997, the ITU-T started working on a new standard, known as H.26L, in response to the demand for an efficient compression solution. From the very beginning, H.26L was intended to outperform all previous standards without forcing strong changes on the classical coding scheme. Because of its successful development, in late 2001, the ITU-T and ISO/IEC decided to join their efforts in the so-called Joint Video Team (JVT). As a result of this collaboration, a single technical design was approved in mid 2003. For the ITU-T, it constituted a new and separate recommendation known as H.264, a name we will use further on in this document. For the ISO/IEC, the new set of video coding tools was integrated as a new part (Part 10) of the MPEG-4 standard, referred as MPEG-4 AVC (advanced video coding). After the closure of the standard, the JVT group has started working in a FRExt allowing the treatment of larger resolutions, bit depths, and chroma formats. The FRExt is expected to be approved as amendment of the H.264 standard by July 2004.

3. Overview of the standard goals

As stated in its specifications [4], the H.264 standard has been designed to be generic and serve a wide range of applications, bitrates, resolutions, qualities, and services. Within the JVT terms of reference (ToR) [5], the following high-level requirements were imposed.

(i) *Simplification back-to-basics approach* with the adoption of simple and clean solutions using well-known building blocks, targeting a simplified design that avoids any excessive quantity of optional features or profile configurations.

(ii) *High compression performance* having the capability of 50% or greater bitrate savings compared to previous standards at all bitrates, for a similar degree of encoder

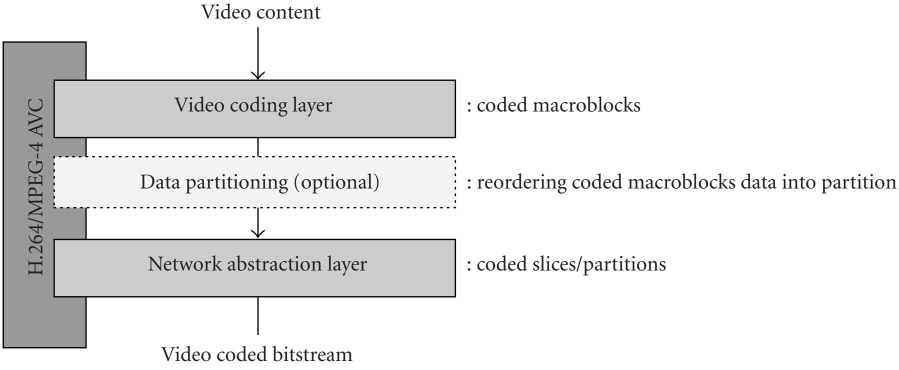

Figure 2: Layered structure of the H.264/MPEG-4 AVC encoder.

optimization (see Section 8 for results on final verification testing). Complexity scalability in encoder and decoder should ensure asymmetry of encoder and decoder processing complexity with scalability between amount of encoder processing and achievable quality.

(iii) *Improved network friendliness* to ease packetization, information priority control, and application to video streaming services. Flexible application to delay constraints appropriate to a variety of services and low delay for real-time conversational services and higher delay to optimize compression for storage or server-based applications.

(iv) *Enhanced error and packet loss resilience* tools for real-time applications over error-prone channels. Furthermore, there should be a full specification of decoding to resolve the mismatch problem typically due to the use of noninteger transforms.

To achieve these requirements, the H.264 has been designed in a layered structure, as illustrated in Figure 2. The video coding layer (VCL), presented in Section 4 efficiently represents the video content. Its goal is to improve the coding efficiency and promote simple solutions. Furthermore, to enable efficient implementations, a reduced number of profiles and levels have been defined, specifying subsets of the algorithmic features and degrees of capability. We present both of these concepts in Section 5. The network abstraction layer (NAL), presented in Section 6, provides the appropriate syntax to packet the VCL for its conveyance by the transport layers or storage media. Its goal is to improve the network friendliness and flexibility. We devote Section 7 to map the different profiles with the most common applications targeted by the standard. We conclude this overview by summarizing in Section 8 the conclusions of the analysis of performance done by the JVT. Interested readers can find a more detailed presentation of the standard in [6].

4. Video coding layer

The Video Coding Layer (VCL) is specified to efficiently represent the content of the video data. The strength of H.264 relies on the achievement of high-performance video coding on a classical hybrid coding configuration, as illustrated in Figure 3. Further on, we focus on presenting the new features in the VCL. We first briefly discuss the picture and slice

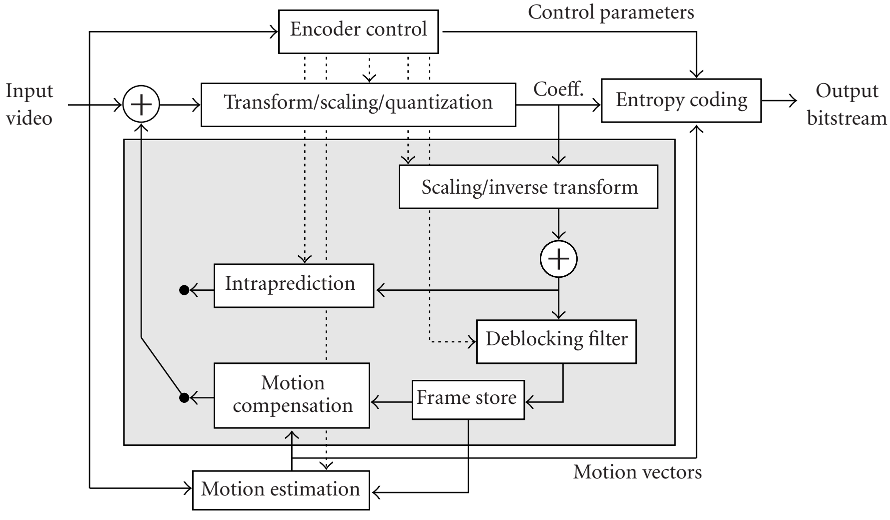

FIGURE 3: H.264 encoder scheme (nonnormative).

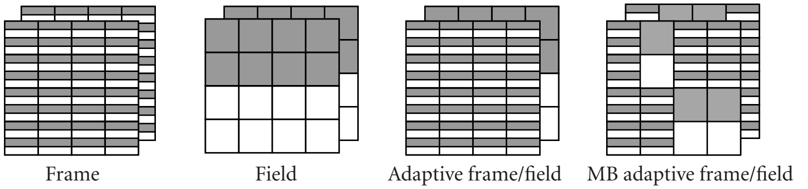

FIGURE 4: Picture coding modes within H.264.

coding types in H.264, then we present the new coding features affecting the blocks and macroblocks within the picture.

The H.264 standard has been defined to encode both progressive and interlaced sequences in YUV 4:2:0 format. The VCL supports four picture coding modes, as illustrated in Figure 4. In *frame-based* coding, a picture is created by interleaving both top and bottom lines. In *field-based* coding, a picture is created by adding in a sequential order first the top and then the bottom fields. At the picture level, *adaptive frame/field* coding allows the encoder to select one from of the previous two coding modes. The last coding mode is *MB adaptive frame/field* coding, in which the frame is scanned by MB pairs. As a result, a different coding type (frame/field) can be selected for each MB pair. This mode is particularly suited for interlaced content with a mixture of static and moving regions, encoded frame and field, respectively. Note that although the picture format is signaled, it has no impact on the set of available coding tools.

At a lower level, *slices* are the smallest self-contained video coding units. Slices are composed of an integer number of continuous MBs or MB pairs, accessed in raster scan order.

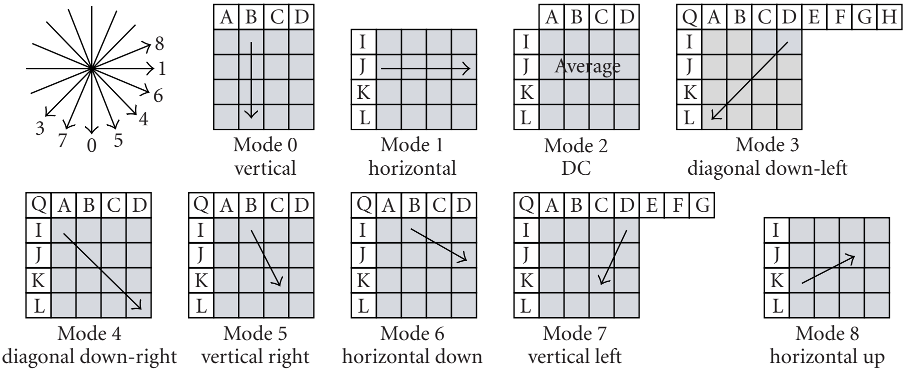

Figure 5: Intra prediction coding in H.264.

Within a slice, MBs are coded with interdependence. In H.264, five slice coding types are defined (see Table 1):

(1) In intra slices (I), MBs are predicted only from decoded samples within the same slice.
(2) In predictive slices (P), inter prediction is performed from previously decoded pictures with at most one motion vector per block.
(3) In bipredictive slices (B), inter prediction is conducted from previously decoded pictures with at most two motion vectors per block.
(4) The intra- and the predictive switching slices (SI, SP) [7] have the characteristic of allowing identical reconstruction of slices even when different reference frames are being used.

4.1. *Intra prediction coding*

In H.264, if an MB is encoded in intramode, a prediction is derived based on previously encoded and reconstructed surrounding available samples. The prediction is then subtracted from the current block prior to encoding. Intra prediction of a luma MB can be formed by means of (1) a single prediction for an entire 16×16 MB, with 4 modes available (vertical, horizontal, DC, plane); (2) sixteen individual predictions on 4×4 blocks, with 9 modes available (DC and 8 directional) as illustrated in Figure 5. Chroma intra prediction supports a single prediction type for each 8×8 region. In this case, only 4 modes (vertical, horizontal, DC, plane) are available.

4.2. *Inter prediction coding*

Similar to other major standards, inter prediction coding in H.264 relies on a block-based motion compensation. New features include the use of variable block sizes and motion vectors with 1/4 pixel precision, the extension of the bidirectional prediction with a larger number of intercoding modes, and the support of multiple reference frames.

MB and sub-MB partitions. Intercoded 16×16 MBs can be broken into smaller MB partitions of sizes 16×8, 8×16, or 8×8, as illustrated on the left side of Figure 6. Additionally, the 8×8 partitions, also known as sub-MBs, can be further broken into sub-MB partitions

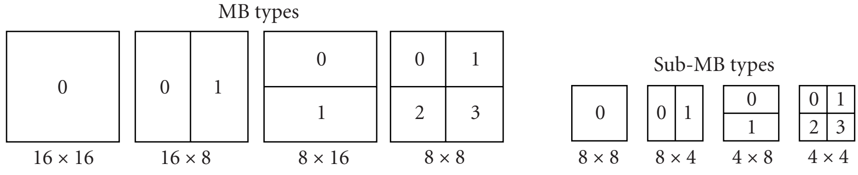

FIGURE 6: Tree-structured hierarchical MB and sub-MB partitions.

of sizes 8×4, 4×8, and 4×4. The choice of the partition size has a significant impact on compression efficiency. In general, a large partition size is appropriate for homogeneous areas, while small partition sizes are beneficial for encoding details and contours.

Subpixel motion vectors. Each partition in an intercoded MB is predicted from an area of the same size in a reference picture, indicated by a motion vector. H.264 supports 1/4 pixel precision for motion vectors, which outperforms both the integer and the half pixel compensation, at the expense of increased complexity. In luma, the interpolated samples at half positions are generated first by means of a 6-tap FIR filter with coefficients $1/32(1, -5, 20, 20, -5, 1)$. Then, pixels at quarter positions are obtained by averaging samples at integer and half positions. Corresponding 1/8 chroma samples are obtained by linear interpolation.

Inter prediction modes. H.264 supports four different inter prediction modes: predictive, bipredictive, skip, and direct mode. In *predictive mode* (P), blocks are estimated from previously decoded reference pictures, using at most one motion vector and reference picture per block. In *bipredictive mode* (B), blocks can be predicted as the weighted average of two distinct reference frames [8]. H.264 introduced the term bipredictive instead of using the term bidirectional, as in the previous standards, to highlight two important differences: (1) no restrictions apply to the temporal order of the references, both may come from the past or the future; (2) B pictures can be used as reference pictures. In *skip mode*, no motion information is encoded. Motion vectors are derived as the median of the motion vectors of neighboring MBs. The prediction is done using the most recent picture in the reference picture list. The last mode, referred as *direct mode* applies only to bipredicted partitions. As for the skip mode, both motion vectors and reference picture indices are derived at the decoder. The direct mode can be *spatial* or *temporal* depending on whether the information used for prediction is derived from the same picture or from colocated partitions.

Multiple reference pictures. The MBs in the current picture can be predicted from multiple reference pictures [9]. Multiple reference pictures are most helpful for sequences with chaotic motion. This new feature mainly increases the computational burden at the encoder, while requires an additional usage of memory. Much of the gain from multiple reference pictures is achieved by using 2–5 reference frames. Hence, it has been shown that using 5 reference frames for prediction can yield 5–10% in bitrate savings as compared to using only 1 reference frame.

4.3. Transform

H.264 is unique in the use of a spatial 4×4 integer transform [10] as approximation of the floating point DCT. The new transform allows all operations to be performed with 16-bit integer arithmetic, avoiding mismatches between encoders and decoders. Furthermore, the transform has been formulated to be multiply-free, so it can be computed using only shift and addition operations. Equations (1) provide the exact formulation for the forward and inverse 4×4 integer transform. Such equations have been derived from the 4×4 DCT, where the nonrational coefficient $\sqrt{2}-1$ is approximated by 1/2 as a fixed point value, and the final scaling matrix with $a = 1/2$ and $b = \sqrt{2/5}$ is integrated into the quantization step:

$$Y = \left(\begin{bmatrix} 1 & 1 & 1 & 1 \\ 2 & 1 & -1 & -2 \\ 1 & -1 & -1 & 1 \\ 1 & -2 & 2 & -1 \end{bmatrix} \begin{bmatrix} & & & \\ & X & & \\ & & & \end{bmatrix} \begin{bmatrix} 1 & 2 & 1 & 1 \\ 1 & 1 & -1 & -2 \\ 1 & -1 & -1 & 2 \\ 1 & -2 & 1 & -1 \end{bmatrix}\right) \otimes \begin{bmatrix} a^2 & \frac{ab}{2} & a^2 & \frac{ab}{2} \\ \frac{ab}{2} & \frac{b^2}{4} & \frac{ab}{4} & \frac{b^2}{4} \\ a^2 & \frac{ab}{2} & a^2 & \frac{ab}{2} \\ \frac{ab}{2} & \frac{b^2}{4} & \frac{ab}{2} & \frac{b^2}{4} \end{bmatrix}$$

$$X' = \begin{bmatrix} 1 & 1 & 1 & \frac{1}{2} \\ 1 & \frac{1}{2} & -1 & -1 \\ 1 & -\frac{1}{2} & -1 & 1 \\ 1 & -1 & 1 & -\frac{1}{2} \end{bmatrix} \left(\begin{bmatrix} & & & \\ & Y & & \\ & & & \end{bmatrix} \otimes \begin{bmatrix} a^2 & ab & a^2 & ab \\ ab & b^2 & ab & b^2 \\ a^2 & ab & a^2 & ab \\ ab & b^2 & ab & b^2 \end{bmatrix}\right) \begin{bmatrix} 1 & 1 & 1 & 1 \\ 1 & \frac{1}{2} & -\frac{1}{2} & -1 \\ 1 & -1 & -1 & 1 \\ \frac{1}{2} & -1 & 1 & -\frac{1}{2} \end{bmatrix}. \tag{1}$$

Another differentiating trait between H.264 and previous standards is the presence of three transform modes depending on the type of the residual that is to be encoded. The 4×4 *residual transform* applies to all 4×4 blocks of residual data based on the integer approximation of the DCT previously described. The 4×4 *luma DC coefficient transform* applies only to luma MBs predicted in 16×16 intramode. After transforming all the 4×4 blocks within the residual MB, the resulting array of DC coefficients is further transformed by a 4×4 Hadamard transform. The 2×2 *chroma DC coefficient transform* applies to all the 8×8 blocks of residual chroma. A 2×2 Hadamard transform is computed over the array of DC coefficients resulting from the 4×4 residual transform.

4.4. In-loop deblocking filter

H.264 includes a deblocking filter [11] within the decoding loop, as illustrated in Figure 3. The deblocking filter is fully defined by the standard, and must be computed at both the encoder and the decoder. As an in-loop filter, it improves the quality of the decoded picture prior to motion estimation, showing a performance significantly superior to post-processing filters. The deblocking filter operates on the horizontal and vertical edges between blocks of 4×4 pixels inside the prediction loop to remove artifacts caused by quantization.

Being highly content adaptive, the filtering procedure mainly removes blocking artifacts and does not unnecessarily soften the visual content. At the slice level, the global filtering strength is adjusted to the individual characteristics of the video sequence. At the edge level, the filtering strength is made dependent on the inter-/intramode decision, the motion vectors, and the coded residuals. Finally, at the sample level, quantizer-dependent thresholds have been defined to turn off the filtering process for every individual sample.

4.5. *Entropy coding*

In contrast with other coding standards, two entropy coding schemes, namely, the content adaptive binary arithmetic coding (CABAC) and the content adaptive variable length coding (CAVLC), are supported in H.264, which is the outcome of a trade-off between complexity and coding efficiency. Hence, CABAC provides higher coding efficiency than CAVLC but at the expense of higher complexity.

In CABAC [12], context modeling provides conditional probabilities by utilizing context models and intersymbol redundancy. Arithmetic codes combine code words to permit noninteger number of bits to be assigned to each symbol. This is especially helpful for symbol probabilities much greater than 0.5, which occur often when context modeling is used. A fraction of a bit versus minimum 1 bit for VLC can be used in this case. Adaptivity is achieved by taking into account the cumulative probabilities of already coded symbols, which leads to a better fit of the arithmetic codes to the current symbol statistics. For example, statistics may vary for different sequences and for different bitrates.

CAVLC was proposed to improve the performance of the VLC without incurring the complexity of CABAC. CAVLC adds context models only to the transform coefficient coding. Other elements, such as motion vectors, MB types, and so forth, are coded with exp-Golomb codes (VLC). CAVLC coding separates runs and levels, which allows better adaptivity and thereby better coding efficiency. It also results in low complexity with low demand on memory. Zigzag scanning is still used, but in the transmission of the coefficient data (levels and runs), the scanning is done in reverse order. To improve efficiency, CAVLC switches between a reduced number of VLC tables. The selection takes into account the statistics in the previously coded blocks.

5. Profiles and levels

The H.264 standard has been designed to be generic and serve a wide range of applications, bitrates, resolutions, qualities, and services. To enable efficient implementations, a limited number of subsets of the entire syntax are also stipulated by the standard by means of profiles. Furthermore, to bound the variation in performance within a profile, reasonable limits are imposed by establishing a hierarchy of levels.

5.1. *Profiles*

The standard has defined three different profiles: the *baseline*, the *main* profile, and the *extended* profile. However, unlike previous standards, profiles in H.264 are not fully hierarchical. While the baseline is a subset of the extended profile, some tools in baseline are not present in main profile. Table 2 summarizes the set of tools available in each profile. In this section, we present only the new features unique to different profiles.

5.1.1 *Baseline*

Baseline supports some low-delay error resiliency tools combined with a subset of relatively low complexity coding efficiency tools [13]. To reduce the error effect and facilitate error recovery, flexible macroblock ordering (FMO) [14] is supported, whereby the sender can transmit MBs in non-raster scan order. The pattern of MBs partitioning the picture is represented by means of an MB allocation map. Figure 7 illustrates, as an example, three of the

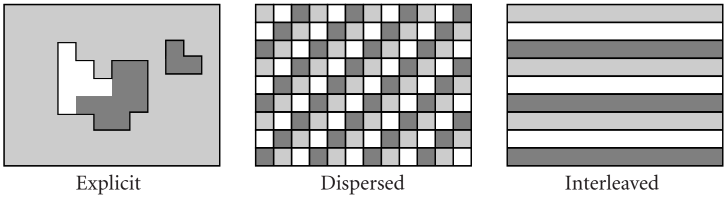

Figure 7: Example of three different MB allocation maps allowed by the FMO.

Table 2: Selection of algorithmic features according to the profile.

Visual tools	Baseline	Main	Extended
1/4 pixel precision	•	•	•
VLC entropy coding	•	•	•
CABAC entropy coding		•	
I pictures	•	•	•
P pictures	•	•	•
B pictures		•	•
SI/SP pictures			•
Flexible macroblock ordering	•		
Arbitrary slice ordering	•		
Data partitioning			•
Weighted prediction		•	•
Interlace picture-level adaptation		•	•
Interlace MB-level adaptation		•	
Redundant pictures	•		•

seven different types of MB allocation maps allowed by the standard. The explicit map is the most flexible mode in which a slice group identifier needs to be sent for each MB. The dispersed map facilitates error concealment when a entire slice group is lost. In interleaved map, slice groups are interleaved in raster scan order. Besides the FMO, H.264 provides another new error resilient feature, referred as arbitrary slice order (ASO). If used, slices may not be coded following the raster-scan order.

5.1.2 Main profile

Main profile supports all tools that tend to maximize the coding efficiency, such as inter biprediction and CABAC. Furthermore, for the first time, a weighted prediction (WP) tool is incorporated into a video coding standard. WP improves coding efficiency by applying a multiplicative weighting factor and an additive offset to the motion compensated prediction to form a WP, which is particularly useful for fading sequences. Two methods are provided for the WP tool. In the *explicit* mode (P and B slices), a weighting factor and offset may be coded in the slice header for each allowable reference picture index. In the *implicit* mode (B slices), the weighting factors are not coded but are derived based on the relative picture order count (POC) distances of the two reference pictures.

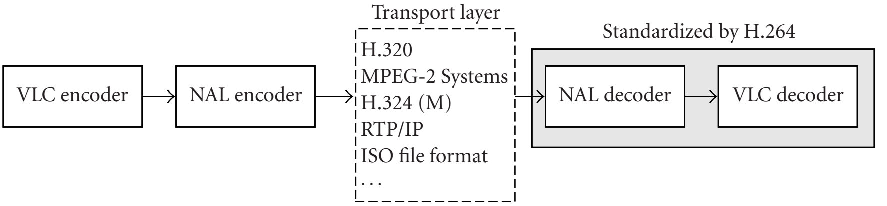

Figure 8: Relationship between the NAL and the transport layer.

5.1.3 *Extended profile*

The extended profile includes tools to support coding efficiency, error resiliency, and re-synchronization. A new picture type referred as *switching* pictures (SI or SP) is incorporated into H.264. An SP picture makes use of motion-compensated predictive coding to exploit temporal redundancy in the sequence similar to P pictures. The difference between SP and P pictures is that SP pictures allow identical picture reconstruction even when they are predicted from different reference frames. SI are similar to I and have the same features as SP.

5.2. *Levels*

Applying to all the previous profiles, H.264 also defines a set of levels and sublevels, which establish different degrees of capability. There are *profile-independent* level limits, as the maximum frame rate or the picture height and width, and *profile-specific* level limits, as the number of slices per picture, just to cite an example. According to the maximal frame size a level can support, levels are sorted as follows: level 1–QCIF @ 15 fps; level 2–CIF @ 30 fps; level 3–SDTV @ 25 fps; level 4–HDTV @ 30 fps; and level 5–Digital Cinema @ 30 fps. Table 3 illustrates an example of maximum bitrates according to level limits for some example frame sizes. Note that decoders supporting a given level shall also be capable of decoding bitstreams using all lower levels.

6. Network abstraction layer

The network abstraction layer (NAL) is specified to efficiently represent the encoded video data, allowing an easy integration into a variety of protocol and multiplex architectures [15]. Figure 8 illustrates the relationship between the NAL and the transport layer. The NAL provides both header information and the appropriate format for the conveyance of encoded video data by transport layers or storage media. The new concepts introduced by the NAL are related to the definition of NAL units and parameter sets, to which we devote the next two sections.

6.1. *NAL units*

According to the NAL specification, all the encoded data must be contained in NAL units. A NAL unit is a syntax structure consisting of a one-byte header and a raw byte sequence payload (RBSP). NAL units can be directly used as payload in packet-based networks or can be mapped into a bitstream-oriented transport layer. NAL units are self-contained in the sense that they are independently decodable. Only their size must be conveyed externally.

Table 3: Maximum bitrates according to level limits for some example frame sizes.

Level	Format	Width	Height	Limits	Max bitrate
1	QCIF	176	144	15 fps	64 Kbps
1.1	QVGA	320	240	10 fps	192
1.2	CIF	352	288	15 fps	384
1.3	CIF	352	288	30 fps	768
2	CIF	352	288	30 fps	2 Mbps
2.1	625-HHR	352	576	25 fps	4
2.2	625 SD	720	576	12.5 fps	4
3	625 SD	720	576	25 fps	10 Mbps
3.1	720p HD	1280	720	30 fps	14
3.2	720p HD	1280	720	60 fps	20
4	1080 HD	1920	1088	30 fps	20 Mbps
4.1	1080 HD	1920	1088	30 fps	50
4.2	1080 HD	1920	1088	60 fps	50
5	16 VGA	2460	1920	30 fps	135 Mbps
5.1	4 K×2K	4096	2048	30 fps	240

Depending on the encoded payload, the standard has defined up to thirteen different NAL unit types, among those we would like to introduce the following.

Coded slice. This NAL unit conveys all the information required for decoding a slice. This concerns both slice header information and encoded data.

Coded data partition. When data partitioning is used, the MB data of a slice is organized in three partitions: partition A contains header information as well as motion vectors; partition B contains intratexture information; and partition C contains intercoded texture. According to the standard, each partition is conveyed in its own NAL unit, namely, DPA, DPB, and DPC. This should allow, for example, the DPA transport to be done with higher QoS than DPB and DPC.

Instantaneous decoder refresh. An IDR picture is a special I or SI picture delivered as a decoder refresh point. It reinitializes the POC, resets the decoded pictures buffer, and guaranties that all the later coded pictures can be decoded without inter prediction from any picture decoded prior to the IDR picture.

Supplemental enhancement information. SEI messages are delivered into a NAL unit synchronous with the video data content to assist mainly in the processes of decoding and display. However, SEI is not limited to this purpose and can transmit any arbitrary data. Note that SEI is not required to decode VCL data correctly, and decoders are not required to process this information for conformance with the standard. Examples of SEI messages are display information, control information, error resilience issues, and so forth.

Sequence and picture parameter set. SPS and PPS NAL units have been defined to convey all the information relevant to more than one slice. Because of the novelty of this concept, we devote the next section to its presentation.

6.2. The parameter sets

With the layered nature of recent video coding standards, the loss of a packet containing header information renders useless all the following packages with related data. To address

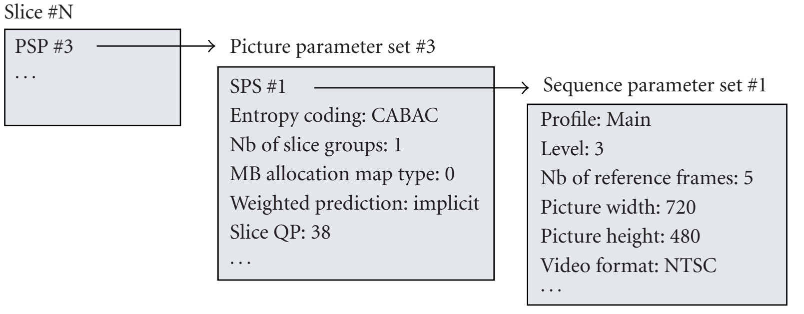

Figure 9: Asynchronous transmission of nested parameter sets.

this problem, several strategies have been designed. One strategy consists in checking for packet integrity at the decoder end and invoking error concealment when missing or corrupted packages are detected. The main drawback of such approach, however, concerns the difficulty of predicting header changes. A more robust strategy proposes the use of error resiliency tools when the transmitter is aware of possible header losses. However, this is done at a high cost in both bitrate and design complexity.

To overcome the previous drawbacks, the H.264 standard takes a new approach proposing to convey asynchronously all the information relevant to more than one slice in the nested parameter set NAL units. At this stage, the slice layer was identified as the appropriate smallest self-contained unit (unless data partitioning is used) because the size of the slice can be adjusted to the MTU size of the most demanding system. The slice header contains a reference to a picture parameter set (PPS) to be used for the decoding of its VLC data, and some parameters that change dynamically. At its turn, each PPS references a sequence parameter set (SPS), which contains information that varies slowly along a video coding session. Figure 9 illustrates an example.

The advantages of such approach are twofold: on the one hand, it decouples the transmission of MB data from relevant header information; and, on the other hand, it allows using different logic channels or even a different out-of-band protocols to convey reliably the most valuable information. Typically, establishment could be done during the capability exchange or in a session announcement. However, for those applications where no control protocol is available, special NAL unit types are specified to set up and update the SPS and the PPS in-band. Note that the use of in-band and out-of-band PS transmission is mutually exclusive.

6.3. Network adaptation

Although the mapping of the encoded data into transport networks is not specified by the standard, it is within its terms of reference to provide appropriate mechanisms, syntax, and interfaces to facilitate the gateway design. Hence, unlike previous video coding standards, H.264 has taken into account the transmission over packet networks in the video codec design from the very beginning. Systems using packet networks as the RTP/IP can transmit NAL units directly by using them as payloads.

Similarly, for all transport protocols that are not packet-based, such as H.320 for video conferences or MPEG-2 systems for broadcasting, the H.264 standard defines a byte-stream format to transmit a sequence of NAL units as an ordered stream of bytes. A byte-oriented framing is defined to identify the NAL unit boundaries and allow the correct decoding: NAL units are encapsulated by start codes of three bytes. Since H.264 contains two different entropy coding modes, a start-code-emulation-free environment would be very difficult to achieve. Instead, an emulation prevention code is specified by using byte stuffing.

7. Applications

The H.264 standard is intended to cover a large range of applications including, but not limited to, the following: CATV, cable TV on optical networks, copper, and so forth; DBS, direct broadcast satellite video services; DSL, digital subscriber line video services; DTTB, digital terrestrial television broadcasting; ISM, interactive storage media (optical discs, etc.); MMS, multimedia messaging service; MSPN, multimedia services over packet networks; RTC, real-time conversational services (video conference, videophone, etc.); RVS, remote video surveillance; and SSM, serial storage media (digital VTR, etc.).

Furthermore, requirements from the most typical applications have been considered when creating the three different profiles.

7.1. Baseline profile: low delay conversational services

Low delay conversational applications are expected to adopt the baseline profile of the H.264 standard, in replacement of the H.263 standard currently in use. Key applications in this area include video conferencing, internet video chat, and mobile video phones [16]. The Baseline has been conceived to satisfy all the needs of such kind of applications: it includes a powerful set of error resiliency tools; allows low complexity implementations by avoiding the handling of interlaced coding modes; and ensures low delay processing. Furthermore, the Baseline profile should benefit from a royalty-free policy, as demanded by the ITU-T community in the ToR [5].

7.2. Main profile: television broadcasting, streaming, and storage

The Main profile of the H.264 standard targets entertainment-quality broadcasting, streaming, and storage applications where latency is allowed. Small resolutions cover applications such as delivery of stored or live video content over the internet and other networks including 3G networks; SD resolutions target digital storage and broadcast, in-home servers, and camcorders; and HD resolutions are devoted to digital broadcast of HDTV and storage of HD material as, for example, in HD-DVDs.

The Main profile achieves a maximal compression performance by using B frames and CABAC at the cost of higher delays and increased complexity. No error resilience tools are included in this profile. Its key features are the adaptive WP, which is specially suited for fading sequences, and the MB frame/field adaptive coding mode, which gives bitrate savings up to 15% depending on the content. It is expected that the Main profile replaces progressively the MPEG-2 standard, since it performs significantly better at the same quality (PSNR). Similarly, there is an important quality improvement when sequences are encoded at the same bitrate. See Section 8 for more details.

7.3. Extended profile: internet streaming and communications

Despite of previous standards, H.264 has been conceived to be transmitted over packet-based networks, providing a seamless and easy integration of the coded video into all

Table 4: JVT verification testing.

Test	Resolutions	Format	Input rate	Bitrate	Compared codecs
MD baseline	QCIF, CIF	p	10–15 fr/s	96–768 Kbps	AVC Baseline @L2, MPEG-4 SP @L3
MD main	QCIF, CIF	p	8–15 fr/s	96–768 Kbps	AVC Main @L2, MPEG-4 ASP @L3
SD main	SD	i	50–60 fi/s	1.5–5 Mbps	AVC Main @L3, MPEG-2 TM5 and HiQ
HD main	HD	p, i	25–60 fr/s, 60 fi/s	6–20 Mbps	AVC Main @L4, MPEG-2 TM5 and HiQ

current protocols and multiplex architectures. Typical applications include fixed and wireless video transmission over the internet protocol (IP), as for 3G networks of mobile systems [17]. In this field, the extended profile of the H.264 standard combines successfully coding efficiency with an enhanced set of error resilience tools. However, and as for all the previous applications, extensive research needs to be conducted in order to exploit all the new capabilities of the standard.

8. Study of performance

Formal *subjective* verification tests were carried out by the JVT after closing the standardization process. The goal was to evaluate the compression performance of H.264 compared with that of previous MPEG standards, as commonly used in the intended application areas. Results of this study are documented in [18] and provide an authoritative study of the performance of the H.264 standard. In this section, we summarize the conclusions driven from the study and we refer the reader to [19] for an objective analysis based on rate distortion curves.

JVT verification tests were conducted at the laboratories FUB/ISCTI (Italy), NIST (USA), and TUM (Germany) from October to December 2003. Four sets of tests were defined based on application areas and latency issues. The *multimedia definition (MD) baseline profile* test targeted QCIF and CIF resolutions encoded at 1 Mbps or less. The test compared H.264 baseline profile at level 2 against MPEG-4 part 2 single profile (SP) at level 3. The *MD main profile* test targeted the same resolutions and bitrates as the MD baseline test, but was compared to MPEG-4 part 2 advanced single profile (ASP) for applications where latency is allowed. The *standard definition (SD) main profile* test aimed at encoding SD material at 8 Mbps or less, comparing H.264 Main profile at level 3 against two different MPEG-2 encoders: the MPEG-2 TM5 reference software, at similar level of maturity as the current H.264 implementations; and a real-time high-quality commercial encoder (MPEG-2 HiQ). Finally, the *high definition (HD) main profile* test focused on HDTV material encoded at 20 Mbps or less. The test compared the H.264 Main profile at level 4 against MPEG-2. Pre- and postfiltering were allowed for all tests depending on common industry practice. Table 4 summarizes the testing conditions for the four different sets.

The JVT verification test group concluded that H.264 provides a significant coding efficiency improvement over the codecs to which it was compared. It should be noticed, however, that there is also a substantial increase in complexity and computational cost due to the new intra- and inter prediction modes and use of multiple reference pictures. The performance analysis shows that H.264 achieved a coding efficiency improvement of 1.5 times or greater in 78% (66 out of 85) of the statistically conclusive cases, out of which 77% (51 out of 66) show improvements of 2 times or greater. A summary of the results presented in [18]

Table 5: Summary of results for the H.264 verification test.

MD baseline (QCIF)

Bitrates	Foreman	Paris	Head	Zoom
192 Kbps	T	—	—	—
96 Kbps	2 x	2 x	2 x	2 x
48 Kbps	2 x	2 x	> 2 x	1 x
24 Kbps	> 1 x	> 1 x	—	> 1 x

MD baseline (CIF)

Bitrates	Foreman	Paris	Head	Zoom
192 Kbps	T	T	T	—
96 Kbps	T	T, 2 x	T	—
48 Kbps	> 2 x	2 x	—	—
24 Kbps	2 x	2 x	2 x	2 x

MD main (QCIF)

Bitrates	Football	Mobile	Husky	Tempete
192 Kbps	—	—	—	T
96 Kbps	2 x	2 x	> 1 x	> 2 x
48 Kbps	2 x	2 x	2 x	2 x
24 Kbps	2 x/1 x	2 x/1 x	2 x	2 x

MD main (CIF)

Bitrates	Football	Mobile	Husky	Tempete
768 Kbps	> 1 x	T	—	T
384 Kbps	1 x	> 2 x	2 x	T, 2 x
192 Kbps	> 1 x	4 x	2 x	2 x
96 Kbps	> 1 x	> 2 x	2 x	2 x

SD main (MPEG-2 HiQ)

Bitrates	Football	Mobile	Husky	Tempete
6 Mbps	—	T	—	T
4 Mbps	1.5 x	T	1.5 x	T
3 Mbps	1.3 x	2.0 x	1 x/1.3 x	T
2.25 Mbps	> 1.3 x	2.7 x	1.3 x	T
1.5 Mbps	> 1.5 x	4.0 x	> 1.5 x	T, 2 x

SD main (MPEG-2 TM5)

Bitrates	Football	Mobile	Husky	Tempete
6 Mbps	—	T	—	T
4 Mbps	1.5 x	T	> 1.5 x	T
3 Mbps	1.3 x	> 2.0 x	2.0 x	T
2.25 Mbps	1.8 x	> 2.7 x	1.8 x	T
1.5 Mbps	2.0 x	> 4.0 x	2.7/2 x	T, 4 x

HD main (MPEG-2 HiQ)

	720(60p)		1080(30i)		1080(25p)	
Bitrates	Crew	Harbour	Stockholm	New mob & cal	River bed	Vintage car
20 Mbps	T	T	—	T	T	T
10 Mbps	2 x	T	1 x	T, 2 x	> 1 x	T, 2 x
6 Mbps	1.7 x	T, 3.3 x	not part of test	not part of test	> 1.7 x	1.7 x

HD main (MPEG-2 TM5)

	720(60p)		1080(30i)		1080(25p)	
Bitrates	Crew	Harbour	Stockholm	New mob & cal	River bed	Vintage car
20 Mbps	T	T	—	T	T	T
10 Mbps	2 x	T	2 x	T, 2 x	> 1 x	T, 2 x
6 Mbps	1.7 x	T, 1.7 x	not part of test	not part of test	> 1.7 x	1.7 x

is provided in Table 5. The numbers in the tables indicate the coding efficiency improvement achieved by the H.264 codec where the codecs being compared provide statistically equivalent picture quality. The letter "T" indicates that H.264 achieved visual transparency.

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Volume 2004, No. 3, 1 March 2004
Contents and Abstracts

New Insights into the RLS Algorithm

Jacob Benesty and Tomas Gänsler

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The recursive least squares (RLS) algorithm is one of the most popular adaptive algorithms that can be found in the literature, due to the fact that it is easily and exactly derived from the normal equations. In this paper, we give another interpretation of the RLS algorithm and show the importance of linear interpolation error energies in the RLS structure. We also give a very efficient way to recursively estimate the condition number of the input signal covariance matrix thanks to fast versions of the RLS algorithm. Finally, we quantify the misalignment of the RLS algorithm with respect to the condition number.

Using Mel-Frequency Cepstral Coefficients in Missing Data Technique

Zhang Jun, Sam Kwong, Wei Gang, and Qingyang Hong

http://dx.doi.org/10.1155/S1110865704309030

Filter bank is the most common feature being employed in the research of the marginalisation approaches for robust speech recognition due to its simplicity in detecting the unreliable data in the frequency domain. In this paper, we propose a hybrid approach based on the marginalisation and the soft decision techniques that make use of the Mel-frequency cepstral coefficients (MFCCs) instead of filter bank coefficients. A new technique for estimating the reliability of each cepstral component is also presented. Experimental results show the effectiveness of the proposed approaches.

Classification of Acoustic Emissions Using Modified Matching Pursuit

Samuel P. Ebenezer, Antonia Papandreou-Suppappola, and Seth B. Suppappola

http://dx.doi.org/10.1155/S1110865704311029

We propose methodologies to automatically classify time-varying warning signals from an acoustic monitoring system that indicate the potential catastrophic structural failures of reinforced concrete structures. Since missing even a single warning signal may prove costly, it is imperative to develop a classifier with high probability of correctly classifying the warning signals. Due to the time-varying nature of these signals, various time-frequency classifiers

are considered. We propose a new time-frequency decomposition-based classifier using the modified matching pursuit algorithm for an actual acoustic monitoring system. We investigate the superior performance of the classifier and compare it with existing classifiers for various sets of acoustic emissions, including warning signals from real-world faulty structures. Furthermore, we study the performance of the new classifier under different test conditions.

3D Visualization of Radar Backscattering Diagrams Based on OpenGL

Yulia V. Zhulina

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A digital method of calculating the radar backscattering diagrams is presented. The method uses a digital model of an arbitrary scattering object in the 3D graphics package "OpenGL" and calculates the backscattered signal in the physical optics approximation. The backscattering diagram is constructed by means of rotating the object model around the radar-target line.

Bearing Fault Detection Using Artificial Neural Networks and Genetic Algorithm

B. Samanta, Khamis R. Al-Balushi, and Saeed A. Al-Araimi

http://dx.doi.org/10.1155/S1110865704310085

A study is presented to compare the performance of bearing fault detection using three types of artificial neural networks (ANNs), namely, multilayer perceptron (MLP), radial basis function (RBF) network, and probabilistic neural network (PNN). The time domain vibration signals of a rotating machine with normal and defective bearings are processed for feature extraction. The extracted features from original and preprocessed signals are used as inputs to all three ANN classifiers: MLP, RBF, and PNN for two-class (normal or fault) recognition. The characteristic parameters like number of nodes in the hidden layer of MLP and the width of RBF, in case of RBF and PNN along with the selection of input features, are optimized using genetic algorithms (GA). For each trial, the ANNs are trained with a subset of the experimental data for known machine conditions. The ANNs are tested using the remaining set of data. The procedure is illustrated using the experimental vibration data of a rotating machine with and without bearing faults. The results show the relative effectiveness of three classifiers in detection of the bearing condition.

Neural-Network-Based Time-Delay Estimation

Samir Shaltaf

http://dx.doi.org/10.1155/S1110865704309261

A novel approach for estimating constant time delay through the use of neural networks (NN) is introduced. A desired reference signal and a delayed, damped, and noisy replica of it are both filtered by a fourth-order digital infinite impulse response (IIR) filter. The filtered signals are normalized with respect to the highest values they achieve and then applied as input for an NN system. The output of the NN is the estimated time delay. The network is first trained with one thousand training data set in which each data set corresponds to a randomly chosen constant time delay. The estimated time delay obtained by the NN is an accurate estimate of the exact time-delay values. Even in the case of noisy data, the estimation error obtained was a fraction of the sampling time interval. The delay estimates obtained by the NN are comparable to the estimated delay values obtained by the cross-correlation technique. The main advantage of using this technique is that accurate estimation of time delay results from performing one pass of the filtered and normalized data through the NN. This estimation process is fast when compared to the classical techniques utilized for time-delay estimation. Classical techniques rely on generating the computationally demanding cross-correlation function of the two signals. Then a peak detector algorithm is utilized to find the time at which the peak occurs.

A New Algorithm for Joint Range-DOA-Frequency Estimation of Near-Field Sources

Jian-Feng Chen, Xiao-Long Zhu, and Xian-Da Zhang

http://dx.doi.org/10.1155/S1110865704310152

This paper studies the joint estimation problem of ranges, DOAs, and frequencies of near-field narrowband sources and proposes a new computationally efficient algorithm, which employs a symmetric uniform linear array, uses eigenvalues together with the corresponding eigenvectors of two properly designed matrices to estimate signal parameters, and does not require searching for spectral peak or pairing among parameters. In addition, the proposed algorithm can be applied in arbitrary Gaussian noise environment since it is based on the fourth-order cumulants, which is verified by extensive computer simulations.

Soft and Joint Source-Channel Decoding of Quasi-Arithmetic Codes

Thomas Guionnet and Christine Guillemot

http://dx.doi.org/10.1155/S1110865704308085

The issue of robust and joint source-channel decoding of quasi-arithmetic codes is addressed. Quasi-arithmetic coding is a reduced precision and complexity implementation of arithmetic coding. This amounts to approximating the distribution of the source. The approximation of the source distribution leads to the introduction of redundancy that can be exploited for robust decoding in presence of transmission errors. Hence, this approxi-

mation controls both the trade-off between compression efficiency and complexity and at the same time the redundancy (*excess rate*) introduced by this suboptimality. This paper provides first a state model of a quasi-arithmetic coder and decoder for binary and M-ary sources. The design of an error-resilient soft decoding algorithm follows quite naturally. The compression efficiency of quasi-arithmetic codes allows to add extra redundancy in the form of markers designed specifically to prevent desynchronization. The algorithm is directly amenable for iterative source-channel decoding in the spirit of serial turbo codes. The coding and decoding algorithms have been tested for a wide range of channel signal-to-noise ratios (SNRs). Experimental results reveal improved symbol error rate (SER) and SNR performances against Huffman and optimal arithmetic codes.

A Novel Pseudoerror Monitor

Peng Wang and Wee Ser

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The error rate (ER) is a crucial criterion in evaluating the performance of a digital communication system. Many ER estimation methods have been described in the literature. Among them, the pseudoerror monitoring solution has attracted special attention due to its consistent performance in different environments and distinctive blind estimation capability, that is, estimating the ER without needing any prior knowledge of the transmitted information. In this paper, a novel pseudoerror monitor (PEM) design, the kernel PEM, is developed. Incorporating the strength of the probability density function (pdf) approximation technique, the proposed design has remarkable advantage of being able to produce statistically consistent ER estimate within a much shorter observation time. Simulation results are given in support of this claim.

Subband Adaptive Array for DS-CDMA Mobile Radio

Xuan Nam Tran, Takanori Omata, Tetsuki Taniguchi, and Yoshio Karasawa

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We propose a novel scheme of subband adaptive array (SBAA) for direct-sequence code division multiple access (DS-CDMA). The scheme exploits the spreading code and pilot signal as the reference signal to estimate the propagation channel. Moreover, instead of combining the array outputs at each output tap using a synthesis filter and then despreading them, we despread directly the array outputs at each output tap by the desired user's code to save the synthesis filter. Although its configuration is far different from that of 2D RAKEs, the proposed scheme exhibits relatively equivalent performance of 2D RAKEs while having less computation load due to utilising adaptive signal processing in subbands. Simulation programs are carried out to explore the performance of the scheme and compare its performance with that of the standard 2D RAKE.

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Special Issue on Biometric Signal Processing

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

This special issue brings together researchers working on biometric signal processing and its applications, with a particular emphasis on person authentication and identification. In this special issue, we are pleased to present new techniques as well as developments of the different signal processing techniques and their applications. We have three papers on speaker verification, two papers on fingerprint matching, four papers on human face detection and recognition, two papers on signature verification, and one paper on gait recognition.

Speaker verification and fingerprint pattern recognition are among the very first applications in biometric signal processing. The first paper by Bimbot et al. is a tutorial paper that provides an overview of a state-of-the-art text-independent speaker verification system. A modular scheme of the training and test phase of the system is introduced. Gaussian mixture model and ceptral analysis, which are the dominant techniques for speaker verification, are explained in detail in this paper. Other speaker modeling alternatives, scoring normalization, and the extension of speaker verification techniques to other applications are also covered. This is a useful paper for researchers working in this field. The second paper by Mak et al. considers a new channel compensation approach to telephone-based speaker verification. This direction of speaker verification has attracted much attention recently because of the proliferation of eBanking and eCommerce, which require the verification of a speaker over the telephone. The paper proposes to combine a handset selector with stochastic feature transformation to reduce the distortion caused by the limited bandwidth of the telephone network. In addition, a divergence-based handset selector with out-of-handset rejection capability to identify the unseen handsets is proposed. The last paper on speaker verification is by Besacier et al., and it presents the investigation of speaker verification over the Internet at the protocol level and at the speech signal level. At the protocol level, the paper recommends the transmission of data models or features instead of raw biometric data in order to reduce the transmission time, and the use of encryption/decryption for enhancing data security. At the signal level, the paper shows that packet loss is not a major problem for text-independent speaker authentication. However, the use of a low bit rate coder will greatly degrade the performance. The next two papers are on fingerprint segmentation and recognition. The paper by Chen et al. proposes a novel algorithm for the block feature-based segmentation of fingerprints. Its major contribution is an integrative approach to feature analysis and segmentation. Adding morphological postprocessing, the method could significantly improve the quality of segmenting fingerprints with greatly reduced misclassification. The paper by Yin et al. focuses on an accurate estimation of the ridge distance in fingerprint feature analysis. Its major contribution is a balanced effort on both algorithm development and performance evaluation. Due to the lack of much published work on this topic, the paper could significantly motivate fertile scientific discussions.

Research on human face recognition has been growing rapidly over the last two decades. To identify a person in an open environment, human face recognition is the most natural approach. This is because to collect useful data for recognition, face recognition has the advantage of being nonintrusive, requiring little cooperation from the person being identified. The paper by Jiang addresses the issues of detecting human faces in a complex airport environment. The paper presents a new variant of the AdaBoost to detect human faces, namely, S-AdaBoost, which uses the AdaBoost distribution weight as a dividing tool to separate the input face space into inlier and outlier face spaces; dedicated classifiers are then used to handle the inliers and outliers in their corresponding face spaces. This is an effective approach to locating human faces in a complex background. The accuracy of detecting a human face and locating its respective facial features has a direct impact on the performance of the face recognition algorithms to be used. The next three papers are also on face recognition. The paper by Perronnin et al. proposes a novel approach to face recognition by modeling the transformation between face images of the same person. The transformation is approximated by means of a collection of local transformations with a constraint to make neighboring transformations consistent with each other. Local transformations and neighboring constraints are embedded within a probabilistic framework using 2D hidden Markov models (HMMs). Another major contribution of this paper is the introduction of the Turbo-HMM, which is an efficient technique to approximate intractable 2D HMMs. The next two papers on face recognition consider the optimal conversion of a color image to a monochromatic image and the combination of different face recognition results, respectively, to improve face recognition performance, instead of considering a new face recognition algorithm. The paper by Jones III et al. proposes optimal methods to convert a color image to a monochromatic image for face recognition. Actually, this issue has not been considered in the current face recognition algorithms. Three approaches—Karhunen-Loève analysis, the linear regression of color distribution, and a genetic algorithm—are explored to determine the optimal conversion. The color-conversion methods are independent of the face recognition approach being used, but can improve its recognition performance. The other paper by Huang et al. presents a way to achieve a better recognition performance level by combining the classifier outputs based on different face recognition techniques. The paper proposes three methods to combine the classifiers, namely, the normalization of the classifier output, the selection of classifier(s) for recognition, and the weighting of each classifier.

Signature verification is also a commonly used biometric identification technique. Signatures have been widely used in bank and credit card transactions as a means of authentication, and most computers or hand-held devices are also equipped with I/O to allow handwriting input. In addition, a signature or a piece of handwriting may be changed by the user, but that is impossible with fingerprints, face, irises, and so forth. This special issue has two papers on signature and handwriting verification. The paper by Vielhauer and Steinmetz presents an approach to derive biometric hashes based on handwriting. The paper investigates the degree to which each of the statistical feature parameters contributed to the overall intrapersonal stability and interpersonal value space. A feature correlation method for feature analysis and selection is also proposed. The next paper by Coetzer et al. is on offline signature verification. The paper proposes to use the discrete Radon transform first to extract global features of a scanned signature, and then to feed the features into the HMM in order to model the signature. Most of the existing signature verification approaches utilize local features. It is likely that the algorithm proposed in the paper can be incorporated with other existing signature verification methods based on local features to achieve a significant improvement.

The last paper in this issue is by BenAbdelkader et al. They study human identification at a distance using gait recognition. This research has recently attracted growing interest from computer vision researches even though it is still at its infancy. The paper describes a novel gait recognition technique based on the image self-similarity of a walking person. The major advantages of the method are that it is correspondence-free, works well with low-resolution video, and is robust to variation in clothing, lighting, and to segmentation errors.

In summary, this special issue presents a wide range of different biometric features and techniques for applications, such as speaker verification, fingerprint recognition, face recognition, signature verification, and gait recognition. We can foresee that developments in this field will become even more rapid in the future. We hope that the techniques presented in this issue will be of great use to researchers in this field and will provide them with possible directions for the development of biometric technologies. We wish to thank all the authors for their contributions and all the reviewers for their diligent efforts in evaluating and commenting on the papers.

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Contents and Abstracts

A Tutorial on Text-Independent Speaker Verification

Frédéric Bimbot, Jean-François Bonastre, Corinne Fredouille, Guillaume Gravier, Ivan Magrin-Chagnolleau, Sylvain Meignier, Teva Merlin, Javier Ortega-García, Dijana Petrovska-Delacrétaz, and Douglas A. Reynolds

http://dx.doi.org/10.1155/S1110865704310024

This paper presents an overview of a state-of-the-art text-independent speaker verification system. First, an introduction proposes a modular scheme of the training and test phases of a speaker verification system. Then, the most commonly speech parameterization used in speaker verification, namely, cepstral analysis, is detailed. Gaussian mixture modeling, which is the speaker modeling technique used in most systems, is then explained. A few speaker modeling alternatives, namely, neural networks and support vector machines, are mentioned. Normalization of scores is then explained, as this is a very important step to deal with real-world data. The evaluation of a speaker verification system is then detailed, and the detection error trade-off (DET) curve is explained. Several extensions of speaker verification are then enumerated, including speaker tracking and segmentation by speakers. Then, some applications of speaker verification are proposed, including on-site applications, remote applications, applications relative to structuring audio information, and games. Issues concerning the forensic area are then recalled, as we believe it is very important to inform people about the actual performance and limitations of speaker verification systems. This paper concludes by giving a few research trends in speaker verification for the next couple of years.

Stochastic Feature Transformation with Divergence-Based Out-of-Handset Rejection for Robust Speaker Verification

Man-Wai Mak, Chi-Leung Tsang, and Sun-Yuan Kung

http://dx.doi.org/10.1155/S1110865704308048

The performance of telephone-based speaker verification systems can be severely degraded by linear and nonlinear acoustic distortion caused by telephone handsets. This paper proposes to combine a handset selector with stochastic feature transformation to reduce the distortion. Specifically, a Gaussian mixture model (GMM)-based handset selector is trained to identify the most likely handset used by the claimants, and then handset-specific stochastic feature transformations are applied to the distorted feature vectors. This paper also proposes a divergence-based handset selector with out-of-handset (OOH) rejection capability to identify the "unseen" handsets. This is achieved by measuring the *Jensen difference* between the selector's output and a constant vector with identical elements. The resulting handset selector is combined with the proposed feature transformation technique for telephone-based speaker verification. Experimental results based on 150 speakers of the HTIMIT corpus show that the handset selector, either with or without OOH rejection capability, is able to identify the "seen" handsets accurately (98.3% in both cases). Results also

demonstrate that feature transformation performs significantly better than the classical cepstral mean normalization approach. Finally, by using the transformation parameters of the seen handsets to transform the utterances with correctly identified handsets and processing those utterances with unseen handsets by cepstral mean subtraction (CMS), verification error rates are reduced significantly (from 12.41% to 6.59% on average).

Voice Biometrics over the Internet in the Framework of COST Action 275

Laurent Besacier, Aladdin M. Ariyaeeinia, John S. Mason, Jean-François Bonastre, Pedro Mayorga, Corinne Fredouille, Sylvain Meignier, Johann Siau, Nicholas W. D. Evans, Roland Auckenthaler, and Robert Stapert

http://dx.doi.org/10.1155/S1110865704310012

The emerging field of biometric authentication over the Internet requires both robust person authentication and secure computer network protocols. This paper presents investigations of vocal biometric person authentication over the Internet, both at the protocol and authentication robustness levels. As part of this study, an appropriate client-server architecture for biometrics on the Internet is proposed and implemented. It is shown that the transmission of raw biometric data in this application is likely to result in unacceptably long delays in the process. On the other hand, by using data models (or features), the transmission time can be reduced to an acceptable level. The use of encryption/decryption for enhancing the data security in the proposed client-server link and its effects on the transmission time are also examined. Furthermore, the scope of the investigations includes an analysis of the effects of packet loss and speech coding on speaker verification performance. It is experimentally demonstrated that whilst the adverse effects of packet loss can be negligible, the encoding of speech, particularly at a low bit rate, can reduce the verification accuracy considerably. The paper details the experimental investigations conducted and presents an analysis of the results.

Segmentation of Fingerprint Images Using Linear Classifier

Xinjian Chen, Jie Tian, Jiangang Cheng, and Xin Yang

http://dx.doi.org/10.1155/S1110865704309194

An algorithm for the segmentation of fingerprints and a criterion for evaluating the block feature are presented. The segmentation uses three block features: the block clusters degree, the block mean information, and the block variance. An optimal linear classifier has been trained for the classification per block and the criteria of minimal number of misclassified samples are used. Morphology has been applied as postprocessing to reduce the number of classification errors. The algorithm is tested on FVC2002 database, only 2.45% of the blocks are misclassified, while the postprocessing further reduces this ratio. Experiments have shown that the proposed segmentation method performs very well in rejecting false fingerprint features from the noisy background.

Ridge Distance Estimation in Fingerprint Images: Algorithm and Performance Evaluation

Yilong Yin, Jie Tian, and Xiukun Yang

http://dx.doi.org/10.1155/S1110865704310103

It is important to estimate the ridge distance accurately, an intrinsic texture property of a fingerprint image. Up to now, only several articles have touched directly upon ridge distance estimation. Little has been published providing detailed evaluation of methods for ridge distance estimation, in particular, the traditional spectral analysis method applied in the frequency field. In this paper, a novel method on nonoverlap blocks, called the statistical method, is presented to estimate the ridge distance. Direct estimation ratio (DER) and estimation accuracy (EA) are defined and used as parameters along with time consumption (TC) to evaluate performance of these two methods for ridge distance estimation. Based on comparison of performances of these two methods, a third hybrid method is developed to combine the merits of both methods. Experimental results indicate that DER is 44.7%, 63.8%, and 80.6%; EA is 84%, 93%, and 91%; and TC is 0.42, 0.31, and 0.34 seconds, with the spectral analysis method, statistical method, and hybrid method, respectively.

Robust Face Detection in Airports

Jimmy Liu Jiang, Kia-Fock Loe, and Hong Jiang Zhang

http://dx.doi.org/10.1155/S1110865704310206

Robust face detection in complex airport environment is a challenging task. The complexity in such detection systems stems from the variances in image background, view, illumination, articulation, and facial expression. This paper presents the S-AdaBoost, a new variant of AdaBoost developed for the face detection system for airport operators (FDAO). In face detection application, the contribution of the S-AdaBoost algorithm lies in its use of AdaBoost's distribution weight as a dividing tool to split up the input face space into inlier and outlier face spaces and its use of dedicated classifiers to handle the inliers and outliers in their corresponding spaces. The results of the dedicated classifiers are then nonlinearly combined. Compared with the leading face detection approaches using both the data obtained from the complex airport environment and some popular face database repositories, FDAO's experimental results clearly show its effectiveness in handling real complex environment in airports.

A Probabilistic Model for Face Transformation with Application to Person Identification

Florent Perronnin, Jean-Luc Dugelay, and Kenneth Rose

http://dx.doi.org/10.1155/S1110865704308012

A novel approach for content-based image retrieval and its specialization to face recognition are described. While most face recognition techniques aim at modeling faces, our goal is to model the *transformation* between face images of the same person. As a global face transformation may be too complex to be modeled directly, it is approximated by a collection of local transformations with a constraint that imposes consistency between neighboring transformations. Local transformations and neighborhood constraints are embedded within a probabilistic framework using two-dimensional hidden Markov models (2D HMMs). We further introduce a new efficient technique, called turbo-HMM (T-HMM) for approximating intractable 2D HMMs. Experimental results on a face identification task show that our novel approach compares favorably to the popular eigenfaces and fisherfaces algorithms.

Optimization of Color Conversion for Face Recognition

Creed F. Jones III and A. Lynn Abbott

http://dx.doi.org/10.1155/S1110865704401073

This paper concerns the conversion of color images to monochromatic form for the purpose of human face recognition. Many face recognition systems operate using monochromatic information alone even when color images are available. In such cases, simple color transformations are commonly used that are not optimal for the face recognition task. We present a framework for selecting the transformation from face imagery using one of three methods: Karhunen-Loève analysis, linear regression of color distribution, and a genetic algorithm. Experimental results are presented for both the well-known eigenface method and for extraction of Gabor-based face features to demonstrate the potential for improved overall system performance. Using a database of 280 images, our experiments using these methods resulted in performance improvements of approximately 4% to 14%.

Face Recognition Using Local and Global Features

Jian Huang, Pong C. Yuen, J. H. Lai, and Chun-hung Li

http://dx.doi.org/10.1155/S1110865704312187

The combining classifier approach has proved to be a proper way for improving recognition performance in the last two decades. This paper proposes to combine local and global facial features for face recognition. In particular, this paper addresses three issues in combining classifiers, namely, the normalization of the classifier output, selection of classifier(s) for recognition, and the weighting of each classifier. For the first issue, as the scales of each classifier's output are different, this paper proposes two methods, namely, linear-exponential normalization method and distribution-weighted Gaussian normalization method, in normalizing the outputs. Second, although combining different classifiers can improve the performance, we found that some classifiers are redundant and may even degrade the recognition performance. Along this direction, we develop a simple but effective algorithm for classifiers selection. Finally, the existing methods assume that each classifier is equally weighted. This paper suggests a weighted combination of classifiers based on Kittler's combining classifier framework. Four popular face recognition methods, namely, eigenface, spectroface, independent component analysis (ICA), and Gabor jet are selected for combination and three popular face databases, namely, Yale database, Olivetti Research Laboratory (ORL) database, and the FERET database, are selected for evaluation. The experimental results show that the proposed method has 5–7% accuracy improvement.

Handwriting: Feature Correlation Analysis for Biometric Hashes

Claus Vielhauer and Ralf Steinmetz

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In the application domain of electronic commerce, biometric authentication can provide one possible solution for the key management problem. Besides server-based approaches, methods of deriving digital keys directly from biometric measures appear to be advantageous. In this paper, we analyze one of our recently published specific algorithms of this category based on behavioral biometrics of handwriting, the biometric hash. Our interest is to investigate to which degree each of the underlying feature parameters contributes to the overall intrapersonal stability and interpersonal value space. We will briefly discuss related work in feature evaluation and introduce a new methodology based on three components: the intrapersonal scatter (deviation), the interpersonal entropy, and the correlation between both measures. Evaluation of the technique is presented based on two data sets of different size. The method presented will allow determination of effects of parameterization of the biometric system, estimation of value space boundaries, and comparison with other feature selection approaches.

Offline Signature Verification Using the Discrete Radon Transform and a Hidden Markov Model

J. Coetzer, B. M. Herbst, and J. A. du Preez

http://dx.doi.org/10.1155/S1110865704309042

We developed a system that automatically authenticates offline handwritten signatures using the discrete Radon transform (DRT) and a hidden Markov model (HMM). Given the robustness of our algorithm and the fact that only global features are considered, satisfactory results are obtained. Using a database of 924 signatures from 22 writers, our system achieves an equal error rate (EER) of 18% when only high-quality forgeries (skilled forgeries) are considered and an EER of 4.5% in the case of only casual forgeries. These signatures were originally captured offline. Using another database of 4800 signatures from 51 writers, our system achieves an EER of 12.2% when only skilled forgeries are considered. These signatures were originally captured online and then digitally converted into static signature images. These results compare well with the results of other algorithms that consider only global features.

Gait Recognition Using Image Self-Similarity

Chiraz BenAbdelkader, Ross G. Cutler, and Larry S. Davis

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Gait is one of the few biometrics that can be measured at a distance, and is hence useful for passive surveillance as well as biometric applications. Gait recognition research is still at its infancy, however, and we have yet to solve the fundamental issue of finding gait features which at once have sufficient discrimination power and can be extracted robustly and accurately from low-resolution video. This paper describes a novel gait recognition technique based on the image self-similarity of a walking person. We contend that the similarity plot encodes a projection of gait dynamics. It is also correspondence-free, robust to segmentation noise, and works well with low-resolution video. The method is tested on multiple data sets of varying sizes and degrees of difficulty. Performance is best for fronto-parallel viewpoints, whereby a recognition rate of 98% is achieved for a data set of 6 people, and 70% for a data set of 54 people.

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Special Issue on MIMO Communications and Signal Processing

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The topic of multiple-input multiple-output (MIMO) systems is one that has attracted a significant amount of attention in the research community over the past decade or so. MIMO systems refer to wireless systems that are equipped with multiple antenna elements on either side of a communication link. Propelled by the startling discovery in the mid 1990's that the capacity of MIMO systems grows roughly proportionally with the minimum number of antenna elements on each side of the wireless link, the field has undergone an explosive growth in both the academic and the industrial communities that has led to many further important advances. These advances have brought about not only the definition of new subareas of focused research, but also, equally importantly, a reconsideration of older techniques and a cross-fertilization of ideas from several other overlapping fields.

One of the research areas that has both affected strongly MIMO systems and has been equally affected by them is that of signal processing, as many of the developed/demonstrated MIMO transceiver architectures are based on advanced signal processing techniques. On the transmitter side, one can view most space-time coding/spatial multiplexing techniques as solving a problem of space-time signal design. On the receiver side, various flavors of multiuser detectors, space-time decoders, and related techniques for MIMO channel estimation and tracking (e.g., including blind/semiblind processing) are also typically derived in a signal processing framework.

More recent research on MIMO systems has started to focus on new areas of interest. At the link level, such areas are the handling of cochannel (e.g., in-cell and out-of-cell) interference; the development of precoding techniques at the transmitter to preempt adverse channel effects; and the design and use of efficient receiver-to-transmitter feedback mechanisms to improve the link throughput. In parallel, many studies have focused on the application of such MIMO techniques to specific transmission formats (dictated by different air interfaces) such as CDMA, OFDM, and so forth. Moving up the protocol stack, the interaction of MIMO techniques with MAC layer procedures such as adaptive retransmission and scheduling is an area that has started producing important know-how, especially regarding the suitability of MIMO techniques in high-speed data systems. Moving beyond wireless links, architecting an entire wireless network that uses MIMO connections poses a number of important questions, both at a fundamental level (e.g., MIMO network capacity) and at a practical level (e.g., MIMO network design).

With all of the above in mind, this special issue aims at giving a well-rounded snapshot of recent advances that cover most of these topics, with a special emphasis on signal processing methodologies as a design tool. As progress in the field is both rapidly emerging and voluminous, it has clearly not been our intent to provide an exhaustive coverage of all MIMO topics but rather a good selection of recent studies that are indicative of the progress in the field.

The papers included in this special issue address a broad range of issues arising in the development and application of MIMO techniques. These contributions range from general space-time coding and processing techniques and analytical methodologies to specific implementation issues arising in particular wireless standards and environments and to fundamentals of wireless MIMO networks. Among other topics, they touch upon the areas of transmitter and receiver design, blind and training-based techniques, link-level and system-level studies, open- and closed-loop systems, physical layer and higher layer issues, and wireless LAN and cellular applications.

The specific contributions of the papers in this issue are summarized in the following paragraphs.

Invited paper

In their invited paper, Jafar, Foschini, and Goldsmith present an in-depth analysis of the so-called "PhantomNet" wireless network concept. In such a network, the best possible service is provided to new users joining the network without affecting existing users. The problem is addressed in its full generality, that is, assuming multiple cells, users, and antennas, and results are obtained for both uplink and downlink communication. Optimality is sought in terms of the multiuser capacity region. This leads to a high degree of generality of the presented results and solutions. Furthermore, despite the inherent differences between the two directions of communication (and the resulting differences between the corresponding solutions), the authors demonstrate a remarkable symmetry between the uplink and downlink problems.

Channel estimation and multiuser detection in MIMO systems

In the first paper of this section, J. Du and Y. Li study the problem of channel estimation for D-BLAST OFDM systems. The authors propose a layerwise channel estimation algorithm that takes advantage of the D-BLAST structure. Further performance improvements are realized by introducing a subspace tracking scheme.

In the next paper, Buzzi, Grossi, and Lops study the problem of blind multiuser detection in asynchronous DS-CDMA systems equipped with multiple antennas. Several novel blind schemes are proposed and their performance is evaluated, showing their multiple access interference suppression capability, despite the absence of channel state information.

Another blind detection scheme that is specifically tailored to space-time differentially encoded systems is presented in the paper by Zhang and Ilow. Their proposed receiver algorithm is based on constant modulus characteristics of signaling and it is suitable for a rich multipath environment. The scheme requires no channel estimation and can work with small numbers of signal samples.

In their paper, Y. Du and Chan examine a technique for speeding up the search for an optimal multiuser detection solution using a genetic algorithm. The authors first study the objective function of the genetic algorithms. Then they propose two detectors to generate the seed chromosome of the initial population. Their results show that the proposed scheme not only reduces the computational complexity of finding the detector, but also improves performance.

MIMO systems, space-time coding, and beamforming

In the first paper of this section, C. Li and Xiaodong Wang compare the performance of three well-known MIMO techniques: BLAST, space-time block coding (STBC), and linear precoding/coding (i.e., beamforming) in the context of WCDMA. The authors study the signal-to-noise properties analytically, and the bit error rate performance via simulations. They also consider a subspace method for implementing the linear precoding method (which requires channel knowledge at the transmitter). The authors evaluate the trade-offs between BLAST and STBC in terms of data-rate and diversity in this situation (see also the following paper in this section) and demonstrate that subspace-based beamforming can be effectively realized in WCDMA systems.

In the next paper, Mecklenbräuker and Rupp consider a new STBC scheme that extends the well-known Alamouti codes to the situation in which the number of transmit antennas is an arbitrary power of two. Further solutions for arbitrary even numbers of transmit antennas are also presented, which offer improved orthogonalization properties while preserving high diversity. The authors also consider schemes that trade off the properties of Alamouti and BLAST-type systems (see also Li's and Wang's paper above) to achieve a continuous trade-off between quality of service and data rate. The appropriate trade-off can be selected using only the number of transmit antennas. Implications of these techniques for UMTS are also discussed.

MIMO systems and interference

In his paper, Blum studies the problem of maximum system mutual information in MIMO systems that employ antenna selection in the presence of interference. This leads to optimal signaling covariance matrices for the interesting case of limited channel feedback required for antenna selection.

The paper by Song and Blostein studies the effect of colored space-time interference on MIMO systems, emphasizing the problems of channel estimation, data detection, and interference correlation estimation. The focus is on the case of one dominant interferer and the quantification of its impact on the performance of a generalized BLAST ordered data detection algorithm. The authors show that exploiting the interference's spatio-temporal nature can result in important gains.

MIMO techniques in current/emerging air interfaces

In the first paper of this section, J. Liu and J. Li study some practical issues arising in the application of MIMO OFDM to high-rate wireless LAN systems. The authors propose signaling and corresponding synchronization, channel estimation, and detection schemes that are backward compatible with the existing 802.11a standard. They also propose the use of a BLAST-type data transmission scheme and a simple LS-based soft detector to reduce the complexity of the receiver.

In the next paper, Hansen, Affes, and Mermelstein revisit the problem of multiuser detection in CDMA networks. The authors apply an interference subspace rejection technique to the downlink of networks in which the spreading factors or modulation used by the interferer may not be known. The schemes proposed in the paper require no prior knowledge of these factors. A new code allocation scheme is also proposed to reduce the complexity of the proposed interference cancellation schemes.

The paper by González-López, Míguez, and Castedo presents a maximum likelihood channel estimation scheme that is suitable for turbo equalization in a space-time coded system. The authors apply their scheme to GSM-based transmission in a subway tunnel. Their experiment shows a significant reduction in the required training sequence length.

In the final paper of this section, Leus, Petré, and Moonen propose novel transmit diversity and corresponding space-time chip equalization techniques for DS-CDMA systems. Their proposed scheme is shown to achieve both maximal antenna diversity and maximal multipath diversity.

Resource allocation and feedback in multiple antenna systems

In the first paper of this section, Han, Farrokhi, and K. J. Ray Liu revisit the problem of jointly optimizing power control and beamforming to minimize the cochannel interference. The authors optimize the bit error rate directly in calculating the power and beamforming vector. Both the power control and beamforming algorithms are updated iteratively and are shown to converge.

In their paper, Chung, Lozano, Huang, Sutivong, and Cioffi study closed-loop MIMO systems. In order to achieve the closed-loop capacity, the authors propose to use a low rate feedback channel to provide rate and power information to the transmitter. Two joint rate and power allocation schemes are proposed and studied by the authors. Their results show that the performance loss due to the quantization of power is marginal, and that the MIMO system demonstrates an average rate close to capacity with the low-rate feedback channel and strong coding scheme.

Higher layer issues in MIMO systems

In the final paper of the special issue, Zheng, Lozano, and Haleem propose an ARQ scheme based on the BLAST system. The authors suggest the use of separate ARQ for each layer of the BLAST transmission. This multiple ARQ structure not only improves the throughput, but also facilitates the interference cancellation.

We believe that the included papers present an excellent sampling of state-of-the-art research in the field of MIMO communications and signal processing. We would like to thank all of the authors for their timely contributions and we anticipate that these papers will make this special issue a useful reference that will act as a catalyst for further exciting research in the field of MIMO systems.

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Contents and Abstracts

PhantomNet: Exploring Optimal Multicellular Multiple Antenna Systems

Syed A. Jafar, Gerard J. Foschini, and Andrea J. Goldsmith

http://dx.doi.org/10.1155/S1110865704312072

We present a network framework for evaluating the theoretical performance limits of wireless data communication. We address the problem of providing the best possible service to new users joining the system without affecting existing users. Since, interference-wise, new users are required to be invisible to existing users, the network is dubbed PhantomNet. The novelty is the generality obtained in this context. Namely, we can deal with multiple users, multiple antennas, and multiple cells on both the uplink and the downlink. The solution for the uplink is effectively the same as for a single cell system since all the base stations (BSs) simply amount to one composite BS with centralized processing. The optimum strategy, following directly from known results, is successive decoding (SD), where the new user is decoded before the existing users so that the new users' signal can be subtracted out to meet its invisibility requirement. Only the BS needs to modify its decoding scheme in the handling of new users, since existing users continue to transmit their data exactly as they did before the new arrivals. The downlink, even with the BSs operating as one composite BS, is more problematic. With multiple antennas at each BS site, the optimal coding scheme and the capacity region for this channel are unsolved problems. SD and dirty paper (DP) are two schemes previously reported to achieve capacity in special cases. For PhantomNet, we show that DP coding at the BS is equal to or better than SD. The new user is encoded before the existing users so that the interference caused by his signal to existing users is known to the transmitter. Thus the BS modifies its encoding scheme to accommodate new users so that existing users continue to operate as before: they achieve the same rates as before and they decode their signal in precisely the same way as before. The solutions for the uplink and the downlink are particularly interesting in the way they exhibit a remarkable simplicity and an unmistakable, near-perfect, up-down symmetry.

D-BLAST OFDM with Channel Estimation

Jianxuan Du and Ye (Geoffrey) Li

http://dx.doi.org/10.1155/S1110865704311054

Multiple-input and multiple-output (MIMO) systems formed by multiple transmit and receive antennas can improve performance and increase capacity of wireless communication systems. Diagonal Bell Laboratories Layered Space-Time (D-BLAST) structure offers a low-complexity solution for realizing the attractive capacity of MIMO systems. However, for broadband wireless communications, channel is frequency-selective and orthogonal frequency division multiplexing (OFDM) has to be used with MIMO techniques to reduce system complexity. In this paper, we investigate D-BLAST for MIMO-OFDM systems. We develop a layerwise channel estimation algorithm which is robust to channel variation by exploiting the characteristic of the D-BLAST structure. Further improvement is made by subspace tracking to considerably reduce the error floor. Simulation results show that the layerwise estimators require 1 dB less signal-to-noise ratio (SNR) than the traditional blockwise estimator for a word error rate (WER) of 10^{-2} when Doppler frequency is 40 Hz. Among the layerwise estimators, the subspace-tracking estimator provides a 0.8 dB gain for 10^{-2} WER with 200 Hz Doppler frequency compared with the DFT-based estimator.

Timing-Free Blind Multiuser Detection for Multicarrier DS/CDMA Systems with Multiple Antennae

Stefano Buzzi, Emanuele Grossi, and Marco Lops

http://dx.doi.org/10.1155/S1110865704310127

The problem of blind multiuser detection for an asynchronous multicarrier DS-CDMA system employing multiple transmit and receive antennae over a Rayleigh fading channel is considered in this paper. The solutions that we develop require prior knowledge of the spreading code of the user to be decoded only, while no further information either on the user to be decoded or on the other active users is required. Several combining rules for the observables at the output of each receive antenna are proposed and assessed, and the implications of the different options are studied in depth in terms of both detection performance and computational complexity. A closed form expression is also derived for the conditional error probability and a lower bound for the near-far resistance is provided. Results confirm that the proposed blind receivers can cope with both multiple access interference suppression and channel estimation at the price of a limited performance loss as compared to the ideal linear receivers which assume perfect channel state information.

Signal Reception for Space-Time Differentially Encoded Transmissions over FIR Rich Multipath Channels

Zhan Zhang and Jacek Ilow

http://dx.doi.org/10.1155/S1110865704312229

With sophisticated signal and information processing algorithms, air interfaces with space-time (ST) coding and multiple reception antennas substantially improve the reliability of wireless links. This paper proposes a new receiver algorithm for differential ST coded transmissions over the finite-impulse-response (FIR) rich multipath fading channels. The symbol detection introduced in this paper is a deterministic subspace-based approach in a multiple-input and multiple-output (MIMO) system framework. The receiver (i) operates in a blind fashion without estimating the channel or its inverse and (ii) is able to work with a small number of signal samples and hence can be applied in the quasistatic channels. The proposed scheme employs multiple antennas at both sides of the transceiver and exploits both the antenna diversity and the multiple constant modulus (MCM) characteristics of the signaling. The receiver is able to blindly mitigate the intersymbol interference (ISI) in a rich multipath propagation environment, and this has been verified through the extensive Monte Carlo simulations.

Improved Multiuser Detectors Employing Genetic Algorithms in a Space-Time Block Coded System

Yinggang Du and Kam Tai Chan

http://dx.doi.org/10.1155/S1110865704312163

Enhanced genetic algorithms (GA) applied in space-time block coded (STBC) multiuser detection (MUD) systems in Rayleigh flat-fading channels are reported in this paper. Firstly, an improved objective function, which is designed to help speed up the search for the optimal solution, is introduced. Secondly, a decorrelating detector (DD) and a minimum mean square error (MMSE) detector have been added to the GA STBC MUD receiver to create the seed chromosome in the initial population. This operation has improved the receiver performance further because some signal information has been intentionally embedded in the initial population. Simulation results show that the receiver employing the improved objective function and the DD or MMSE detector can converge faster with the same bit error rate (BER) performance than the receiver with the initial population chosen randomly. The total signal-to-noise ratio (SNR) improvement contributed by these two modifications can reach 4 dB. Hence the proposed GA receiver is a promising solution of the STBC MUD problem.

Performance Comparisons of MIMO Techniques with Application to WCDMA Systems

Chuxiang Li and Xiaodong Wang

http://dx.doi.org/10.1155/S1110865704309029

Multiple-input multiple-output (MIMO) communication techniques have received great attention and gained significant development in recent years. In this paper, we analyze and compare the performances of different MIMO techniques. In particular, we compare the performance of three MIMO methods, namely, BLAST, STBC, and linear precoding/decoding. We provide both an analytical performance analysis in terms of the average receiver SNR and simulation results in terms of the BER. Moreover, the applications of MIMO techniques in WCDMA systems are also considered in this study. Specifically, a subspace tracking algorithm and a quantized feedback scheme are introduced into the system to simplify implementation of the beamforming scheme. It is seen that the BLAST scheme can achieve the best performance in the high data rate transmission scenario; the beamforming scheme has better performance than the STBC strategies in the diversity transmission scenario; and the beamforming scheme can be effectively realized in WCDMA systems employing the subspace tracking and the quantized feedback approach.

Generalized Alamouti Codes for Trading Quality of Service against Data Rate in MIMO UMTS

Christoph F. Mecklenbräuker and Markus Rupp

http://dx.doi.org/10.1155/S1110865704310061

New space-time block coding schemes for multiple transmit and receive antennas are proposed. First, the well-known Alamouti scheme is extended to $N_T = 2^m$ transmit antennas achieving high transmit diversity. Many receiver details are worked out for four and eight transmit antennas. Further, solutions for arbitrary, even numbers ($N_T = 2k$) of transmit antennas are presented achieving decoding advantages due to orthogonalization properties while preserving high diversity. In a final step, such extended Alamouti and BLAST schemes are combined, offering a continuous trade-off between quality of service (QoS) and data rate. Due to the simplicity of the coding schemes, they are very well suited to operate under UMTS with only very moderate modifications in the existing standard. The number of supported antennas at transmitter alone is a sufficient knowledge to select the most appropriate scheme. While the proposed schemes are motivated by utilization in UMTS, they are not restricted to this standard.

Maximum MIMO System Mutual Information with Antenna Selection and Interference

Rick S. Blum

http://dx.doi.org/10.1155/S1110865704311170

Maximum system mutual information is considered for a group of interfering users employing single user detection and antenna selection of multiple transmit and receive antennas for flat Rayleigh fading channels with independent fading coefficients for each path. In the case considered, the only feedback of channel state information to the transmitter is that required for antenna selection, but channel state information is assumed at the receiver. The focus is on extreme cases with very weak interference or very strong interference. It is shown that the optimum signaling covariance matrix is sometimes different from the standard scaled identity matrix. In fact, this is true even for cases without interference if SNR is sufficiently weak. Further, the scaled identity matrix is actually that covariance matrix that yields worst performance if the interference is sufficiently strong.

Channel Estimation and Data Detection for MIMO Systems under Spatially and Temporally Colored Interference

Yi Song and Steven D. Blostein

http://dx.doi.org/10.1155/S1110865704311194

The impact of interference on multiple-input multiple-output (MIMO) systems has recently attracted interest. Most studies of channel estimation and data detection for MIMO systems consider spatially and temporally white interference at the receiver. In this paper, we address channel estimation, interference correlation estimation, and data detection for MIMO systems under both spatially and temporally colored interference. We examine the

case of one dominant interferer in which the data rate of the desired user could be the same as or a multiple of that of the interferer. Assuming known temporal interference correlation as a benchmark, we derive maximum likelihood (ML) estimates of the channel matrix and spatial interference correlation matrix, and apply these estimates to a generalized version of the Bell Labs Layered Space-Time (BLAST) ordered data detection algorithm. We then investigate the performance loss by not exploiting interference correlation. For a (5, 5) MIMO system undergoing independent Rayleigh fading, we observe that exploiting both spatial and temporal interference correlation in channel estimation and data detection results in potential gains of 1.5 dB and 4 dB for an interferer operating at the same data rate and at half the data rate, respectively. Ignoring temporal correlation, it is found that spatial correlation accounts for about 1 dB of this gain.

A MIMO System with Backward Compatibility for OFDM-Based WLANs

Jianhua Liu and Jian Li

http://dx.doi.org/10.1155/S111086570431022X

Orthogonal frequency division multiplexing (OFDM) has been selected as the basis for the new IEEE 802.11a standard for high-speed wireless local area networks (WLANs). We consider doubling the transmission data rate of the IEEE 802.11a system by using two transmit and two receive antennas. We propose a preamble design for this multi-input multi-output (MIMO) system that is backward compatible with its single-input single-output (SISO) counterpart as specified by the IEEE 802.11a standard. Based on this preamble design, we devise a sequential method for the estimation of the carrier frequency offset (CFO), symbol timing, and MIMO channel response. We also provide a simple soft detector based on the unstructured least square approach to obtain the soft information for the Viterbi decoder. This soft detector is very simple since it decouples the multidimensional QAM symbol detection into multiple one-dimensional QAM symbol—and further PAM symbol—detections. Both the sequential parameter estimation method and the soft detector can provide excellent overall system performance and are ideally suited for real-time implementations. The effectiveness of our methods is demonstrated via numerical examples.

High Capacity Downlink Transmission with MIMO Interference Subspace Rejection in Multicellular CDMA Networks

Henrik Hansen, Sofiène Affes, and Paul Mermelstein

http://dx.doi.org/10.1155/S111086570431111X

We proposed recently a new technique for multiuser detection in CDMA networks, denoted by interference subspace rejection (ISR), and evaluated its performance on the uplink. This paper extends its application to the downlink (DL). On the DL, the information about the interference is sparse, for example, spreading factor (SF) and modulation of interferers may not be known, which makes the task much more challenging. We present three new ISR variants which require no prior knowledge of interfering users. The new solutions are applicable to MIMO systems and can accommodate any modulation, coding, SF, and connection

type. We propose a new code allocation scheme denoted by DACCA which significantly reduces the complexity of our solution at the receiving mobile. We present estimates of user capacities and data rates attainable under practically reasonable conditions regarding interferences identified and suppressed in a multicellular interference-limited system. We show that the system capacity increases linearly with the number of antennas despite the existence of interference. Our new DL multiuser receiver consistently provides an Erlang capacity gain of at least 3 dB over the single-user detector.

Maximum Likelihood Turbo Iterative Channel Estimation for Space-Time Coded Systems and Its Application to Radio Transmission in Subway Tunnels

Miguel González-López, Joaquín Míguez, and Luis Castedo

http://dx.doi.org/10.1155/S1110865704311066

This paper presents a novel channel estimation technique for space-time coded (STC) systems. It is based on applying the maximum likelihood (ML) principle not only over a known pilot sequence but also over the unknown symbols in a data frame. The resulting channel estimator gathers both the deterministic information corresponding to the pilot sequence and the statistical information, in terms of a posteriori probabilities, about the unknown symbols. The method is suitable for Turbo equalization schemes where those probabilities are computed with more and more precision at each iteration. Since the ML channel estimation problem does not have a closed-form solution, we employ the expectation-maximization (EM) algorithm in order to iteratively compute the ML estimate. The proposed channel estimator is first derived for a general time-dispersive MIMO channel and then is particularized to a realistic scenario consisting of a transmission system based on the global system mobile (GSM) standard performing in a subway tunnel. In this latter case, the channel is nondispersive but there exists controlled ISI introduced by the Gaussian minimum shift keying (GMSK) modulation format used in GSM. We demonstrate, using experimentally measured channels, that the training sequence length can be reduced from 26 bits as in the GSM standard to only 5 bits, thus achieving a 14% improvement in system throughput.

Space-Time Chip Equalization for Maximum Diversity Space-Time Block Coded DS-CDMA Downlink Transmission

Geert Leus, Frederik Petré, and Marc Moonen

http://dx.doi.org/10.1155/S1110865704309297

In the downlink of DS-CDMA, frequency-selectivity destroys the orthogonality of the user signals and introduces multiuser interference (MUI). Space-time chip equalization is an efficient tool to restore the orthogonality of the user signals and suppress the MUI. Furthermore, multiple-input multiple-output (MIMO) communication techniques can result in a significant increase in capacity. This paper focuses on space-time block coding (STBC) techniques, and aims at combining STBC techniques with the original single-antenna DS-CDMA downlink scheme. This results into the so-called space-time block coded DS-CDMA downlink schemes, many of which have been presented in the past. We focus on a new

scheme that enables both the maximum multiantenna diversity and the maximum multipath diversity. Although this maximum diversity can only be collected by maximum likelihood (ML) detection, we pursue suboptimal detection by means of space-time chip equalization, which lowers the computational complexity significantly. To design the space-time chip equalizers, we also propose efficient pilot-based methods. Simulation results show improved performance over the space-time RAKE receiver for the space-time block coded DS-CDMA downlink schemes that have been proposed for the UMTS and IS-2000 W-CDMA standards.

Joint Power Control and Blind Beamforming over Wireless Networks: A Cross Layer Approach

Zhu Han, Farrokh R. Farrokhi, and K. J. Ray Liu

http://dx.doi.org/10.1155/S1110865704311030

Traditional joint power control and beamforming achieve the targeted signal-to-interference-noise ratio (SINR) at the receivers by assuming the knowledge of the measurements of channel parameters and SINR. Blind beamforming is an effective technique for beamforming and channel estimation without the need of training sequences, thus not consuming extra bandwidth. In this paper, we propose a novel joint power control and blind beamforming algorithm that reformulates the power control problem in such a way that it does not need any prior knowledge and additional measurements in the physical layer. In contrast to the traditional schemes that optimize SINR and, as a result, minimize bit error rate (BER), our proposed algorithm achieves the desired BER by adjusting a quantity available from blind beamforming. By sending this quantity to the transmitter through a feedback channel, the transmit power is iteratively updated in a distributed manner in the wireless networks with cochannel interferences (CCIS). Our proposed algorithm is more robust to estimation errors. We have shown in both analysis and simulation that our algorithm converges to the desired solution. In addition, a Cramer-Rao lower bound (CRB) is derived to compare with the performance of our proposed joint power control and blind beamforming system.

Approaching the MIMO Capacity with a Low-Rate Feedback Channel in V-BLAST

Seong Taek Chung, Angel Lozano, Howard C. Huang, Arak Sutivong, and John M. Cioffi

http://dx.doi.org/10.1155/S1110865704312035

This paper presents an extension of the vertical Bell Laboratories Layered Space-Time (V-BLAST) architecture in which the closed-loop multiple-input multiple-output (MIMO) capacity can be approached with conventional scalar coding, optimum successive decoding (OSD), and independent rate assignments for each transmit antenna. This theoretical framework is used as a basis for the proposed algorithms whereby rate and power information for each transmit antenna is acquired via a low-rate feedback channel. We propose the successive quantization with power control (SQPC) and successive rate and power quantization (SRPQ) algorithms. In SQPC, rate quantization is performed with continuous power

control. This performs better than simply quantizing the rates without power control. A more practical implementation of SQPC is SRPQ, in which both rate and power levels are quantized. The performance loss due to power quantization is insignificant when 4–5 bits are used per antenna. Both SQPC and SRPQ show an average total rate close to the closed-loop MIMO capacity if a capacity-approaching scalar code is used per antenna.

Multiple ARQ Processes for MIMO Systems

Haitao Zheng, Angel Lozano, and Mohamed Haleem

http://dx.doi.org/10.1155/S1110865704311042

We propose a new automatic repeat request (ARQ) scheme for MIMO systems with multiple transmit and receive antennas. The substreams emitted from various transmit antennas encounter distinct propagation channels and thus have different error statistics. When per-antenna encoders are used, separating ARQ processes among the substreams results in a throughput improvement. Moreover, it facilitates the interference cancellation in certain MIMO techniques. Quantitative results from UMTS simulations demonstrate that the proposed multiple ARQ structure yields more than 30% gain in link throughput.

EURASIP JASP FORTHCOMING SPECIAL ISSUES

Object-Based and Semantic Image and Video Analysis

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

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

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

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.

Model-Based Sound Synthesis

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

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

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

Cross-Layer Design for Communications and Signal Processing Systems

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

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.

Special Issue on

DSP in Hearing Aids and Cochlear Implants

CALL FOR PAPERS

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.

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

- Feedback cancellation
- Noise reduction
- Source separation
- Adaptive directionality systems
- Speech detection and recognition
- Auditory scene analysis
- Binaural signal processing
- Filterbanks and compression
- CI stimulation strategies
- Psychoacoustically motivated signal processing algorithms
- DSP architectures, complexity, parallel implementation

Authors should follow the EURASIP JASP manuscript format described at the journal site http://asp.hindawi.com/. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP JASP's web submission system at http://www.mstracking.com/asp/, according to the following timetable.

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Special Issue on

Applications of Signal Processing in Astrophysics and Cosmology

CALL FOR PAPERS

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.

The submission of each paper will be assigned to at least one referee of signal processing background and to at least one referee of astrophysics background. In the choice of the accepted papers, the associate editors necessitate the novelty in the astrophysics application and rigour in the signal processing terminology and methods. Special preference will be given to articles which present novel signal processing techniques fuelled by astrophysical problems.

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

- Signal separation in astrophysical radiation maps
- Parametric modelling of astrophysical data
- Astrophysical time series analysis and spectral analysis
- Feature extraction and pattern recognition in astrophysical images
- Compression of astrophysical data
- Noise filtering
- Restoration of astrophysical images
- Transform domain analysis of astrophysical data
- Nonlinear techniques
- Data mining in astrophysical data bases

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Special Issue on

Advances in Interferometric Synthetic Aperture Radar Processing

CALL FOR PAPERS

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.

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

- Spatial/spatial-temporal phase unwrapping
- Multibaseline/multifrequency unwrapping
- Multipass differential interferometry
- Polarimetric interferometry
- Multibaseline/multifrequency/multi-incidence angle model inversion
- 3D SAR tomography
- Multibaseline/multifrequency along-track interferometry
- Joint along-track cross-track interferometry
- Multiplatform interferometry

- Wideband interferometry
- Ground-based interferometry
- Multistatic interferometry

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Special Issue on

Analysis/Synthesis Methods for Wired and Wireless Multimedia On-Demand Applications

CALL FOR PAPERS

Multimedia on-demand applications offer diverse services and content, and promise to be a lasting success in professional and consumer markets. The impact, however, of wired and especially wireless versions of these applications has so far been limited due to lack of bandwidth, problematic quality of service, high cost of media production and transmission, and the constrained resources of user devices. Compact data representation, replacement, and reconstruction at desired quality levels by analysis/synthesis methods can address these limitations by providing solutions for improving compression efficiency, increasing content quality, and adapting the content for various platforms.

In the context of typical multimedia on-demand scenarios (encode content once, transmit and decode many times), analysis/synthesis methods involve several or all of the following components: content analysis, relevancy analysis, partial content removal and/or replacement, reconstruction of the replaced content by synthesis, augmentation of the content by synthesis (e.g., video background analysis, replacement, and synthesis; audio and video advertisements analysis, replacement, and synthesis; synthesis of personalized footage).

This special issue focuses on such analysis/synthesis methods for multimedia on-demand services. Original contributions are invited on the following:

- Methods for mono/multimedia analysis and mono/ multimedia synthesis (e.g., audio analysis; joint audio-video analysis; image synthesis; joint animation and text synthesis; audio analysis for image synthesis; and video analysis for text and audio synthesis)
- Low level, intermediate level, and high level analysis methods (e.g., low level: texture, shape, color, and motion feature extraction; intermediate level: pitch, linear predictor coefficients extraction; high level: semantic analysis, relevancy analysis)
- Procedural, nonprocedural, and hybrid synthesis methods (e.g., procedural: specialized emulators for classes of texture, shape, and sound; nonprocedural: statistical sampling synthesis methods)
- Fixed/variable (ontogenetic) architectures for analysis and synthesis methods (e.g., growing structures, shrinking structures)

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

- Augmentation for multimedia on-demand
- Compression for multimedia on-demand
- Low-complexity spatial and temporal scalability
- Error concealment for multimedia on-demand
- Audio-in-video and video-in-video data hiding
- Latency reduction in multicast on-demand systems
- Efficient use of available quality-of-service guarantees

Authors should follow the EURASIP JASP manuscript format described at the journal site http://asp.hindawi.com/. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP JASP's manuscript tracking system at http://www.mstracking.com/ asp/, according to the following timetable:

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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

Authors should follow the EURASIP JASP manuscript format described at the journal site http://asp.hindawi.com/. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP JASP manuscript tracking system at http://www.mstracking.com/asp/, according to the following timetable:

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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

Authors should follow the EURASIP JASP manuscript format described at the journal site http://asp.hindawi.com/. Prospective authors should submit an electronic copy of their complete manuscript through the EURASIP JASP manuscript tracking system at http://www.mstracking.com/asp/, according to the following timetable:

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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.

Special Issue on

Ad Hoc Networks: Cross-Layer Issues

CALL FOR PAPERS

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.

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

- Energy-aware protocols
- Power control
- Topology management
- Self-organization algorithms
- Localization and neighbour discovery techniques
- Mobility management
- Routing strategies
- Interdependencies between transport, network, and MAC protocols
- Error control schemes
- Scheduling
- Synchronization

- Physical layer issues
- Smart antenna beamforming
- Security in ad hoc networks

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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
- Handover and QoS negotiation
- Link adaptation
- Dynamic resource allocation
- Dynamic frequency allocation
- Channel adaptive networking and routing

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Genomic Signal Processing and Statistics

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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