EURASIP

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European Association
for Signal, Speech,
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Contents

CALLS FOR PAPERS

EUSIPCO Periodicity

In the last months, the EURASIP AdCom has discussed the pros and cons of having our major conference, EUSIPCO, organized on an annual basis, instead of biennially. It is a common feeling that, currently, there already exists a myriad of conferences and workshops dealing with EURASIP related topics. Furthermore, there is always a conservative viewpoint that pushes us to think "if it works, better don't touch it" and biennial EUSIPCOs have worked extremely well since their very beginning, back in 1980.

On the other hand, various facts show us that having an annual event would be possible and positive for the association. First of all, the AdCom has been contacted by several members asking for an annual EUSIPCO. The success of the so-called EURASIP miniconferences that take place in central-east European locations on the odd years between EUSIPCOs helps us to think that a yearly event is feasible. We believe that the success of previous EUSIPCO does not depend on its biannual nature but on the interest of our community for the event. Second, nowadays, the number of bids we have received for hosting future EUSIPCOs allows us to complete a calendar until 2010, already assuming an annual event! Finally, we believe that a major annual event will reinforce the visibility of EURASIP, the links among their members, and their awareness of the association evolution.

After analyzing different possibilities, the AdCom has decided to go for a yearly EUSIPCO. With this new periodicity, EUSIPCO will absorb the EURASIP miniconferences. This way, after EUSIPCO 2004 in Vienna, we have already agreed that EUSIPCO 2005 will be in Istanbul (thanks to Professor Bulent Sankur and his committee for their very fast and positive answer). Moreover, we are now in the process of contacting the chairmen of the various EUSIPCO proposals we have already received to establish an annual EUSIPCO calendar. Very likely, one of the near future EUSIPCO locations will be in a central-east European country.

It is clear to the AdCom that there is some risk in this decision and that the final success of this project does not only depend on us. It will strongly depend on the work of future EUSIPCO organizing committees and, very specially, on the support that our research community will give to an annual EUSIPCO. The AdCom believes that the project is paramount for the future of EURASIP and, therefore, we have committed on giving it the maximum possible support.

Ferran Marqués
EURASIP President

Calendar of Events

Year	Date	Event	Location	EURASIP Involvement	Chairperson/Information
2003	December 11–12	ISCA Workshop on Multimodal User Authentication	Santa Barbara, USA	Cooperation	Jean-Luc Dugelay authentication@research.panasonic.com
2004	April 21–23	5th International Workshop on Image Analysis for Multimedia Interactive Services (WIAMIS 2004)	Lisboa, Portugal	Cooperation	F. Pereira http://www.img.lx.it.pt/WIAMIS2004/
	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 CSNDSP, Int. Symposium on Communication Systems, Networks and DSP	Newcastle, UK	Cooperation	T. Boukouvalas http://www.shu.ac.uk/ocr/csndsp/
	September 7–10	12th European Signal Processing Conference (EUSIPCO-2004)	Vienna, Austria	Sponsor	W. Mecklenbräuker http://www.nt.tuwien.ac.at/eusipco2004/
	September 13–15	11th Intl. Workshop on Systems, Signals and Image Ambient Multimedia Processing (IWSSIP-2004)	Poznan, Poland	Cooperation	M. Domanski http://iwssip2004.et.put.poznan.pl

Sergios Theodoridis
Workshops/Confs Coordinator EURASIP

Report on ISSPA 2003, Paris, July 2003

The 7th meeting, organized by the L2TI laboratory of the University of Paris 13 and the TSI laboratory of the ENST-Paris in collaboration with the SPRC laboratory of the Queensland University of Technology, took place the 1st, 2nd, 3rd, and 4th of July 2003 at “Les Salons de l’Aveyron” in Paris.

1. Background and general conference

With this 7th meeting, ISSPA further established itself as a premier conference dedicated to signal and image processing research and applications. It was initiated in 1985 to promote the signal processing research activities in the Asia-Pacific region and to strengthen the collaboration between the Asia-Pacific countries and the rest of the world. The first meeting was held in 1987 in Brisbane, Queensland, Australia. Subsequent meetings in 1990, 1992, 1996, and 1999 were also held in the state of Queensland, Australia. By then ISSPA had established an international character and reputation; hence the steering committee decided to organize the following ISSPA meeting in another country. In August 2001, the 7th ISSPA meeting took place in Kuala Lumpur, Malaysia; it was the first time an ISSPA meeting was held outside Australia. Following the very successful meeting of Kuala Lumpur in 2001, the steering committee decided to entertain bids from other countries for the next meeting. Six countries were considered, namely, Australia, Canada, France, Egypt, Malaysia, and Turkey. A vote by the participants confirmed Paris as the clear favorite to hold the 2003 meeting.

Every two years, ISSPA gathers researchers and engineers from universities, governmental laboratories, big industrial groups, and businesses, involved in the development of new applications and services relying on signal processing techniques. Through ISSPA, the organizers aim to offer a unique forum for a targeted international audience of 250 to 350 participants, to facilitate and encourage close interactions, collaborations, and exchanges between researchers. With the long-time support of the IEEE Society, and more recently of the EURASIP society, ISSPA has succeeded in bringing together researchers from the Asia Pacific region, Europe, and America for almost two decades now, all along valuing the cultural and scientific diversity of both hosts and participants.

2. Technical program

A major innovation of ISSPA 2003 was the use of the blind review process for the submitted papers. This process worked very well to ensure that papers were reviewed fairly. This decision was not without difficulty because of the large number of submissions (603 papers). ISSPA 2003 has set a new record in terms of submitted papers and participant countries (over 54 countries). Despite the large number of submissions, we have accommodated 337 papers (56%) for the final program so as to preserve the essential character of ISSPA, where the interaction and exchange of ideas are facilitated by the human size of the symposium. We warmly express our acknowledgement to the program committee members and reviewers for their diligence in reviewing the papers.

The following plenary talks were given to and well received by a full audience.

1. Tentative title *MUVIS: EU COST 211quat framework for content-based indexing and retrieval for multimedia databases*, by Professor Moncef Gabbouj from Tampere University of Technology, Tampere, Finland.
2. Tentative title *Microarrays and genomic signal processing*, by Professor Al Hero from the University of Michigan, Ann Arbor, USA.
3. Tentative title *Large random matrices and performance of large CDMA systems*, by Professor Philippe Loubaton from Université de Marne la Vallée, France.

ISSPA 2003 program also included 12 special sessions organized by well-recognized experts in different key and hot topics of signal and image processing as well as a student session to encourage Ph.D. and Masters students to publish and describe the main results of their respective research work in a two-pages length paper. In recognition of the SARS effect on the signal processing community, which resulted in ICASSP 2003 being canceled, ISSPA has accepted a number of papers from ICASSP 2003 to be presented at ISSPA 2003.

3. Tutorial sessions

The tutorial program, organized by S. Bouzerdoum (Tutorial Chair) from ECU, Australia, attracted a number of proposals from different parts of the world: USA, Canada, France, Algeria, Finland, Japan, Singapore, and Australia. After a careful review of the submitted proposals, ten tutorials were initially accepted in the final program, then further reduced to four after taking account of participants interests; these four are listed as follows:

1. *Genomic signal processing*, by Ioan Tabus, Tampere University of Technology, Tampere, Finland.
2. *Existing and emerging image and video coding standards based on wavelets*, by Professor Beatrice Pesquet-Popescu, Ecole Nationale Supérieure des Télécommunications (ENST), and Stéphane Pateux, IRISA/INRIA, Rennes, France.
3. *Introduction to directional filter banks and wavelets with application to image analysis, detection, tracking, enhancement, and compression*, by Professor Mark J. T. Smith, Purdue University, USA.
4. *Neural networks for signal processing and communications*, by Professor Mohamed Ibnkahla, Queen's University, Kingston, Canada.

The final program covered a broad range of signal and image processing areas, ranging from genomic signal processing to image and video coding based on wavelets. The tutorials attracted great interests of conference attendees, and the presenters were highly commended for their efforts. Overall, the tutorial program was a success.

More details can be found at http://www-l2ti.univ-paris13.fr/~isspa2003/.

The next ISSPA meeting, ISSPA 2005, is scheduled to be held in Australia in 2005.

B. Boashash
Steering Committee Chair

A. Beghdadi
Symposium Chair

K. Abed-Meraim
Technical Chair

Report of the 4th EURASIP Conference Focused on Video/Image Processing and Multimedia Communications (EC-VIP-MC 2003)

The 4th EURASIP Conference focused on Video/Image Processing and Multimedia Communications (EC-VIP-MC 2003) was organised by the Department of Radiocommunications and Microwave Engineering, Faculty of Electrical Engineering and Computing, University of Zagreb, Croatia, during the 2nd, 3rd, 4th, and 5th of July 2003.

The European Association for Speech, Signal and Image Processing (EURASIP) initiated this type of conference in central European locations (on the odd years between the EUSIPCO conferences) in order to start a new tradition of conferences, each devoted to a specific area of discipline. The first conference was Signal Analysis and Prediction conference organised in Prague, Czech Republic (1997), the second was Digital Signal Processing for Multimedia Communications and Services conference organized in Krakow, Poland (1999), and the third in Budapest, Hungary (2001).

The 4th EURASIP Conference focused on Video/Image Processing and Multimedia Communications was sponsored by the EURASIP and cosponsored by IEEE Region 8, IEEE Croatia Section, University of Zagreb, and Faculty of Electrical Engineering and Computing, University of Zagreb. The International Program Committee of the EC-VIP-MC 2003 consists of 66 members who are outstanding researchers and professors from different universities and companies.

The EC-VIP-MC 2003 conference programme consists of three keynote talks, one invited lecture, and 19 sessions, where 131 papers written by 277 authors were presented. The authors of the papers presented in the EC-VIP-MC 2003 conference are prominent researchers coming from the following 41 countries: Australia, Austria, Belgium, Brazil, Bulgaria, Canada, China, Croatia, Czech Republic, Finland, France, Germany, Greece, Hungary, India, Iran, Ireland, Israel, Italy, Japan, Korea, Latvia, Lebanon, Macedonia, Mexico, New Zealand, Pakistan, Poland, Portugal, Romania, Serbia and Montenegro, Singapore, Slovak Republic, Slovenia, Spain, Sweden, Taiwan, Turkey, United Arab Emirates, United Kingdom, and United States of America.

The first keynote talk was given by Dr. Danilo P. Manic from the Imperial College London, United Kingdom, who replaced Professor Anthony G. Constantinides (First President of the EURASIP 1978–1980). The title of the talk was *On Contours, Corners and Cats: The Human Visual System and Image Processing*. The second keynote talk was given by Professor Touradj Ebrahimi from the EPFL, Switzerland. The title of the talk was *Novel Frontiers in Multimedia Communications - New Media and Interfaces*. The third keynote talk was given by Professor K. R. Rao (one of the founders of the Discrete Cosine Transform in 1975) from the University of Texas at Arlington, USA. The title of the talk was *H.264/MPEG-4 Part 10 Advanced Video Coding*. The invited lecture was given by Professor Jiri Jan (EURASIP Administrative Committee Nominated Officer for Central Europe) from the Brno University of Technology, Czech Republic. The title of the lecture was *Signal and Image Data Processing in Ultrasonic Imaging*.

The EC-VIP-MC 2003 conference provided all participants with the latest results of research, development, and application of methods and techniques in the rapidly progressing field of the video/image processing and multimedia communications. More information about the EC-VIP-MC 2003 conference, participants feedback, photos, and so forth, can be found at http://www.vcl.fer.hr/ec2003/.

Next EURASIP conference of this type, 5th EURASIP Conference, will focus on Speech and Image Processing, Multimedia Communications and Services and will be held from 29 June to 2 July 2005 in the beautiful Smolenice Castle, Slovak Republic. More information is available at http://www.ktl.elf.stuba.sk/ec2005/.

Mislav Grgic
EC-VIP-MC 2003
Program Co-Chair

Exploration of Visual Data

Xiang Sean Zhou, Yong Rui, Thomas S. Huang
Kluwer Academic Publishers, 2003, 187 pp, ISBN 1-4020-7569-3

Irek Defée

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Visual information processing is increasingly becoming widespread as multimedia becomes common in everyday life. Vision is an essential sense for humans and while its capabilities are prodigious, they are accompanied by deception of blitz and effortless operation. Research into visual processing revealed hidden complexities, and despite long and extensive effort, there are many basic problems which are not solved. In their suitably titled book *Exploration of Visual Data*, Zhou et al. begin with the list of challenges and state that "the excitement of the early successes has worn off." In the state-of-the-art section they give concise but highly informative, outline of representative problems and related literature references. The book itself can be characterized as a monograph of authors own solutions put on the background of state of the art. The books is based on earlier research in challenging problems in the processing of visual data.

Chapter 2 of the book provides an overview of visual information representation methods based on classical features like color, texture, and shape plus less common ones based on spatial layout and interest points. Finally, the problem of image segmentation is briefly outlined. These topics are covered very economically with relevant collection of references.

In Chapter 3, the authors begin with their own original contributions. They propose a new set of structural features for describing content of images. The features are based on edges and use simple intuitive concept of water-filling of edge structure which literally corresponds to water-flow throughout edge lines with forking and looping. Based on these concepts, primitives are defined such as water filling time, fork count, and loop count. These in turn are used to construct structural primitives such as maximum filling time and fork count, these measures should normally correspond to salient object in the image. In similar way, one can define many other features such as maximum fork count, filling time and fork histograms, and so forth. The most important aspect of these features is their experimental validity and this is illustrated extensively by comparison with wavelets variances. It is shown that in many cases water-filling features perform better. To the authors credit, one should add that both "good" and "bad" examples are provided. Final conclusion is that water-filling features perform well on images with strong edge structure while they are bad where edges do not contribute much for perceptual meaning.

In Chapter 4, probabilistic local structure model is proposed. This model is based on multidimensional histograms of salient points obtained by applying edge detection and the so-called invariant local jets. The histograms are factorized using independent component analysis and several first components are used. The technique relies on some additional details, experimental results shown indicate that it is robust for object detection and image retrieval under occlusion and rotation. Comparison shows that results obtained are better than those obtained with models based on global features.

In Chapter 5, the authors start shifting attention to more practically oriented topics. First of them is automatic construction of table of content (TOC) for video. This is a very difficult problem as it ultimately depends on high-level semantic knowledge. But the authors propose several algorithms for solving it basing on the detection of video shots, group of shots, and scenes. This is based on detecting similarities of color histograms and grouping similar shots closely located in time. On top of these, scene structure is constructed by forming initial groups of shots and assigning video shots to them based on similarities. Algorithms are presented to show how to do this and the process of determination of grouping thresholds and other parameters is outlined. Results of experimental evaluations presented show that in most cases proper TOC is constructed as compared to the TOC produced by humans. This is very interesting since the algorithms do not use any semantics and are based only on simple histograms.

Chapter 6 is devoted to nonlinear sampling of video for streaming applications. The idea is to sample key frames for video which could be used in synchronized ‘slide-show’ type presentation with audio. This would provide enough information about the content with minimum resources. Key-frame selection is challenging since it ultimately correspond to minimal number of frames conveying semantics of the content. The authors’ proposal for solving this problem is to start from detecting shot boundaries by saliency scores based on a combination of color and motion. Next the total number of frames is optimized taking into account different criteria like buffer size and network bandwidth. In the end, experimental evaluation shows that performance is adequate for low bit rate communication systems.

Chapter 7 is central to the book. It deals with relevance feedback for visual retrieval, which means learning classification from human user. Relevance feedback is an answer to the algorithmic intractability of high-level semantic concepts by learning from human operator performing retrieval tasks. The hope is that a system can emulate the performance of human operator if suitable machine learning procedure is used on a large set of training examples. In this chapter, previous approaches are well referred and a new method based on them is proposed. In the method, three features are used for the construction of feature vectors: color moments, wavelet texture, and water-filling edges. Generalized Euclidean distance is used as dissimilarity measure between linear combination of feature vectors. A procedure is presented for finding transformation which will optimize distances between feature vectors. Several other procedures are also outlined. Experimental tests and comparisons with other methods show good performance and significant improvement of classification with feedback relevance.

Last chapter of the book, Chapter 8, concerns the unification of keywords with low-level contents. Keywords have relation to high-level semantics, so unifying them with features makes visual retrieval based on semantics possible. This is proposed to be done using relevance feedback based on keywords for generating semantic classes on image database. Concept similarity matrices are built for associating relevant keywords and images. Then, neural network is used for producing semantic classes. Examples show that this is a viable approach.

This book touches problems on the forefront of multimedia research revolving around semantics of perceptual information. Confronting with prodigious capabilities of human perception shows that there is still a long way to go in this area. Current methods may seem ad hoc and simplistic but, on the other hand, approaches presented in the book show that usable results can be obtained with relatively straightforward tools based on simple computational features. Results obtained are illustrated on practical examples, and in many cases not only good but also problematic performance is recorded.

The book is highly advisable to researchers looking for reference and review sources plus inspiration on promising future research directions in multimedia. While the field is difficult and early excitement has gone, the authors prove that, with creativity, it is always possible to advance.

Digital Image Watermarking: An Overview

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1. Introduction

During the last years, the world is rapidly becoming digital in many aspects. Among the most interesting of these aspects is multimedia. Many new technologies have come into our lives along with multimedia and one of the most recent is watermarking. Digital technology has offered users easy ways to create, process, and distribute digital assets. However, these facilities this may become disadvantages at the same time since it is also much easier to illegally copy and manipulate them. In 2000, the Motion Picture Association of America (MPAA) has calculated lost revenues for American motion picture companies from worldwide piracy to be of the amount $2.5 billion a year [1]. Copyright protection is a very important subject but not the only one in which watermarking appears as one of the very promising solutions. In this communication, we are attempting to go back to the early beginning of this exciting new research field, make an overview of its history, discuss the applications and stakes involved, classify the proposed methods, present the problems and benchmarking tools and finally, try to peak into the future.

2. Definitions

But what exactly is the relationship between digital media and watermarks? one may wonder, bearing in mind the notion of watermarks from their analog counterparts. In the analog world, watermarks were used to prove authenticity of banknotes, books, artwork, and so forth. Actually, first paper watermarks date back to the 13th century. The move to the digital world is quite similar: some identification, copyright, or ownership data are embedded into the digital media in an imperceptible (usually) way. These data may identify the creator, the rightful owner, or perhaps the copyright holder. The applications of such a technology are numerous and quite important.

Although first related publications date as back as 1979, it was only in the mid 90s that researchers' attention has been drawn to the subject. The scientific community has recognized the potential of this new technology from the early works of Cox and Kilian [2, 3], Pitas and Kaskalis [4, 5], and others. From a humble number of roughly 220 papers in 1998, on the day of writing this paper, NEC's digital library [6] counts 738 documents for the word "watermark" while the Google search engine gives approximately 136,000 hits for the term "watermarking." There is a great forum for those working in the field [7] while the number of books on that subject, after the first two classics from Katzenbeisser and Petitcolas [8] in 2000 and Cox et al [9] in 2001, has increased until now to thirteen.

The science of hiding pieces of information into other information is called Steganography. There are references as old as Homer's "Iliad" [10] or Herodotus "Histories" [11], while more modern forms include invisible inks, microdots, and so forth. Watermarking is a form of steganography but the main difference is that while in steganography we are more interested in the capacity of the medium that conveys the secret information, in watermarking we also demand robustness. That means that even if someone is aware that the medium contains additional information, it would be impossible or at least extremely difficult for him to remove it without severely destroying the watermarked work.

3. Applications and requirements

As it was earlier mentioned, there are several applications for this new technology, and different kinds of watermarks are well suited to different applications. In general, watermarking applies to the following:

(i) *Intellectual property rights protection/management*. This is one of the most interesting applications of watermarking and of great economic importance for both companies and individuals. The idea is to give the creator/producer the ability to digitally "sign" his work and use his watermark as a piece of evidence in a court of law. It can also be used as a proof of rightful ownership or to ensure a legitimate transaction.

(ii) *Authentication*. As digital media can be easily maliciously manipulated, it is essential to provide means to authenticate them. This could simply mean a message from corresponding software declaring that the image has been tampered, or in a more sophisticated application, the self restoration of the original content of the media file.

(iii) *Broadcast monitoring*. Companies buy time from media networks to play their commercials. Traditionally, someone was watching the program to verify that the spot has been played at the arranged time. A computer system could examine media for watermarked commercials in a much more efficient way.

(iv) *Medical imagery*. It would be quite helpful to watermark medical images to avoid tampering, misuse, to embed data concerning the doctor or the patient, and so forth.

(v) *Illegal transaction identification*. By watermarking the medium with a characteristic serial number or other piece of information that identifies the parts that take part in the medium's transaction, it is possible to identify the sources of illegal copies and put them out of business.

(vi) *Copy control*. With the assistance of special engineered multimedia recording/reproduction/copy devices, the recording or copy of the copyrighted material as well as the playback of pirated media will be prevented.

These are some but not the only applications that exist for watermarking. One thing for sure is that we will certainly see more sophisticated and different kinds of applications in the future for such a rapidly evolving new technology.

Requirements are different depending on the application field. In general, some of the issues that watermarkers care about are stated as follows

(i) *Imperceptibility*. The watermark is embedded into the image to serve a specific purpose, neither to distract the viewer, nor to alter the image quality.

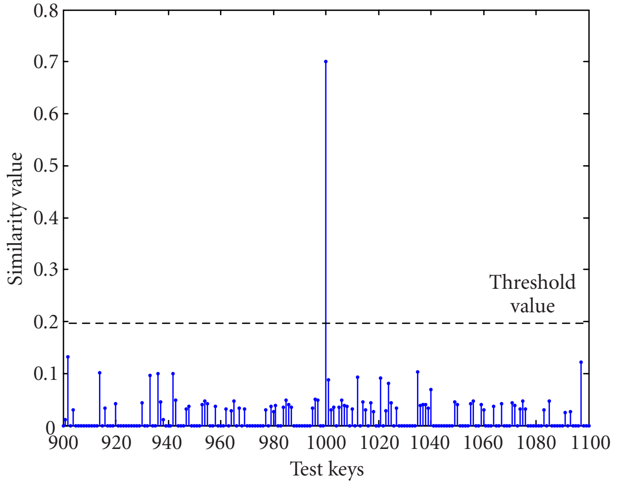

FIGURE 1: Typical detection diagram based on similarity metric. Testing 200 random watermarks produced by keys in the range 900–1100. For the key that was used to watermark the image (1000), the similarity axis exhibits a prominent peak, verifying the watermark's existence.

(ii) *Robustness*. The watermark should be able to survive accidental or malicious attempts of removal and common image processing tasks.

(iii) *Blind detection*. It means that the watermark should be retrieved without using the original image.

(iv) *Multiple casting*. It should be possible to embed without problems more than one watermark, for example, the creator's and the owner's.

(v) *Trustworthy retrieval*. There are two kinds of errors in the retrieval process. One is that the image is watermarked and we find out that it is not and the other is that the image is not watermarked and we say it is. There should be minimal possibilities of error for both cases.

(vi) *Payload* (or capacity). It concerns the amount (bits) of information the image is capable of storing.

(vii) *Unambiguity*. Retrieval of the watermark should unambiguously identify the owner.

4. Classification categories of techniques

A common differentiation of watermarking techniques is between *visible* and *invisible* watermarking. Historically, visible watermarks appeared first and are still in use by magazines and TV networks but there are many cases in which they are not as practical and efficient as invisible ones. Invisibility is achieved by embedding the secret information into the frequency domain of the image or in other embedding spaces in which the human eye is not sensitive. Another classification of watermarks is between *readable* and *detectable*. For the case of detectable watermarks, the detection program's output is a numerical value; if this value exceeds a predefined threshold, the image is declared watermarked, else it is assumed that there is no watermark present (Figure 1). Thus the whole procedure is automated (ma-

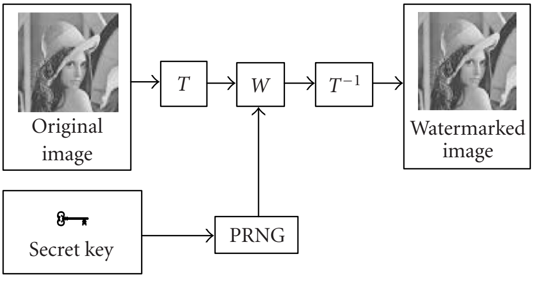

FIGURE 2: Typical frequency domain, invisible watermark embedding T: mathematical transform, T^{-1}: inverse transform, W: watermarking method, and PRNG: pseudorandom noise generator.

chine performed). In the case of readable watermarks, (in which human interaction is usually required), the embedded information is of readable form, a text or a logo image, for example. The advantage of this category is that even if we do not have full extraction of the hidden information, we might still be able to identify the mark. For example, a logo image, might be corrupted by "noise" during the extraction but it can still be recognizable, or some letters in a text message might be wrongly extracted but the message can still be understandable.

Using or not the original image distinguishes between *informed* and *blind* techniques. In the case of informed technique, the original image is used in the detection phase. It is evident that this provides additional robustness but on the other hand it is quite impractical in many cases. Blind watermarking means that we do not need the original or any additional information (other than a detection key, for example) to prove that the work is watermarked. Efficiency is not as high as in the case of informed techniques but it is highly practical, thus making this kind of techniques the most interesting ones for research.

Embedding space is another discriminating element. In the early years of watermarking, there were two choices: *spatial* and *frequency* domain. In the first case, the insertion strategy was usually to alter the least significant bit (LSB) of the pixels' intensity according to some specific way. Generally, spatial domain techniques are quite imperceptible and can be proved efficient in some simple image processing tasks but they are not as robust as frequency domain techniques. In the later case (Figure 2), the image is usually transformed (as a whole or in smaller square blocks) using a well-known mathematical transform (e.g. DCT, DFT, Wavelet, or others) and then a set of carefully selected coefficients is altered according to a modification rule. Then, the inverse transform takes place in order to obtain the watermarked image. Most of the time, the watermark is a pseudorandom noise sequence that is modulated with the original coefficients. Although simple, this idea has been proved efficient in a great variety of image processes. Since the watermark is embedded into the frequency spectrum of the image, it is also invisible and quite robust providing that the altered coefficients are carefully selected. Different frequency bands have different advantages. The aid of the human visual system (HVS) model is of great importance for the selection of the coefficients.

In 1999s ICIP in Kobe, Kutter et al [12] introduced the term "second generation watermarking schemes." This is indeed another classification method for watermarking techniques. As first generation techniques, we define the early watermarking works that used

Table 1: Classification overview.

Discriminating factor	Categories
Visibility of watermark	Visible versus invisible
Detector's output	Readable versus detectable
Need for original image	Informed versus blind
Domain of embedding	Spatial versus frequency
Special image feature exploitation	first generation versus second generation

a strict embedding strategy either in spatial or in frequency domain and did not take into account the specific characteristics of the image carrier. In *Watermarking Glossary* [13], second generation schemes are defined as "watermarking systems that employ some form of feature detection." After extracting some image features, it is easier to implement more sophisticated schemes that are image adaptive, thus achieving better results in terms of robustness, invisibility, and so forth. In order for these features to be efficient, they should be invariant to noise, filtering, and geometrical transformations to ensure that the scheme will be able to survive against intentional or accidental removal attempts. An overview of the techniques' classification can be found in Table 1.

5. Attacks

Any attempt to thwart or compromise the purpose of a watermarking system, intentional or not, is called "attack." This definition includes malicious attempts to remove the watermark, make it illegible, cast more than one different watermark to confuse the detection system as well as simple image processing tasks like filtering, compression, geometric manipulations, and so forth, that might have similar effects on the watermark. Herein, we put them under microscope.

Common image processing operations. This is the easiest way for the nonspecialist who intends to remove or tamper the watermark. It can be performed by means of the most common image processing software. It includes

(i) resampling/requantization,
(ii) filtering (lowpass, highpass, median, Wiener, and others),
(ii) dithering,
(vi) histogram manipulations (stretching, equalization, etc.),
(v) color manipulations (palette changing, color reduction),
(vi) noise addition (Gaussian, uniform, etc.)

Lossy compression. The JPEG and JPEG2000 standards perform both lossless and lossy compressions. In the later case, part of the image information is lost in order to achieve higher compression ratios. This works destructively for the watermark too [14, 15, 16, 17, 18].

Geometric Distortions. Rotation, translation, scaling, cropping, alone or combined, form one of the most difficult opponents of a watermarking system [19, 20, 21, 22, 23].

Other attacks. This category includes more sophisticated, intentional attacks [24] sampled as follows.

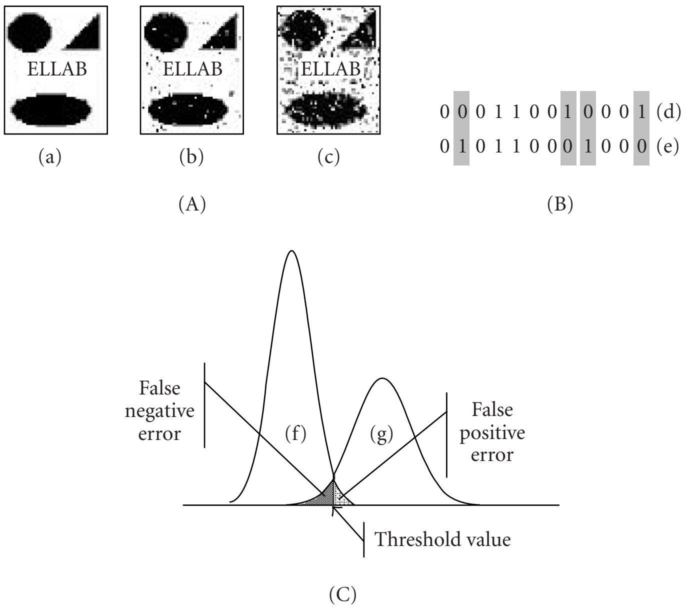

Figure 3: A. logo watermarks: (a) original watermark, (b), and (c) watermarks after attacks. B. binary digit watermarks: (d) original bits and (e) bits recovered after attacks (error bits are marked in gray). C. PDFs of numerical detectors: (f) no watermark detected (g) watermark succesfully detected (shape and position of (g) changes according to attacks).

(i) Printing and rescanning.
(ii) Collusion: according to this attack, a set of watermarked versions of the same image is used to produce a new one (usually by averaging) in which no watermark can be retrieved.
(iii) Copy: an attack in which an adversary copies a legitimate watermark from one work to another.
(iv) Stirmark: the image is slightly stretched, sheared, shifted, and/or rotated by an unnoticeable random amount and then resampled.
(v) Mosaic: splitting the watermarked image into sufficiently small pieces in order to desynchronize/confuse the mark detector. This is also very effective for schemes that use the whole images for detection, without block partitioning.

Errors caused by attacks differ according to the type of the watermark used (Figure 3). For a logo-formed watermark, the error could be the introduction of noise. For a binary string watermark, error means wrong decoding of some bits. That is why, for both of these cases, multiple embedding is a must. For numeric detectors, the problems have to do with the choice of the threshold value [25]. Generally, there are two types of errors in watermarking systems:

(i) *false negative* (or type I): although the image is watermarked, the system fails to detect so;

(ii) *false positive* (or type II): the image is not watermarked with the specific key and the system declares that it is.

6. Benchmarking tools

While new watermarking methods are introduced day by day, the need for evaluation has become quite urgent. There are so many schemes that those interested to use this new technology are completely confused. Different applications have different requirements and there is no algorithm that may cover all of them. Petitcolas and Kutter [26, 27, 28] were ones of the first that realized the necessity for categorization of the needs and development of efficient benchmarking tools. Not only they defined the framework into which a benchmarking tool should be constructed but it was Petitcolas who in 1997 introduced *Stirmark* [29, 30], a tool which is still the most popular benchmarking tool among the watermarking community.

In its first form, *Stirmark* was simply introducing random bilinear geometric distortions to desynchronize watermarking algorithms. Two years later, in 1999, version 3.1 of Stirmark had become the most complete benchmarking tool and a point of reference for corresponding software. The current version is 4.0, in which watermarking methods developers may define new attacks of their own or embed their watermarking libraries and evaluate them on the basis of evaluation profiles that they can specify according to the needs/requirements that they have to satisfy.

Another great tool is *Checkmark* provided by the Computer Vision Group, University of Geneva, Switzerland. Pereira has created a tool [31, 32] that runs under the Matlab environment, with a large number of attacks, some of which are reprogrammed versions of Stirmark originals while others are completely new. It is very flexible and easy to tweak and add, for example, new kind of attacks. There are different benchmarking profiles according to six different application fields. The output results come in quite elegant html pages. Unfortunately, there are no news from the development of the suite (last version and corresponding results were released on December 2001).

Optimark [33, 34] is another Stirmark based, feature rich, benchmarking utility. The attacks have been reprogrammed from Stirmark. Some of its specific features include a graphical interface, evaluation based on numerous parameters like embedding/decoding time, payload, correct message decoding percentage, ROC curves, false positive and negative probabilities, and many more. Like all packages, Optimark distinguishes between methods that output a float number or a binary output, with different characteristics for each case. The development of this suite has been partially supported by the Certimark EU project [35].

7. The future

By now, the reader must have been convinced that watermarking research is currently a very active, really "hot", and challenging new technology. New research topics include informed coding (in which the embedded message is a code word dependent on the cover work) [36], informed embedding (similar to informed coding, prior analysis of cover work is used to perceptually shape the watermark and optimize it for robustness) [36, 37], asymmetric or public key watermarking (in which a private key is used for embedding and a public key for detection) [38, 39], zero-knowledge detection (in which the detection process is substituted by an interactive cryptographic protocol) [40], application of multidimensional nested codes [41, 42], and many more.

But on the other hand, what are the achievements so far? Are there any commercial products out there? Are there any profitable companies in the area? What are the prospects? In the past, other promising technologies have also been hailed with enthusiasm by the research community but at the long run failed to convince the market. There are very few companies that work in the field and offer some commercial watermarking products but it is this year, in the first quarter of 2003, that the leading supplier of digital watermarking technologies and a leader in secure personal identification systems, Digimarc, has announced total revenues of \$21.7 million and net income of \$29 000. This marked the company's first profitable quarter as a public company, culminating five consecutive quarters of continuous improvement in financial and operating performance. To be honest, different watermarking applications seem to have different futures ahead. Broadcast monitoring and authentication for example, appear as the most future-profitable applications while there is great skepticism about copyright protection. But even in that case, as Miller states in [43], "the question in practice is not whether watermarking is a magic bullet or whether we can make something that's unbreakable under the most stringent assumptions, but whether we can make something that's economically valuable under realistic assumptions." It is the authors' opinion, and many more share it, that time has come for the industries involved to get profit out of this technology. Of course, the technology behind commercial implementations is not yet as advanced as current scientific research but after all, this is always expected. New challenges keep coming, like the new compression standard JPEG2000 or novel attack schemes we can not even imagine, but the fact remains that with more exceptional scientists, and researchers coming every day into the field and in collaboration with a wealthy industrial branch, watermarking has definitely a great future ahead.

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We encourage all EURASIP members to ensure that details of recent Ph.D. awards are added to the EURASIP website under the Ph.D. Links section http://www.eurasip.org. We are also introducing a regular section within this newsletter, highlighting recent awards added to the database in the following form.

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Author: Maria G. Jafari
Advisor: Jonathon A Chambers
Granting Institution: University of London (King's College)
Award Date: February 2003
Information: maria.jafari@kcl.ac.uk

Please send future submissions to:
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chambersj@cardiff.ac.uk
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Signal Processing is an interdisciplinary journal presenting the theory and practice of signal processing. Its primary objectives are the following:

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Editors-in-Chief

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Subjects

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

Editor-in-Chief

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Special Issue on Digital Audio for Multimedia Communications

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Interest in digital processing of audio signals has been reinvigorated by the introduction of multimedia communication via the Internet and digital audio broadcasting systems. These new applications demand high bandwidth and require innovative solutions to an old problem: how to achieve high quality at low bit rates. Often this problem is addressed by transmission schemes in which only part of the original audio data is transmitted to other sources: voices or channels. The output must be reconstructed at the receiver from purely synthetic or incomplete data. Additionally, the global networked audio community must solve a new class of problems concerning the protection of audio streams and documents. Accordingly, robust methods are sought for enforcing security, privacy, ownership, and authentication of audio data. Furthermore, the maintenance of audio archives—our cultural heritage—requires the development of efficient techniques for the restoration of corrupted audio documents.

This special issue provides a sample of the new directions in digital audio research. In audio synthesis, real-time computation of physical models of acoustic instruments is now possible due to the steady progress of Moore's law. In the paper by B. Bank et al., a review of piano synthesis is given. The synthesis is described in terms of structured audio and the structured audio orchestral language (SAOL) which is included in MPEG-4. Through the use of filtering and interpolation, P. A. A. Esquef et al. describe the use of the frequency-zooming analysis method to derive an ARMA model for synthesizing stringed instruments. Model-based computation of string sounds can be used to create more expressive synthesis of string sounds by offering a wide space of controllable parameters.

Multichannel audio promises to bring more realistic reproduction to the listener. In the paper by A. Mouchtaris et al., a small number of microphone signals are resynthesized into a larger number of "virtual microphones," thereby reducing the transmission bandwidth while enhancing the final rendering. In the paper by D. Yang et al., a high-performance scheme based on the MPEG advanced audio coding system that allows for the efficient transmission of multiple audio channels at scalable bit rates is proposed.

Watermarking and data-hiding techniques try to prevent unauthorized use of audio resources and additionally make it possible to include additional metadata in the audio stream. In their paper, M. F. Mansour and A. H. Tewfik introduce a new method for robust scale and shift invariant data hiding based on wavelet transforms. The paper by M. Steinebach and J. Dittmann addresses the problem of authenticating audio streams by embedding content-related data that allow the decoder to check for integrity.

Quality networked speech communication poses not only bandwidth but also privacy concerns. In their paper, C. R. N. Athaudage et al. propose a new method for efficiently encoding the spectral information in a low rate speech coder. The authors exploit the possibility of increasing the coding gain at the cost of introducing a substantially higher coding delay. Real-time software applications designed for securing speech transmission over the Internet are reviewed in the paper by A. Aldini et al.

In denoising or noise-reduction problems, a time varying filter can be applied to the corrupted audio signal. Earlier work on a minimum mean square error (MMSE) estimator by Ephraim and Malah is quite expensive to compute. In P. J. Wolfe and S. J. Godsill's paper, a Bayesian estimator that is easier to compute and easier to understand is derived.

The guest editors would like to thank the authors and the reviewers of the papers for their contributions in maintaining clarity, coherence, and consistency in this special issue.

Gianpaolo Evangelista
Mark Kahrs
Emmanuel Bacry

Volume 2003, No. 10, 1 September 2003

Contents and Abstracts

Physically Informed Signal Processing Methods for Piano Sound Synthesis: A Research Overview

Balázs Bank, Federico Avanzini, Gianpaolo Borin, Giovanni De Poli, Federico Fontana, and Davide Rocchesso

http://dx.doi.org/10.1155/S1110865703304093

This paper reviews recent developments in physics-based synthesis of piano. The paper considers the main components of the instrument, that is, the hammer, the string, and the soundboard. Modeling techniques are discussed for each of these elements, together with the implementation strategies. Attention is focused on numerical issues, and each implementation technique is described in light of its efficiency and accuracy properties. As the structured audio coding approach is gaining popularity, the authors argue that the physical modeling approach will have relevant applications in the field of multimedia communication.

Frequency-Zooming ARMA Modeling for Analysis of Noisy String Instrument Tones

Paulo A. A. Esquef, Matti Karjalainen, and Vesa Välimäki

http://dx.doi.org/10.1155/S1110865703304020

This paper addresses model-based analysis of string instrument sounds. In particular, it reviews the application of autoregressive (AR) modeling to sound analysis/synthesis purposes. Moreover, a frequency-zooming autoregressive moving average (FZ-ARMA) modeling scheme is described. The performance of the FZ-ARMA method on modeling the modal behavior of isolated groups of resonance frequencies is evaluated for both synthetic and real string instrument tones immersed in background noise. We demonstrate that the FZ-ARMA modeling is a robust tool to estimate the decay time and frequency of partials of noisy tones. Finally, we discuss the use of the method in synthesis of string instrument sounds.

Virtual Microphones for Multichannel Audio Resynthesis

Athanasios Mouchtaris, Shrikanth S. Narayanan, and Chris Kyriakakis

http://dx.doi.org/10.1155/S1110865703304032

Multichannel audio offers significant advantages for music reproduction, including the ability to provide better localization and envelopment, as well as reduced imaging distortion. On the other hand, multichannel audio is a demanding media type in terms of transmission requirements. Often, bandwidth limitations prohibit transmission of multiple audio channels. In such cases, an alternative is to transmit only one or two reference

channels and recreate the rest of the channels at the receiving end. Here, we propose a system capable of synthesizing the required signals from a smaller set of signals recorded in a particular venue. These synthesized "virtual" microphone signals can be used to produce multichannel recordings that accurately capture the acoustics of that venue. Applications of the proposed system include transmission of multichannel audio over the current Internet infrastructure and, as an extension of the methods proposed here, remastering existing monophonic and stereophonic recordings for multichannel rendering.

Progressive Syntax-Rich Coding of Multichannel Audio Sources

Dai Yang, Hongmei Ai, Chris Kyriakakis, and C.-C. Jay Kuo

http://dx.doi.org/10.1155/S1110865703304044

Being able to transmit the audio bitstream progressively is a highly desirable property for network transmission. MPEG-4 version 2 audio supports fine grain bit rate scalability in the generic audio coder (GAC). It has a bit-sliced arithmetic coding (BSAC) tool, which provides scalability in the step of 1 Kbps per audio channel. There are also several other scalable audio coding methods, which have been proposed in recent years. However, these scalable audio tools are only available for mono and stereo audio material. Little work has been done on progressive coding of multichannel audio sources. MPEG advanced audio coding (AAC) is one of the most distinguished multichannel digital audio compression systems. Based on AAC, we develop in this work a progressive syntax-rich multichannel audio codec (PSMAC). It not only supports fine grain bit rate scalability for the multichannel audio bitstream but also provides several other desirable functionalities. A formal subjective listening test shows that the proposed algorithm achieves an excellent performance at several different bit rates when compared with MPEG AAC.

Time-Scale Invariant Audio Data Embedding

Mohamed F. Mansour and Ahmed H. Tewfik

http://dx.doi.org/10.1155/S1110865703304135

We propose a novel algorithm for high-quality data embedding in audio. The algorithm is based on changing the relative length of the middle segment between two successive maximum and minimum peaks to embed data. Spline interpolation is used to change the lengths. To ensure smooth monotonic behavior between peaks, a hybrid orthogonal and nonorthogonal wavelet decomposition is used prior to data embedding. The possible data embedding rates are between 20 and 30 bps. However, for practical purposes, we use repetition codes, and the effective embedding data rate is around 5 bps. The algorithm is invariant after time-scale modification, time shift, and time cropping. It gives high-quality output and is robust to mp3 compression.

Watermarking-Based Digital Audio Data Authentication

Martin Steinebach and Jana Dittmann

http://dx.doi.org/10.1155/S1110865703304081

Digital watermarking has become an accepted technology for enabling multimedia protection schemes. While most efforts concentrate on user authentication, recently interest in data authentication to ensure data integrity has been increasing. Existing concepts address mainly image data. Depending on the necessary security level and the sensitivity to detect changes in the media, we differentiate between fragile, semifragile, and content-fragile watermarking approaches for media authentication. Furthermore, invertible watermarking schemes exist while each bit change can be recognized by the watermark which can be extracted and the original data can be reproduced for high-security applications. Later approaches can be extended with cryptographic approaches like digital signatures. As we see from the literature, only few audio approaches exist and the audio domain requires additional strategies for time-flow protection and resynchronization. To allow different security levels, we have to identify relevant audio features that can be used to determine content manipulations. Furthermore, in the field of invertible schemes, there are a bunch of publications for image and video data but no approaches for digital audio to ensure data authentication for high-security applications. In this paper, we introduce and evaluate two watermarking algorithms for digital audio data, addressing content integrity protection. In our first approach, we discuss possible features for a content-fragile watermarking scheme to allow several postproduction modifications. The second approach is designed for high-security applications to detect each bit change and reconstruct the original audio by introducing an invertible audio watermarking concept. Based on the invertible audio scheme, we combine digital signature schemes and digital watermarking to provide a public verifiable data authentication and a reproduction of the original, protected with a secret key.

Model-Based Speech Signal Coding Using Optimized Temporal Decomposition for Storage and Broadcasting Applications

Chandranath R. N. Athaudage, Alan B. Bradley, and Margaret Lech

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A dynamic programming-based optimization strategy for a temporal decomposition (TD) model of speech and its application to low-rate speech coding in storage and broadcasting is presented. In previous work with the spectral stability-based event localizing (SBEL) TD algorithm, the event localization was performed based on a spectral stability criterion. Although this approach gave reasonably good results, there was no assurance on the optimality of the event locations. In the present work, we have optimized the event localizing task using a dynamic programming-based optimization strategy. Simulation results show that an improved TD model accuracy can be achieved. A methodology of incorporating the optimized TD algorithm within the standard MELP speech coder for the efficient compression of speech spectral information is also presented. The performance evaluation results revealed that the proposed speech coding scheme achieves 50%–60% compression of speech spectral information with negligible degradation in the decoded speech quality.

On Securing Real-Time Speech Transmission over the Internet: An Experimental Study

Alessandro Aldini, Marco Roccetti, and Roberto Gorrieri

http://dx.doi.org/10.1155/S1110865703304019

We analyze and compare several soft real-time applications designed for the secure transmission of packetized audio over the Internet. The main metrics we consider for the purposes of our analysis are (i) the computational load due to the coding/decoding phases, and (ii) the computational overhead of the encryption/decryption activities, carried out by the audio tools of interest. The main result we present is that an appropriate degree of security may be guaranteed to real-time audio communications at a negligible computational cost if the adopted security strategies are integrated together with the playout control mechanism incorporated in the audio tools.

Efficient Alternatives to the Ephraim and Malah Suppression Rule for Audio Signal Enhancement

Patrick J. Wolfe and Simon J. Godsill

http://dx.doi.org/10.1155/S1110865703304111

Audio signal enhancement often involves the application of a time-varying filter, or suppression rule, to the frequency-domain transform of a corrupted signal. Here we address suppression rules derived under a Gaussian model and interpret them as spectral estimators in a Bayesian statistical framework. With regard to the optimal spectral amplitude estimator of Ephraim and Malah, we show that under the same modelling assumptions, alternative methods of Bayesian estimation lead to much simpler suppression rules exhibiting similarly effective behaviour. We derive three of such rules and demonstrate that, in addition to permitting a more straightforward implementation, they yield a more intuitive interpretation of the Ephraim-Malah solution.

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Special Issue on
Signal Processing for Acoustic Communication Systems

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

The aim of this special issue is to highlight innovative research in signal processing for acoustic communication systems, thus paving the way for future developments in the field. This was the philosophy behind the decision to prepare a special issue of the EURASIP Journal on Applied Signal Processing devoted to this area. Out of sixteen submitted papers, nine have been finally selected by the guest editors, taking into account the evaluation via standard international peer review process. The selected papers cover a wide range of acoustic communication systems ranging from double-talk detectors to blind signal separation. Double-talk detectors are vital to the operation and performance of acoustic echo cancellers. There is a need for an extension of double-talk detection to multidelay block frequency domain adaptive filters, which have been introduced to make a proper selection between processing delay and complexity. In their paper, Benesty and Gänsler define an extended cross-correlation vector in the frequency domain which fits very well with an echo canceller based on the multidelay block frequency domain structure. Since background noise is able to cause low speech intelligibility and hence low overall system performance, the next two papers deal with speech enhancement systems. Lotter, Beniem, and Vary introduce in their contribution two short-time spectral amplitude estimators for speech enhancement with multiple microphones, while Guérin, Le Bouquin-Jeannès, and Faucon present in their paper a two-microphone speech enhance system, dedicated to remove noise in a hands-free car kit.

In their contribution, Gannot and Moonen present a novel approach for multimicrophone speech dereverberation, which is an important subject when two people are far apart in a highly reverberant room since they cannot have a conversation easily. This reverberation is due to strong echoes and decreases the intelligibility of recorded speech. An essential requirement of acoustic communication systems using array processing techniques is the ability to locate and track audio sources. One of the techniques is based on time-delay estimation. Doclo and Moonen present two adaptive algorithms for robust time-delay estimation in acoustic environments where a large amount of additive background noise and reverberation is present. A beamformer is a processor used in conjunction with an array of sensors to provide a versatile form of spatial filtering. The objective of a beamformer is to estimate the signal arriving from a desired direction in the presence of noise and interfering signals. Cohen, Gannot, and Berdugo present a novel approach for real-time multichannel speech enhancement in environments of nonstationary noise and time-varying acoustic transfer functions. The proposed system integrates adaptive beamforming, identification of acoustic transfer functions, soft signal detection, and multichannel postfiltering. The next two papers deal with blind signal separation (BSS) techniques. BSS has been proposed to recover source signals from their measurements (the microphone signals). These techniques are termed blind as the acoustic transfer functions from the sources to the microphones are unknown and there are no reference signals to compare the recovered source signal against. Despite of its high complexity and low convergence properties, independent component analysis is a concept that is frequently used for BSS. Saruwatari, Kurita, Takeda, Itakura, Nishikawa, and Shikano describe a new method of BSS combining subband independent component analysis and beamforming which can solve the slow convergence properties. Since many conventional BSS algorithms can hardly be implemented in real time due to high-computational complexity, this is the topic of the paper by Yin, Sommen, and He.

In their contribution, they aim at reducing the computational complexity by proposing a new mixing model for multispeaker-multimicrophone environment. Araki, Makino, Hinamoto, Mukai, Nishikawa, and Saruwatari give an interpretation of BSS from a physical point of view by showing that BSS is equivalent to two sets of adaptive beamformers.

In the coming years, it is expected that acoustic communication systems will become even more important both in research and industry. We hope that this special issue will further stimulate work on signal processing in this area.

Kees Janse
Walter Kellermann
Marc Moonen
Piet Sommen

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Contents and Abstracts

A Multidelay Double-Talk Detector Combined with the MDF Adaptive Filter

Jacob Benesty and Tomas Gänsler

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The multidelay block frequency-domain (MDF) adaptive filter is an excellent candidate for both acoustic and network echo cancellation. There is a need for a very good double-talk detector (DTD) to be combined efficiently with the MDF algorithm. Recently, a DTD based on a normalized cross-correlation vector was proposed and it was shown that this DTD performs much better than the Geigel algorithm and other DTDs based on the cross-correlation coefficient. In this paper, we show how to extend the definition of a normalized cross-correlation vector in the frequency domain for the general case where the block size of the Fourier transform is smaller than the length of the adaptive filter. The resulting DTD has an MDF structure, which makes it easy to implement, and a good fit with an echo canceler based on the MDF algorithm. We also analyze resource requirements (computational complexity and memory requirement) and compare the MDF algorithm with the normalized least mean square algorithm (NLMS) from this point of view.

An Integrated Real-Time Beamforming and Postfiltering System for Nonstationary Noise Environments

Israel Cohen, Sharon Gannot, and Baruch Berdugo

http://dx.doi.org/10.1155/S1110865703305050

We present a novel approach for real-time multichannel speech enhancement in environments of nonstationary noise and time-varying acoustical transfer functions (ATFs). The proposed system integrates adaptive beamforming, ATF identification, soft signal detection, and multichannel postfiltering. The noise canceller branch of the beamformer and the ATF identification are adaptively updated online, based on hypothesis test results. The noise canceller is updated only during stationary noise frames, and the ATF identification is carried out only when desired source components have been detected. The hypothesis testing is based on the nonstationarity of the signals and the transient power ratio between the beamformer primary output and its reference noise signals. Following the beamforming and the hypothesis testing, estimates for the signal presence probability and for the noise power spectral density are derived. Subsequently, an optimal spectral gain function that minimizes the mean square error of the log-spectral amplitude (LSA) is applied. Experimental results demonstrate the usefulness of the proposed system in nonstationary noise environments.

Subspace Methods for Multimicrophone Speech Dereverberation

Sharon Gannot and Marc Moonen

http://dx.doi.org/10.1155/S1110865703305049

A novel approach for multimicrophone speech dereverberation is presented. The method is based on the construction of the null subspace of the data matrix in the presence of colored noise, using the generalized singular-value decomposition (GSVD) technique, or the generalized eigenvalue decomposition (GEVD) of the respective correlation matrices. The special Silvester structure of the filtering matrix, related to this subspace, is exploited for deriving a total least squares (TLS) estimate for the acoustical transfer functions (ATFs). Other less robust but computationally more efficient methods are derived based on the same structure and on the QR decomposition (QRD). A preliminary study of the incorporation of the subspace method into a subband framework proves to be efficient, although some problems remain open. Speech reconstruction is achieved by virtue of the matched filter beamformer (MFBF). An experimental study supports the potential of the proposed methods.

Exploiting Acoustic Similarity of Propagating Paths for Audio Signal Separation

Bin Yin, Piet C. W. Sommen, and Peiyu He

http://dx.doi.org/10.1155/S1110865703306031

Blind signal separation can easily find its position in audio applications where mutually independent sources need to be separated from their microphone mixtures while both room acoustics and sources are unknown. However, the conventional separation algorithms can hardly be implemented in real time due to the high computational complexity. The computational load is mainly caused by either direct or indirect estimation of thousands of acoustic parameters. Aiming at the complexity reduction, in this paper, the acoustic paths are investigated through an acoustic similarity index (ASI). Then a new mixing model is proposed. With closely spaced microphones (5–10 cm apart), the model relieves the computational load of the separation algorithm by reducing the number and length of the filters to be adjusted. To cope with real situations, a blind audio signal separation algorithm (BLASS) is developed on the proposed model. BLASS only uses the second-order statistics (SOS) and performs efficiently in frequency domain.

Robust Adaptive Time Delay Estimation for Speaker Localization in Noisy and Reverberant Acoustic Environments

Simon Doclo and Marc Moonen

http://dx.doi.org/10.1155/S111086570330602X

Two adaptive algorithms are presented for robust time delay estimation (TDE) in acoustic environments with a large amount of background noise and reverberation. Recently, an adaptive eigenvalue decomposition (EVD) algorithm has been developed for TDE in highly reverberant acoustic environments. In this paper, we extend the adaptive EVD algorithm to noisy and reverberant acoustic environments by deriving an adaptive stochastic gradient algorithm for the generalized eigenvalue decomposition (GEVD) or by prewhitening

the noisy microphone signals. We have performed simulations using a localized and a diffuse noise source for several SNRs, showing that the time delays can be estimated more accurately using the adaptive GEVD algorithm than using the adaptive EVD algorithm. In addition, we have analyzed the sensitivity of the adaptive GEVD algorithm with respect to the accuracy of the noise correlation matrix estimate, showing that its performance may be quite sensitive, especially for low SNR scenarios.

A Two-Sensor Noise Reduction System: Applications for Hands-Free Car Kit

Alexandre Guérin, Régine Le Bouquin-Jeannès, and Gérard Faucon

http://dx.doi.org/10.1155/S1110865703305098

This paper presents a two-microphone speech enhancer designed to remove noise in hands-free car kits. The algorithm, based on the magnitude squared coherence, uses speech correlation and noise decorrelation to separate speech from noise. The remaining correlated noise is reduced using cross-spectral subtraction. Particular attention is focused on the estimation of the different spectral densities (noise and noisy signals power spectral densities) which are critical for the quality of the algorithm. We also propose a continuous noise estimation, avoiding the need of vocal activity detector. Results on recorded signals are provided, showing the superiority of the two-sensor approach to single microphone techniques.

Blind Source Separation Combining Independent Component Analysis and Beamforming

Hiroshi Saruwatari, Satoshi Kurita, Kazuya Takeda, Fumitada Itakura, Tsuyoki Nishikawa, and Kiyohiro Shikano

http://dx.doi.org/10.1155/S1110865703305104

We describe a new method of blind source separation (BSS) on a microphone array combining subband independent component analysis (ICA) and beamforming. The proposed array system consists of the following three sections: (1) subband ICA-based BSS section with estimation of the direction of arrival (DOA) of the sound source, (2) null beamforming section based on the estimated DOA, and (3) integration of (1) and (2) based on the algorithm diversity. Using this technique, we can resolve the low convergence problem through optimization in ICA. To evaluate its effectiveness, signal-separation and speech-recognition experiments are performed under various reverberant conditions. The results of the signal-separation experiments reveal that the noise reduction rate (NRR) of about 18 dB is obtained under the nonreverberant condition, and NRRs of 8 dB and 6 dB are obtained in the case that the reverberation times are 150 milliseconds and 300 milliseconds. These performances are superior to those of both simple ICA-based BSS and simple beamforming method. Also, from the speech-recognition experiments, it is evident that the performance of the proposed method in terms of the word recognition rates is superior to those of the conventional ICA-based BSS method under all reverberant conditions.

Multichannel Direction-Independent Speech Enhancement Using Spectral Amplitude Estimation

Thomas Lotter, Christian Benien, and Peter Vary

http://dx.doi.org/10.1155/S1110865703305025

This paper introduces two short-time spectral amplitude estimators for speech enhancement with multiple microphones. Based on joint Gaussian models of speech and noise Fourier coefficients, the clean speech amplitudes are estimated with respect to the MMSE or the MAP criterion. The estimators outperform single microphone minimum mean square amplitude estimators when the speech components are highly correlated and the noise components are sufficiently uncorrelated. Whereas the first MMSE estimator also requires knowledge of the direction of arrival, the second MAP estimator performs a direction-independent noise reduction. The estimators are generalizations of the well-known single channel MMSE estimator derived by Ephraim and Malah (1984) and the MAP estimator derived by Wolfe and Godsill (2001), respectively.

Equivalence between Frequency-Domain Blind Source Separation and Frequency-Domain Adaptive Beamforming for Convolutive Mixtures

Shoko Araki, Shoji Makino, Yoichi Hinamoto, Ryo Mukai, Tsuyoki Nishikawa, and Hiroshi Saruwatari

http://dx.doi.org/10.1155/S1110865703305074

Frequency-domain blind source separation (BSS) is shown to be equivalent to two sets of frequency-domain adaptive beamformers (ABFs) under certain conditions. The zero search of the off-diagonal components in the BSS update equation can be viewed as the minimization of the mean square error in the ABFs. The unmixing matrix of the BSS and the filter coefficients of the ABFs converge to the same solution if the two source signals are ideally independent. If they are dependent, this results in a bias for the correct unmixing filter coefficients. Therefore, the performance of the BSS is limited to that of the ABF if the ABF can use exact geometric information. This understanding gives an interpretation of BSS from a physical point of view.

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Volume 2003, No. 12, 1 November 2003
Contents and Abstracts

Spatially Adaptive Intensity Bounds for Image Restoration

Kaaren L. May, Tania Stathaki, and Aggelos K. Katsaggelos

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Spatially adaptive intensity bounds on the image estimate are shown to be an effective means of regularising the ill-posed image restoration problem. For blind restoration, the local intensity constraints also help to further define the solution, thereby reducing the number of multiple solutions and local minima. The bounds are defined in terms of the local statistics of the image estimate and a control parameter which determines the scale of the bounds. Guidelines for choosing this parameter are developed in the context of classical (nonblind) image restoration. The intensity bounds are applied by means of the gradient projection method, and conditions for convergence are derived when the bounds are refined using the current image estimate. Based on this method, a new alternating constrained minimisation approach is proposed for blind image restoration. On the basis of the experimental results provided, it is found that local intensity bounds offer a simple, flexible method of constraining both the nonblind and blind restoration problems.

A Fast and Efficient Topological Coding Algorithm for Compound Images

Xin Li

http://dx.doi.org/10.1155/S1110865703306043

We present a fast and efficient coding algorithm for compound images. Unlike popular mixture raster content (MRC) based approaches, we propose to attack compound image coding problem from the perspective of modeling location uncertainty of image singularities. We suggest that a computationally simple two-class segmentation strategy is sufficient for the coding of compound images. We argue that jointly exploiting topological properties of image source in classification and coding stages is beneficial to the robustness of compound image coding systems. Experimental results have justified the effectiveness and robustness of the proposed topological coding algorithm.

Algorithm of the Radar Imaging by Using the Wideband Signals with the Distorted Signal Phases

Yulia V. Zhulina

http://dx.doi.org/10.1155/S1110865703307012

The problem of restoring an image by its Fourier transform is considered when the Fourier transform contains phase distortions. The nature of these distortions and their values are arbitrary. The criterion for the quality of the phase distortion estimates is suggested. It can be used to select the image which is mostly like the true one. The nature of the true image is also arbitrary. The only condition for the true image is that it is real and positive for all the points of the restored area. The other condition for the task is that the recovered image is calculated as the absolute value of the inverse Fourier transform. The algorithm for the search of the compensating phases satisfying the criterion is not considered for the general case; however, the task of the radar imaging based on the wideband signal and the time synthesis of the aperture is treated in detail. The physical basis for the task is a wideband pulse radar signal reflected by a moving object. As a result, a two-dimensional aperture is synthesized along the range, due to the super resolution, and along the velocity, according to the motion of the object. The radar signals are received by a single receiver. The image is reconstructed on the basis of these signals by using the maximum likelihood technique. The method uses the coherent processing of the signals. In practice, the coherence can be destroyed (due to some atmospheric turbulence or equipment instability, due to some inaccuracy in defining the motion). We assume that the objects to be observed are located at the far zone. For this task and on the basis of the suggested criterion, we develop an approximate algorithm for searching the best compensating phases in the radar signal. The quality of the images is tested with the help of simulation.

Signal Processing of Ground Penetrating Radar Using Spectral Estimation Techniques to Estimate the Position of Buried Targets

Shanker Man Shrestha and Ikuo Arai

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Super resolution is very important for the signal processing of GPR (ground penetration radar) to resolve closely buried targets. However, it is not easy to get high resolution as GPR signals are very weak and enveloped by the noise. The MUSIC (multiple signal classification) algorithm, which is well known for its super-resolution capacity, has been implemented for signal and image processing of GPR. In addition, conventional spectral estimation technique, FFT (fast Fourier transform), has also been implemented for high-precision receiving signal level. In this paper, we propose CPM (combined processing method), which combines time-domain response of MUSIC algorithm and conventional IFFT (inverse fast Fourier transform) to obtain a super-resolution and high-precision signal level. In order to support the proposal, detailed simulation was performed analyzing SNR (signal-to-noise ratio). Moreover, a field experiment at a research field and a laboratory experiment at the University of Electro-Communications, Tokyo, were also performed for thorough investigation and supported the proposed method. All the simulation and experimental results are presented.

Nonstationary Interference Excision in Time-Frequency Domain Using Adaptive Hierarchical Lapped Orthogonal Transform for Direct Sequence Spread Spectrum Communications

Li-ping Zhu, Guang-rui Hu, and Yi-Sheng Zhu

http://dx.doi.org/10.1155/S1110865703306055

An adaptive hierarchical lapped orthogonal transform (HLOT) exciser is proposed for tracking, localizing, and rejecting the nonstationary interference in direct sequence spread spectrum (DSSS) communications. The method is based on HLOT. It utilizes a fast dynamic programming algorithm to search for the best basis, which matches the interference structure best, in a library of lapped orthogonal bases. The adaptive HLOT differs from conventional block transform and the more advanced modulated lapped transform (MLT) in that the former produces arbitrary time-frequency tiling, which can be adapted to the signal structure, while the latter yields fixed tilings. The time-frequency tiling of the adaptive HLOT can be time varying, so it is also able to track the variations of the signal time-frequency structure. Simulation results show that the proposed exciser brings significant performance improvement in the presence of nonstationary time-localized interference with or without instantaneous frequency (IF) information compared with the existing block transform domain excisers. Also, the proposed exciser is effective in suppressing narrowband interference and combined narrowband and time-localized impulsive interference.

Modeling Nonlinear Power Amplifiers in OFDM Systems from Subsampled Data: A Comparative Study Using Real Measurements

Ignacio Santamaría, Jesús Ibáñez, Marcelino Lázaro, Carlos Pantaleón, and Luis Vielva

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A comparative study among several nonlinear high-power amplifier (HPA) models using real measurements is carried out. The analysis is focused on specific models for wideband OFDM signals, which are known to be very sensitive to nonlinear distortion. Moreover, unlike conventional techniques, which typically use a single-tone test signal and power measurements, in this study the models are fitted using subsampled time-domain data. The in-band and out-of-band (spectral regrowth) performances of the following models are evaluated and compared: Saleh's model, envelope polynomial model (EPM), Volterra model, the multilayer perceptron (MLP) model, and the smoothed piecewise-linear (SPWL) model. The study shows that the SPWL model provides the best in-band characterization of the HPA. On the other hand, the Volterra model provides a good trade-off between model complexity (number of parameters) and performance.

Nonlinear System Identification Using Neural Networks Trained with Natural Gradient Descent

Mohamed Ibnkahla

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We use natural gradient (NG) learning neural networks (NNs) for modeling and identifying nonlinear systems with memory. The nonlinear system is comprised of a discrete-time linear filter H followed by a zero-memory nonlinearity $g(\cdot)$. The NN model is composed of a linear adaptive filter Q followed by a two-layer memoryless nonlinear NN. A Kalman filter-based technique and a search-and-converge method have been employed for the NG algorithm. It is shown that the NG descent learning significantly outperforms the ordinary gradient descent and the Levenberg-Marquardt (LM) procedure in terms of convergence speed and mean squared error (MSE) performance.

Hammerstein Model for Speech Coding

Jari Turunen, Juha T. Tanttu, and Pekka Loula

http://dx.doi.org/10.1155/S1110865703307048

A nonlinear Hammerstein model is proposed for coding speech signals. Using Tsay's nonlinearity test, we first show that the great majority of speech frames contain nonlinearities (over 80% in our test data) when using 20-millisecond speech frames. Frame length correlates with the level of nonlinearity: the longer the frames the higher the percentage of nonlinear frames. Motivated by this result, we present a nonlinear structure using a frame-by-frame adaptive identification of the Hammerstein model parameters for speech coding. Finally, the proposed structure is compared with the LPC coding scheme for three phonemes /a/, /s/, and /k/ by calculating the Akaike information criterion of the corresponding residual signals. The tests show clearly that the residual of the nonlinear model presented in this paper contains significantly less information compared to that of the LPC scheme. The presented method is a potential tool to shape the residual signal in an encode-efficient form in speech coding.

Performance Estimation for Lowpass Ternary Filters

Brenton Steele and Peter O'Shea

http://dx.doi.org/10.1155/S1110865703308030

Ternary filters have tap values limited to -1, 0, or $+1$. This restriction in tap values greatly simplifies the multipliers required by the filter, making ternary filters very well suited to hardware implementations. Because they incorporate coarse quantisation, their performance is typically limited by tap quantisation error. This paper derives formulae for estimating the achievable performance of lowpass ternary filters, thereby allowing the number of computationally intensive design iterations to be reduced. Motivated by practical communications systems requirements, the performance measure which is used is the worst-case stopband attenuation.

The Fractional Fourier Transform and Its Application to Energy Localization Problems

Patrick J. Oonincx and Hennie G. ter Morsche

http://dx.doi.org/10.1155/S1110865703305086

Applying the fractional Fourier transform (FRFT) and the Wigner distribution on a signal in a cascade fashion is equivalent to a rotation of the time and frequency parameters of the Wigner distribution. We presented in ter Morsche and Oonincx, 2002, an integral representation formula that yields affine transformations on the spatial and frequency parameters of the n-dimensional Wigner distribution if it is applied on a signal with the Wigner distribution as for the FRFT. In this paper, we show how this representation formula can be used to solve certain energy localization problems in phase space. Examples of such problems are given by means of some classical results. Although the results on localization problems are classical, the application of generalized Fourier transform enlarges the class of problems that can be solved with traditional techniques.

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Special Issue on
Signal Processing for Broadband Access Systems: Techniques and Implementations

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With the rapid growth of internet access and voice/data-centric communications, many access technologies have been developed to meet the stringent demand of high-speed data transmission and bridge the wide bandwidth gap between ever-increasing high-data-rate core networks and bandwidth-hungry end-user networks. To make efficient utilization of the limited bandwidth of existing access routes and cope with the adverse channel environment, many standards have been proposed for a variety of broadband access systems over different access environments (twisted pairs, coaxial cables, optical fibers, and fixed or mobile wireless access). These access environments may create different channel impairments and dictate unique sets of signal processing algorithms and techniques to combat specific impairments. In the design and implementation domain of those systems, many research issues arise.

Motivated by this design trend, the aim of this special issue is to present state-of-the-art signal processing techniques and implementation issues for broadband access systems over different access channels. We have selected nine high-quality papers covering the algorithm development, design, implementation, and application aspects of current and emerging wireline/wireless access technologies. In terms of technical contents, the nine papers can be loosely grouped into the areas of (1) efficient DSP algorithms for high-quality and secure data transmissions, and (2) implementation of DSP processing engines (e.g., FFT, FEC codec, and DSP core) for access-transceiver designs. These designs can be readily applied to emerging access systems such as xDSL, cable modem, FTTC, FTTH, 10G/Gigabit Ethernet, Bluetooth, wireless LAN, broadband wireless access, digital audio/video broadcasting, and so forth.

The first three papers focus on advanced DSP algorithm development. The first paper by Lakhzouri et al. considers the problem of line-of-sight (LOS) detection in WCDMA for mobile positioning. Mobile phone positioning in a cellular network with accurate position information has attracted great interest (e.g., FCC-E911 in USA and the coming E112 in the European Union). One method for locating the mobile station (MS) in two dimensions requires the measurement of LOS distance between the MS and at least three base stations. However, in many cases, the non-LOS (NLOS) signal components arrive with delay less than one chip at the receiver, thus obscuring the LOS signals. In this paper, the extended Kalman filter (EKF) is used to jointly estimate the delays and complex channel coefficients. The technique can provide an accurate decision whether LOS component is present or not, which is suitable for WCDMA receiver designs. The second paper by Martin et al. deals with the time-domain equalization (TEQ) algorithm that plays a crucial role in reducing intercarrier interference (ICI) and intersymbol interference (ISI) in multicarrier communication systems. This paper analyzes two TEQ design methods for cost-effective real-time implementation: minimum mean squared error (MMSE) and maximum shortening SNR (MSSNR) methods, which can significantly reduce the computational complexity in performing channel shortening compared with existing TEQ approaches, without degrading performance. The third paper by H. C. Chen et al. proposes a new signal security system and its VLSI architecture for real-time multimedia data transmission applications. By defining first two bit-circulation functions for one-dimensional binary array transformations, the authors exploit a chaotic system in generating a binary sequence to control the bit-circulation functions to perform the successive data transformation on the input data. The proposed chaotic security design features low computational complexity and regular operations, which leads to high feasibility for easy integration with commercial multimedia storage and transmission applications.

The next six papers deal with the VLSI algorithms and implementations of various important processing modules in modern access systems such as FFT, FEC codec, and DSP core. Due to the prevalence of multimode/multistandard communication systems, it is desirable to have a configurable/programmable DSP processing engine for various access systems. The first two papers in this category are the variable-length FFT/IFFT design by Kuo et al. and the configurable Viterbi decoder design by Benaissa and Zhu. In the FFT design, the authors design and implement a variable-length FFT/IFFT processor by using the techniques of cached-memory FFT architecture as well as the mixed-scaling-rotation CORDIC (MSR-CORDIC) scheme for the butterfly processing element. In the online reconfigurable Viterbi decoder design, the authors propose an area-efficient ACS architecture, in which the constraint length and traceback depth can be dynamically reconfigured. In addition, a scheduling program is used to systematically determine the maximum level of pipelining (speed-up) that can be applied to the decoder in an area-efficient/foldable architecture with in-place path-metric updating. This enables the exploration of the trade-off of decoding speed (throughput) versus area (number of ACS units) for a range of constraint lengths.

The next two papers discuss the advanced FEC algorithms and implementations for modern communication systems. The sixth paper by Kong and Parhi discusses the interleaved convolutional code that can further randomize the error bursts and can be used for burst-error correction. In this paper, an area-efficient high-speed Viterbi decoder architecture is proposed to decode $(n, 1, mI)$ interleaved convolutional code. This paper shows that hardware complexity reduction can be achieved with higher interleaving degree, which leads to area-efficient design for interleaved Viterbi decoder. The next paper by Y. Chen and Parhi demonstrates low-complexity decoding of block turbo-coded system with antenna diversity. This work tries to reduce the decoding complexity of space-time block turbo-coded system with low performance degradation. It considers two block turbo-coded systems with antenna diversity, including the simple serial concatenation of error-control code with space-time block code (STBC), and the recently proposed transmit antenna diversity scheme using FEC techniques. The overall decoding complexity is approximately ten times less than that of the near-optimum block turbo decoder but with only 0.5 dB loss of coding gain at the BER of 10^{-5} over an AWGN channel.

The final two papers are related to the communication DSP core implementations for access systems. The eighth paper, authored by Lee et al. presents new application-specific DSP (ASDSP) instructions and hardware accelerator to efficiently implement Reed-Solomon (RS) encoding and decoding. The ASDSP architecture can implement various programmable primitive polynomials, and significantly reduce the number of clock cycles compared with existing DSP chips. Thus, hardwired RS codecs can be replaced by ASDS accelerators. The final paper by Tsao et al. proposes a low-power embedded DSP core for communication systems. The features of the DSP core include parameterized data path, dual MAC unit, subword MAC, and optional function-specific blocks for accelerating communication system modulation operations. It is also a parameterized design so that the users can select the parameters and special functional blocks based on the characteristics of their applications, and then generate a DSP core in a very short time-to-the-market period.

Overall, the nine papers cover different aspects of advanced signal processing techniques and low-complexity/reconfigurable/programmable DSP implementations for broadband access systems. We thank the authors, the reviewers, the publisher, and the Editor-in-Chief for their tremendous amount of efforts to make this special issue successful. We hope the readers will find the results presented in this special issue helpful in their own design and implementation problems for communication systems.

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Contents and Abstracts

Extended Kalman Filter Channel Estimation for Line-of-Sight Detection in WCDMA Mobile Positioning

Abdelmonaem Lakhzouri, Elena Simona Lohan, Ridha Hamila, and Markku Renfors

http://dx.doi.org/10.1155/S1110865703306018

In mobile positioning, it is very important to estimate correctly the delay between the transmitter and the receiver. When the receiver is in line-of-sight (LOS) condition with the transmitter, the computation of the mobile position in two dimensions becomes straightforward. In this paper, the problem of LOS detection in WCDMA for mobile positioning is considered, together with joint estimation of the delays and channel coefficients. These are very challenging topics in multipath fading channels because LOS component is not always present, and when it is present, it might be severely affected by interfering paths spaced at less than one chip distance (closely spaced paths). The extended Kalman filter (EKF) is used to estimate jointly the delays and complex channel coefficients. The decision whether the LOS component is present or not is based on statistical tests to determine the distribution of the channel coefficient corresponding to the first path. The statistical test-based techniques are practical, simple, and of low computation complexity, which is suitable for WCDMA receivers. These techniques can provide an accurate decision whether LOS component is present or not.

Efficient Channel Shortening Equalizer Design

Richard K. Martin, Ming Ding, Brian L. Evans, and C. Richard Johnson Jr.

http://dx.doi.org/10.1155/S1110865703307024

Time-domain equalization is crucial in reducing channel state dimension in maximum likelihood sequence estimation and intercarrier and intersymbol interference in multicarrier systems. A time-domain equalizer (TEQ) placed in cascade with the channel produces an effective impulse response that is shorter than the channel impulse response. This paper analyzes two TEQ design methods amenable to cost-effective real-time implementation: minimum mean squared error (MMSE) and maximum shortening SNR (MSSNR) methods. We reduce the complexity of computing the matrices in the MSSNR and MMSE designs by a factor of 140 and a factor of 16 (respectively) relative to existing approaches, without degrading performance. We prove that an infinite-length MSSNR TEQ with unit norm TEQ constraint is symmetric. A symmetric TEQ halves FIR implementation complexity, enables parallel training of the frequency-domain equalizer and TEQ, reduces TEQ training complexity by a factor of 4, and doubles the length of the TEQ that can be designed using fixed-point arithmetic, with only a small loss in bit rate. Simulations are presented for designs with a symmetric TEQ or target impulse response.

Design and Realization of a New Signal Security System for Multimedia Data Transmission

Hun-Chen Chen, Jiun-In Guo, Lin-Chieh Huang, and Jui-Cheng Yen

http://dx.doi.org/10.1155/S1110865703309011

We propose a new signal security system and its VLSI architecture for real-time multimedia data transmission applications. We first define two bit-circulation functions for one-dimensional binary array transformation. Then, we exploit a chaotic system in generating a binary sequence to control the bit-circulation functions defined for performing the successive transformation on the input data. Each eight 8-bit data elements are regarded as a set and are fed into an 8×8 binary matrix being transformed on each row and each column of the matrix by these two bit-circulation functions such that the signal can be transformed into completely disordered data. The features of the proposed design include low computational complexity, regular operations suitable for low-cost VLSI implementation, high data security, and high feasibility for easy integration with commercial multimedia storage and transmission applications. We have performed Matlab simulation to verify the functional correctness of the proposed system. In implementing the system, a low-cost VLSI architecture has been designed, verified, and physically realized based on a 0.35 μm CMOS technology. The implementation results show that the proposed signal security system can achieve 117 Mbytes/s data throughput rate that is fast enough for real-time data protection in multimedia transmission applications.

VLSI Design of a Variable-Length FFT/IFFT Processor for OFDM-Based Communication Systems

Jen-Chih Kuo, Ching-Hua Wen, Chih-Hsiu Lin, and An-Yeu (Andy) Wu

http://dx.doi.org/10.1155/S1110865703309060

The technique of orthogonal frequency division multiplexing (OFDM) is famous for its robustness against frequency-selective fading channel. This technique has been widely used in many wired and wireless communication systems. In general, the fast Fourier transform (FFT) and inverse FFT (IFFT) operations are used as the modulation/demodulation kernel in the OFDM systems, and the sizes of FFT/IFFT operations are varied in different applications of OFDM systems. In this paper, we design and implement a variable-length prototype FFT/IFFT processor to cover different specifications of OFDM applications. The cached-memory FFT architecture is our suggested VLSI system architecture to design the prototype FFT/IFFT processor for the consideration of low power consumption. We also implement the twiddle factor butterfly processing element (PE) based on the coordinate rotation digital computer (CORDIC) algorithm, which avoids the use of conventional multiplication-and-accumulation unit, but evaluates the trigonometric functions using only add-and-shift operations. Finally, we implement a variable-length prototype FFT/IFFT processor with TSMC 0.35 μm 1P4M CMOS technology. The simulations results show that the chip can perform (64–2048)-point FFT/IFFT operations up to 80 MHz operating frequency which can meet the speed requirement of most OFDM standards such as WLAN, ADSL, VDSL ($256 \sim 2K$), DAB, and $2k$-mode DVB.

A Novel High-Speed Configurable Viterbi Decoder for Broadband Access

Mohammed Benaissa and Yiqun Zhu

http://dx.doi.org/10.1155/S1110865703310054

A novel design and implementation of an online reconfigurable Viterbi decoder is proposed, based on an area-efficient add-compare-select (ACS) architecture, in which the constraint length and traceback depth can be dynamically reconfigured. A design-space exploration to trade off decoding capability, area, and decoding speed has been performed, from which the maximum level of pipelining against the number of ACS units to be used has been determined while maintaining an in-place path metric updating. An example design with constraint lengths from 7 to 10 and a 5-level ACS pipelining has been successfully implemented on a Xilinx Virtex FPGA device. FPGA implementation results, in terms of decoding speed, resource usage, and BER, have been obtained using a tailored testbench. These confirmed the functionality and the expected higher speeds and lower resources.

Interleaved Convolutional Code and Its Viterbi Decoder Architecture

Jun Jin Kong and Keshab K. Parhi

http://dx.doi.org/10.1155/S1110865703309126

We propose an area-efficient high-speed interleaved Viterbi decoder architecture, which is based on the state-parallel architecture with register exchange path memory structure, for interleaved convolutional code. The state-parallel architecture uses as many add-compare-select (ACS) units as the number of trellis states. By replacing each delay (or storage) element in state metrics memory (or path metrics memory) and path memory (or survival memory) with I delays, interleaved Viterbi decoder is obtained where I is the interleaving degree. The decoding speed of this decoder architecture is as fast as the operating clock speed. The latency of proposed interleaved Viterbi decoder is "decoding depth (DD) $\times$ interleaving degree (I) + extra delays (A)," which increases linearly with the interleaving degree I.

Low-Complexity Decoding of Block Turbo-Coded System with Antenna Diversity

Yanni Chen and Keshab K. Parhi

http://dx.doi.org/10.1155/S1110865703305116

The goal of this paper is to reduce the decoding complexity of space-time block turbo-coded system with low performance degradation. Two block turbo-coded systems with antenna diversity are considered. These include the simple serial concatenation of error control code with space-time block code, and the recently proposed transmit antenna diversity scheme using forward error correction techniques. It is shown that the former performs better when compared to the latter in terms of bit error rate (BER) under the same spectral efficiency (up to 7 dB at the BER of 10^{-5} for quasistatic channel with two transmit and two receive antennas). For the former system, a computationally efficient decoding approach is proposed

for the soft decoding of space-time block code. Compared to its original maximum likelihood decoding algorithm, it can reduce the computation by up to 70% without any performance degradation. Additionally, for the considered outer code block turbo code, through reduction of test patterns scanned in the Chase algorithm and the alternative computation of its extrinsic information during iterative decoding, extra 0.3 dB to 0.4 dB coding gain is obtained if compared with previous approaches with negligible hardware overhead. The overall decoding complexity is approximately ten times less than that of the near-optimum block turbo decoder with coding gain loss of 0.5 dB at the BER of 10^{-5} over AWGN channel.

Design of Application-Specific Instructions and Hardware Accelerator for Reed-Solomon Codecs

Jung H. Lee, Jaesung Lee, and Myung H. Sunwoo

http://dx.doi.org/10.1155/S1110865703309138

This paper presents new application-specific digital signal processor (ASDSP) instructions and their hardware accelerator to efficiently implement Reed-Solomon (RS) encoding and decoding, which is one of the most widely used forward error control (FEC) algorithms. The proposed ASDSP architecture can implement various programmable primitive polynomials, and thus, hardwired RS codecs can be replaced. The new instructions and their hardware accelerator perform Galois field (GF) operations using the proposed GF multiplier and adder. Therefore, the proposed digital signal processor (DSP) architecture can significantly reduce the number of clock cycles compared with existing DSP chips. The proposed GF multiplier was implemented using the Faraday 0.25 μm standard cell library and it can perform RS decoding at a rate up to 228.1 Mbps at 130 MHz.

Low-Power Embedded DSP Core for Communication Systems

Ya-Lan Tsao, Wei-Hao Chen, Ming Hsuan Tan, Maw-Ching Lin, and Shyh-Jye Jou

http://dx.doi.org/10.1155/S1110865703309059

This paper proposes a parameterized digital signal processor (DSP) core for an embedded digital signal processing system designed to achieve demodulation/synchronization with better performance and flexibility. The features of this DSP core include parameterized data path, dual MAC unit, subword MAC, and optional function-specific blocks for accelerating communication system modulation operations. This DSP core also has a low-power structure, which includes the gray-code addressing mode, pipeline sharing, and advanced hardware looping. Users can select the parameters and special functional blocks based on the character of their applications and then generating a DSP core. The DSP core has been implemented via a cell-based design method using a synthesizable Verilog code with TSMC 0.35 μm SPQM and 0.25 μm 1P5M library. The equivalent gate count of the core area without memory is approximately 50 k. Moreover, the maximum operating frequency of a 16×16 version is 100 MHz (0.35 μm) and 140 MHz (0.25 μm).

Biometric Signal Processing

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

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

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

Genomic Signal Processing

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

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

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

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

Multimedia over IP and Wireless Networks

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

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

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

MIMO Communications and Signal Processing

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

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

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

Object-Based and Semantic Image and Video Analysis

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

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

Cross-Layer Design for Communications and Signal Processing Systems

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

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.

Special Issue on

Advances in Sensor Array Processing Technology

CALL FOR PAPERS

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

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

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

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

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

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Special Issue on

Advances in Intelligent Vision Systems: Methods and Applications

CALL FOR PAPERS

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

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

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

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

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

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Special Issue on DSP-Enabled Radio

CALL FOR PAPERS

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. Prospective papers should be unpublished and present novel contributions either in terms of fundamental research or from an applications perspective.

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

- Hardware/software architectures
- DSP algorithms and architectures
- Digital front-end reconfiguration
- Baseband DSP reconfiguration
- Simulation and design software
- Integrated development environments
- Rapid prototyping
- Synchronisation, carrier recovery, and equalisation
- Multiantenna systems and MIMO

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Special Issue on

Trends in Brain-Computer Interfaces

CALL FOR PAPERS

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.

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

- EEG-based BCI systems
- Cortical activity-based BCI systems
- BCI systems based on MEG, fMRI, etc.
- Preprocessing and feature extraction techniques for BCI systems
- Neural activity processing
- Protocols and evaluation methodologies for BCI systems
- Machine learning applied to BCI systems
- Applications of BCI systems in rehabilitation, entertainment, robotics, etc.

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Special Issue on

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

CALL FOR PAPERS

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.

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

- MIMO structures
- Smart antennas
- Multiuser detection
- Interference cancellation
- Space-time coding
- Channel decoding
- Channel identification/equalization
- Synchronization
- Power control
- CDMA/MC-CDMA/OFDM, etc.

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Special Issue on

Signal Analysis Tools for Optical Information Processing

CALL FOR PAPERS

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.

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

- Fourier optics and image optical processing
- Novel material, devices, and architectures for spatial and temporal optical signal processing
- Fractional transforms for optical signal analysis
- Phase-space analysis (e.g., Wigner-Ville analysis) of optical signals and fields
- Joint time-frequency representations (e.g., Wavelet analysis) of optical signals
- Ultrafast optical pulse processing and shaping
- Optical filter design and analysis
- Signal processing tools for the analysis and synthesis of photonic devices
- Applications of optical signal processing for optical communications, optical computing, pattern recognition, etc.

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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

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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

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

CALL FOR PAPERS

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

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

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

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

- Spreading Sequences Design
- Multiple access design

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

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Special Issue on

Multiuser MIMO Networks

CALL FOR PAPERS

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

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

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

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

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

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Special Issue on
Optical Wireless Communications

CALL FOR PAPERS

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

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

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

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

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

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

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

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Special Issue on

Advanced Signal Processing Algorithms for Wireless Communications

CALL FOR PAPERS

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.

Topics of interest include, but are not limited to, applications of advanced signal processing algorithms in wireless communication subsystems such as synchronization, equalization, and sequence estimation, based on such techniques as:

- The expectation-maximization algorithm
- The SAGE algorithm
- The Baum-Welch algorithm
- Per-Survivor processing
- Kalman filters and their extensions
- Hidden Markov modeling
- Sequential Monte Carlo filters
- Stochastic approximation algorithms
- Monte Carlo expectation Maximization

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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.

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- Physical layer issues
- Smart antenna beamforming
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