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The European Association for Signal Processing

Tampere University of Technology

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TueAmPS1
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OPENING CEREMONY AND PLENARY SESSION
Chair: Jaakko Astola, *Tampere University of Technology, Finland*

08:00

MOBILE INFORMATION SOCIETY AND DSP

Yrjö Neuvo, Nokia Corporation

The number of cellular phones sold annually exceeds that of PC's and cars together. The shift to second-generation digital cellular standards has made cellular phones the major application for digital signal processing. In the next few years the 3rd generation cellular standards will be taken in use. This will call for sophisticated DSP algorithms for the radio channel processing implemented in very high performance but low power processors. Simultaneously, it is expected that various forms of multimedia and advanced speech applications will take off. These applications will rely heavily on advanced signal processing algorithms. This presentation gives an overview of the current and future cellular technologies and highlights the expectations that this development places on both DSP algorithms and hardware.

WedAmPS1
Iso Sali
Multimedia Security
Chair: M. Gabbouj *Tampere University of Technology, Finland*

08:30

MULTIMEDIA SECURITY: DOES SIGNAL PROCESSING HAVE A PLACE IN SECURING THE DIGITAL WORLD?

Edward J. Delp, Purdue University West Lafayette, USA

With the recent growth of networked multimedia systems, techniques are needed to prevent (or at least deter) the illegal copying, forgery and distribution of digital media elements such as audio, images and video. The problem is that a copy of a media element is identical to the original. It is also desirable to determine where and by how much a multimedia element has been changed from the original. One classical approach to the problem of protecting "digital content" has been the use of cryptographic methods such as encryption and authentication. Recently other methods have been proposed to improve one's claim of ownership over a digital media element by inserting a small amount of distortion directly into the media element. This distortion, known as a digital watermark, can be used to uniquely identify the owner of the content. Digital watermarking research is mainly the domain of the signal processing community that is very naive when considering real world security problems. Also the attitude of the academic community that it has the "right" to copy and reproduce any multimedia element is curious at best and perhaps conveys the wrong message to our students. The use of the "Lena" image is a good example of this. In this talk I will survey the current state of multimedia security and describe important research issues. I will also express some concern that perhaps we in the signal processing community might want to understand the "security scenario" before making grandiose claims about how our latest watermarking technique will save the world!

ThuAmPS1
Iso Sali
Speech Processing
Chair: P. Haavisto *Nokia Research Center, Finland*

08:30

REPRESENTING SPEECH

W. Bastiaan Kleijn, KTH Royal Institute of Technology

The properties of the speech production process and the auditory periphery have led to the usage of similar speech signal representations for various processing tasks such as speech and speaker recognition, speech synthesis, and speech coding. The representation is generally divided into a description of the vocal-tract transfer function and the excitation source. For recognition purposes, the biased characterization of the vocal-tract transfer function by a time sequence of low-dimension cepstral vectors performs well. For coding and synthesis, we argue that for the vocal-tract transfer function autoregressive (AR) models are more effective than filter banks, while for the excitation source pitch-synchronous filter banks and modulation-domain filters are most effective. A clear trend exists towards the exploitation of the time variation of both the vocal-tract transfer function and the excitation source.

FriAmPS1
Iso Sali
A Perfect Fit
Chair: I. Hartimo *Helsinki University of Technology, Finland*

08:30

SPLINES: A PERFECT FIT FOR SIGNAL PROCESSING

Michael Unser, Swiss Federal Institute of Technology, Switzerland

Splines, which were invented by Schoenberg more than fifty years ago [1], constitute an elegant framework for dealing with interpolation and discretization problems. They are widely used in computer-aided design and computer graphics, but have been neglected in signal and image processing applications, mostly as a consequence of what I call the "bad press" phenomenon. Thanks to some recent research efforts in signal processing and wavelet-related techniques, the virtues of splines have been revived in our community [2]-there is now compelling evidence (several independent studies [3-5]) that splines offer the best cost-performance tradeoff among available interpolation methods. In this talk, I will argue that the spline representation is ideally suited for all processing tasks that require a continuous model of signals or images. I will show that most forms of spline fitting (interpolation, least squares approximation, smoothing splines) can be performed most efficiently using recursive digital filters. I will discuss the connection between splines and Shannon's sampling theory. I will also look at their multiresolution properties which make them prime candidates for constructing wavelet bases and computing image pyramids. I will provide multiple illustrations of their use in image processing; these include zooming and visualization, geometric transformation, registration, contour detection, as well as snakes and contour modeling.

VOLUME I

TueAmOR1
Hall A
SOURCE SEPARATION
Chair: P. Comon, *I3S Laboratory, France*

09:20

BLIND SEPARATION OF INSTANTANEOUS MIXTURE OF SOURCES VIA THE GAUSSIAN MUTUAL INFORMATION CRITERION

Dinh Tuan Pham, Laboratory of Modeling and Computation, IMAG, C.N.R.S.

In this paper, a method for blind separation of instantaneous mixture of colored sources is proposed. It is based on the minimization a Gaussian mutual information criterion, which leads to the problem of joint approximate diagonalization, without orthogonality constraint, of a set of estimated spectral density matrices. It is shown that separation is achievable (up to a scaling and a permutation) if no pair of sources can have proportional spectral densities. An efficient algorithm for the joint approximate diagonalization of positive matrices is briefly described and on-line processing methods are discussed. Theoretical results on the consitency and the asymptotic normality of the separator are given and some simulations are performed showing good agreement with the theory. It is seen that nearly optimal performance can be attained by jointly diagonalizing only a few spectral matrices.

09:40

DIFFERENTIAL SOURCE SEPARATION: CONCEPT AND APPLICATION TO A CRITERION BASED ON DIFFERENTIAL NORMALIZED KURTOSIS

Yannick Deville, Mohammed Benali, University of Toulouse, France

This paper concerns the underdetermined case of the blind source separation problem, i.e. the situation when the number of observed mixed signals is lower than the number of sources. The general concept that we propose in this case consists of a differential source separation approach, which uses optimization criteria based on differential parameters, so as to make some sources invisible in these criteria and to perform an exact separation of the other sources only. We illustrate this partial source separation concept on a new criterion based on the "differential normalized kurtosis" that we introduce to this end. We then validate the performance of this criterion by means of experimental tests. Differential normalized kurtosis Differential source separation Higher-order statistics Partial source separation Underdetermined linear instantaneous mixtures

10:00

A NEW DIRECTION FINDING METHOD WITH UNKNOWN SENSOR GAIN AND PHASE

Pascal Chargé, Yide Wang and Joseph Saillard IRCCyN/SETRA, Ecole polytechnique - University of Nantes, France

In this paper we present an original subspace-based method for direction finding by an array of uncalibrated isotropic sensors. This iterative method has been developed in order to estimate both directions-of-arrival of non circular radiating signal sources and unknown gain and phase of sensors. The non circular sources assumption leads to an extended array data model. One of the benefits of this approach is that the proposed procedure works even in the case when the number of sources is larger than the number of sensors. Some computer simulation results are provided to illustrate the performance of the algorithm.

10:50

SOLUTION OF HIGH-DIMENSIONAL LINEAR SEPARATION PROBLEMS

F. Herrmann and A. K. Nandi, The University of Liverpool, UK

Blind source separation (BSS) has been one of the emerging research topics within the signal processing community in recent years. Particularly, the maximum squared kurtosis has been found to be a suitable criterion for many technical applications of BSS. Conventionally an elementary Givens rotation estimator is applied to all source pairs in a Jacoby-like algorithm. However, those methods suffer from an escalation of computational expenses as soon as the number of sources becomes large. This paper introduces a novel eigenvector deflation method. It allows the separation of complex and high-dimensional mixtures without such performance penalty.

11:10

OPTIMIZATION OF A SECOND-ORDER STATISTICS BLIND SEPARATION ALGORITHM FOR GAUSSIAN SIGNALS

A. Yeredor, Tel-Aviv University, Israel

The Second-Order Blind Identification (SOBI) algorithm (Belouchrani et al., 1997) is aimed at blind separation of static mixtures of stationary source signals with distinct spectra. It uses approximate joint diagonalization of empirical correlation matrices to estimate the mixing matrix. We show that SOBI's performance can be improved by transforming the joint diagonalization into a properly weighted nonlinear Least Squares problem. In the case of Gaussian sources, the optimal weights can be estimated consistently from the empirical correlation matrices. We demonstrate the substantial improvement by analysis and simulations.

11:30

FAST AND ROBUST DEFLATIONARY SEPARATION OF COMPLEX VALUED SIGNALS

Ella Bingham and Aapo Hyvärinen, Helsinki University of Technology, Finland

A fast and robust algorithm for the separation of complex valued signals is presented. It is assumed that the original, complex valued source signals are mutually statistically independent, and that the mixing process is linear. The problem is solved by the independent component analysis (ICA) model. ICA is a statistical method for transforming an observed multidimensional random vector into components that are mutually as independent as possible. Our fast, fixed-point type algorithm is capable of separating complex valued, linearly mixed source signals in a deflationary manner. The computational efficiency of the algorithm is shown by simulations. Also, a theorem on the local consistency of the estimator given by the algorithm is presented.

11:50

BLIND SOURCE SEPARATION APPLICATION TO SMART EDDY CURRENTS SENSOR FOR METAL TAG RECOGNITION

Regis Huez, Fabien Belloir and Alain Billat Université de Reims, France

We present in this paper a Blind Source Separation (BSS) application to the separation of metal objects located inside the ground. We first describe a smart sensor based on the eddy currents and its application to the identification of metal tags. This system is sensitive to the presence of metal objects near the metal tag which can alter the sensor response. We show how using the BSS techniques permit one to relese from defect perturbations. We describe the sensor modifications allowing the BSS application. The last part of this paper concerns the results obtained from a metal tag perturbed by a can.

12:10

ANALYTICAL BLIND DISCRETE SOURCE SEPARATION

Olivier Grellier and Pierre Comon, I3S Laboratory

In this paper, a new blind source separation algorithm is described. Its main feature is to be analytical, in other words it does not suffer from local minima. The proposed method uses the discrete character of digital sources, which yields a polynomial system. The estimation of the sources is shown to be equivalent to the computation of rank-one tensors that are found by means of the old method by Macaulay for the computation of resultant. Finally, computer simulations are presented and the performances are compared to the analytical CM algorithm by van der Veen.

TueAmOR2
Room 200
Biomedical Signal Processing
Chair: J. Malmivuo *Tampere University of Technology, Finland*

09:20

MODELLING SLEEP WITH GAUSSIAN MIXTURE MODEL BASED ON EYE MOVEMENTS AND DELTA-ACTIVITY

Mikko Koivuluoma, Alpo Värri, Tampere University of Technology, Finland
Arthur Flexer, Austrian Research Center of Artificial Intelligence, Austria

The automatic sleep analysis is difficult task. The signals needed to perform the analysis are at least EEG and EOG. In this paper, the analysis is made mainly by using the EOG signal, and only the part of the EEG signal, the delta-band. The Gaussian mixture model (GMM) is calculated from the signals after detection of the eye movements and the processing of the EEG signal. The results show the eye movements and the delta-activity can be used to make discrimination between different sleep stages, although some additional signals, for example EMG, might improve the analysis.

09:40

DETECTION AND LOCALIZATION OF EPILEPTIC BRAIN ACTIVITY USING AN ARTIFICIAL NEURAL NETWORK FOR DIPOLE SOURCE ANALYSIS

Rik Van de Walle, Gert Van Hoey, Paul Boon, Michel D'Havé, Ghent University, Belgium
Bart Vanrumste, Ignace Lemahieu, Kristl Vonck, Ghent University Hospital, Belgium

We present a method for the detection of focal epileptic spikes in the EEG (electroencephalogram). The method is based on the dipole source localization technique and provides a source location estimate for each detected spike. An artificial neural network is used for performing dipole source localization, in order to be able to perform the detection method in real-time. The sensitivity and specificity of the method is studied on the basis of real patient EEG. It is observed that the method is able to operate with a sensitivity and specificity of 80% and 60%, respectively. We conclude that the method is suitable for real-time preprocessing of long-term EEG recording, in order to extract the relevant EEG epochs for subsequent source analysis.

10:00

SIGNAL FIDELITY REQUIREMENTS IN DETERMINING MEASURES OF CARDIAC FUNCTION - ADVANCING TO MULTI-CHANNEL IMPEDANCE CARDIOGRAPHY

Kauppinen, Hyttinen and Malmivuo, Tampere University of Technology, Finland
Kööbi and Kaukinen, Tampere University Hospital, Finland

Cardiac output (CO) is one of the core parameters in assessing the status of the heart. Impedance cardiography (ICG) has been introduced in the 1960's as a simple and noninvasive measurement of CO. However, measured data remain controversial, leading sometimes to errors and discouraging utilization of the approach. This work presents an investigation of the measurement properties of new ICG measurement configurations based on computerized application of the lead field theory in realistically shaped volume conductor models of the human thorax. Preliminary clinical experience was acquired by making recordings in healthy volunteers and valve patients. To improve reliability of ICG, a large number of different measurements should be taken and analyzed instead of a single signal as in conventional ICG. Comparison of simulations with parameters derived from clinical data correlated favorably, indicating the potential of the theoretical method in developing ICG measurement configurations. Moreover, clinical data suggested the likelihood that determination of different parameters - whether standard ones such as CO or derived indices approximating cardiac function - can be based more solidly on physiological principles and not only on experimental data, although the multiregional signal origin remains. For more advanced experiments and signal analysis, a multi-channel ICG device should be implemented.

10:50

ANALYSIS OF VOCAL DISORDERS IN A FEATURE SPACE

Lorenzo Matassini, Max-Planck-Institut, Germany

This paper provides a way to classify vocal disorders for clinical applications, thanks to the idea of geometric signal separation in a feature space. It is well known that the human voice source generates complex signals including subharmonics and toroidal oscillations. Typical chaotic quantities - like the entropy and the dimension of the attractor - together with autocorrelation function, power spectrum and other conventional measures are analysed in order to provide entries for the feature vectors. We report on a successful application of the geometrical signal separation in distinguishing between normal and disordered phonation. Both qualitative and quantitative results are presented. This approach can be applied as far as the post-operatory evolution and possible rehabilitation are concerned, which are commonly performed on a subjective basis only. Finally, glottal functionality can be analysed by means of objective indexes other than visual inspection of the spectrogram.

11:10

ECG SIGNAL DENOISING USING WAVELET DOMAIN WIENER FILTERING

Nikolay Nikolaev and Atanas Gotchev, Institute of Information Technology and Tampere International Center for Signal Processing, Finland

A two-stage algorithm for suppression of electro-myogram (EMG) artifacts from the electrocardiogram (ECG) using Wavelet Domain Wiener Filtering has been investigated. An improvement of the traditional technique is proposed by involving Time-frequency dependent threshold for calculation of the pilot estimate in the first stage. The appropriate choice of the wavelet basis functions used in each stage has been stressed. The strong relationship between the wavelet function's support and the ECG morphology has been emphasized. The preliminary assumptions have been argued by experiments on a wide range database. They have shown that an appropriate choice of the decomposing wavelets for the two algorithm stages can considerably improve the quality of the denoised signal.

11:30

COMPARISON OF FUZZY REASONING AND AUTOASSOCIATIVE MLP IN SLEEP SPINDLE DETECTION

E. Huupponen, A. Värri, M. Lehtokangas, J. Saarinen, Tampere University of Technology, Finland
J. Hasan, S-L. Himanen, Tampere University Hospital, Finland

Sleep spindles are important short-lasting waveforms in the sleep EEG. They are the hallmarks of the so-called Stage 2 sleep. Automated methods for spindle detection presented in literature typically use some form of fixed spindle amplitude threshold. The problem with that approach is that it is poor against inter-subject variability in spindle amplitudes. In this work a spindle detection method without an amplitude threshold was considered. Two versions of the method were compared as fuzzy reasoning and an Autoassociative Multilayer Perceptron (A-MLP) network were both employed for the classification between sleep spindles and non-spindle EEG activities. A novel training procedure was developed to remove inconsistencies from the training data of the A-MLP. This improvement of training data was found to have a positive effect on the method performance on the test data. However, in this comparison the fuzzy reasoning produced a better spindle detection result, probably due to the small size of the A-MLP.

11:50

ANALYSIS OF SINGING DURING EPILEPTIC SEIZURES BASED ON PARTIAL COHERENCE

R. Le Bouquin Jeannès ans G. Faucon, Université de Rennes, France
F. Bartolomei and P. Chauvel, Faculté de médecine, France

This paper deals with the analysis of singing during epileptic seizures. This phenomenon is rare and seems to appear when the origin of the seizure is the temporal lobe. Analyzing the relations

between signals belonging to different structures may contribute to define the role of each of them and derive a neural network involved during singing production. To study the relations along time, we choose to estimate the coherence function and the partial coherence function. The results obtained show that some coherences increase during singing phenomenon, and differences appear between given coherences and the same coherences conditioned on a third signal. This last result inform us of the contribution of a signal in the relation between two others.

12:10

MATHEMATICAL MODEL OF A SMALL NEURON

Marja-Leena Linne, Tuula O. Jalonen, University of Tampere, Finland

Cerebellum is important in controlling, fine-tuning and predicting movements. Here a mathematical model of the cerebellar granule neuron was developed to help to explain the dynamics of the non-linear ionic currents underlying the neuronal excitability. The model neuron was able to reproduce the experimentally recorded firing pattern, when six voltage- and time-dependent ion channel types and simplified calcium dynamics were implemented. A model of cerebellar learning, based on most recent physiological data from single neurons, may also be used for engineering applications, for example for adaptive movement prediction when designing robots.

TueAmSS1
Small Audiotorium
3rd Generation Mobile Communications
Chair: M. Latva-aho *Oulu University, Finland*

09:20

INVESTIGATION OF MULTIPLE ACCESS INTERFERENCE WITHIN UTRA-TDD

Gordon J R Povey, Elektrobit, UK

This paper highlights inter-cell and inter-operator interference which can occur within a UTRA-TDD system. Interference simulation results show that when the network has frame synchronisation and operates with the same asymmetry in all cells, there will be interference between the base stations and adjacent mobiles only (i.e. MS-BS and BS-MS interference). When synchronisation is lost, or different asymmetries are used, the results indicate the extent of the additional interference between adjacent base stations (BS-BS interference) and between adjacent mobile terminals (MS-MS interference). Further results for inter-operator interference shows that the interference at the mobiles caused by synchronised operators is dominated by interference from the base station, whereas for unsynchronised networks the MS-MS interference becomes significant.

09:40

PERFORMANCE OF MULTIUSER DETECTION IN TD-CDMA UPLINK

Heli Väätäjä, Pauli Kuosmanen, Tampere University of Technology, Finland
Markku Juntti, University of Oulu, Finland

Paper compares the performance of linear and interference cancellation type of receivers in UTRA TDD uplink, which is based on TD-CDMA. Overview of the physical layer of the UTRA TDD mode is given. Performance of the zero-forcing detector, linear minimum mean squared error detector and one-, two-, and three-stage HD-PIC are compared with the conventional Rake receiver. Simulations show that the advanced receivers give a clear gain compared to the conventional Rake receiver in UTRA TDD. The difference in the performance between linear and interference cancellation type of receivers is small in the operation area of 5-10% BER. The usage of variable spreading factors in UTRA TDD uplink favors the usage of the LMMSE type of detector. Simulations show that as the spreading factor gets lower, the difference in performance between LMMSE and HD-PIC receivers becomes more distinct.

10:00

MMSE ADAPTIVE RECEIVER FOR UTRA TDD

D. García-Alís, R.W. Stewart, University of Strathclyde, UK

A modified implementation of the minimum mean squared error (MMSE) receiver to be used in a UTRA time division duplex (TDD) environment is presented. Simulations are carried out in a multipath fading environment and BER values are obtained, proving that this structure reduces the mulitple access interference (MAI) that appears due to the loss of orthogonality between orthogonal codes in multipath channels. The performance of the proposed structure is compared with that of a traditional correlator, a linear equaliser and a RAKE receiver.

10:50

KALMAN-FILTER BASED CHIP ESTIMATOR FOR WCDMA DOWNLINK DETECTION

S. Werner, Helsinki University of Technology, Finland
M.L.R. de Campos, COPPE/Federal Univ. of Rio de Janeiro, Brazil
J.A. Apolinario Jr. Escuela, Politecnica del Ejercito, Ecuador

In the WCDMA downlink, the multipath channel destroys orthogonality between users and causes multiple access interference (MAI). This channel-induced MAI can be combated by channel equalization to approximately restore orthogonality. This paper presents a chip-estimator method based on Kalman filtering suitable for WCDMA downlink. The method does not require a training sequence or the knowledge of other-users' spreading codes. Simulation results show considerable performance improvement using the proposed estimator compared with that of the RAKE receiver.

11:10

SINR MAXIMIZING EQUALIZER RECEIVER FOR DS-CDMA

Massimiliano Lenardi, Dirk T.M. Slock, Institut Eurecom, France

The conventional receiver for DS-CDMA communications is the RAKE receiver which is a (linear) matched filter (MF), matched to the operations of spreading, pulse shape filtering and channel filtering. Such a MF maximizes the Signal-to-Interference-plus-Noise Ratio (SINR) at its output if the interference plus noise is white noise. This may be approximately the case if user-dependent scrambling (aperiodic spreading) is used. However, if no scrambling (hence the spreading is periodic) or only cell-dependent scrambling is used, then the interference exhibits cyclostationarity with symbol period and hence is far from white noise. In that case, the SINR at the output of a RAKE receiver can be far from optimal in the sense that other linear receivers may perform much better. In this paper we propose a restricted class of linear receivers that have the same structure as a RAKE receiver, but the channel MF and the pulse shape MF get replaced by equalizer filters that are designed to maximize the SINR at the output of the receiver. The complexity of the equalizer filters is variable and can possibly be taken to be as low as in the

RAKE receiver, while its adaptation(s) guarantees improved performance with respect to the RAKE receiver. The adaptation of the SINR maximizing equalizer receiver can be done in a semi-blind fashion at symbol rate, while requiring the same information (channel estimate) as the RAKE receiver.

11:30

BIT-INTERLEAVED TURBO-CODED MODULATIONS FOR MOBILE COMMUNICATIONS

Stéphane Yves Le Goff, Etisalat College of Engineering, United Arab Emirates

This paper deals with the study of bandwidth-efficient coding schemes for mobile communications. They are designed from the bit-interleaved coded modulation approach. Since turbo coding technique is utilized for correcting errors, these schemes are termed bit-interleaved turbo-coded modulations (BITCMs). Their structure is presented in a detailed manner and their bit error rate (BER) performance over Rayleigh fading channels is investigated for spectral efficiencies ranging from 2 to 7 bit/s/Hz. Simulation results show that BITCMs are attractive for mobile communications and, in particular, achieve performance being within 1.1 dB of their respective channel capacity at a BER of 10-5.

11:50

IDENTIFICATION OF NONLINEAR SATELLITE MOBILE CHANNELS USING VOLTERRA FILTERS

Souad Meddeb and J.Y. Tourneret, ENSEEIHT / TESA, France

This paper addresses the problem of nonlinear Satellite Universal Mobile Telecommunication System (S-UMTS) identification. The satellite channel is structured as a time-invarying (TIV) filter, followed by a zero-memory nonlinearity (ZMNL) and a TIV linear filter. The nonlinearity distortion is due to the on-board satellite amplifier. The mobile channel is modeled by a Time-varying (TV) structure which describes the multipath effects. This paper shows that a specific class of TV Volterra series (with odd kernels) can model this specific satellite mobile bandpass channel. A least squares estimate of the TV Volterra kernels with finite memory is derived for any arbitrary channel input.

12:10

THE POWER SPECTRUM OF A GENERALIZATION OF FULL-RESPONSE CPM

Jeffrey O. Coleman, Naval Research Laboratory

The power spectrum is derived for a generalization of full-response continuous phase modulation (CPM). The derivation is simpler than those previously published for CPM, the result is cleaner, and the more-general signal class may enable improvement on CPM at negligible cost.

TueAmSS2
Studio
Color Image Processing
Chair: S. Sangwine *The University of Reading, UK*

09:20

ADAPTIVE FILTERS FOR COLOR IMAGE PROCESSING: A SURVEY

A.N. Venetsanopoulos and K.N. Plataniotis, University of Toronto, Canada

Processing color images using nonlinear filters has received increased attention lately due to its importance in applications, such as multimedia technologies and telecommunications. The objective of this paper is twofold: (1) to introduce adaptive filtering techniques to the reader who is just beginning in this area and (2) to provide a review for the reader who may be well versed in nonlinear signal processing. The perspective of the topic offered here is one that comes primarily from work done in order statistics and adaptive filters for color image processing. Hence, many of the techniques and publications cited here relate to image filtering with the emphasis placed primarily on filtering algorithms based on fuzzy concepts, multidimensional scaling, and order statistics-based designs. It should be noted, however, that color image processing is a very broad field and thus contains many other approaches that have been developed from different perspectives, such as transform domain filtering, classical least-square approaches and neural networks, just to name a few. In this paper, we present a general formulation based on fuzzy concepts, which allows the use of adaptive weights in the filtering structure and we discuss different filter designs. The strong potential of fuzzy adaptive filters for color image processing is illustrated with several examples.

09:40

ENHANCEMENT OF MULTICHANNEL SIGNALS BY HISTOGRAM MODIFICATION

Akira Taguchi and Yoshinori Ikada, Musashi Institute of Technology

While histogram modification has traditionally been used for the enhancement of color images, it has usually been restricted to processing the luminance component. In the case of the RGB color space, we perform the independent histogram equalization of the R, G, and B components. However, these methods can't use the full color space. Several RGB histogram explosion algorithms are proposed. The histogram decimation (HD) method is also one of the RGB histogram explosion algorithms. We point out the defects of the HD method and propose the modified HD method, in order to conquer the defects of the HD method. The proposed method realize almost an ideal enhancement result (i.e., the distribution of transformed signals' histogram is almost uniform distribution).

10:00

COLOUR-SENSITIVE EDGE DETECTION USING HYPERCOMPLEX FILTERS

Carolyn J. Evans and Stephen J. Sangwine, The University of Reading Whiteknights, UK
Todd A. Ell, 5620 Oak View Court, Minnesota, USA

Colour edge detection procedures usually aim to find significant discontinuities between all adjacent homogeneous image regions, no matter what the colour. In this paper we present a colour edge detection procedure designed to find edges between homogeneous regions of particular colours. Initially the three-dimensional RGB colour space is modelled as the imaginary subspace of a four-dimensional, hypercomplex space. A hypercomplex filter is then used to highlight edges between a region of colour C_1 to a region of colour C_2. A 4-D, thresholding criterion is derived to identify the image pixels belonging to these edges. We demonstrate that this procedure can successfully find particular coloured edges in both synthetic and natural images.

10:50

A HISTOGRAM BASED MULTICHANNEL FILTER

Fotinos, G. Economou, E. Zigouris and S. Fotopoulos, University of Patras, Greece

A multidimensional filtering technique is proposed using fuzzy logic ideas and based on local statistics. The local multivariate histogram of the mutlichannel image is computed using the Parzen estimation technique. The maximum and minimum of the histogram are used as parameters, to describe the signal shape. The method is organized around a fuzzy control system. Experimental results, in true color images, show that the proposed technique suppresses different types of noise and preserve the image details better than other popular techniques.

11:10

COLOR IMAGE WAVELET COMPRESSION USING VECTOR MORPHOLOGY

Martha Saenz, Edward J. Delp, Purdue University West Lafayette, USA
Rusen Öktem, Karen Egiazarian, Tampere University of Technology, Finland

In this paper we explore a wavelet compression scheme for color images that uses binary vector morphology to aid in the encoding of the locations of the wavelet coefficients. This is accomplished by predicting the significance of coefficients in the sub-bands. This approach fully exploits the correlation between color components and the correlation between and within sub-bands of the wavelet coefficients. This compression scheme produces images that are comparable in quality to those of color zerotree tree encoders at the same data rate but is computationally less complex.

11:30

OBJECTIVE METRIC FOR COLOUR IMAGE COMPARISON

Didier Coquin, Philippe Bolon, Alexandru Onea, LAMII/CESALP - Université de Savoie, France.

In this paper, a method for colour image comparison is proposed. An indexation step is performed using a Kohonen neural network, with the SOM (Self Organizing Map) algorithm. Therefore, a colour image in the Red, Green, Blue (RGB) space is interpreted as a 256 colour image. The comparison step, between two indexed images, is based on a global dissimilarity measure, which is an extension of Baddeley's distance, adapted to this colour point set. Experimental results obtained with real images are presented.

11:50

A NOVEL TECHNIQUE FOR DATA HIDING IN COLOR PALETTED IMAGES

Lale Akarun, Nilay Özdilek, B. Uygar Öztekin, Bogazici University, Turkey

The size of an uncompressed image depends on the resolution of the image and the number of colors in it. Gray-scale images typically contain 256 or fewer gray levels. In a color image, the color of each pixel is represented by three bytes, one each for red, blue and green. This leads to a potential of more than 16 million colors. However, the eye can discern only about 10,000 distinct colors. Moreover, common images usually contain fewer colors. Many images, therefore, contain much fewer colors, which form a palette. Many file formats treat the information in such a color image as a table containing the palette and individual pixels as pointers to that table. In this paper, a novel technique for hiding data using the palettes of color images is proposed. The technique can be implemented without interfering with the image data, by using unused entries or visually indistinguishable colors in the image palette.

12:10

FANN-BASED SUBSAMPLING OF COLOR VIDEO SEQUENCES

Adriana Dumitras and Faouzi Kossentini, University of British Columbia, Canada

In this paper, we present a subsampling method for color video sequences using feedforward artificial neural networks. In our method, subsampling of the high detail and smooth regions of the video chrominance frames is performed using pattern matching, by extracting different geometrical information for each region. Experimental results show that our method outperforms spatial subsampling obtained via lowpass filtering and decimation both objectively and subjectively. Moreover, our method significantly outperforms other FANN--based subsampling methods, especially at very low bit rates.

TueAmSS3
Hall B
Multimedia Indexing, Browsing and Retrieval
Chair: S. Panchanathan *Arizona State University, USA*

09:20

AN INDEXING AND BROWSING SYSTEM FOR HOME VIDEO

Wei-Ying Ma and HongJiang Zhang, Hewlett-Packard Laboratories

In this paper, we present a content-based video indexing and browsing system for home video. In addition to traditional video parsing to extract temporary structure and abstraction of a video, we also utilize face detection, tracking and recognition techniques to create an organization based on the existence of people in a video. The entire parsing process is designed in the way that it takes full advantage of the information in MPEG compressed data so that it can be performed very efficiently. The system also provides semi-automatic tools for semantic annotation and fully automatic content-based retrieval.

10:00

MULTIPLE MEDIA CUES FOR MPEG-7

B.J. Brown, K. Derom, A. Lindsay, C. Saraceno, Starlab Brussels, Belgium

This work presents a methodology to extract and represent the semantic content of audio-visual documents. A collection of diverse tools is used to extract low level, signal based descriptions. Joint audio and visual analysis is utilized to automatically extract higher level semantic features. High-level, hand-annotated, descriptors are also used. The hand annotated descriptors are used for retrieval purpose as well as to enhance the results of the automatic procedure, i.e. to allow the system to learn how high level semantic information are linked to low level automatically extracted features through user's input. We draw upon MPEG-7's collection of Descriptors to provide some targets for our audio and visual analysis methods. Selected MPEG-7 Description Schemes, such as the {\it textual description}, the {\it description of persons}, and the {\it description of the structural aspects of the content} of the AV document \cite{MDSWD}, provide some of the larger containment structures for our features.

10:50

CONTENT-BASED DESCRIPTION OF IMAGES FOR RETRIEVAL IN LARGE DATABASES: MUVIS

Mejdi Trimeche, Faouzi Alaya Cheich, Moncef Gabbouj and Bogdan Cramariuc Tampere University of Technology, Finland

The rapid increase in the size of digital image and video collections is urging for the development of efficient browsing and search tools that skip the subjective task of keyword indexing, paving the way for the ambitious and challaging idea of content based description of imagery. With this goal in mind the MUVIS system attempts to provide an integrated solution for the indexing and retrieval of images based on their content within large databases. In this paper we describe briefly the overall structure of the system and expose the promising results obtained so far, demonstrating the similarity retrieval capabilities of the system based on seperate image features.

11:10

CONTENT-BASED ACCESS TO MULTIMEDIA EDUCATIONAL LIBRARIES

Paulo Villegas, Elena Nistal, Telefónica I+D, Spain
Isidro Aguado, Miguel Roser, Telefónica de España, Spain
Narciso García, Guillermo Cisneros, Universidad Politécnica de Madrid, Spain

The availability of huge amounts of digital multimedia information introduces new challenges for the efficient representation of the data and the access to it. This document describes a system designed to allow such an access for a concrete type of information (educational video material) with a defined architecture (a client-server model over standard IP networks). The objective is to create a platform that supports applications focused to ease distance learning by the creation of multimedia courses, stored in servers and accessed by client platforms through networks with different bandwidths. Systems with capabilities like the ones described in this paper are deemed to achieve an increasing importance due to the foreseeable on-line availability of vast amounts of multimedia content. The work currently undergoing in standardization bodies (like ISO/MPEG) will make possible the necessary interoperability among systems. We begin the paper by describing the system architecture and the way the video content is processed, then proceed to give an overview of some details on the description scheme used for meta-information, and finally comment the procedure used by the client to access and recover the content.

11:30

SEMANTIC VIDEO INDEXING USING MPEG MOTION VECTORS

A. Bonzanini, R. Leonardi, P. Migliorati, University of Brescia, Italy

With the diffusion of large video databases and "electronic program guides", the problem of semantic video indexing is of great interest. In literature we can find many video indexing algorithms, based on various types of low-level features, but the problem of semantic video

indexing is less studied and surely it is a great challenging one. In this paper we present a particular semantic video indexing algorithm based on motion information extracted from MPEG compressed bit-stream. This algorithm is an example of solution to the problem of finding a semantic event (scoring of a goal) in case of specific categories of TV programmes.

11:50

A NEW ALGORITHM FOR SHOT BOUNDARY DETECTION

Yousri Abdeljaoued, Touradj Ebrahimi, EPFL Signal Processing Laboratory, Switzerland
Charilaos Christopoulos, Ignacio Mas Ivars, MediaLab, Sweden

This paper presents a new approach to the detection of shot boundaries. The video is characterized through the tracking of feature points, extracted from the video sequence. The rate of the feature points that are lost or initiated is used as a criterion for shot boundary detection.

12:10

PERFORMANCE-OPTIMIZED FEATURE ORDERING IN CONTENT-BASED IMAGE RETRIEVAL

Horst Eidenberger, Austrian Libraries Network, Austria
Christian Breiteneder, Institute for Software Technology, Austria

We present a method to improve the performance of content-based image retrieval (CBIR) systems. The idea is based on the concept of query models, which generalizes the notion of similarity in multi-feature queries. In a query model features are organized in layers. Each succeeding layer has to investigate only a subset of the image set the preceding layer had to examine. For the purpose of performance acceleration we group features into two types: features for quick elimination of rather not similar images and features for the detailed analysis of result set candidates. Performance optimization is based on a model for predicting the number of images to be retrieved and on a model describing relationships between features. Results in our test environment show significant reduction of query execution time.

TueAmPO1
Exhibition Hall
Speech Analysis and Enhancement
Chair: P. Alku *Helsinki University of Technology, Finland*

SPEECH SIGNAL RECOVERY IN WHITE NOISE USING AN ADAPTIVE KALMAN FILTER

M. Gabrea and D. O'Shaughnessy, INRS-Téléecommunications, Montreal, Canada

This paper deals with the problem of Adaptive Noise Cancellation (ANC) when only a corrupted speech signal with an additive Gaussian white noise is available for processing. All the approaches based on the Kalman filter proposed in the past, in this context, operate in two steps: they first estimate the noise variances and the parameters of the signal model and secondly estimate the speech signal. We propose a new method to estimate the noise variances. This estimation is made by reformulating and adapting the Mehra approach.

DETECTING RELEVANT ACOUSTIC EVENTS FOR PILOTING IMPROVEMENT OF INTELLIGIBILITY

Vincent Colotte and Yves Laprie, Loria – France

This paper presents a speech signal transformation which slows down speech signals selectively and enhances some important acoustic cues. This transformation can be used for hearing aids and also for second language acquisition by facilitating oral comprehension. Selective slowing down relies on the use the TD-PSOLA synthesis method. The strategy used to control slowing down exploits a spectral variation function which locates rapid spectral changes. The enhancement simply consists of amplifying stop bursts and unvoiced fricatives. These acoustic cues are detected automatically through the examination of energy criteria. This approach was evaluated in the context of second language acquisition, more precisely by evaluating improvements in oral comprehension through the transformation, the detection and a perceptive experiment. The detection of relevant events gives good results and experiments show that the oral comprehension is improved.

ALL-POLE SPECTRAL MODELLING OF VOICED SPEECH WITH A HIGHLY COMPRESSED SET OF PARAMETERS

Susanna Varho and Paavo Alku, Helsinki University of Technology, Finland

Quantisation of low-order all-pole models for the speech spectrum is analysed in the present study by comparing a new linear predictive method, Regressive Linear Prediction with Triplets (RLPT), to conventional linear prediction (LP). Effects of scalar quantisation with LARs (log-area-ratios) were analysed using both the all-pole spectra and the residuals of the prediction. It appeared that with only four bits describing the predictors, RLPT is able to model two resonances of the speech spectrum while conventional LP typically models only the overall

structure of the spectrum.

SINGLE-CHANNEL NOISE REDUCTION WITH PITCH-ADAPTIVE POST-FILTERING

Jan Tilp, Darmstadt University of Technology, Germany

Single-channel noise reduction for speech enhancement is often applied in cellular and in hands-free telephones. For speech distortions to be minimal, single-channel systems based on spectral subtraction cannot entirely eliminate environmental noise. A relatively high spectral noise floor has to remain in the speech signal. For further reduction of annoying noise components, pitch-adaptive post-filtering is investigated in this paper. The basic idea is to attenuate during voiced parts of speech the spectral valleys between the pitch frequency and the harmonics. Simulation results for a spectral-subtraction scheme in conjunction with a pitch-adaptive post-filter are given. A frequency-domain subband filtering scheme is shown to be capable of enhancing speech signals disturbed by car noise.

SPEECH ENHANCEMENT BASED ON NOISE ADAPTIVE NONLINEAR MICROPHONE ARRAY

Hiroshi Saruwatari, Kazuya Takeda, Shoji Kajita, Fumitada Itakura, Nagoya University

This paper describes an improved complementary beamforming microphone array with a new noise adaptation algorithm for efficient speech enhancement. Complementary beamforming is based on two types of beamformers designed to obtain complementary directivity patterns each other. In this system, two directivity patterns of the beamformers are adapted to the noise directions so that the expectation values of each noise power spectrum are minimized. Using this technique, we can realize the directional nulls for each noise even when the number of sound sources exceeds that of microphones. To evaluate the effectiveness, speech enhancement experiments are performed using both computer simulation and actual devices in real acoustic environments with a two-element array and three sound sources. Compared with the conventional spectral subtraction method cascaded with the adaptive beamformer, it is shown that (1) the proposed array improves the signal-to-noise ratio by more than 6dB under nonreverberant condition, (2) improvement of 5.5dB is obtained when the reverberation time is 0.15sec, and (3) improvement of 4.3dB is obtained when the reverberation time is 0.30sec.

A TWO-SENSOR VOICE ACTIVITY DETECTION AND SPEECH ENHANCEMENT BASED ON COHERENCE WITH ADDITIONAL ENHANCEMENT OF LOW FREQUENCIES USING PITCH INFORMATION

Alexandre Guérin, Université de Rennes, France

This paper proposes a robust Two-Sensor Voice Activity Detector and a Speech Enhancement algorithm based on the Magnitude Square Coherence operator which expresses the normalized spectral cross-correlation of the two microphone signals. The zero-phase enhancement filter is derived from a modified power spectral subtraction called Cross-Power Spectral Subtraction.

The estimation of the power spectral densities is optimised to prevent the emergence of spectral subtraction technique main drawbacks : musical noise and reverberation. In a second part, the low frequencies are especially considered : indeed, the observations lead to the conclusion that this part of the speech spectrum, which mainly corresponds to voiced sections, is often under-estimated and removed by the gain filter due to the low SNR in these regions. Therefore, a pitch tracking algorithm is used and the knowing of the pitch frequency if combined with the gain filter to improve the estimation of the low frequencies.

A NEW APPROACH TO DEREVERBERATION AND NOISE REDUCTION WITH MICROPHONE ARRAYS

J.L. Sanchez-Bote, J. Gonzalez-Rodriguez, and J. Ortega-Garcia, Universidad Politecnica de Madrid, Spain

In this paper the speech enhancement abilities of a new array-based processor have been tested. The proposed system works in three cascade stages. First, the signals are time aligned with the estimated direction of the desired sound source. Second, the signal is decomposed in its all-pass and minimum-phase components using cepstral processing. In this moment, beamforming and liftering in cepstral domain is performed, where the output signal reverberation is reduced. The third part consists in a noise canceller, based in the optimum Wiener filtering and Coherence evaluation. The processor have been tested both with a real database and in simulated conditions of noise and reverberation. To evaluate its performance, we have considered the subjective perception of the output signal and objective measurements of LAR distances, Real Cepstrum distances and Signal to Noise Ratios

SPEECH ENHANCEMENT THROUGH BINAURAL NEGATIVE FILTERING

Pedro Gómez, Agustín Alvarez, Rafael Martínez, Víctor Nieto and Victoria Rodellar Universidad Politécnica de Madrid, Spain

Through the present paper the use of Negative Beamforming for Speech Enhancement is explored. This method may be used to eliminate or enhance a specific signal using a binaural array, and its extension to three-microphone arrays is also shown. The fundamentals of the technique are reviewed, showing that its performance is better than traditional microphone arrays as far as the angular selectivity is concerned. A structure to control and improve this selectivity is presented in arrays with three microphones. Results obtained from recordings produced in a real situation are shown and commented. Applications of this technique may be found in improving Noise Robust Speech Recognition Methods for Security Systems or Domotic Control.

A POSSIBILITY FOR IMPROVEMENT OF DIFFERENT SPEECH PROCESSING SYSTEMS

Goran S. Jovanovic, Institute for Applied Mathematics and Electronics, FR Yugoslavia

In this paper we investigate the possibility to improve performances of different speech processing systems by focusing the analysis on the speech signal segments named sub-phonemic acoustic targets. Presented is theoretical background necessary to realize this analysis. It includes appropriate symbolic description of the speech signal at word and sub-word level and formulation of IFC-guided analysis of the speech signal to localize sub-phonemic acoustic targets. New experimental evidence 'covering' several phonemic speech segments belonging to the same word confirmed the possibility to reduce speech signal processing only on the portions corresponding to sub-phonemic acoustic targets. Possible new research projects are suggested.

FREQUENCY-DOMAIN SEPARATION OF CONVOLVED NON-STATIONARY SIGNALS WITH ADAPTIVE NON-CAUSAL FIR FILTERS

Kyungmin Na, Kyung Jin Lee and Soo-Ik Chae, Seoul National University

In the frequency domain, a convolutive signal separation problem can be decomposed into multiple instantaneous separation problems that can be solved independently by an instantaneous separation algorithm. In this case, however, permutation indeterminacy in each frequency bin is a crucial problem. To solve this problem, a constrained gradient method has been suggested recently. In this paper, we propose an alternative procedure that realizes the same constrained gradient method with reduced computation. To invert a non-minimum phase system, non-causal finite impulse response (FIR) filters are employed in our realization. In addition, we derive a new instantaneous separation algorithm by minimizing the Hadamard inequality criterion with the natural gradient method. The algorithm has the equivariant property for uniform performance and belongs to orthogonal learning algorithms by nature. An improved separation was achieved for real-world speech-speech signals.

A MULTIRESOLUTION APPROACH TO CLOSED GLOTTIS INTERVAL DETERMINATION

Atanas Gotchev, Karen Egiazarian, and Tapio Saramaki Affiliation, Tampere University of Technology, Finland

We consider the closed glottis interval determination problem from the point of view of getting a multiresolution signal representation and searching for maxima in successive scales. Within this framework, an octave band decomposition, using half-band finite impulse response (FIR) filters is very promising, since those filters are less constrained than the orthogonal and bi-orthogonal wavelet filters. A number of experiments with synthesized signals argue the superior performance of the multiresolution approach in comparison with the traditional covariance approaches.

TueAmPO2
Exhibition Hall
Image Filtering, Restoration and Enhancement
Chair: J. Biemond ***Technical University of Delft, The Netherlands***

WAVELET TRANSFORM-BASED NOISE REDUCTION SCHEMES TO IMPROVE THE NOISE SENSITIVITY OF THE NAS-RIF ALGORITHM FOR BLIND IMAGE DECONVOLUTION

Supakorn Siddhichai and Jonathon A. Chambers, Imperial College of Science, UK

The nonnegativity and support constraints recursive inverse filtering (NAS-RIF) algorithm for blind image deconvolution is extremely sensitive to the presence of additive noise. We therefore propose three novel noise-reduction schemes based upon the discrete wavelet transform (DWT) denoising technique to incorporate within the NAS-RIF algorithm. Simulation studies show that such methods have the potential to increase the range of signal-to-noise ratios (SNR) over which the NAS-RIF algorithm can be applied.

A NEW APPROACH WITH IFS FOR IMAGE RESTORATION

Miki Haseyama, Megumi Takezawa, Junichi Miura, and Hideo, Hokkaido University, Japan

This paper proposes a new image-restoration method based on Iterated Function System (IFS). The proposed method can restore images contaminated by impulsive noise according to self-similarity represented by the IFS parameters. Since the IFS is usually used for image coding, it has never been applied to image restoration; and it cannot be utilized for the image restoration as it is. In order to adapt the IFS for image restoration, this paper reforms the conventional criterion for the computation of the IFS parameters to suit for image restoration, and as preprocessing we apply an epsilon-filter in which a median filter is embedded to contaminated images prior to computing the IFS parameters. Some simulation results are presented to demonstrate the effectiveness of this method.

DIRECTIONAL DIFFERENCE-BASED IMPULSE DETECTOR AND MEDIAN FILTER

Yuuhei Hashimoto, Yoshinobu Kajikawa and Yasuo Nomura, Kansai University, Japan

A novel median-based filter having the effectiveness in the removal of impulse noise with high probability is proposed in this paper. The proposed filter is a kind of switching scheme-based median filters, and consists of two main parts: one detects impulse noise (impulse detecotor), another removes impulse noise (noise removal filter). First, we propose a novel impulse detector based on the directional difference algorithm we have proposed. Next, a novel median-based filter, where the output of the directinal difference filter is introduced into the filtering operation, is proposed. Experiment results demonstrate that the proposed impulse detector and noise removal filter are better than the conventional detectors and median-based filters, respectively.

Moreover, we show through the restoration images that the proposed filter has effectiveness even for highly corrupted images.

SCATTER COMPENSATION IN DIGITAL RADIOGRAPHY

Vesa Onnia, Vesa Varjonen, Mari Lehtimäki, Mikko Lehtokangas and Jukka Saarinen, Tampere University of Technology, Finland

When an X-ray image is taken, interactions between tissues of the patient and X-rays cause scattered radiation. The detection of the scattered radiation causes degradation of the image quality. Very common technique for reducing scatter is the antiscatter grid. The grid is effective, but it can not remove all scattering. Another drawback of the grid is that the dose level must be increased, because of the attenuation caused by the grid. Larger the dose level is, larger the health risk became for the patient. Imaging device could be simplified and the dose level decreased if effects of the scattering could be reduced using computational image processing methods. This paper addresses the problem of the scatter compensation from the digital X-ray images. Our algorithm is based on maximum likelihood expectation maximization (MLEM) algorithm derived in. Modified version of this algorithm is presented in this paper. MLEM algorithm increases noise. Because of this SUSAN filter was used after MLEM. Our algorithm reduced scatter to 21% from its original value in the skull image. Also contrast and signal to noise ratio (SNR) were improved.

A NEW METHOD OF THE ENHANCEMENT OF GEL ELECTROPHORESIS IMAGES

Bogdan Smolka, Andrzej Swierniak, Artur BAL, Andrzej Chydzinski, Konrad Wojciechowski, Silesian Technical University, Poland

The gel electrophoresis is a method of separation of substances possessing electrical charge using the difference of their mobility in the electric field. The rate of migration of a fragment of a protein or DNA depends on the intensity of the constant electric field, on its resultant charge and the friction coefficient. The effect of the electrical force is dependent on the fragment shape, its volume, mass and the viscosity of the medium. The fragments which are smaller than the gel pores migrate easily, whereas the larger fragments are almost motionless. The problem with the analysis of gel images is that the bands can overlap, some of the bands are very weak and they are hardly visible. The gel images are very often corrupted by different kinds of noise, which makes their analysis difficult. As the standard software for analysing gel images are not very reliable, a human interaction is very often necessary. For this inspection, additional special tools are needed. In this paper the concept of a walking particle performing a self-avoiding random walk is introduced for the enhancement of gel electrophoresis images. Self-avoiding random walk (SAW) is a special walk along an m-dimensional lattice such that adjacent pairs of edges in the sequence share a common vertex of the lattice, but no vertex is visited more than once and in this way the trajectory never intersects itself. Using this approach a new algorith has been developed.The novel operator has the ability of image smoothing, while preserving image edges. It also eliminates impulse noise and performed in an iterative manner can be viewed as a segmentation algorithm.

A NOVEL TEMPORAL CONCEALMENT APPROACH USING A MESH-BASED MOTION COMPENSATION SCHEME

Luigi Atzori, Cristina Perra, University of Cagliari, Italy
Francesco G.B. De Natale, University of Trento, Italy

Concealment techniques aim at masking the visual effect of data losses and errors, by exploiting either spatially or temporally correlated information. Both temporal and spatial approaches present drawbacks: the first is in general inefficient in handling complex or fast objects' motion, while the second is computationally expensive and is not able to recover high-frequency contents and small details. In this paper a new solution is proposed that combines temporal and spatial approaches. The technique first replaces the lost block with the best matching pattern in a previously decoded frame, using the border information, and then applies a mesh-based warping that reduces the artifacts caused by fast movements, rotations or deformations. Experimental results show that significant improvements can be achieved in comparison with traditional spatial or temporal concealment approaches, in terms of both subjective and objective reconstruction quality.

BLIND IDENTIFICATION OF DEGRADATIONS USING ASCENDANT HIERARCHICAL CLASSIFICATION

Kacem Chehdi, Benoît Vozel, Marie-Paule Carton-Vandecandelaere, Nathalie Berric Lasti – Enssat, University of Rennes

Filtering and restoration aim to improve the quality of an image to facilitate its interpretation. Most of these algorithms require a priori information about the degradation affecting the image. In the context of blind image processing, when no a priori information is available, the nature of the degradation has to be identified from the observed image. Noise, blur and combinations of both are the sources of degradation considered here. We present a blind identification procedure which involves the computation of local statistics and the execution of a non parametric ascendant hierarchical classification algorithm. The method, tested on a set of 90 degraded image, allows to identify the nature of the degradation that produces the predominant effects. This result is essential in the definition of a strategy to decide how to process an image. When integrated in a filtering and restoration module, this step directs images to the most appropriate algorithm, for example a filter for noisy images and a restoration for blurred images. In the case of a combination of both sources of degradation, depending on the predominance of one source or not, a simple process or a combination of elementary processes can be applied.

ADAPTIVE MORPHOLOGY APPLIED TO GRAY LEVEL OBJECT TRANSFORMATION

O. Lavialle, D. Delord, P.Baylou, Equipe Signal/Image ENSERB et GDR-ISIS-CNRS, France

In this paper, we introduce a new approach to compute a directional transformation of gray-tone objects. The method is based on the definition of elementary forces exerted between the pixels. The dilation and the erosion are then obtained respectively by increasing and decreasing the membership function of the pixel according to the resultant force intensity. In order to emphasize

the anisotropic effect of this force, we introduce a penalization function depending on the angular dispersion of the resultants. The effects of adaptive dilation and erosion are illustrated on both synthesized and real objects.

ADAPTIVE DIRECTIONAL MORPHOLOGICAL OPERATORS

Romulus Terebes, Olivier Lavialle, Pierre Baylou, Equipe Signal/Image ENSERB et GDR-ISIS-CNRS, France
Monica Borda, Technical University of Cluj-Napoca, Romania
Ioan Nafornita, Politehnica University Timisoara

This paper proposes a new method of nonlinear filtering based on the classical morphological operators. The originality of our method consists in assigning to each pixel an adaptive structuring element depending on the local orientation of the image. Several examples of applications are presented and a comparison with the classical operators is done. Our method can be applied for early vision tasks.

A VERY EFFICIENT ALGORITHM FOR CORRECTION OF SKEWED DOCUMENTS

Sheung-On Choy, Yuk-Hee Chan, Shiu-Wing Hui and Wan-Chi Siu, The Hong Kong Polytechnic University, Hong Kong

This paper proposes a fast and efficient algorithm for rotating binary images. As compared to conventional algorithms, the proposed one saves around 90% of memory consumption. Moreover, it requires no floating-point multiplication, and the major operations in it are just memory access and simple integer addition. These make it very suitable for correction of skewed document images.

TueAmPO3
Exhibition Hall
Architectures and Implementations
Chair: L. Wanhammar *Linköping University, Sweden*

NEUROMATRIX(R) NM6403 DSP WITH VECTOR/MATRIX ENGINE

Dmitri Fomine, Vladimir Tchernikov, Pavel Vixne and Pavel Chevtchenko, Research Center MODULE, Moscow, Russia

The paper describes the architecture of the NeuroMatrix(R) NM6403 DSP designed for image processing, signal processing and neural networks emulation. The paper includes a brief description of the processor structure and its instruction set. The NM6403 is the first DSP based on NeuroMatrix(R) Core (NMC) comprises an original 32-bit VLIW RISC processor and a 64-bit SIMD Vector co-processor. In contrast to other modern general purpose DSPs and microprocessors with SIMD units such as: Texas Instruments c64xx, Intel Pentium MMX, Motorola AltiVec PowerPC G4 and Analog Devices TigerSHARC, the new DSP performs variable bit-length vector/matrix arithmetic, logic and saturation operations. The main NMC operation is matrix by vector multiplication. The NM6403 supports shared memory mode for two 64-bit external data buses. Two byte-width communication ports simplify the multiprocessor systems design. The NM6403 has been designed by RC "Module" (www.module.ru) and produced by Samsung 0.5mm CMOS technology. The peak performance - up to 14.400 MMACs has been achieved at a 50MHz clock rate, 3.3V operating voltage and PBGA256 package.

SCALABLE DSP REALIZATION OF WAVELET TRANSFORM IN IMAGE CODING

Kaisa Haapala, Pasi Kolinummi, Timo Hämäläinen and Jukka Saarinen, Tampere University of Technology, Finland

A scalable realisation of a two dimensional (2D) fast wavelet transform (FWT) is presented and compared to an earlier implementation of discrete wavelet transform (DWT), which uses matrix multiplication method. Parallelisation and mapping possibilities are analysed. The main emphasis is in minimising communication requirements and utilising local communication. Measured performance figures verify good performance and illustrate the benefits of parallel computation in Wavelet transform. The computation time of the FWT is only a fraction of the computation time of the previous implementation.

SCALABLE DSP IMPLEMENTATION OF DCT-BASED MOTION ESTIMATION ALGORITHM

Miia Viitanen, Pasi Kolinummi, Timo Hämäläinen and Jukka Saarinen, Tampere University of Technology, Finland

A parallel implementation of a DCT-based motion estimation algorithm (DXT-ME) for 2D images/signals is presented for a parallel scalable DSP system called PARNEU. PARNEU was used to test the performance of the parallelism. The DCT-based motion estimation can be used for video coding instead of the more frequently used full search block-matching approach (BKM-ME). The DCT-based system has lower computational complexity compared to the BKM-ME and it can result in a higher system throughput. Data parallel implementation scales very well and the performance measurements are promising compared to the traditional methods.

AN EFFICIENT RNS ARCHITECTURE FOR THE COMPUTATION OF DISCRETE WAVELET TRANSFORMS ON PROGRAMMABLE DEVICES

Javier Ramirez, Antonio Garcia and Antonio Lloris, University of Granada Campus Universitario Fuentenueva, Spain
Pedro G. Fernandez, University of Jaen Escuela Politecnica Superior, Spain

In this paper, a new RNS architecture to compute the 1-D DWT is introduced. It makes use of the relation between the coefficients of the low-pass and high-pass decomposition filters for orthogonal wavelet families to compute the transform with a minimum number of modular multipliers. These multipliers are based on pipelined LUTs (Look-Up Tables) and are used in consecutive cycles by each filter. Two modular adder trees operating at half of the sampling rate and controlled by out-of-phase clocks compute the approximation and detail sequences. A 6-tap Daubechies analysis filter bank was synthesized at structural level using VHDL. Altera FLEX10K FPL devices were considered to map binary 2's complement arithmetic and RNS solutions and to conduct performance simulations. Thus, a significant throughput improvement of up to 75% is achieved for the proposed RNS architecture using up to 8-bit length modulus set. It is shown that the selection of three 7-bit channels is optimum for an FPL implementation while four 6-bit RNS channels would be the best choice for a VLSI architecture in order to reduce chip area.

DESIGN AND EFFICIENT IMPLEMENTATION OF NARROW-BAND SINGLE FILTER FREQUENCY MASKING FIR FILTERS

Oscar Gustafsson, Håkan Johansson, and Lars Wanhammar, Linköping University, Sweden

In this paper a new structure for frequency masking FIR filters is introduced. By using identical model and masking filters (except for the periodicity) folding can be used to efficiently implement the filter by mapping all subfilters to a single hardware structure. The resulting structure will for most cases require fewer multipliers and adders compared to conventional narrow-band frequency masking techniques. The cost is an increase in the number of delay elements.

SYSTOLIZING THE ADAPTIVE DECISION FEEDBACK EQUALIZER USING A SYMBOLIC STATE SPACE FORMULATION

Mrityunjoy Chakraborty, Suraiya Pervin and Anindya S. Dhar, Indian Institute of Technology, India

A systolic array architecture for the adaptive decision feedback equalizer (ADFE) is proposed in this paper which is based on an algebra developed earlier by Kung et al. Two basic processing cells that are computationally equivalent and easy to realize are used to construct the main body of the array. To satisfy all the desirable features of a systolic array, the array needs to be operated at a clock speed twice that of input. The increase in clock speed can, however, be exploited to reduce the total number of adders and multipliers by about 50%.

BRIDGING NETWORK TRAFFIC BETWEEN WIRELESS AND WIRED LANS IN WINDOWS NT

Mauri Kuorilehto, Marko Hännikäinen, Timo Hämäläinen, and Jukka Saarinen, Tampere University of Technology, Finland

This paper presents two different alternatives for interconnecting Local Area Networks (LANs) in the data link layer. The solutions use a Windows NT workstation as a bridge between separate networks. Interconnection is provided by either implementing an intermediate driver or a protocol driver into the Windows NT networking stack. Both alternatives screen traffic according to the address fields of network frames and therefore unnecessary traffic can be avoided. Bridge will be used for interconnections between an Ethernet wired LAN, a standard IEEE 802.11 wireless LAN, and TUTWLAN that is a wireless LAN designed at Tampere University of Technology. The protocol driver is concluded to be a more practical alternative for implementing the bridging functionality in TUTWLAN because of its simpler structure and estimated higher performance.

A SYSTOLIC SERIAL SQUARER OF CONTINUOUS OPERATION

K. Z. Pekmestzi, N. Moshopoulos, P. Kalivas, National Technical University of Athens, Greece

A systolic serial squarer for unsigned numbers, which operates without zero words inserted between successive data words, outputs the full product and has immediate response, is presented. The systolic form is obtained by merging two adjacent multiplier cells, and the continuous operation is achieved by dividing the squaring procedure in two pipelined stages.

VERIFYING EXTERNAL DATA MEMORY INTERFACE FOR H.263 VIDEO DSP WITH MEMORY SIMULATOR

J. Alakarhu, J. Niittylahti, T. Sihvo and J. Tanskanen, Tampere University of Technology, Finland

In this paper, we present the simulator-based method to estimate the time required by the external data memory accesses in the H.263 video encoding. Different frame rates and picture resolutions are considered. The Video DSP structure considered here consists of several parallel on-chip DSP units and it is optimized for the H.263 video encoding. Execution time is coarsely divided between control/non-sequential processing, parallel processing, and external data memory traffic. To evaluate the performance in early design phase, one must find out the time required by each part. The memory simulator method described here gives an estimate of the time required by the external memory accesses. With this estimate, one can also make sure that the proposed partitioning between internal and external data memories is correct and the required memory bandwidth for the external data memory is not too high.

PIPELINED ARCHITECTURE FOR INVERSE DISCRETE COSINE TRANSFORM

Jari Nikara, Jarmo Takala, Jukka Saarinen, and Jaakko Astola, Tampere University of Technology, Finland
David Akopian, Nokia Mobile Phones, Finland

In this paper, a pipelined architecture for inverse discrete cosine transform (IDCT) is presented. Pipeline architectures are popular in parallel fast Fourier transform implementations but they are rare in IDCT implementations due to the irregularities in fast IDCT algorithms. The proposed architecture is derived by applying vertical projection to in-place IDCT algorithm. The resulting structure is modular and easy to pipeline. The word width requirements in the internal arithmetic are estimated to fulfil the requirements set by IEEE standard for 8*8 inverse cosine transform.

ANALOG ADAPTIVE MEDIAN FILTERS

Alejandro Díaz-Sánchez, National Institute for Astrophysics, Optics and Electronics, Mexico
Jaime Ramírez-Angulo, New Mexico State University, USA

The implementation of analog adaptive median filters for image processing is presented. The adaptive median filter is based on transconductance comparators, which saturation current is adapted to act as a local weight operator. Transistor level simulations, using 1.2mm parameters, have shown excellent results in high removing incidence noise. An image with 249 X 209 corrupted with 35% salt and pepper noise, is used to test the adaptive median filter. All the simulations were made using BSIM3 level 49 model and 1.2 mm MOSIS parameters.

EFFICIENT IMPLEMENTATION OF FIR FILTERS USING BIT-LEVEL OPTIMIZED CARRY-SAVE ADDITIONS

Kei-Yong Khoo, Zhan Yu and Alan N. Willson, Jr. , University of California, USA

A widely used technique to implement high-speed fixed-coefficients FIR filters in VLSI circuits is to use carry-save additions (CSA) to defer the time-consuming carry-propagations. This paper shows that by exploiting the unequal wordlength of the partial products generated by the fixed-coefficients, significant savings can be achieved by optimizing the carry-save representation at the bit-level. The key in our proposed technique is to allow for more than two signals per bit position, so that we gain flexibility in the bit-level implementation of CSA arrays. We have applied our algorithms to the optimization of high-speed digital FIR filters and have achieved 15% to 30% savings (weighted cost) in the overall filter implementation array in comparison to the standard carry-save implementation.

A REAL-TIME MONO-CHIP TRACKING ACTIVE IR TARGETS

Marius Vasiliu and Francis Devos, Paris-Sud University, France

We propose an original VLSI architecture to be implemented on a single chip focal plane array, dedicated to the detection and the tracking of cooperative active targets in unfriendly and noisy environment. The system we describe can detect IR targets very fast (in less than 1 ms) at signal-to-noise ratios close or below 0 dB. The basic principle is an asynchronous detection of pseudo-random IR target signal at each pixel level. Each pixel uses an analog circular convolution mechanism to extract useful information from input signal. The form and the length of pseudo-random patterns to be tracked are user defined and can be dynamically changed from outside of focal plane array. The spatial and the temporal resolution of the system can be balanced inside of some hardware limits, with respect of user requirements.

SYSTOLIC IMPLEMENTATION OF THE STEEPEST DESCENT ALGORITHM AND ITS PERFORMANCE IN CDMA RECEIVERS

R. Baghaie, Helsinki University of Technology, Finland

In code division multiple access receivers, in order to suppress the multiple access interference, different multiuser detectors such as the decorrelating and the linear minimum mean square error detectors can be utilized. In order to alleviate the implementation complexity, iterative implementation of such detectors have been reported. Two most popular iterative algorithms used in such detectors are the Steepest Descent (SD) and the Conjugate Gradient (CG) algorithms. In this paper, we first compare fixed-point and floating-point performances of these algorithms. Furthermore, hardware implementations of the algorithms are compared. Based on these comparisons, for the implementation of the SD algorithm a systolic architecture is proposed.

TuePmOR1
Hall B
Image Filtering, Restoration and Enhancement
Chair: G. Sicuranza *University of Trieste, Italy*

14:00

IMAGE SHARPENING USING PERMUTATION WEIGHTED MEDIANS

Marco Fischer, Jose L. Paredes and Gonzalo R. Arce, University of Delaware, USA

A class of robust weighted median (WM) sharpening algorithms is developed in this paper. Unlike traditional linear sharpening methods, weighted median sharpeners are shown to be less sensitive to noise or to image artifacts introduced by compression algorithms. The WM sharpening structure is extended to include data dependent weights under the framework of permutation weighted medians leading to sharpeners that have tunable robustness properties. A statistical analysis of the various algorithms is presented. Experiments illustrate the sharpening of images with compression artifacts and of images with background film-grain noise

14:20

A TUNABLE FUZZY FILTER BASED ON DIFFERENCES BETWEEN PIXELS

Mitsuji Muneyasu, Kouichiro Asou, and Takao Hinamoto, Hiroshima University, Japan
Akira Taguchi, Musashi Institute of Technology, Japan

This paper presents a new fuzzy filter structure for edge-preserving smoothing of an image corrupted by impulsive and white Gaussian noise. This filter structure is expressed as an adaptive weighted mean filter that uses fuzzy control. The coefficients of the proposed filter can be varied adaptively by fuzzy control laws based on differences between pixels in the window. The parameter of this filter can be adjusted by learning. Finally, simulation results demonstrate the effectiveness of the proposed technique.

14:40

SIGNAL AND IMAGE ENHANCEMENT BY A GENERALIZED FORWARD-AND-BACKWARD ADAPTIVE DIFFUSION PROCESS

G. Gilboa, Y.Y. Zeevi, N. Sochen, Technion-Israel Institute of Technology and The University of Tel-Aviv

Signal and image enhancement in the presence of noise is considered in the context of the scale-space approach. A modified dynamic process, based on the action of an adaptive diffusion equation, is presented. The nonlinear diffusion coefficient is locally adjusted according to image features such as edges, textures and moments, and, as such, can also reverse its sign, i.e. switches from a forward to a backward (inverse) diffusion process according to a given set of critera. This

results in a generalized forward-and-backward adaptive diffusion process that enhances features such as transients and singularities in the one-dimensional case, and edges in images, while locally denoising smoother segments of the signal or image. Advantages afforded by the generalized adaptive diffusion process are illustrated by examples of both one-dimensional signals and images.

15:00

NON LINEAR IMAGE ENHANCEMENT IN THE SPACE SCALE DOMAIN

L.Capodiferro, Fondazione Ugo Bordoni
M. Ciotti, G. Jacovitti, University of Rome, Italy

In this work, we present a technique designed for improving the subjective quality of still and video images presenting lack of sharpness. The technique employs a wavelet image representation based on edges extracted at different resolution levels by smoothed differential operators [1], and makes use of non-linear processing in the multiresolution edge domain. In this contribution, after a concise illustration of the theoretical basis supporting the principle of Lipschitz coefficient restoration, the developed algorithms are described in some detail. Optimal non-linear amplification laws are determined for specific and general uses. Examples of sharpness restoration for different kind of image degradation are provided.

15:50

AN IMPLEMENTATION FOR SWITCHING MEDIAN FILTERS BASED ON COUNTER PROPAGATION NETWORKS

Mitsuji Muneyasu, Takashi Ochiai and Takao Hinamoto, Hiroshima University, Japan

This paper proposes an implementation for switching median filters using counter propagation networks. The proposed filter consists of an impulse detector based on the counter propagation networks and a median filter. The median filter only removes the impulsive noise marked by the impulse detector and isn't applied to the other pixels. By using the proposed method, we can achieve to reduce the impulsive noise and keep the edges and details in an image. For the impulsive noise with high probability, this filter can be used repeatedly, then the noise can eliminate efficiently. Finally, several examples are given to illustrate the utility of the proposed technique.

16:10

IMPROVING MPEG-4 CODING PERFORMANCE BY JOINTLY OPTIMIZING BOTH COMPRESSION AND BLOCKING EFFECT ELIMINATION

Wan-Fung Cheung and Yuk-Hee Chan Affiliation,The Hong Kong Polytechnic University, Hong Kong

In most current image/video coding systems, the compression stage and the restoration stage operate separately and hence they cannot make use of each other to optimise the overall coding performance. In this research, we suggest modifying the basic structure of the encoding systems such that the contribution of the restoration to be performed can be taken into account in the compression by jointly optimising the two processes. An example is also provided to show how this idea works successfully when MPEG-4 deblocking filter is exploited in the post-processing stage.

16:30

SELF-AVOIDING RANDOM WALK APPROACH TO IMAGE ENHANCEMENT

Bogdan Smolka, Konrad Wojciechowski, Artur Bal, Andrzej Chydzinski, Andrzej Swierniak, Silesian Technical University, Poland

The paper presents a new technique of image enhancement. The described algorithm enables the suppression of noise and contrast enhancement. The interesting feature of this new algorithm is that its iterative usage leads to the image segmentation. The algorithm is based on a concept of a virtual particle, which performs a special kind of random walk - the so called self-avoiding random walk. The transition probabilities of the random walk are determined by the median distribution. Segmentation effect obtained using this method, together with its ability of the algorithm to eliminate impulse and Gaussian noise makes the new method an interesting pre-processing tool. To some extent this method is similar to anisotropic diffusion, but the major advatage of the proposed method is its ability to supress strong noise.

16:50

REJECTION OF SCRATCHES FROM VIDEO BY USE OF FILTERS WITH PREDICTION ERROR PROCESSING

Jamal K. Abbas, Marek Domañski, Poznañ University of Technology, Poland

This paper deals with denoising of color video sequences using improved median-based filters appropriate for real-time implementations. In the paper, it is proven that application of prediction error processing results in improved efficiency of video restoration. The basic idea is to predict a pixel value by using a nonlinear filter, and then to calculate the prediction error by a comparison with the input value that maybe is corrupted. The output value is a sum of the prediction and the prediction error processed according to a nonlinear function. The experimental results prove that such filters are appropriate for rejection of scratches from video even in the interlaced format. The computational overhead is less than 10% as compared to classic median-based filters.

17:10

A NEW LINEAR MODEL FOR IMAGE REPRESENTATION FOR USE WITH KALMAN FILTER RESTORATION

Tamer O. Diab, Banha higher Institute of Tech, Egypt
Ahmed M. Darwish, Cairo University, Egypt

When a Kalman filter is applied to an image for restoration purposes, the model of the original image affects the accuracy of the restoration. The model for effective restoration depends on the correlation of the original image and the variance of the noise. If these parameters are unknown, they have to be estimated from the observed image. In this paper, a method to estimate the unknown parameters in the image restoration process is proposed along with a method that identifies the region of support (number of pixels used for estimation and their positions). Three schemes were developed and used to represent the image and to identify the parameters for the filter. The approach has been tested on numerous images. Results show superior performance compared to methods and implementations previously reported in the literature both in terms of computational complexity and signal to noise ratio.

TuePmOR2
Hall A
Signal Interpretation
Chair: R. Tourki ***Ecole Nationale d'Ingenieurs de Monastir, Tunisia***

14:00

RECOGNITION OF ACOUSTIC NOISE MIXTURES BY COMBINED BOTTOM-UP AND TOP-DOWN PROCESSING

Jukka Sillanpää, Anssi Klapuri, Jarno Seppänen and Tuomas Virtanen, Tampere University of Technology, Finland

In this paper, a system is described for the recognition of mixtures of noise sources in acoustic input signals. The problem is approached by utilizing both bottom-up signal analysis and top-down predictions of higher-level models. The developments are made using musical signals as test material. Validation experiments are presented both for self-generated sound mixtures and for real musical recordings.

14:20

MDL BASED DIGITAL SIGNAL SEGMENTATION

Ciprian Doru Giurcaneanu, Ioan Tabus and Jorma Rissanen Tampere University of Technology, Finland

The segmentation of a signal based on a piecewise polynomial model is reexamined here in light of the recent advances in applying the MDL principle to the exponential density family. The critical case of discriminating between short segments is handled very efficiently by using the exact MDL formula for small sample size, whereas MAP or AIC methods result in drastic over-segmentation. The simulation result for ECG segmentation shows that the piecewise linear approximation obtained with the proposed method preserves well the location of the QRS complex.

14:40

DEFINITION OF INSTANTANEOUS FREQUENCY ON REAL SIGNALS

Sokol Saliu, Chalmers Lindholmen University College, Sweden

This paper introduces a definition of the instantaneous frequency (IF) on real signals. Basically, the definition avoids some pitfalls associated with the corresponding definition on analytic signals. It readily applies to the class of monocomponent frequency modulated signals and also accommodates the class of amplitude and frequency modulated signals. An interpretation of the IF is given for the latter and an extension of it to instantaneous parameters is discussed.

15:00

EMG-BASED SIGNAL PROCESSING SYSTEM FOR INTEPRETING ARM GESTURES

Osamah A. Alsayegh, Kuwait Institute for Scientific Research, Kuwait

This paper presents an electromyographic-based (EMG-based) signal processing system for interpreting human arm gestures. To retain a constraint-free user's environment, EMG sensing is limited to three arm muscles. EMG signals are processed to attain parameters that are related to the muscles temporal activities. The attainment of these parameters through time constructs a unique signature for each particular gesture. Experimental investigation was carried out to examine the system's reliability in recognizing 12 arm gestures. The results show that the system can recognize the 12 gestures with a success rate of 96%.

TuePmSS1
Studio
Use of Optimization in Filter Design and Related Applications
Chair: A. Antoniou ***University of Victoria, Canada***

14:00

DESIGN OF DIGITAL FILTERS AND FILTER BANKS BY OPTIMIZATION: A STATE OF THE ART REVIEW

W.-S. Lu and A. Antoniou, University of Victoria, Canada

The many advancements in the area of numerical optimization in conjunction with the ever-increasing power of computers have made optimization-based filter design an increasingly important field of research. In this paper, several recent optimization methods for the design of FIR and IIR digital filters and filter banks are reviewed.

14:40

DESIGN OF DIGITAL FILTERS AND FILTER BANKS BY OPTIMIZATION: APPLICATIONS

Tapio Saramäki and Juha Yli-Kaakinen, Tampere University of Technology, Finland

This paper emphasizes the usefulness and the flexibility of optimization for finding optimized digital signal processing algorithms for various constrained and unconstrained optimization problems. This is illustrated by optimizing algorithms in six different practical applications. The first four applications include optimizing nearly perfect-reconstruction filter banks subject to the given allowable errors, minimizing the phase distortion of recursive filters subject to the given amplitude criteria, optimizing the amplitude response of pipelined recursive filters, and optimizing the modified Farrow structure with an adjustable fractional delay. In the last two applications, optimization algorithms are used as intermediate steps for finding the optimum discrete values for coefficient representations for various classes of lattice wave digital filters and linear-phase FIR filters. For the last application, linear programming is utilized, whereas for the first five ones the following two-step strategy is applied. First, a suboptimum solution is found using a simple design scheme. Second, this start-up solution is improved by using a general-purpose nonlinear optimization algorithm, giving the optimum solution. Three alternatives are considered for constructing this general-purpose algorithm.

15:50

OPTIMAL IIR FILTERS WITH VARIABLE DELAY

Andrzej Tarczynski, University of Westminster

A new method of designing IIR filters with tuneable delay is presented. The proposed variable delay filters consist of a number of parallel fixed-coefficient IIR filters whose out-puts are weighted and added up. The weights are affected only by the required delay. In this paper they are chosen as simple polynomial functions of the delay but other selection are also possible. The fixed-coefficient filters are designed using Weighted Least Squares (WLS) approach. In order to guarantee stability of the designed filters the WLS optimisation criterion has been modified by linearly combining it with some time-domain components. These components are selected in such a way that all the local minima of the modified cost correspond to stable filters. The proposed method is illustrated with a design of a band-pass filter whose bulk delay can be modified by up to +/-0.5 of the sample time.

16:10

OPTIMAL FILTERS FOR RANGE-TIME SIDELOBE SUPPRESSION

Dan Scholnik, Naval Research Laboratory Washington, USA

Powerful convex optimization techniques can be applied to the design of the system pulse in radar systems. The flexibility of direct optimization allows control over such parameters as the beamwidth, maximum sidelobe level, and total sidelobe energy. Instead of designing the entire receive filter, a short filter is optimized in cascade with a matched filter, allowing the use of pulse-compression waveforms of arbitrary length. The extension to multiple-pulse synthetic-wideband waveforms is also presented.

16:30

EFFICIENT DESIGN OF WAVEFORMS FOR ROBUST PULSE AMPLITUDE MODULATION USING MEAN SQUARE ERROR CRITERIA

Timothy N. Davidson, McMaster University

The design of a pulse shaping filter which provides maximal robustness to an unknown frequency-selective channel is formulated as a convex optimization problem from which an optimal filter can be efficiently obtained. Robustness is measured by the worst-case mean square error of the data estimate over a class of deterministically bounded channels, and the optimization is subject to a constraint on the bandwidth of the filter. The design technique allows efficient exploration of design trade-offs between bandwidth, performance in an ideal channel and robustness to unknown channel distortion. It is used to design chip waveforms with superior performance to the waveform specified in a recent standard for digital mobile telephony.

16:50

A GENERAL-PURPOSE OPTIMIZATION TECHNIQUE FOR DESIGNING TWO-CHANNEL FIR FILTER BANKS

Robert Bregovic and Tapio Saramäki, Tampere University of Technology, Finland

An efficient general-purpose optimization algorithm is proposed for designing two-channel FIR filter banks. This technique can be used for optimizing two-channel FIR filters in all the alias-free cases proposed in the literature. The generalized problem is to minimize the maximum of the stopband energies of the two analysis filters subject to the given passband and transition band constraints and the given allowable reconstruction error. There-fore, in addition to the perfect-reconstruction filter banks, nearly perfect-reconstruction banks can be optimized in a controlled manner. The optimization is carried out in two steps. In the first step, a good starting-point filter banks for further optimization is generated using an existing design scheme for the selected class of filter banks. The second step involves optimizing the filter bank with the aid of the proposed algorithm. Several examples are included illustrating the efficiency of the proposed approach.

17:10

OPTIMAL DESIGN OF TWO-CHANNEL BIORTHOGONAL FILTER BANKS

W.-S. Lu and A. Antoniou, University of Victoria, Canada

A method for the design of signal-adapted 2-channel biorthogonal filter banks of finite length is presented. The design problem is formulated as a constrained optimization problem and is solved by converting it into an iterative line-search problem through a first-order parameterization of the perfect reconstruction constraint. It is shown that for the filter banks with analysis and synthesis filters of different lengths, a refinement of the algorithm is possible that leads to a solution in a very small neighborhood of a local minimize, which satisfies the PR constraint precisely.

TuePmSS2
Room 200
Perspectives on Nonlinear Filter Design
Chair: E. Dougherty *Texas A&M University, USA*

14:00

NONLINEAR ORDER STATISTIC FILTER DESIGN: METHODOLOGIES AND CHALLENGES

Moncef Gabbouj and Jaakko Astola, Tampere University of Technology, Finland

Linear filtering techniques have serious limitations in dealing with signals that have been created or processed by a system exhibiting some degree of nonlinearity or, in general, situations where the relevance of information cannot be specified in frequency domain. In image processing many of these characteristics are often present and it is no wonder that image processing is the field where nonlinear filtering techniques have first shown clear superiority over linear filters. Despite this success, nonlinear filtering, excepting a few special cases, is still more of an art than an established systematic engineering discipline. Considering the immensity of the area, it can never be such but still much can be done to establish and clarify the relations of different techniques and the assumptions behind them. The present paper is a reflection on the methodologies and challenges of nonlinear filter design with special emphasis on order statistics, polynomial and rational filter classes.

14:40

DESIGN OF STATISTICALLY OPTIMAL MORPHOLOGICAL OPERATORS

Junior Barrera, University of Sao Paulo
Edward R. Dougherty, Texas A & M University

This paper recalls a general framework for the design of optimal nonlinear digital filters: the statistically optimal morphological operators. Optimal filters are designed using the algebraic decomposition structure of Mathematical Morphology and observations of input-output pairs of signals or images.

15:00

DESIGN OF MORPHOLOGICAL FILTERS USING GENETIC ALGORITHMS

Stephen Marshall, David Greenhalgh, University of Strathclyde
Neal Harvey, Los Alamos National Laboratory

This paper will describe a new tighter bound for the convergence of genetic algorithms. It will also present an approach to the design of morphological filters using genetic algorithms. It uses iterative search techniques to probe the solution space in a way which models evolution in nature. Results using soft morphological filters designed by genetic algorithms in film restoration will be shown. The GA method is simple in its approach and bypasses the need for highly complex models of the process. As well as providing an interesting perspective to the non linear design debate, solutions found in this way may prove to be complementary to more analytical approaches by both confirming and even prompting new solutions by these routes.

15:50

OPTIMAL PREDICTIVE DESIGN OF BOOLEAN AND ORDER STATISTICS BASED FILTERS

Ioan Tabus, Ciprian Doru Giurcaneanu and Jaakko Astola, Tampere University of Technology, Finland

This paper first reviews several techniques for optimal predictive design of Boolean and L-filters, which proved successful in lossless audio and image compression. We show that two well known nonlinear predictors, namely the Median Adaptive Predictor and the Gradient Adaptive Predictor, which are used in state of the art lossless compression are particular cases of our FSM-L predictors. Including the GAP predictor parameters as initial conditions in our adaptive FSM-L prediction scheme is shown to consistently improve the compression performance, with no increase in the overall complexity.

16:10

USING MODELS OF THE HUMAN VISUAL SYSTEM IN THE DESIGN OF STACK FILTERS FOR THE ENHANCEMENT OF COLOR IMAGES

Jr-Jen Huang and Edward J. Coyle, Purdue University

A technique is developed for utilizing models of the Human Visual System to improve the design of filters for the enhancement of color images. The technique uses an image fidelity measure based on models of the human visual system - such as the Visible Differences Predictor (VDP) - in a nested loop training algorithm. In the inner loop of the algorithm, a stack filter is trained under a Weighted Mean Absolute Error (WMAE) criterion to remove noise. In the outer loop, the VDP is used to train the weights in the WMAE to ensure that the filter that the overall algorithm converges to is one that produces output images that are as visually satisfying as possible. The stack filters resulting from this VDP-driven, WMAE approach perform much better on images than filters trained under the standard mean absolute error criterion. This fact is

demonstrated with color images. The robustness of the resulting filters to variations in both the image and the noise is discussed.

16:30

PARAMETRIC DESIGN OF NONLINEAR OPTIMAL FILTERS

Edward Dougherty, Texas A&M University, USA

The design of optimal nonlinear filters for signal processing has involved mostly nonparametric esti-mation. This paper discusses the elements of para-metric design for analog signal processing. It pro-vides details for the class of conjunctive homothetic granulometric bandpass filters. These are nonlinear and their optimization in the signal-union-noise model for random sets can be completely character-ized in terms of the granulometric size density in-duced by the granulometry from which the optimal filter is to be selected.

16:50

NONLINEAR SIGNAL PROCESSING FILTERS: A UNIFICATION APPROACH

L. Yaroslavsky, Tel Aviv University, Israel

The paper addresses the problem of classification of nonlinear filters for digital signal processing and of unification of approaches to their analysis, design and implementation. It is assumed that the filters work in a moving window and, in each position of the window, produce an output signal sample as an estimate obtained from input signal samples within the window. The keys of the classification and unification are notions of estimation and neighborhood building operations. On the base of thorough analysis of a large variety of known filters, a set of estimation and multi stage (in general) neighborhood building operations is introduced that can serve as building blocks for the filter design. In conclusion, application of the approach to the filter design and implementations is briefly discussed.

TuePmSS3
Small Audiotorium
Intelligent Processing for Communication Terminals
Chair: J. Tasic, M. Ansorge and Carlos Bousono-Calzon ***Univ. of Ljubljana, Slovenia; Univ. of Neuchâtel, Switzerland; Univ. Carlos III, Spain***

14:00

ON HOW STORAGE AT HOME WILL CHANGE HOW WE WATCH TELEVISION

E. Persoon, Philips, The Netherlands

The availability of cheap mass storage will in the future make it possible to watch television programs at the moment we want it. To achieve this the storage of the required programs should be automated. Also we should find a way to make the retrieval of desired programs or scenes easier. Moreover it will also be possible to watch programs in a non-linear fashion as well as faster than real time. To make the system work the total content provision chain should be adapted to provide a typology of the broadcasted programs.

14:40

EFFICIENT IMAGE-INTENSIVE COMMUNICATION SYSTEM SUPPORTING PERSONALISED USER REQUIREMENTS

Urban Burnik, Jurij F. Tasic, University of Ljubljana, Slovenia

The paper presents an optimised approach to image encoding related to real--time imaging applications. A proposal is made how adaptive allocation and compression techniques can be applied to provide an optimal level of service quality. The criteria of optimality is to be set by a currently available network connection, while providing a user with desired level of imaging services. The problem, described in the paper, is how to provide optimal image compression and transmission parameters, which employ currently available bandwidth and maintain perceptual degradation at a minimum level.

15:00

ADAPTIVE COMBINATION OF LMS AND LOGISTIC-LINEAR EQUALIZERS TO IMPROVE THE SPEED-PERFORMANCE COMPROMISE

Manel Martínez-Ramón, José-Luis Sancho and Aníbal R. Figueiras-Vidal, Universidad Carlos III de Madrid, Spain
Antonio Artés-Rodríguez, Universidad de Alcalá, Spain

The least Mean Squares algorithm (LMS) has been the most usual solution to channel equalization: it is easy to implement and it is relatively simple to obtain the appropriate parameters to assure stability and convergence. The convergence speed of LMS is high and it is robust in nonstationary channels. Furthermore, the computational burden of this algorithm is very low, and it can be implemented in low computational power processors, while nonlinear algorithms cannot. The main drawback of this algorithm is that its final Bit Error Rate (BER) is sensitive to the adaptation step and to the effect of well-classified samples which are far from the classification border. Also, in nonlinear channels its performance is reduced. A sigmoidal nonlinearity applied to the output of the filter can reduce the misadjustment due to well-classified samples, and also reduce the dependence from the adaptation step, altough it makes the equalizer much less robust. We introduce a new scheme based on an adaptive combination of an standard LMS equalizer and a sigmoidal output based equalizer, which benefits from the advantages of both equalizers reducing their drawbacks.

15:50

LAYERED AGENT SYSTEM ARCHITECTURE FOR PERSONALIZED RETRIEVAL OF INFORMATION FROM INTERNET

Matev Pogaènik, Jurij F. Tasiè, University of Ljubljana, Slovenia

The article focuses on the architecture of the agent system designed for personalized data retrieval from the Internet. As a consequence of the explosion of the World Wide Web, an increasing amount of information is stored in repositories all over the Web. We suggest three-layered architecture of the agent system including user side layer, which contains user profile and personalization module, information source layer and an intermediary layer offering accompanying services such as authentication and service registration. Such architecture enables inclusion of different information sources, personalization of search results and later enhancements of the system.

16:10

JOINT SOURCE-CHANNEL DECODING WITH ITERATIVE ALGORITHMS

Norbert Goertz and Ulrich Heute, Christian-Albrechts-University of Kiel, Germany

Ideas from iterative decoding of concatenated channel codes are adopted for joint source-channel decoding. Extrinsic information from a soft-in/soft-out channel decoder is used as a-priori information for the new soft-in/soft-out source decoder. The source decoder processes soft channel outputs, extrinsic information from the channel decoder, and source-statistics and it computes extrinsic information for the data-bits that again is used as a-priori information for the channel decoder. In this novel iterative approach to joint source-channel decoding the redundancies within the data-bits of the source encoder are used as an additional source of information in contrast to iterative channel decoding, where further information on the data-bits is drawn from the redundancy-bits of some additional, independent channel-coding scheme.

16:30

CARRIER PHASE ESTIMATION SUITED TO TURBO-CODE BASED COMMUNICATION SYSTEM

Catherine Morlet, Marie-Laure Boucheret, ENST site de Toulouse, France
Isabelle Buret, Alcatel Space Industries, France

In this paper, we describe a new carrier phase estimator, called TD estimator, suited to transmission schemes based on convolutional turbo-codes. The foreseen application is multimedia telecommunication systems with geostationary satellites at Ka-band. For these systems, the on-board receiver has to operate at low SNR to fulfill up-link link budgets and the critical point is the carrier phase recovery. We show that the proposed TD estimator has an operating point of at least Ec/N0=0dB and that the degradation due to frequency deviation and phase noise impairments of this algorithm is less than 1dB for recursive systematic convolutional codes on which turbo-codes are based.

16:50

VIDEO OBJECT EXTRACTION AND SHAPE CODING USING B-SPLINES

Janez Zaletelj, Jurij Tasic Organization, University of Ljubljana, Slovenia

This work presents a system for semi-automatic video object extraction and efficient shape coding using cubic B-splines. In the framework of object-based layered coding of image sequences, shape information is essential for content-based access to video objects, and its efficient encoding needs to be investigated. We present a rate and distortion controlled algorithm for video object shape approximation by variable number of cubic B-spline segments and motion compensated inter-frame coding of B-spline control points. Video object masks are generated using a semi-automatic tracking algorithm. Given an outline of the object in the first frame of the sequence, color and shape models for video object are generated. The adjustment of object's shape and position in the following frames is based on motion estimation and local color information in the neighborhood of the object's boundary. Rate-distortion efficiency of the proposed shape coding algorithm is compared to MPEG-4 context arithmetic encoding.

17:10

INFLUENCE OF GSM SPEECH CODING ON THE PERFORMANCE OF TEXT-INDEPENDENT SPEAKER RECOGNITION

S. Grassi, A. Dufaux, M. Ansorge, and F. Pellandini, University of Neuchatel, Switzerland
L. Besacier, University Joseph Fourier, France

We have investigated the influence of GSM speech coding in the performance of a text-independent speaker recognition system based on Gaussian Mixture Models (GMM). The performance degradation due to the utilization of the three GSM speech coders was assessed, using three transcoded databases, obtained by passing the TIMIT through each GSM coder/decoder. The recognition performance was also assessed using the original TIMIT and its 8 kHz downsampled version. Then, different experiments were carried out in order to explore feature calculation directly from the GSM EFR encoded parameters and to measure the degradation introduced by different aspects of the coder.

TuePmPO1
Exhibition Hall
Speech Recognition and Speaker Identification
Chair: O. Viikki *Nokia Research Center, Finland*

A MODEL FOR SPEECH RECOGNITION SUBJECTED TO PARTIAL AND TEMPORAL CORRUPTION WITH UNKNOWN, TIME-VARYING NOISE STATISTICS

Ji Ming, Philip Hanna, Darryl Stewart, Peter Jancovic and Jack Smith, The Queen's University of Belfast, Ireland

This paper proposes a new statistical approach, namely the probabilistic union model, for speech recognition subjected to unknown, time-varying, burst noise during the utterance. The model characterizes the partially and randomly corrupted observations based on the union of random events. We have tested the new model using the TIDIGITS database, corrupted by various type of additive abrupt noise. The experimental results show that the new model offers robustness to partial and temporal corruption, requiring little or no knowledge about the noise characteristics.

A COMPARATIVE STUDY OF SEVERAL PARAMETERIZATIONS FOR SPEAKER RECOGNITION

Marcos Faundez Zanuy Escola, Universitaria Politecnica de Mataro, Spain

This paper presents an exhaustive study about the robustness of several parameterizations, in speaker verification and identification tasks. We have studied several mismatch conditions: different recording sessions, microphones, and different languages (it has been obtained from a bilingual set of speakers). This study reveals that the combination of several parameterizations can improve the robustness in all the scenarios for both tasks, identification and verification. In addition, two different methods have been evaluated: vector quantization, and covariance matrices with an arithmetic-harmonic sphericity measure.

MAXIMUM A POSTERIORI LINEAR REGRESSION FOR SPEAKER ADAPTATION WITH THE PRIOR OF MEAN

Chih-Heng Lin, Wern-Jun Wang, Chunghwa Telecommunication Laboratories

For speaker adaptation, we propose a new method to estimate the linear regression transformations by maximum a posteriori criterion. Let the relationship between the mean parameters of adapted model and the mean parameters of the speaker independent model be represented by sets of linear transformations like that of maximum likelihood linear regression approach. We estimate the transformations by means of the prior distributions of mean. The experiments on Mandarin speech recognition show the proposed approach is superior to the MLLR approach when only little data is available for speaker adaptation.

FEATURE CONCATENATION FOR SPEAKER IDENTIFICATION

R. D. Zilca, Research and Development Division, Amdocs Israel
Y. Bistritz, Tel Aviv University, Israel

The use of feature vectors obtained by concatenation of different features for text independent speaker identification from clean and telephone speech is studied. The composite feature vectors are examined with GMM and VQ models used to classify speakers. Linear discriminant analysis (LDA), a statistical tool designed to select a reduced set of features for best classification, is applied to enhance performance. The use of LDA for reducing the size of composite feature vector was found satisfactory for clean speech but not for telephone speech. On the other hand, using LDA in the not conventional manner – as a nonsingular transformation (i.e. without size reduction) - improved the performance of composite features in both clean and the telephone speaker identification experiments.

A COMPUTATIONALLY SCALABLE SPEAKER RECOGNITION SYSTEM

W. M. Campbell, C. C. Broun, Motorola Human Interface Laboratory Tempe, USA

Computationally scalable speaker recognition systems are highly desirable in practice. To achieve this objective, we use a two-stage architecture for text-prompted speaker recognition. In this system, the input speech is first segmented on subword boundaries using a Viterbi alignment. The second stage applies a polynomial classifier to each subword for verification. Through a simple approximation, the scoring criterion for the polynomial classifier is made highly scalable. The resulting combination of speaker independent segmentation and a scalable recognition system results in a system which can perform speaker recognition on a large population with minimal computation.

MARKOV RANDOM FIELD LINEAR REGRESSION

Xintian WU and Yonghong Yan, Oregon Graduate Insitute of Science and Technology

This paper outlines the Markov Random Field Linear Regression (MRFLR) algorithm, which combines the transformation-based adaptation and dependency-modeling technique together. The hypothesis is that the adaptation performance can be improved by explicitly modeling the correlations among acoustic parameters and applying such constraints to the transformation-matrix estimation. The correlations are modeled by Markov Random Field, and the incorporation of the correlations is under the Maximum A Posteriori framework. Experimental results show that MRFLR has significant improvement over Maximum Likelihood Linear Regression when only small amounts of adaptation data are available.

A COMPETITION-BASED MEASUREMENT FOR HMM SPEAKER VERIFICATION

Yong Gu and Trevor Thomas, Vocalis Group Plc., UK

Score normalisation with cohort speaker models has been widely used in HMM-based speaker verification. Most of the proposed methods are based on the framework of the hypothesis testing. Based on this framework an overall average of all cohort scores is often used for normalisation, which leads a log likelihood ratio (LLR) for verification. In this paper we use a competition-based criterion to define the measurement. Based on this criterion a measurement is proposed, which reflects the competitiveness of claimed model against cohort models for a given testing utterance. The evaluation is carried out on the YOHO database. A comparison with conventional LLR is given. The results show that with this proposed measurement can improve the speaker verification (SV) performance significantly. The SV equal error rate can be reduced by about 30%.

HIGH ORDER STATISTICS FOR ROBUST SPEECH/NON-SPEECH DETECTION

Arnaud Martin, Lamia Karray and André Gilloire organisation, France Télécom

In noisy environments, a robust speech/non-speech detection is necessary for speech recognition. This paper presents a new method for speech/non-speech detection using third-order moments. The analysis of the energy third-order moment behaviour gives useful information on energy distribution. We present the estimation of this statistic to take into account the non-stationarity of signal. The new algorithm is compared to the one based on noise and speech statistics presented in previous works. We compare the results of the segmentation and recognition tests on two databases, one recovered on Public Switched Network, the other one on GSM. The results show that the new algorithm outperforms the one based on noise and speech statistics only, especially in the case of noisy environments.

INTRODUCTION OF A RELIABILITY MEASURE IN MISSING DATA APPROACH FOR ROBUST SPEECH RECOGNITION

Philippe Renevey and Andrzej Drygajlo, Swiss Federal Institute of Technology, Switzerland

This paper addresses the problem of robust speech recognition in noisy conditions in the framework of hidden Markov models (HMMs) and missing feature imputation techniques. It presents a new statistical approach to the detection and estimation of unreliable features based on a probabilistic measure and Gaussian mixture model (GMM) representing clean speech distribution. In the estimation process, the GMM is compensated using parameters of the statistical model of additive background noise. The GMM means are used to estimate the clean speech features. The GMM imputed values and the noisy signal are combined proportionally to a probabilistic reliability measure to estimate the clean speech. The reliability measure allows us to avoid to use a hard decision threshold to decompose the data into reliable and unreliable features and consequently reduces the risk of missclassification. GMM based technique is less complex than the corresponding HMM based estimation and gives similar improvement in the recognition performance. Once unreliable features are replaced by the estimated clean speech features, the entire set of spectral features is transformed to the MFCC (Mel Frequency Cepstral Coefficient)

feature domain. The MFCCs which are characterized by a higher baseline recognition rate are used for final recognition using continuous density hidden Markov models (CDHMMs) with diagonal covariance matrices.

MAP-BASED CONTEXT DEPENDENT TONE RECOGNITION METHOD OF CHINESE SPEECH

Li Ming, Liu Jian and Yu Tiecheng, Institute of Acoustics, Academia Sinica

This paper presents a new context dependent tone recognition method. First we suggest that there be more than five tone modes in Chinese continuous speech. We get all new tone modes by grouping all tone feature vectors to a specific number of categories. Secondly, we recognize a sentence with the new tone modes and get the new tone sequence. Finally, we find out each original tone of the sentence which has the maximum a posteriori probability for the corresponding new tone and its context new tones. In a ten-person test set, which includes about 20,000 tones, we achieve a higher average recognition rate with the MAP-based context dependent tone method than that with the conventional context dependent tone method.

TuePmPO2
Exhibition Hall
Multirate Filtering
Chair: M. Gelgon *Nokia Research Center, Finland*

LOW DELAY QMF BANK DESIGN USING INITIAL FILTER APPROACH

Faris Al-Naimy, M. J. Nigam and Alaa Kasim, University of Roorkee, India

Investigation have been carried out on the iterative method proposed by Wu-Sheng [11] to design an efficient QMF banks with approximately constant group delay. Simulations show that it is quite sensitive to the choice of the initial filter. A new and important benefit of manipulating the initial filter design of QMF banks is its ability to design QMF banks with low and approximately constant group delay. By using a linear minimum phase efficient method proposed by Ivan [4] as the initial filter, improvements in the most of design specification in [11] have been obtained. The constraint of the transition band is added in the initial condition. Computer simulation shows that the proposed approach produce better results than those reported previously. However, the proposed approach requires more computational efficiency compared with existing methods.

SOME BENEFITS OF ALIASING IN TIME SERIES ANALYSIS

P.M.T. Broersen and S. de Waele, Delft University of Technology, The Netherlands

The problem of sampling a signal with interval T is present in preparing continuous-time processes for discrete-time signal processing algorithms and in down-sampling a discrete-time signal to a larger time scale. The important issue is whether invariance in time or in frequency domain is preferred. The time domain approach preserves the covariance function at time shifts KT, while the frequency domain approach tries to preserve the part of the original spectrum up to frequency pi/T. This requires the use of non-ideal anti-aliasing filters, which will be harmful in time series analysis because the product of the process and the anti-aliasing filter is modeled.

IMPULSE RESPONSE APPROXIMATION FOR ARBITRARY SAMPLING RATE CONVERSION

Gennaro Evangelista, Ruhr-Universitaet, Bochum, Germany

Converting a digital signal from one sampling rate to another, where the according clocks are independent of each other, is revisited. The state-of-the-art approach approximates an ideal continuous impulse response in a suboptimal manner. For a new approach an ideal impulse response is specified and approximated with a polynomial according to a least squares-criterion. This approach is compared with the state-of-the-art approach concerning expenditure: storage and multiplication.

ARE NONUNIFORM PRINCIPAL COMPONENT FILTER BANKS OPTIMAL?

Sony Akkarakaran and P.P.Vaidyanathan, California Institute of Technlogy, USA

The notion of a principal component filter bank (PCFB) for a given class of uniform filter banks (FB's) has been well studied. Recent work by the authors has shown that PCFB's are optimal orthonormal FB's whenever the minimization objective is a concave function of the vector of subband variances of the FB. This result gives a unified explanation of PCFB optimality for progressive transmission, compression, noise suppression, and as shown more recently, for use in DMT (discrete multitone modulation) systems. This paper generalizes such results to nonuniform FB's. We propose two distinct definitions of nonuniform PCFB's. Each definition results in PCFB optimality for certain types of concave objectives whose form is somewhat more restricted than in the case of uniform FB's. We study existence of the defined PCFB's, and observe that it can be very delicate: Small perturbations of the input spectra can sometimes destroy the existence of nonuniform PCFB's.

TIME-VARYING RECURSIVE FILTERS FOR DECIMATION AND INTERPOLATION

Tim Hentschel, Gerhard Fettweis, Dresden University of Technology, Germany

Decimation and interpolation require filtering. In order to reduce the hardware effort, FIR filters with linear phase are implemented as polyphase filters often. Still, the number of coefficients cannot be reduced by this method. Comb filters implemented as cascaded-integrator-comb filters are hardware-efficient filters. Due to the periodicity of the comb section the number of registers (i.e., the state space) can be reduced, while the separation of poles and zeros provides a means to avoid coefficient multipliers at all. However, due to the necessarily perfect pole-zero cancellation these filters can only realize zeros at integer multiples of the lower sample rate. The presented time-varying recursive filter structures exploit the advantage of CIC-filters of implementing the transfer-zeros for only one period of a periodic comb section and thus, reduce complexity considerably. Moreover, they provide perfect pole-zero cancellation for arbitrary zero placement.

SENSITIVITY PROPERTIES OF A NOVEL CLASS OF IIR DECIMATION FILTERS FOR LOFARGRAM ANALYSIS IN PASSIVE SONAR SYSTEMS

Marleusa Corrêa Gonçalves, Antonio Petraglia and William Soares-Filho, COPPE - EE Universidade Federal do Rio de Janeiro, Brasil

A new class of low sensitivity and efficient decimation filters based on the polyphase decomposition of IIR transfer functions is proposed. Comparisons with conventional IIR elliptic filters are presented. An illustrative example for application in passive sonar systems using real noisy data is provided.

FAST FOURIER-WEYL TRANSFORM

E.Rundblad-Labunets and V.Labunets, State Technical University, Russia

We discuss an important attribute of a discrete wavelet transform derived using a group-theoretic approach. We use discrete affine group AFF(GF(p)) of the Galois field GF(p) for generating wavelet atoms. We develop a common method of computing the wavelet distribution based on the fast Fourier-Weyl transform.

TuePmPO3
Exhibition Hall
Biomedical Image Processing
Chair: A. Värri *Tampere University of Technology, Finland*

ANALYSIS AND OPTIMISATION OF 3D RECONSTRUCTION METHOD OF THE AORTA FROM A TOMOGRAPHIC IMAGES SEQUENCE

William Puech, Vincent Ricordel, Université de Toulon et du Var, France
Guy Passail, Centre Hospitalier Intercommunal de Fréjus - St Raphaël, France

The aim concerns 3D reconstruction of the aorta from sequence of X rays scanner cuts. We analyse methods used in the medical imaging services, essentially based on threshold techniques and images subtraction. We propose improving techniques of these methods by using semi-automatic extraction of only one anatomic structure. Our method leads to use a model of active contours.

THE NEW APPROACH TO THE MICROSCOPE IMAGE CLASSIFICATION OF BRAIN AMYLOIDOSIS DISEASES BY FILTERING PROCESS OF AMYLOID PLAQUE

Malgorzata Napieralska, Mariusz Zubert, Andrzej Napieralski, Technical University of Lodz
Aleksander Grams, Pawel Liberski, Tomasz Sobow, Jacek Grabowski, Medical University of Lodz

This paper presents the first approach to application of the two-dimensional image processing in recognition of brain amyloidosis diseases which could also lead to modification of the Creutzfeld-Jakob disease diagnosis. The authors propose to create special universal amyloid plaque computer pattern and special multivariate medical image segmentation techniques based on collected images and statistical information. The recognition image procedure is divided into 3-dimensional statistical colour and morphological shape identifications. Presented method is universal and can be applied to recognition the wide class of microscope objects. The further application of a created diagnostic system based on an amyloid plaque pattern for screening animals is also feasible (especially in cases of "mad cow disease" which is one of the major health conserve in Europe). It is possible to improve object recognition by the system using fuzzy logic or systolic networks.

AUTOMATED DETECTION OF PROSTATE CANCER NUCLEI

Metehan MakinacI and Kemal Ozmehmet, Dokuz Eylul Universitesi, Turkey

In this work, a cross-disciplinary approach was presented for the solution of a biomedical pattern recognition problem: cancer diagnosis. Specifically, prostate specimen images were analyzed. Images were acquired by a microscope, video camera and a digitizing board and stored on a computer. The software required to indicate the areas of interest of the tissue image to the

pathologist, was developed on the object-oriented basis. User interface was designed for the required functions. Discriminatory power of shape feature were evaluated statistically. The sensitivity and specificity ratios for different threshold levels of the feature set were calculated.

DISCRETIZED RADON TRANSFORM AND ITS SPECTRAL PROPERTIES

F. Boschen, A. Kummert, University of Wuppertal, Germany

The sampling theorem also known as Shannon theorem is the basis for digital signal processing and computer based algorithms. A great number of publications is devoted to sampling and reconstruction of signals. Spectral properties of the underlying continuous signals and different kinds of applications require a specific approach in designing an optimal sampling grid. For doing this, in the domain of multidimensional signal processing a greater degree of freedom can be utilized. In this paper the spectral properties of the projection signal of a tomograph with respect to the bandwidth is analysed and an optimal sampling grid for projection data is presented.

WAVELET AND HMM ASSOCIATION FOR ECG SEGMENTATION

Ronan Le Page, Jean-Marc Boucher, Ecole Nationale Superieure des Telecommunications de Bretagne, France
Karine Provost, Jean-Christophe Cornily, Jean-Jacques Blanc, Centre Hospitalier Universitaire de la Cavale Blanche, France

This paper presents a multiscale Hidden Markov Model (HMM) to improve an automatic segmentation of an electrocardiographic signal (ECG). While the HMM describes the dynamical mean evolution of cardiac cycle, the use of wavelet analysis in association with the HMM leads to take into account local singularities and to obtain better segmentation results. This was tested on a learning base composed of 130 patients in order to segment a part of ECG signals:the P-wave. Good segmentation of P-waves is a difficult task due to the weakness of this waves comparatively to other parts of ECG signals. Using wavelets and HMM leads to obtain P-wave segmentation results closer to manual segmentations.

FAST IMPLEMENTATION OF POLYNOMIAL-BASED INTERPOLATORS FOR BIOMEDICAL IMAGE ROTATION AND SCALING

Saul R. Dooley, Philips Research Laboratories, UK
Tariq S. Durrani and Robert W. Stewart, University of Strathclyde, UK

This paper proposes the use of a polynomial interpolator structure (the Farrow structure) which is efficiently realizable in hardware, for high-quality geometric transformation of 2-D and 3-D images. Polynomial based interpolators such as B-splines are exactly implementable in the Farrow structure framework. Using this structure, the advantage in computational complexity is demonstrated to be very significant, typically saving around 80% of the computational cost of on-line direct interpolation.

DIAGNOSTIC COMPRESSION OF BIOMEDICAL VOLUMES

Alberto Signoroni and Riccardo Leonardi, University of Brescia, Italy

In this work we deal with lossy compression of biomedical volumes. By force of circumstances, diagnostic compression is bound to a subjective judgment. However, with respect to the algorithms, there is a need to shape the coding methodology so as to highlight beyond compression three important factors: the medical data, the specific usage and the particular end-user. Biomedical volumes may have very different characteristics which derive from imaging modality, resolution and voxel aspect ratio. Moreover, volumes are usually viewed slice by slice on a lightbox, according to different cutting direction (typically one of the three voxel axes). We will see why and how these aspects impact on the choice of the coding algorithm and on a possible extension of 2D well known algorithms to more efficient 3D versions. Crosscorrelation between reconstruction error and signal is a key aspect to keep into account; we suggest to apply a non uniform quantization to wavelet coefficients in order to reduce slice PSNR variation. Once a good "neutral" coding for a certain volume is obtained, non uniform quantization can also be made space variant in order to reach more objective quality on Volumes of Diagnostic Interest (VoDI), which in turns can determine the diagnostic quality of the entire data set.

SEGMENTATION OF MEDICAL IMAGES USING A MIXTURE MODEL AND MORPHOLOGICAL FILTERING

Qosai Kanafani and Azeddine Beghdadi, Universite Paris, France

In this work a practical solution for segmenting 3D MR images is proposed. This method is based on a mixture model and Expectation Maximization (EM) algorithm. Here, we only focus on image segmentation which is used as a first step in our 3D compression and visualization system we are developing. A pretreatment based on gray-level thresholding followed by a morphological filtering is employed, then a stochastic segmentation method based on a finite mixture model is used. The obtained results confirm that statistical segmentation based on mixture model combined with Bayesian decision rule is a powerful tool for segmenting MR images.

CORTICAL SULCI DETECTION AND TRACKING

Christophe Renault, Michel Desvignes, Marinette Revenu, GREYC – ISMRA, France

Automatic labelling and identification of cerebral structure, like cortical sulci, are useful in neurology, surgery planning, etc… We propose a cortical sulci valley detection. The aim of the method is to achieve the sulci medial surface. The method applied on MRI data, is based on geometrical features (curvature) which doesn't require the accurate segmentation of the cerebral cortex. We use a sub-voxel precision tracking. The minimum curvature vector in each point allows successive displacement along the valley of sulci. Partial derivatives provide the differential characteristics.

TuePmPO4
Exhibition Hall
Modulation and Detection
Chair: P. Handel ***Royal Institute of Technology, Sweden***

STOCHASTIC PROCESS SHIFT KEYING: A NOVEL MODULATION METHOD FOR SECURE DIGITAL COMMUNICATIONS

Arnt-Børre Salberg and Alfred Hanssen, University of Tromsø, Norway

We present a new digital modulation technique that introduces security in digital communications. The basic principle is to transmit realizations of a stochastic process in such a manner that the transmitted waveform appears noiselike. In this paper, we have chosen to express the transmitted waveform in a subspace formalism. This allows for an elegant geometrical interpretation of the waveform, and it naturally suggests a simple and accurate matched subspace detector for the receiver. The technique is demonstrated by numerical simulations, and a comparison with an optimal Neyman-Pearson detector shows that our simple subspace detector yields a high-quality and reliable receiver for the modulated signal.

AN EQUAL GAIN COMBINING DPSK MC-CDMA SYSTEM FOR FAST FADING MOBILE DOWNLINK CHANNELS

A. McCormick, P. Grant and J. Thompson, The University of Edinburgh, UK

This paper describes a multi-carrier code division multiple access downlink system which can be used in mobile or broadcast radio communication applications. It employs a novel form of differential phase shift keying, in which the phase shift keying is applied in the transmitter after all the users symbols have been combined. Theoretical performance predictions and results of simulations are presented and comparison is made with other equivalent MC-CDMA and DS-CDMA downlink systems.

SUPPRESSION OF NARROW FREQUENCY BANDS IN MULTICARRIER TRANSMISSION SYSTEMS

Robert Baldemair, Vienna University of Technology, Austria

A method to enhance capacity in Discrete Multitone Transmission (DMT) systems is presented. This paper addresses the problem, how to reduce the transmit power spectral density (PSD) of the DMT signal in specific narrow frequency bands below a certain level . One example is VDSL signalling in the presence of amateur radio. Only a small amount of egress is allowed inside these very narrow frequency bands. If a DMT system operates in such a situation, in the conventional system all tones inside these frequency bands and a lot of adjacent tones must be loaded with zeros to meet the requirements on the egress. In our proposal we devote some unused tones to reduce the PSD inside these frequency bands. With this method the number of idle tones can be reduced substantially. To demonstrate the increase of the number of usable

tones, we will show a realistic example in the context of VDSL.

ON DIGITAL ESTIMATION OF THE INSTANTANEOUS FREQUENCY BEYOND THE FOLDING FREQUENCY

Ewa Hermanowicz and Miroslaw Rojewski, Technical University of Gdansk, Poland

The aim of this paper is to demonstrate the ability of the proposed time domain and frequency domain instantaneous frequency (IF) estimators to track with high accuracy the very dynamically varying IF of a discrete-time signal. Our experiments with the analysis of small birds songs show that both these estimators enable to follow the IF which extends beyond the signal spectral range as well as beyond the spectral folding frequency.

SELECTIVE PARTIAL PARALLEL INTERFERENCE CANCELLATION FOR PERSONAL CDMA COMMUNICATIONS

Filippo Belloni, University of Oulu, Finland
Romano Fantacci and Simone Morosi, University of Florence, Italy

This paper deals with a cancellation multiuser detector for CDMA communication systems. Proposed receiver is supposed to be used at the end of up-link channel, leading to consider multipath fading phenomena. Proposed receiver approach consists in performing a weighted selective cancellation of the co-channel interfering signals, divided in two different groups according to the received power level: signals exceeding a suitable threshold are considered more reliable and, therefore, cancelled with a higher weight. In comparison with previously proposed cancellation receivers described detector shows remarkable improvement of the resistance to multiple access interference and low computational complexity, linear in the number of users.

PERFORMANCE OF CODED MULTISTAGE DETECTION IN CORRELATED RAYLEIGH CHANNELS

Kimmo Kettunen, Helsinki University of Technology, Finland

In this paper, we estimate the performance of coded multistage decectors operating over a correlated Rayleigh fading channel without interleaving. We derive analytical estimates for the single-cell capacity of such systems by estimating the effective signal-to-noise ratio (SNR) after each stage and by calculating the exact symbol error probabilities of the channel decoder in a correlated Rayleigh fading channel. These analytical estimates are compared with the capacity estimates achived through simulations.

TuePmPO5
Exhibition Hall
Pattern Recognition
Chair: B. Macq ***Universite Catolique de Louvain, Belgium***

SUPPORT VECTOR METHOD FOR MINUTIAE DETECTION IN FINGERPRINT IMAGES

Adrian Burian, Marius Tico, Mikko Lehtokangas, Pauli Kuosmanen and Jukka Saarinen, Tampere University of Technology, Finland

Fingerprint minutiae (i.e. ridge endings and ridge bifurcations) form a pattern that is unique to each fingerprint. Almost all automatic fingerprint comparison systems rely on minutiae matching, and hence the minutiae extraction from fingerprint images highly influences the performance of every such system. A minutiae extraction method based on support vector machines is proposed here. The method does not require a ridge thinning processing step in contrast with most of the previously proposed methods of minutiae detection and classification. Because of this, the number of spurious minutiae detected is maintained low, such that a subsequent processing step of spurious minutiae elimination becomes unnecessary.

A POST-CLASSIFICATION SCHEME FOR AN OCR SYSTEM FOR THE NOTATION OF THE ORTHODOX HELLENIC BYZANTINE MUSIC

Velissarios G. Gezerlis and Sergios Theodoridis, University of Athens, Greece

In this paper we present, for the first time, the development of a new OCR system for the off-line optical recognition of the characters of the Orthodox Hellenic Byzantine Music Notation, that has been established for use since 1814. We describe the structure of the new system, and propose algorithms for the recognition of the 71 distinct character classes, based on structural and statistical features. For the classification, a tree-structured classification schema and a simple Nearest Neighbor classifier was used. In this paper, an emphasis is given for the description of the post-classification part, which is used in order to increase the overall performance of the system. The final accuracy achieved by the OCR system, tested on a data base of 18,000 characters is 99.3%. The development of such a system is of great importance to musicologists, especially in our days, which are marked by an increased interest, world wide, for the study and understanding of Eastern type musical forms.

IMPROVING GENERALIZATION ABILITY OF HMM/NNs BASED CLASSIFIERS

Miguel A. Ferrer, Itziar G. Alonso, Carlos M. Travieso and Aníbal R. Figueiras-Vidal, Universidad de Las Palmas de Gran Canaria, Spain

Standard Hidden Markov Models (HMM) have proved to be a very useful tool for temporal sequence pattern recognition, although they present a poor discriminative power. On the contrary Neural Networks (NNs) have been recognized as powerful tools for classification task, but they are less efficient to model temporal variation than HMM. In order to get the advantages of both

HMMs and NNs, different hybrid structures have been proposed. In this paper we suggest a HMM/NN hybrid where the NN classify from HMM scores. As NN we have used a committee of networks. As networks of the committee we have used a Multilayer Perceptron (MLP: a global classifier) and Radial Basis Function (RBF: a local classifier) nets which drawn conceptually different interclass borders. The combining algorithm is the TopNSeg scoring method which sum the top N ranked networks normalized outputs for each class. The test of above architecture with speech recognition, handwritten numeral classification, and signature verification problems show that this architecture works significantly better than the isolated networks.

CLASSIFICATION OF MFSK MODULATED SIGNALS USING THE MEAN OF COMPLEX ENVELOPE

Antti-Veikko Rosti, Tampere University of Technology, Finland
Visa Koivunen, Helsinki University of Technology, Finland

The interest in modulation classification has recently emerged in the research of communication systems. Modulation classification has many important applications in communications, e.g., reconfigurable receivers, spectrum managment and interference cancellation. At the moment, the most attractive single application area is software radio. In this paper we address the problem of classifying digitally modulated signals using cyclostationary statistics. We derive the first-order moments of the complex envelope of digitally modulated signals and verify their periodicity. A novel feature for the classification of the frequency shift keyed signals is proposed. The performance of this feature in distinguishing among different FSK constellations is studied in simulation. Some comparisons to commonly used features are performed.

ACCELEROMETER VIBRATION ANALYSIS IN FINDING OUT VELOCITIES OF PEN INPUT DEVICE

Mikko Haukijärvi and Jukka Yrjänäinen Nokia Research Center

The pen input devices have become a popular input method with personal digital assistants and communication devices. The development of microelectromechanical systems (MEMS) technology has made it possible to put accelerometers - acceleration sensors - inside the pen. The acceleration information can be utilised in order to find out the movements of the pen. This makes it possible to manufacture pen input devices that do not need touch sensitive surfaces. However, due to the complex nature of the acceleration data, i.e. a mixture of movement and gravity data, it is not easy to extract the essential information. This paper presents an idea how the acceleration data could be utilised in finding out the instantaneous velocity of the pen tip. The idea is based on the vibration caused by the friction between the accelerometer pen and the textured surface, on which the pen is drawn. The properties of the vibration relate to the direction and the speed of the pen movement.

EFFICIENT IMPLEMENTATION OF MATCHING PURSUIT USING A GENETIC ALGORITHM IN THE CONTINUOUS SPACE

Jean-Marc Vesin, Swiss Federal Inst. of Technology, Switzerland

In this work we introduce an alternative implementation of matching pursuit (MP) using a genetic algorithm in the continuous space (GACS). MP is an attractive analysis approach in which the signal is sequentially decomposed into a linear expansion of atoms (functions) from a dictionary of waveforms so as to obtain a sparse representation. The main problem with MP is its computation load, due to the necessarily large size of the dictionary. We propose instead to determine the optimal atom at each stage of the decomposition using a GACS, i.e. a genetic algorithm that requires no quantization of the solution parameters. Preliminary simulation results illustrate the potential benefits of this scheme.

A NEW ALGORITHM TO DESIGN OPTIMAL HIDDEN MARKOV MODELS

I. Dologlou, S. Bakamidis and G. Carayannis, Institute for Language and Speech Processing, Greece

A new algorithm for the design of Hidden Markov Models (HMM) from observed symbol and bisymbol probabilities is presented. The algorithm provides a global optimum and makes use of linear vectorial models of sequences of probabilistic vectors estimated during an off-line learning process. Moreover, a method to enhance observed data so as to comply with the constraints of HMM strings is proposed. An optimal estimate of the number of states of the HMM for the given observed data is also provided.

TuePmOR3
Hall A
Blind Methods in Communications
Chair: R. Tourki ***Ecole Nationale d'Ingenieurs de Monastir, Tunisia***

15:50

UNIFIED FORMULATION OF CLOSED-FORM ESTIMATORS FOR BLIND SOURCE SEPARATION IN COMPLEX INSTANTANEOUS LINEAR MIXTURES

Vicente Zarzoso and Asoke K. Nandi, The University of Liverpool, UK

The blind separation of unknown independent source signals from sensor observations is addressed in environments of instantaneous linear mixtures. After pre-whitening the sensor data, the source extraction reduces to the identification of certain parameters defining a unitary transformation. This contribution develops the algebraic devices which allow us: 1) to provide a unified formulation of the closed-form estimators of the separation parameters in the complex-mixture scenario -typical of multi-user digital communications-, and 2) to disclose the remarkable parallelism existing between the real and the complex problem in the context of their analytic solutions.

16:10

BLIND MIMO SYSTEM IDENTIFICATION BASED ON CROSS-POLYSPECTRA

Binning Chen, Athina P. Petropulu, Lieven De Lathauwer and Bart De Moor, Drexel University, USA

In this paper we propose a novel frequency domain approach for the identification of a multiple-input multiple-output (MIMO) system driven by temporally i.i.d. and spatially independent non-Gaussian processes. The system frequency response is obtained based on singular value decomposition of a matrix constructed based on the power-spectrum and slices of polyspectra of the system output. Since more than one HOS slices could be used, the SVD can be replaced by joint diagonalization of a set of matrices. The flexibility to select the HOS slices allows us to bypass the frequency dependent permutation and phase ambiguity problems, which are usually associated with frequency domain SVD.

16:30

PRE-COMBINING ADVANCED BLIND MULTIUSER DETECTOR FOR TIME AND FREQUENCY SELECTIVE WIRELESS CHANNEL

Lorenzo Mucchi, University of Oulu, Finland
Enrico Del Re, Romano Fantacci and Simone Morosi, University of Florence, Italy

This paper deals with an adaptive multiuser detector for DS-CDMA (Direct Sequence Code Division Multiple Access) wireless communication systems, whose main features are low complexity and joint utilization of the time diversity and blind adaptive processing techniques. The proposed multiuser detector, defined as Pre-combining Advanced Blind Adaptive Multiuser Detector (PABA-MUD), a window reprocessing technique together with an original adaptation rule is used in order to improve performance for time varying environment. In particular, pre-combining of different replicas allows deriving a unique adaptive sequence even for multipath signals. The proposed detector shows remarkable near-far resistance and requires knowledge of the desired user's signature waveform, timing and phase. No training sequence is needed. Receiver performance is expressed in terms of bit error rate (BER), which has been derived by simulations under the assumption of a time and frequency-selective Rayleigh fading channel.

16:50

AN ALGEBRAIC ICA ALGORITHM FOR 3 SOURCES AND 2 SENSORS

L. De Lathauwer, B. De Moor and J. Vandewalle, K.U.Leuven

In this paper we develop an algebraic algorithm for Independent Component Analysis with 3 complex-valued sources and 2 sensors. First we consider a generalization of an old theorem by Sylvester, which allows us to relate the problem with the approximation of a 4th-order tensor with Hermitean symmetry by a tensor of rank-1. We present an Alternating Least Squares algorithm for the computation of the result.

17:10

ARE BLIND EQUALIZERS ASSUMING WHITE DATA APPLICABLE TO BLOCK ENCODED SEQUENCES?

Jukka Mannerkoski and Visa Koivunen, Helsinki University of Technology, Finland

Several blind equalization/identification exploit the assumption that transmitted information sequence is white. In real communication systems, redundancy is added to the source sequence in order to be able to detect and correct symbol errors at the receiver. It is not obvious how channel coding affects the assumption of whiteness. In this paper, we analyse the autocorrelation sequences of widely used linear block codes. The codes are presented in terms of Markov model, for which the autocorrelation can be analytically obtained. The encoded data is used as input to a prediction error based blind equalizer proposed in our earlier work. The performance is compared to the case of unencoded data.

VOLUME II

WedAmSS1
Hall B
Blind Methods in Communications
Chair: A.-J. van der Veen ***Technical University of Delft, The Netherlands***

09:20

BLIND SEPARATION OF CONVOLVED MIXTURES FOR CDMA SYSTEMS

Razvan Cristescu and Juha Karhunen, Helsinki University of Technology, Finland
Tapani Ristaniemi, University of Jyväskylä, Finland
Jyrki Joutsensalo, Tampere University of Technology, Finland

This paper presents a new method for symbol estimation in the downlink of a CDMA communication system. Our approach is based on the observation that the slowly-fading CDMA signal model may be expressed as a linear combination of the convolved independent symbol sequences. A blind source separation approach based on maximization of output entropy is used for the blind separation of the sources; the symbols corresponding to the user of interest are determined using a small training sequence. Our method shows good simulation results when compared to traditional symbol separation techniques.

09:40

A SEMIBLIND APPROACH TO OPTIMUM MULTIUSER DETECTION

Joaquín Míguez, Mónica F. Bugallo and Luis Castedo, Universidade da Coruña, Spain

A novel multiuser semiblind demodulation scheme is proposed. Channel estimation is carried out according to the Maximum Likelihood (ML) principle and using the Expectation Maximization (EM) algorithm. Afterwards, the channel state information is used to implement the joint optimum detector with a dynamic programming algorithm (e.g., the Viterbi algorithm). The resulting demodulator is termed semiblind because the statistical features of the received signals are exploited together with the a priori knowledge of a very small number of transmitted symbols. Computer simulations show that the proposed scheme leads to practically optimum performance in Code Division Multiple Access (CDMA) time dispersive channels.

10:00

CHANNEL ESTIMATION AND DEMODULATION OF ASYNCHRONOUS CDMA SIGNALS IN FREQUENCY-SELECTION FADING CHANNELS

C.Carlemalm, A.Logothetis and H.V. Poor, Princeton University, USA

In this paper, we propose an iterative scheme for joint tracking of fading channels and demodulation of multi-user CDMA signals operating asynchronously over multipath fading channels. We use the expectation maximization algorithm to track the time-varying channel and as a by-product we also achieve estimates of the input signals. We show that the maximum a posteriori estimates of the fading channels is iteratively computed by the Kalman smoother. The input signals are demodulated using a hidden Markov model smoother. Our scheme has computational complexity and memory requirement that grow linearly with the data length. Computer simulations illustrate the performance of our proposed detection and estimation method.

10:50

BLIND MULTIUSER DETECTORS FOR DUAL RATE DS-CDMA SYSTEMS OVER FREQUENCY SELECTIVE CHANNELS

Michail K. Tsatsanis and Xuguang Lu, Stevens Institute of Technology
Zhengyuan (Daniel) Xu, UC Riverside, USA

Linearly constrained optimization techniques based on the minimization of the output energy have been proposed for the design of single rate DS-CDMA detectors. The goal of this paper is to apply this approach to dual rate systems where low rate users have symbol periods which are multiples of those of high rate users. It is observed that a high rate (HR) user experiences the same frequency selective communication channel during its M consecutive symbol periods corresponding to one symbol period of low rate (LR) users. A bank of M detectors can be derived to detect those M symbols for a HR user by minimizing the total output power of these detectors subject to a common constraint for all M detectors. The unknown constraint can be obtained by further maximizing the resulting power. To detect a LR user, one possible solution is to similarly design M such detectors in its one symbol period and combine all outputs based on some criterion such as maximum ratio combining. An alternative is to directly apply existing constrained optimization methods. It is shown that the performance of the proposed detectors tends to be close to that of the MMSE detector at high SNR.

11:10

CRAMER-RAO BOUND FOR BLIND CHANNEL ESTIMATION IN MULTI-CARRIER CDMA SYSTEMS

Daniel I. Iglesia, Carlos J. Escudero and Luis Castedo, Universidad de A Coruna, Spain

In this paper we present a study of the effect of considering codes larger than the spreading gain when estimating the channel in a MultiCarrier Code Division Multiple Acces(MC-CDMA)

system. In order to perform this study we calculate the Cramer-Rao Bound (CRB) of channel estimators for a MC-CDMA system. CRB gives a lower bound on the error covariance matrix for a parameter vector (in our case, the channel impulse response vector), and it gives a benchmark against which algorithms performance can be compared.

11:30

PER TONE BLIND SIGNAL SEPARATION FOR A DMT-DS-CDMA SYSTEM

Geert Leus, Piet Vandaele and Marc Moonen, Katholieke Universiteit Leuven – ESAT, Belgium

In this paper, we discuss a per tone blind signal separation method for a discrete multi-tone direct-sequence code-division multiple-access (DMT-DS-CDMA) system. We show that the use of a cyclic prefix reduces the inter-chip interference (ICI). As a result, when we increase the length of the cyclic prefix, using the minimal required amount of temporal smoothing, the computational complexity of the proposed method decreases. Moreover, we demonstrate that this reduction in computational complexity even comes with a performance improvement.

11:50

AN INFORMATION THEORETIC APPROACH TO BLIND MULTIPLE ACCESS INTERFERENCE SUPPRESSION

Mónica F. Bugallo, Joaquín Míguez and Luis Castedo, Universidade da Coruña, Spain

This paper addresses the problem of blind interference cancellation in Direct Sequence Code Division Multiple Access (DS CDMA) systems. The Maximum Likelihood (ML) criterion is used to estimate the coefficients of a linear filter that suppresses both Multiple Access Interference (MAI) an Inter-Symbol Interference (ISI). Block-iterative and adaptive algorithms based on the Space Alternating Generalized Expectation-Maximization (SAGE) approach are proposed and their performance is compared with existing Linearly Constrained Minimum Variance (LCMV) blind receivers.

12:10

BLIND SINGLE-USER ARRAY RECEIVER FOR MAI CANCELLATION IN MULTIPATH FADING CDMA CHANNELS

Li-Ke Huang and Athanassios Manikas, Imperial College of Science, Technology and Medicine, UK

In this paper, a blind subspace-type single-user array direct-sequence code division multiple access (DS-CDMA) receiver is proposed based on a new joint space-time channel estimation technique for frequency-selective channels. In the proposed approach, the spatio-temporal multipath channel parameters associated with the desired user are jointly estimated with the number of identifiable paths not limited by the number of antennas. Then these estimated parameters are employed to efficiently combine the desired multipath rays while, at the same

time, (asymptotically) complete MAI cancellation is achieved.

WedAmSS2
Small Audiotorium
Video Compression: Current and Future Trends
Chair: L. Torres and E. Delp ***UPC, Spain and Purdue University, USA***

09:20

NEW TRENDS IN IMAGE AND VIDEO COMPRESSION

Luis Torres, Polytechnic University of Catalonia Barcelona, Spain
Edward J. Delp, Purdue University West Lafayette, USA

Image and video compression have been the object of intensive research in the last thirty years. The field is now mature as is proven by the large number of applications that make use of this technology. Digital Video Broadcasting, Digital Versatile Disc, and Internet streaming are only a few of the applications that use compression technology. Image and video standards have played a key role in this deployment. Now is time to ask: are there any new ideas that may advance the current technology? Have we reached a saturation point in image and video compression research? Although the future is very difficult to predict, this paper will try to provide a brief overview to where this exciting area is heading.

10:00

IS THERE ANY FUTURE IN IMAGE COMPRESSION RESEARCH?

Narciso Garcia, Universidad Politecnica de Madrid, Spain

The evolution of visual communication systems show a dangerous lack of use of new research as well as the enhancement and adaptation of old schemes. So, almost classical time prediction and spatial DCT still survive against new proposals. In parallel, new ideas coming from the images analysis and computer vision areas can increase the speed-performance rate of that schemes, although not modifying their underlying principles. On the other hand, model-based approaches seem to offer interesting alternatives, but they stem from computer graphics, using different paradigms. Finally, several private-based alternatives are been proposed to protect proprietary visual information in open domains like the Internet. Looking to all these facts, it seems that future research, if any, should be addressed to the joint real-time graphics-video transmission.

10:50

VISUAL DATA REPRESENTATION: RECENT ACHIEVEMENTS AND FUTURE DEVELOPMENTS

Fernando Pereira, Instituto Superior Técnico/Instituto de Telecomunicações, Portugal

This paper intends to review the recent achievements and the future developments in the area of visual data representation with special emphasis on the standardisation efforts. Moreover this

paper will discuss the research topics which should deserve major attention after the recent and emerging milestones in terms of representation standards.

11:10

COMPRESSION FOR RECOGNITION AND CONTENT-BASED RETRIEVAL

Antonio Ortega, Baltasar Beferull-Lozano, Naveen Srinivasamurthy and Hua Xie University of Southern California, USA

Most compression algorithms developed to date aim at achieving the best perceptual quality of the decoded media for the given rate. In this paper we consider several scenarios where the end user of the compressed data is not a human viewer or listener, but rather a known classifier or recognizer. Drawing from applications in speech recognition and image classification, as well as from simple examples, we discuss the new requirements that are imposed on the encoders under these circumstances. Our goal is to motivate the importance, and describe the associated design challenges, of achieving compression optimized for classification/recognition, rather than perceptual quality.

11:30

MVC: ADVANCED LOW BIT RATE CODEC FOR MOBILE MULTIMEDIA

Marta Karczewicz, Joni Vahteri, Jani Lainema and Bogdan Dobrin, Nokia Research Center

MVC is a proprietary low bit rate video coder developed at Nokia Research Center. This video coder contains three major elements distinguishing it from the current video coding standards: Motion compensation scheme utilizing affine motion field model, which enables very accurate prediction. Efficient pixel domain prediction for intra-coded frames utilizing image data from the surrounding reconstructed area. Powerful Multi-Shape Discrete Cosine Transform (DCT) and Karhunen-Loeve Transform (KLT) based scheme for efficient coding of the residual error. The aforementioned features provide substantial improvement for coding efficiency when compared e.g. with H.263 video coding standard; MVC coder achieves 35 % to 50 % bit rate reduction in interframe coding and 20 % to 50 % reduction in intraframe coding.

11:50

A STUDY OF JPEG 2000 STILL IMAGE CODING VERSUS OTHER STANDARDS

Diego Santa-Cruz and Touradj Ebrahimi, Swiss Federal Institute of Technology, Switzerland

JPEG 2000, the new ISO/ITU-T standard for still image coding, is about to be finished. Other new standards have been recently introduced, namely JPEG-LS and MPEG-4 VTC. This paper puts into perspective the performance of these by evaluating JPEG~2000 versus JPEG-LS and MPEG-4 VTC, as well as the older but widely used JPEG. The study concentrates on compression efficiency, although complexity and set of supported functionalities are also evaluated. Lossless compression efficiency as well as the lossy rate-distortion behavior is discussed. The principles behind each algorithm are briefly described and an outlook on the future of image coding is given. The results show that the choice of the ``best" standard depends strongly on the application at hand.

12:10

IMPROVING IMAGE COMPRESSION - IS IT WORTH THE EFFORT ?

Ralf Schaefer, Guido Heising and Aljoscha Smolic, Heinrich-Hertz-Institute, Germany

This paper presents some arguments in favour of further research in video compression because of scarce spectrum resources for mobile applications and the lack of powerful tools for scalable coding. Finally some ideas and results for metadata based compression, improved prediction and scalable schemes are presented.

WedAmSS3
Studio
Nonlinear and Non-Gaussian Adaptive Signal Processing
Chair: C. Cowan *The Queen's University of Belfast, UK*

09:20

ADAPTIVE VOLTERRA PARAMETER ESTIMATION USING A ZERO TOLERANCE OPTIMISATION FORMULATION

Georgios Stathakis, Anthony Constantinides and Tania Stathaki, Communications and Signal Processing Research Group Imperial College, UK

This paper forms a part of a series of recent studies we have undertaken, where the problem of nonlinear signal modelling is examined. We assume that an observed "output" signal is derived from a Volterra filter that is driven by a Gaussian input. Both the filter parameters and the input signal are unknown and therefore the problem can be classified as blind or unsupervised in nature. In the statistical approach to the solution of the above problem we seek for equations that relate the unknown parameters of the Volterra model with the statistical parameters of the "output" signal to be modelled. These equations are highly nonlinear and their solution is achieved through a novel constrained optimisation formulation. The results of the entire modelling scheme are compared with recent contributions.

09:40

SEA CLUTTER & CHAOS: IMPROVED SURROGATE-DATA TESTS

B.Mulgrew, C.P.Unsworth, M.R.Cowper and S.McLaughlin, The University of Edinburgh, UK

Currently there is contention as to the nature of sea clutter for high resolution radar. Conventionally, sea clutter has been modelled as a compound stochastic K-distribution, originally suggested by Ward et al. However, recent work by Haykin et al. has suggested that the clutter can be modelled as a nonlinear deterministic process ,otherwise referred to as a chaotic process. The paper presented here uses a new surrogate test which is designed specifically for this problem. The test is designed with a null hypothesis (H_0= The data can be approximated to by a compound stochastic K-distribution). Therefore, acceptance of such a test will accept the conventional K-distribution as a viable model sea clutter. In addition, a new surrogate statistic is introduced which is used to reject/accept the null hypothesis. This statistic is the normalised mean square error (NMSE) from a predictor and is a statistic which can be applied to any type of time-series. An overview of the method is presented together with results for a number a sea clutter data sets.

10:00

BLIND SEPARATION OF COMPLEX-VALUED SIGNALS BY REAL-VALUED IN-PHASE AND QUADRATURE ROTATIONS

Ira J Clarke, Defence Evaluation and Research Agency, UK

We propose a novel method for applying real-valued independent component analysis (ICA) to complex-valued multi-sensor data that comprises instantaneous linear mixtures of co-channel non-Gaussian independent signals. We examine, for non-ideal practical conditions, the optimality of cumulant-based ICA and blind signal separation approaches in terms of the standard criterion for statistical independence and extend this theory to the complex-valued case. We show that this justifies estimation of in-phase and quadrature mixing parameters and, for finite data, avoids the difficulty of optimising a selected subset of real and complex-valued cumulants. Computational efficiency is also improved.

10:50

STABILITY CONDITIONS FOR ADAPTIVE ALGORITHMS WITH NON-QUADRATIC ERROR CRITERIA

Shin'ichi Koike, NEC Corporation

In this paper, an upper bound of the step size and that of the initial value of the Mean Squared Error (MSE) are derived as stability conditions for FIR adaptive filters with non-quadratic error criteria, i.e., with cost functions of r + 1 st power of the error, where correspondingly r th power of the error is used in the correlation multiplier for tap weight adaptation (r is non-negative). A chart showing the relationship between the MSE and its increment at a discrete time instant is presented, from which two kinds of zeroes, stable and unstable, are solved for r > 1. Stability conditions are discussed in relation to the chart and its zeroes. Unlike the Sign Algorithm (r=0), the LMS Algorithm (r=1), etc., stability for those algorithms with r > 1 is found to crucially depend on the initial value of the error, hence the initial value of the tap weights. Simple formulae of the upper bounds and the zeroes are given for the Least Mean Fourth Algorithm (LMFA, r=3). Simulations with some examples for the LMFA verify the stability conditions derived.

11:10

A NEW APPROACH TO STATIONARY COST FUNCTION ADAPTATION

Corneliu Rusu, Tampere University of Technology, Finland
Colin F. N. Cowan y, The Queen's University of Belfast, Ireland

The goal of this paper is to present a new approach of stationary cost function adaptation algorithm. First the derivation of the algorithm is considered, then computer simulation results are provided.

11:30

A MODIFIED VITERBI ALGORITHM FOR NON-GAUSSIAN INTERFERENCE SCENARIOS

John Hudson, Nortel Networks, UK
Steve McLaughlin, University of Edinburgh Kings buildings, UK

Viterbi sequence detectors and Turbo decoders are widely used in the receivers of mobile communications systems. They are offer an efficient means of implementing a near-maximum likelihood receiver. However, Viterbi sequence estimators are usually based on the assumption of Gaussian noise and/or interference statistics. It is becoming clear that for third generation and enhanced second generation cellular mobile packet systems that Gaussian-ness cannot be taken for granted and in such scenarios it is not clear what performance a Viterbi receiver will offer. This paper explores this issue and develops a modified Viterbi for just such situations and presents performance results.

11:50

STATISTICAL SIGNAL AND ARRAY PROCESSING NONLINEAR PROCESSING OF NONGAUSSIAN AND NONCOHERENT SIGNALS IN THE PRESENCE OF COLORED NOISE

V.A. Potapov, Mints Radiotechnical Institute, Russia

The problem of nonlinear processing of nongaussian and noncoherent signals is common in many applications such as remote sensing, hydroacoustics, radioastronomy and communication technologies. Optimal solution of this problem requires taking into account the space-time statistical properties of signals, medium and noise, which are usually known a priori. One of the best methods is suggested by the theory of conditional Markov processes [1-3]. Using this technique, we may describe the signal fluctuations in the form of stochastic differential equations. According to the standard procedure, to obtain the filtering algorithms, we must write the evolution equation for the conditional density functional. We next split this equation into the differential equations for the desired estimates of signal fluctuations (the desired algorithms) and for the second-order conditional cumulants (describing the filtering errors). The disadvantage of this procedure is that we may write closed equation for the conditional density functional only for the case of signals received in the presence of white noise. In this paper on the base of the new functional approach [4,5], we consider the case of signals, which are observed in the presence of colored noise only. We obtain optimal algorithms for a set of modulated signals and colored noises, including the noise described by Ornstein-Uhlenbeck process. In addition, we numerically verify these algorithms.

12:10

FAST LEAST MEAN M-ESTIMATE ALGORITHMS FOR ROBUST ADAPTIVE FILTERING IN IMPULSE NOISE

Yuexian Zou, Shing-Chow Chan, and Tung Sang Ng, The University of Hong Kong, Hong Kong

Adaptive filters with suitable nonlinear devices are very effective in suppressing the adverse effect due to impulse noise. In a previous work, the authors have proposed a new class of nonlinear adaptive filters using the concept of robust statistics [1, 2]. The robust M-estimator is used as the objective function, instead of the mean square errors, to suppress the impulse noise. The optimal coefficient vector for such nonlinear filter is governed by a normal equation which can be solved by a recursive least squares like algorithm with O(N^2) arithmetic complexity, where N is the length of the adaptive filter. In this paper, we generalize the robust statistic concept to least mean square (LMS) and transform domain LMS algorithms. The new fast nonlinear adaptive filtering algorithms called the least mean M-estimate (LMM) and transform domain LMM (TLMM) algorithms are derived. Simulation results show that they are robust to impulsive noise in the desired and input signals with an arithmetic complexity of order O(N).

WedAmSS4
Hall A
Soft Computing Methods
Chair: J. Lampinen *Helsinki University of Technology, Finland*

09:20

BAYESIAN TECHNIQUES FOR NEURAL NETWORKS - REVIEW AND CASE STUDIES

Jouko Lampinen and Aki Vehtari, Helsinki University of Technology, Finland

We give a short review on Bayesian techniques for neural networks and demonstrate the advantages of the approach in number of industrial applications. Bayesian approach provides a principled way to handle the problem of overfitting, by averaging over all model complexities weighted by their posterior probability given the data sample. The approach also facilitates estimation of the confidence intervals of the results, and comparison to other model selection techniques (such as committee of early stopped networks) often reveals faulty assumptions in the models. In this contribution we review the Bayesian techniques for neural networks and present comparison results from several case studies that include regression problems, classification problems and inverse problems.

09:40

A LEARNING VECTOR QUANTIZATION ALGORITHM FOR PROBABILISTIC MODELS

Jaakko Hollmen and Olli Simula, Helsinki University of Technology, Finland
Volker Tresp, Siemens Corporate Technology

In classification problems, it is preferred to attack the discrimination problem directly rather than indirectly by first estimating the class densities and by then estimating the discrimination function from the generative models through Bayes's rule. Sometimes, however, it is convenient to express the models as probabilistic models, since they are generative in nature and can handle the representation of high-dimensional data like time-series. In this paper, we derive a discriminative training procedure based on Learning Vector Quantization (LVQ) where the codebook is expressed in terms of probabilistic models. The likelihood-based distance measure is justified using the Kullback-Leibler distance. In updating the winner unit, a gradient learning step is taken with regard to the parameters of the probabilistic model. The method essentially departs from a prototypical representation and incorporates learning in the parameter space of generative models. As an illustration, we present experiments in the fraud detection domain, where models of calling behavior are used to classify mobile phone subscribers to normal and fraudulent users. This is an extension of our earlier work in clustering probabilistic models with the Self-Organizing Map (SOM) algorithm to the classification domain.

10:00

AN IRWLS PROCEDURE FOR SVR

F. Perez-Cruz, P. L. Alarcon-Diana and A. Artes-Rodriguez, Universidad de Alcala, Spain
A. Navia-Vazquez, Universidad Carlos III de Madrid, Spain

In this paper we propose an Iterative Re-Weighted Least Square procedure in order to solve the Support Vector Machines for regression and function estimation. Furthermore, we include a new algorithm to train Support Vector Machines, covering both the proposed approach instead of the quadratic programming part and the most advanced methods to deal with large training data sets. Finally, the performance of the method is assessed by selected examples which show that the training time is much shorter and the memory requirements much less than the employed ones by current methods.

10:50

A HARDWARE-FRIENDLY SOFT-COMPUTING ALGORITHM FOR IMAGE RECOGNITION

Masakazu Yagi, Masayoshi Adachi and Tadashi Shibata, The University of Tokyo, Japan

A robust image recognition algorithm has been developed aiming at direct implementation in a bio-inspired hardware accelerator chip. The characteristic features in an original gray-scale image of 64x64-pels (i.e., a 4096-dimension vec-tor) are extracted by a newly-developed Principal Axes Pro-jection (PAP) method and compressed to form a 64-dimension characteristic vector. Despite the large dimension-ality reduction, the essential features in the original image are well retained in the vector representation. As a result, very robust image recognition has become possible by using a simple template matching technique. Although the matching with a large number of templates is computationally very expensive, we rely upon the already-developed vector matching LSI fs featuring about 1000 GOPS performance for template matching [1-3]. The present algorithm has been successfully applied to medical radiograph analysis and handwriting pattern recognition, and its robust nature in rec-ognition tasks has been proven by intensive computer simu-lation.

11:10

PATTERN RECOGNITION WITH NOVEL SUPPORT VECTOR MACHINE LEARNING METHOD

Mikko Lehtokangas, Tampere University of Technology, Finland

The concept of optimal hyperplane has been recently proposed in the context of statistical learning theory. The important property of an optimal hyperplane is that it provides maximum margins to each class to be separated. Obviously, such a decision boundary is expected to yield good generalization. Currently, the support vector machines (SVM) are probably one of the very few models (if not the only ones) that make use of the optimal hyperplane concept. In this study we investigate the basic SVM method and point out some problems that may arise especially in

large scale problems with abundant data. Moreover, we propose a novel SVM type method that aims to avoid the problems found in the basic method. The experimental results demonstrate that the proposed method can give very good classification performance. However, the results also point out another potential problem in the SVM scheme which should be considered in the future studies.

11:30

A HIDDEN MARKOV MODEL FOR METRIC AND EVENT-BASED DATA

Jaakko Hollmen, Helsinki University of Technology, Finland
Volker Tresp, Siemens Corporate Technology

The question of data representation is central to any data analysis problem. Ideally, the representation should faithfully describe the domain to be analyzed and in addition, the model used should be able to process such a representation. In practice, however, the modeler must often compromise how the problem is described, since the class of possible representations is constrained by the model. This problem may be circumvented by extending conventional models to handle more unconventional data representations. These data are often found in industrial environments and especially in telecommunications. In this paper, we consider an extension of hidden Markov models (HMM) for modeling data streams, which switch between metric and event-based representations. In a HMM, the representation of the observed data is constrained by the emission probability density. Since this density can not change its representation once it is fixed, modeling data streams involving different types of data semantics can be difficult. In the extension introduced in this paper, an additional data semantics variable is introduced, which is conditional on the hidden variable. Furthermore, data itself is conditioned on its semantics, which enables correct interpretation of the observed data. We briefly review the essentials of HMMs and present our extended architecture. We proceed by introducing inference and learning rules for the extension. As an application, we present a HMM for user profiling in mobile communications networks, where the data exhibits switching behavior.

11:50

SCALE-BASED CLUSTERING WITH LATENT VARIABLES

F.M. Frattale Mascioli, M. Panella, A. Rizzi, and G. Martinelli, University of Rome "La Sapienza", Italy

The use of clustering systems is very important in those real-word applications where an efficient, both accurate and economical, representation of the data to be processed is necessary. When dealing with statistical models, such a problem is usually related to the estimate of their parameters in the Maximum Likelihood context. At this regard, we propose an EM-based algorithm that uses a hierarchical growing approach, based on a given splitting procedure, to determine in an efficient way the parameters of a mixture of Gaussian clusters. The splitting procedure and the determination of the correct number of clusters are based on a scale-based approach, which imitates the human perception of images. Moreover, each cluster is modelled by means of latent variables, which also ensure a local linear dimension reduction of the data being processed.

12:10

SIGNAL PROCESSING AND SONIFICATION OF SEISMIC ELECTROMAGNETIC RADIATION IN ELF BAND

Seiji Adachi, Hiroshi Yasukawa, Ichi Takumi and Masayasu Hata, Nagoya Institute of Technology, Japan

We develop a signal processing method appropriate for detecting electromagnetic radiation due to the earthquake activities. The radiation is usually accompanied by a background noise that is mainly caused by atmospheric discharges in the tropical regions. Data representing the seismic radiation is presented as sound via the concept of sonification. This is useful to immediately find out anomalous seismic radiations, which are often followed by a disastrous earthquake, from massive data collected from over forty observation stations, It is illustrated that the auditory display is worth for the future earthquake prediction system.

WedAmOR1
Room 200
Speech Production, Perception and Synthesis
Chair: M. Sams *Helsinki University of Technology, Finland*

09:20

REDUCTION OF MUSICAL NOISE GENERATED BY SPECTRAL SUBTRACTION BY COMBINING WAVELET PACKET TRANSFORM AND WIENER FILTERING

Sofia Ben Jebara, Amel Benazza-Benyahia, Abdelsalem Ben, Ecole Sup'erieure des Communications de Tunis, Tunisia

This paper presents a new method for musical noise attenuation induced by spectral subtraction. The proposed technique consists in combining a wavelet packet decomposition with Wiener filtering. Simulations demonstrate how this artifact is reduced without affecting the original speech intelligibility.

09:40

PHYSICALLY BASED FACIAL MODEL

Vili Jussila, Mikko Sams and Kimmo Kaski, Helsinki University of Technology, Finland
Oili Salonen, Helsinki University Central Hospital, Finland

We have constructed a dynamic face model, which is based on the anatomical structures of a real face. Accurate head geometry was obtained from Magnetic Resonance Images (MRIs). An image segmentation program was used to convert the volumetric MRI data to a polygonal representation of the head. The most detailed models we have used have contained approximately 35000 vertices and 70000 polygons. We have constructed separate models for the behavior of the skin, muscles and skull. Contracting the modeled muscles produces facial movements.

10:00

REALIZATION OF A VOWEL-PLOSIVE-VOWEL TRANSITION BY A TUBE MODEL

K. Schnell, A. Lacroix, Goethe-University Frankfurt, Germany

Tube models, realized by lattice filters, can be used for the production of speech signals, since the model simulates the propagation of plane sound waves through the vocal tract. In this contribution the model contains a time dependent glottis coefficient. The parameters of this time dependent model can be estimated by minimization of an error, which describes the spectral distance between the tube model and the speech signal. The estimated vocal tract areas are comparable to areas, which are obtained by NMR or X-ray investigations. A procedure is proposed to model the voiced transition from a vowel to a plosive and to another vowel (VPV).

This is realized by a vocal tract area transition of the tube model. The areas of the constriction of the plosive and the areas of the voiced sounds are estimated from speech signals.

10:50

PITCH SYNCHRONOUS RESIDUAL EXCITED SPEECH RECONSTRUCTION ON THE MFCC

Zbynek Tychtl and Josef Psutka, University of West Bohemia, Czech Republic

Practical applications of speech recognition and dialogue systems bring sometimes a requirement to synthesize or reconstruct the speech from the saved or transmitted mel-frequency cepstral coefficients (MFCCs). Presented paper describes several approaches to the speech reconstruction based on the MFCC parameterization. Approaches differ mainly in their various possible excitations. Let us mention that for designing a MFCC reconstruction module we applied some principles usually used in speech recognition and synthesis process. We suppose the speech reconstructor together with speech synthesizer and recognizer to be a part of a speech dialogue system developed in our department.

11:10

A COMPLETE TEXT-TO-SPEECH SYSTEM FOR THE SLOVENIAN LANGUAGE

Tomaz Sef and Matjaz Gams, "Jozef Stefan" Institute, Slovenia

The Slovenian text-to-speech engine is a modular system consisting of four independent modules (text normalization, grapheme-to-phoneme conversion, prosody generation and segmental concatenation), which are pipelined together. Each module is responsible for one portion of the problem of converting from text into speech. The first two modules comprises such tasks as end-of-sentence detection, abbreviation and number expansion, special formats conversion, morphological and contextual analysis, phonological modeling. In order to generate rules for our synthesis scheme, data was collected by analysing the readings of ten speakers, five males and five females. A two-level approach has been used for duration modelling and so-called superpositional approach at pitch modelling. The system is based on the concatenation of speech units, diphones and some frequently used polyphones, using TD-PSOLA technique.

11:30

PARAMETERIZED VISUAL SPEECH SYNTHESIS AND ITS EVALUATION

Riikka Möttönen, Jean-Luc Olivés, Janne Kulju, and Mikko Sams Helsinki University of Technology, Finland

We have constructed an audio-visual text-to-speech synthesizer for Finnish by combining a dynamic facial model with an acoustic speech synthesizer. The visual speech is based on a letter-to-viseme mapping and the animation is created by linear interpolation between the visemes. A viseme is defined by 12 parameter values. In a recent study we showed that visual speech

increases the intelligibility of both natural and synthetic auditory speech [5]. We have upgraded our visual speech synthesis by adding the tongue model and improving the speech parameters on the basis of the intelligibility study. Here we show data from a new intelligibility study demonstrating the improved performance of the synthesizer. Presenting the visual speech in three-dimensional space did not further improve the intelligibility.

11:50

DEVELOPMENT OF A RULE-BASED SPEECH SYNTHESIS SYSTEM FOR THE JAPANESE LANGUAGE USING A MELP VOCODER

Naofumi Aoki, Hokkaido University, Japan

A rule-based speech synthesis system for the Japanese language, which employs a MELP (Mixed Excitation Linear Prediction) vocoder as its speech synthesizer, was implemented. This paper especially describes some speech synthesis techniques utilized in our system. Since the MELP vocoder developed in this study could effectively gain the naturalness of voiced consonants as well as purely voiced speech, the implemented system could succeed in enhancing the voice quality of synthesized speech more than a system that employed a conventional normal LPC (Linear Predictive Coding) vocoder.

12:10

ROBUST RECURSIVE AR SPEECH ANALYSIS BASED ON QUADRATIC CLASSIFIER WITH SLIDING TRAINING DATA SET AND A HEURISTIC DECISION THRESHOLD

Milan Markovic, Institute of Applied Mathematics and Electronics

A robust recursive procedure for identification of nonstationary AR speech model based on a quadratic classifier with a heuristic decision threshold is proposed and evaluated. A comparative experimental analysis is done through processing natural speech signal with voiced and mixed excitation segments. Obtained results show that the proposed robust procedure based on the quadratic classifier with sliding training data set and the heuristic decision threshold achieves more accurate AR speech parameter estimation and provides improved tracking performance.

WedAmPO1
Exhibition Hall
Higher Order Statistics
Chair: R. Pearson ***ETH Zuerich, Switzerland***

BLIND FOCUSING WIDE-BAND ARRAY PROCESSING

A. Bendjama, S.D.E.M /URA-CNRS, France
S. Bourennane, Institut Fresnel /UMR-CNRS, France

The coherent signal subspace method based on the second-order statistics has been proposed for estimating the directions of arrival (DOA) of multiple wide-band sources. These techniques are ineffective in presence of the Gaussian noise and also when the propagation model is unknown. In this paper, the extension of the focusing algorithms to the noisy data is developed. This extension is based on the higher-order statistics cumulant. Indeed, the use of the fourth order cumulant leads to eliminate the Gaussian noise contribution and then, to improve the high resolution algorithms on the one hand, on the other hand to determine the unbiased focusing matrices needed for estimating the coherent signal subspace. To improve the cumulant matrix estimation, a spatial average of slice cumulant matrices is developed. The performances of our algorithm are studied by the use simulated data.

A CRITERION FOR DERIVING THE SUB/SUPER GAUSSIANITY OF SOURCE SIGNALS FROM THEIR MIXTURES

Yannick Deville and Mohammed Benali, University of Toulouse, France

In this paper, we consider the situation when two linear instantaneous mixtures of two independent source signals are available. We aim at determining whether the source signals are sub- or super-Gaussian, using only their observed mixtures. To this end, we propose a criterion based on the roots of a specific polynomial, whose coefficients depend on the estimated cross-cumulants of the observed signals. The effectiveness of this approach is demonstrated by means of experimental tests.

MATERIAL CLASSIFICATION BY HOS ANALYSIS OF ULTRASONIC SIGNALS

Ramón Miralles and Luis Vergara, Universidad Politécnica de Valencia, Spain

Starting with a theoretical model of a dispersive material illuminated by an ultrasonic pulse we developed a classifier based on the use of third order cumulants. We will compare the results with those obtained with a similar classifier but using only second order statistics. The underlying application is the classification of materials by non-destructive ultrasonic testing.

BISPECTRAL ESTIMATION: A MULTITAPER APPROACH

Yngve Birkelund and Alfred Hanssen, University of Tromsø, Norway

We have compared the bias and variance properties of several multitaper based bispectral estimators, with those of frequency smoothed biperiodograms, and Hanning-tapered/frequency-smoothed biperiodograms. The test processes are a white Gaussian process and a non-Gaussian ARMA(2,2) process. We found that the bias and variance properties of the multitaper bispectral estimators are governed mainly by a quantity we call the total bispectral window. We conclude that the multitaper based bispectral estimator are superior to the conventional non-parametric bispectral estimators.

WAVELET-POLYSPECTRA: PRINCIPLES AND PROPERTIES

Yngvar Larsen and Alfred Hanssen, University of Tromsø, Norway

We present a stringent definition of higher-order evolutionary spectra. On this basis, we define wavelet-polyspectral densities as a way of dealing with non-stationarities in higher-order statistics. We propose a simple wavelet-polyspectral estimator, and we discuss its statistical properties. The proposed wavelet-polyspectral analysis tool is demonstrated by a numerical example. It is concluded that the wavelet-polyspectra have desirable properties for the analysis of data that are simultaneously non-stationary and non-Gaussian/non-linear.

NUMERICAL AND ANALYTICAL SOLUTIONS TO THE DIFFERENTIAL SOURCE SEPARATION PROBLEM

Frederic Abrard, Yannick Deville and Mohammed Benali, University of Toulouse, France

The aim of this paper is to present several algorithms which solve the blind source separation (BSS) problem when stationary noises are added to the source signals in the linear instantaneous mixture context. We use a new criterion based on the differential normalized kurtosis that we developed in a previous paper. The different algorithms are then applied to mixtures of two sources signals and an additive noise.

LOWER ORDER STATISTICS: A NEW APPROACH FOR PROBABILITY DENSITY FUNCTIONS DEFINED ON R+

Jean Marie Nicolas, Alain Maruani, Ecole Nationale Superieure des Telecommunications, France

Probability Density Functions defined on R+ can be successfully modeled with the help of the Mellin Transform : this rather underrated transform is well suited for such functions so that we propose the new definitions of "second kind" characteristic functions based on this transform. By this way, second kind moments and second kind cumulants can also be defined, so that multiplicative noise, which can be seen as a Mellin convolution of Probability Density Function, can be easily analysed. The estimation of PDF parameters can be improved with this new

approach. Indeed, it is possible to deal with lower order statistics so that negative moments can be defined in such a way that the variances of the estimators are reduced. The analytical formulation of this variance is proposed and validated on numerical simulations for the Gamma law. With this new approach, a same estimator variance is reached with a more reduced set of samples than traditional methods.

MULTITIME-FREQUENCY CLASSIFIERS FOR NOISY MODULATIONS

M. Colas, G. Gelle and G. Delaunay, Université de Reims Champagne-Ardenne, France

This communication presents two new classification algorithms based on Time Varying Higher Order Statistic (TVHOS). These two algorithms benefit from the advantageous localization properties provided by TVHOS4 for instantaneous frequency laws disrupted by a multiplicative noise. The classification scheme used is a bank of several normalized TVHOS4 correlators. Simulations illustrate successful performance of the classification algorithms for different situations and especially if the SNR is higher than -6 dB.

WedAmPO2
Exhibition Hall
Speech Coding
Chair: I. Tabus ***Tampere University of Technology, Finland***

NONLINEAR PREDICTIVE MODELS COMPUTATION IN ADPCM SCHEMES

Marcos Faundez Zanuy, Escola Universitaria Politecnica de Mataro, Spain

Recently several papers have been published on nonlinear prediction applied to speech coding. At ICASSP'98 we presented a system based on an ADPCM scheme with a nonlinear predictor based on a neural net. The most critical parameter was the training procedure in order to achieve good generalization capability and robustness against mismatch between training and testing conditions. In this paper, we propose several new approaches that improve the performance of the original system in up to 1.2dB of SEGSNR (using bayesian regularization). The variance of the SEGSNR between frames is also minimized, so the new scheme produces a more stable quality of the output.

MULTIPATH TREE-STRUCTURED VECTOR QUANTIZERS

Jean Cardinal, Université Libre de Bruxelles, Belgium

Tree-structured vector quantization (TSVQ) is a popular mean of avoiding the exponential complexity of full-search vector quantizers. We present two new design algorithms for TSVQ in which more than one path can be chosen at each internal node. The two algorithms differ on the way the paths are chosen. In the first algorithm the number of paths is fixed and the encoding is similar to the M-algorithm for delayed decision coders. In the second algorithm, the paths are chosen adaptively at each node, according to a $(1+\epsilon)-$nearest neighbor rule. We show the performances of the two algorithms on an AR(1) gaussian process, and observe that the adaptive method is the best one. Those methods allow near full-search performances at a fraction of the complexity cost.

THE SIGNAL QUANTIZATION USING NON-LINEAR ADAPTIVE QUANTIZATION FUNCTIONS

Hiroto Saito, Takashi kohama, Shogo Nakamura and Masahide Yoneyama, Tokyo Denki University, Japan

In this paper, we would like to describe a signal quantizaiton algorithm using the GA(Genetic Algorithm) and Probabilistic method. The quantization are realized by using a nonlinear quantization functions. If we apply a quantization for phisical signal(ex. electrocardiogram), the observation data must be reconstructed as faithfully as possible. In the case of encoding or decoding a voice mail, the speech signal is desired to be compressed. The quality and the compression rate are equivalent to a conventional compression such as the CELP(4.8kbp).

SOFT-DECISION DECODING OF BINARY BLOCK CODES IN CELP SPEECH CODING

Tomi Mikkonen and Konsta Koppinen, Tampere University of Technology, Finland

It is shown that the residual codebook in CELP speech coders can be replaced with a spherical code derived from an error-correcting code with low encoding complexity. A fast soft-decision decoding algorithm for binary block codes is modified to search the closest code vector in the spherical code. The quantization is done in the spectral domain by concentrating the quantization power on the most important components according to the LPC-parameters. The Voronoi cells turn out to be more spherical than in a corresponding scalar quantizer. Simulation results indicate that the proposed quantizer performs clealy better than traditional CS-ACELP with signals which are hard for CELP coders. However the proposed method is inferior to CS-ACELP with signals for which the synthesis is highly energy concentrating such as clean male speech.

PREDICTIVE LSF COMPUTATION

Bogdan Dumitrescu, Ioan Tabus, Tampere University of Technology, Finland

This paper presents a new method for computing the line spectral frequencies (LSF). The new features of the method consist in: (1) the use of predictions of the LSFs as starting points for the search of the current LSFs, (2) the fast bracketing of the LSFs working on a grid of points and (3) the refinement of LSFs by a variable number of bisections. The method provides an easy control of the trade off between necessary accuracy and worst case computational time and is easily tunable to different orders of prediction polynomials. The experimental results show an improvement of the average performance with more than 30% with respect to a frequently used method (Kabal-Ramachandran).

DESIGN AND IMPLEMENTATION OF AN LD-CELP CODEC

Fesal Toosy, Omer Bin Abdul Aziz, Shahid Qayyum, Muhammad Asim Rafiq and Abdul Ghafoor, University of Engineering & Technology, Pakistan

This paper discusses the design and implementation of a low bit rate codec along with its performance at different bit rates. The International Telecommunications Union's G.728 CELP speech coder is specifically designed for low coding delay and toll quality speech at a rate of 16kbps. Here, we present the design of a CELP(Code Excited Linear Prediction) algorithm similar to the above coder, but can compress speech up to rates like 6.4kbps(as in the G.723.1 standard). This codec shares the advantages of both the above standards; it has a low delay (up to 1.25ms) and greater compression, even though the techniques used are simpler. This advantage makes this coder suitable for applications like video conferencing, mobile communications etc.

WedAmPO3
Exhibition Hall
Adaptive Signal Processing
Chair: L. Yaroslavski *Tel Aviv University, Israel*

A FAST CONVERGING AND STABLE ADAPTIVE FILTERING ALGORITHM FOR STEREOPHONIC ACOUSTIC ECHO CANCELLATION

Nisachon Tangsangiumvisai and Jonathon A. Chambers Imperial College, London, U.K

A new automatic switching algorithm (ASA) has been introduced in this paper for use in stereophonic acoustic echo cancellation (SAEC) in order to achieve fast convergence rate with low computational cost whilst guaranteeing algorithm stability. This algorithm has been designed to overcome the problem of "abrupt changes in the acoustic echo paths" (ACEP) and "double talk" (DT). Two different detectors, one based upon the estimate of the autocorrelation between the error signal e(n) and e(n-1) and another based upon the energy ratio of the desired signal, d(n), at different conditions, are employed within the ASA to detect ACEP and DT respectively. Simulation results are presented based on real stereophonic signals to support this idea.

IMPROVING PERFORMANCES OF COMPLEMENTARY PAIR LMS ALGORITHM

Radu Ciprian Bilcu, Pauli Kuosmanen, Corneliu Rusu, Tampere University Of Technology, Finland

In this paper we discuses how to improve the behavior of the CP-LMS algorithm that was proposed by Min-Soo Park and Woo-Jin Song [5]. The difference between the new algorithm and CP-LMS algorithm is that the new algorithm uses two LMS algorithms that operate in parallel, one with large and fixed step and another with variable step. The coefficients of the novel variable step algorithm are re-initialized by the coefficients of the algorithm with fixed step whenever possible. In the meantime the step-size of the proposed algorithm is increased or decreased if the algorithm is respectively far or near to the optimum.

ADAPTIVE SINGLE-TONE FREQUENCY ESTIMATION BASED ON AUTOREGRESSIVE MODEL

H. C. So, The Chinese University of Hong Kong, Hong Kong

A novel adaptive filter algorithm based on the linear prediction property of sinusoidal signals is proposed for estimating the frequency of a real tone in white noise, assuming that the noise power is known a priori. Using the least mean square algorithm, the estimator is computationally efficient and it provides explicit frequency measurements on a sample-by-sample basis. Convergence behavior and variance of the estimated frequency are derived. Computer simulations are included to corroborate the theoretical analysis and to demonstrate the capability of the method of tracking time-varying frequencies.

CONTROLING QOS AT THE APPLICATION LEVEL FOR MULTIMEDIA APPLICATIONS USING ARTIFICIAL NEURAL NETWORKS: EXPERIMENTAL RESULTS

George A. Rovithakis, Argyris Matamis and Michael Zervakis, Technical University of Crete, Greece

In this paper we implement a recurrent high order neural network based controller (built into a Client-Server architecture), in order to provide QoS control. The scheme is based on a recently proposed QoS control algorithm with proven stability and robustness properties. Mild assumptions have been used to construct a user satisfaction function that depends on two media characteristics (i.e. color depth and frame rate). The above, along with bandwidth measurements obtained at per frame basis, provide the essential information to the controller, which outputs the appropriate values for the media characteristics in order to achieve the required user satisfaction without violating the available bandwidth constraint. The implementation is performed on a TCP/IP network and the QoS control, which is applied at the application level, is shown to hold a real time property. Experimental results highlight its performance.

PERFORMANCE ANALYSIS OF AN ADAPTIVE ALGORITHM FOR MINOR COMPONENT ANALYSIS

Hideaki Sakai, Shigeyuki Miyagi and Kouji Minamiyama, Kyoto University, Japan

The single minor component extraction algorithm proposed by Douglas et al. is extended to a multiple minor components extraction algorithm by combining the deflation technique and the Gram-Schmidt orthogonalization. The second order analysis for the multiple case is presented by applying the averaging method or the ordinary differential equation (ODE) method. The error covariances of the estimated minor components are derived and the validity of these evaluations is demonstrated by simulations.

COMPARISON OF CDMA SPATIO-TEMPORAL RECEIVERS STRUCTURES FOR MULTI-RATE APPLICATIONS

Marie Le Bot, Philippe Forster and Luc F'ety, CNAM, France

Code division multiple access (CDMA) is the multiple access scheme retained for the third generation wireless communication systems, that must support high data rate transmissions, and operate over jammed Rayleigh fading channels. In the current work, we propose and analyze multi-antenna spatio-temporal adaptive receiving structures of detection at the base station, able to cope with degradations due to multiple access interferences. Their ability to reject part of the jammers, and their resistance to near-far effects is discussed.

STABILITY AND PERFORMANCE OF ADAPTIVE ALGORITHMS FOR MULTICHANNEL BLIND SEPARATION AND DECONVOLUTION

Allen G. Lindgren, University of Rhode Island, USA
Thomas P. von Hoff and August N. Kaelin, ETH Zurich, Switzerland

The problem of blind source separation is reviewed and the stability properties of the classic adaptive algorithms with non-score nonlinearities are derived. In addition to changing the nonlinearity, when the algorithm yields an unstable separating solution, a new stabilization procedure is proposed. The stability findings are then extended to the problems of single-channel and multi-channel blind deconvolution. The method of transposing and/or time-reversing allows a given nonlinearity to yield a stable algorithm for any situation of nonlinearities and source distributions. The method is shown to be effective by theory and simulation. For special cases the steady-state error performance is derived for all algorithms.

ON THE ROBUSTNESS OF ADAPTIVE PREDICTIVE SCHEME FOR TRACKING RANDOMLY TIME-VARYING CHANNELS

S. Ben Jebara, Sup'Com, Tunisia
H. Besbes, Concordia University, Canada
M. Jaidane, L.S.Telecoms, Tunisia

This paper deals with the use of adaptive predictive algorithms in order to improve tracking performances of classical identification scheme. The performances of coupled structure depend on the prediction device. In this paper, we present a new robust algorithm for the predictive scheme. The robustness deals with insensibility of proposed algorithm to the input statistics variations, namely the power and fourth order statistics.We illustrate our contribution by performances evaluation of the proposed structure in double non stationary context: on tracking random walk channel with speech input signal.

SELF-ADAPTIVE EVOLUTION STRATEGIES FOR ARMA MODEL IDENTIFICATION

G. N. Beligiannis, E. N. Demiris and S. D. Likothanassis, University of Patras, Greece

This work presents the application of Evolutionary Computation techniques to the identification (order selection and parameter estimation) of an AutoRegressive Moving Average model (ARMA). Our method combines the effectiveness of the Multi Model Partitioning (MMP) theory with the robustness of the Genetic Algorithms (GAs) in order to give optimum estimations of the noise sequence embedded to the moving average terms of the model. Although the noise sequence's coding is very complicated, the proposed algorithm succeeds better results compared to the classical methods, since it has the ability to search the whole values' range. This is because, in contradiction with all the known classical methods, our algorithm is able to estimate with high precision the unknown parameters even in the case of large order in the moving average terms of the model.

SIMPLIFIED RLS ALGORITHM FOR ADAPTIVE IFIR FILTERING

Leonardo S. Resende, Federal University of Santa Catarina

In this paper we introduce the optimum interpolated Wiener filter, which results from an original and elegant linearly-constrained approach to the interpolated transversal filtering problem. A GSC-type structure for adaptive IFIR filtering is proposed, in which least-square algorithms can be directly employed for updating only the nonzero coefficients of the sparse filter. The performance of the RLS algorithm applied to the AIFIR filtering is illustrated by simulations and compared with that of the LMS algorithm.

WedAmPO4
Exhibition Hall
Image Analysis
Chair: S. Marshall ***University of Strathclyde, UK***

THE VIPER ALGORITHM: A NEW APPROACH FOR DETECTION USING 2-D ADAPTIVE SAMPLING

Davide Avagnina, Letizia Lo Presti and Paolo Mulassano, Politecnico di Torino, Italy

This paper describes a new technique of irregular and adaptive sampling developed to process still images containing objects of interest over a non interesting background. The method allows the identification of interesting elements and is able to choose the sampling interval more adequate to obtain a correct description of them. The not interesting background is roughly sampled and this particular aspect of the method is an important advantage. Objects are first detected by a preliminary sampling phase and then a resampling process begins, until a convergence test in the Fourier domain is satisfied and stops the process. Examples of applications are given.

FONT CLUSTERING AND CLASSIFICATION IN DOCUMENT IMAGES

Serdar Ozturk and Bulent Sankur, Bogazici University, Turkey
A. Toygar Abak, Marmara Research Center, Turkey

Clustering and identification of fonts in document images impacts on the performance of optical character recognition (OCR). Therefore font features and their clustering tendency are investigated. Font clustering is implemented both from shape similarity and from OCR performance points of view. A font recognition algorithm is developed to identify the font group with which a given text was created.

A HIERARCHICAL SEGMENTATION ALGORITHM BASED ON HEPTA-TREE

J.N. Provost, P. Rostaing and C. Collet, Research Institute of the French Naval Academy, France

This paper is concerned with a Hierarchical Markov Random Field (HMRF) algorithm for image segmentation, based on samples belonging to a hexagonal grid. Most of image segmentation algorithms use the topology based on the classical squared grid, because this is an extension from the one-dimensional case. Nevertheless, the squared grid is not optimal according to the Shannon sampling theorem: the optimal one for image sampling is the hexagonal grid. In this paper, we adapt to hexagonal topology a hierarchical image segmentation algorithm developed previously on a squared grid. We present here a new structure, called the hepta-tree, adapted to hexagonal grids. Unsupervised segmentation results are compared on synthetic images issued from the both sampling grids.

DETECTING AND GROUPING WORDS IN TOPOGRAPHIC MAPS BY MEANS OF PERCEPTUAL CONCEPTS

M. Caprioli, P. Gamba Department of Electronics, University of Pavia, Italy

We describe a method for the semi-automated extraction and clustering of characters occurring in scanned topographic maps. The method takes into account the oriented strings that are frequent in topographic maps. We describe the whole application but we emphasize the module devoted to string extraction based on a suitable human vision model related to this particular problem. The experimental results show the capability to exploit this method for semi-automated character extraction to the aim of building or updating data in Geographic Information Systems.

A CONTOUR MATCHING APPROACH FOR ACCURATE NOAA-AVHRR IMAGE NAVIGATION

Francisco Eugenio, Eduardo Suárez, and Eduardo Rovaris, University of Las Palmas of G.C.
Ferran Marqués, Polytechnic University of Catalonia

Although different methods for NOAA AVHRR image navigation have already been established,the multitemporal and multi-satellite character of most studies requires automatic and accurate methods for navigation of satellite images. In the proposed method, a simple Kepplerian orbital model for the NOAA satellites is considered as reference model, and mean orbital elements are given as input to the model from ephemeris data. In order to correct the errors caused by these simplifications, errors resulting from inaccuracies in the positioning of the satellite and failures in the satellite internal clock, an automatic global contour matching approach has been adopted. First, the sensed image is preprocessed to obtain a gradient energy map of the reliable areas (sea-land contours) using a cloud detection algorithm and a morphological gradient operator. An initial estimation of the reliable contour positions is automatically obtained. The final positions of the contours are obtained by means of an iterative local minimization procedure that allows a contour to converge on an area of high image energy (edge). Global transformation parameters are estimated based on the initial and final positions of all reliable contour points. Finally, the performance of this approach is assessed using NOAA 14 AVHRR images from different geographic areas.

BIAS INTRODUCED BY MEAN ORIENTATION ESTIMATION METHODS

J.P. Da Costa, C. Germain, O. Lavialle and P. Baylou, Equipe Signal et Image, ENSERB - GdR Isis, France

Many problems deal with two-dimensional random variables. One of them concerns the estimation of statistical parameters from a gradient vector field on a 2D-lattice. Special interest has been devoted to the extraction of orientation which is an inevitable feature when characterizing the spatial organisation of patterns on textures. Orientations on an image are given by any gradient estimator. The statistical mean is provided by the way of directional statistics. Unfortunately, the way orientation is estimated results in a distribution of correlated vectors. The bad use of the statistical tools may introduce distortions and bias in the statistical distribution of the mean orientation. This phenomenon is pointed out through several simulations involving mean orientation estimates on a Sobel vector field. A theoretical explanation based on variance estimation is also provided. An answer to such a problem is based on the partitioning of the gradient orientation field into independent codings.

SEGMENTATION OF COLOR IMAGES USING A HIERARCHICAL FUZZY REGION MERGING TECHNIQUE

S. Makrogiannis, G. Economou, E. Zigouris and S. Fotopoulos, University of Patras, Greece

A novel hierarchical clustering method is presented in this work. It operates as a part of a split and merge segmentation scheme. The proposed technique incorporates the use of several color features to compare clusters in the RGB space and the flexibility of the fuzzy reasoning approach to accomplish satisfactory segmentation results. The boundary values of the fuzzy sets have been determined by means of a genetic algorithm optimization approach. The segmentation results evaluated subjectively and objectively were compared to a straightforward product cost function.

ROTATION INVARIANT TEXTURE ANALYSIS : A COMPARATIVE STUDY

C. Rosenberger, C. Cariou and K. Chehdi, Ecole Nationale Supérieure de Sciences Appliquées et Technologie, France

The problem of recognizing rotated homogeneous textured images is addressed. The aim is to provide some comparison results between two classical non-parametric techniques - namely Zernike moments and Fourier-Mellin descriptors - and a new parametric approach involving the Wold decomposition of 1-D processes. In order to obtain translation invariance, all these methods start with the computation of the 2-D normalized autocovariance of textures. The techniques and numerical aspects of the computation of invariant features are briefly described. Experiments performed on a texture database show that the parametric model provides encouraging recognition rates comparable with the Zernike moments, along with the important advantage of its parcimony w.r.t. the classical approaches.

VIRTUAL REALITY TOOL FOR VISUALISING RGB COLOUR SPACE USING VRML

Anthony J. Bardos and Stephen J. Sangwine, The University of Reading, UK

This paper presents a novel Virtual Reality tool for visualising image neighbourhoods. It is currently implemented using RGB colour space, but the technique is not limited to Cartesian spaces. The Visualisation Tool is based on Virtual Reality Modeling Language (VRML). Visualisation using virtual reality techniques is a useful tool to help evaluate the effectiveness of colour filters. It can also be used as an aid to developing new ideas for filters. VRML makes Virtual Reality Visualisation Tools relatively trivial to implement and negates the need for expensive software packages.

2-D WOLD DECOMPOSITION: NEW PARAMETER ESTIMATION APPROACH TO EVANESCENT FIELD SPECTRAL SUPPORTS

C. Ramananjarasoa and M. Najim, Equipe Signal et Image ENSERB and GDR ISIS - CNRS, France
O. Alata, Université de Poitiers, France

In the context of parametric model-based methods for image processing, this paper deals with image modeling, when the image is considered as a two-dimensional (2-D) random process. The model we are searching on is based on the 2-D Wold-type decomposition. The core of the contribution of this paper is to provide with a new estimation algorithm of the so called "evanescent field" in the 2-D Wold decomposition framework. This new method is based on a projection approach and requires a set of projection directions which is obtained by using Farey's series. The algorithm performances are illustrated and investigated using Monte Carlo simulations.

PERFORMANCE EVALUATION METRICS FOR OBJECT-BASED VIDEO SEGMENTATION

Çiðdem Eroðlu Erdem and Bülent Sankur Boðaziçi, University of Ýstanbul, Turkey

In this paper, we investigate performance metrics for quantitative evaluation of object-based video segmentation algorithms. The proposed metrics address the case when ground-truth video object planes are available.The metrics are based on weighted pixel misclassification penalty, shape penalty and motion penalty of the objects existing in the scene. The proposed metrics are used to evaluate three essentially different approaches for video segmentation, i.e., an edge-based, a motion clustering based, and a total feature vector clustering based algorithm. The experimental results on real and synthetic image sequences have been shown to agree with the subjective quality assessments of the three leading segmentation algorithms found in the literature.

ORIENTATION ESTIMATION: CONVENTIONAL TECHNIQUES AND A NEW NON-DIFFERENTIAL APPROACH

Rudolf Mester, J.W.Goethe-Universitaet Frankfurt, Germany

The estimation approach discussed in this paper is based on a signal-theoretic and statistical analysis of the notion of orientation. In contrast to other approaches, it does not require the computation of gray value gradients, or the power spectrum of the given signal patch, or quadrature filter outputs, but directly estimates a small central part of the autocovariance function (acf) of the signal and derives the sought orientation from the direction of main curvature in the acf origin. The adaptation to the frequency-dependent signal/noise ratio of the signal is performed here by linearly filtering the acf, but not the signal itself. This is in contrast to the usage of pre-filters which are necessary in conventional approaches in order to compensate for the limited capabilities of the discrete differentiation filters. This shortcut which is possible as a consequence of the Wiener-Lee theorem is the key to considerable savings in computational effort.

ESTIMATION OF VIDEO OBJECT'S RELEVANCE

Paulo Correia and Fernando Pereira, Instituto Superior Técnico - Instituto de Telecomunicações

With the advent of the MPEG-4 and MPEG-7 standards, a further impulse to the representation and description of multimedia information has been given. In particular, object-based coding and description are nowadays possible (or about to become so). In both object-based coding and description environments the estimation of video object's relevance can be very useful. It has a major role in segmentation quality evaluation, responsible for selecting the appropriate segmentation algorithm to use for identification of the objects to work with. Object relevance information is also very valuable for a rate control module of an object-based coder. In description creation applications, relevance can be used directly as an object descriptor, or indirectly to ensure that more relevant objects receive more detailed and complete descriptions. Recognizing the importance of object relevance estimation, this paper proposes an objective metric, automatically calculated, for object relevance evaluation, both when objects are considered individually or within a given context.

CLASSIFICATION OF TWO-DIMENSIONAL SHAPES BASED ON MOMENT INVARIANTS

Predrag Pejnovic, SMT d.o.o,
Milan Markovic, Institute of Applied Mathematics and Electronics
Srdjan Stankovic University of Belgrade, Romania

An experimental analysis of two-dimensional (2D) shape classification method based on moment invariants is presented. Various types of translation, scale and rotation invariants are used to construct feature vectors for classification. The performance is evaluated using five different objects picked up from real scenes with a TV camera. Silhouettes and contours are extracted from nonoccluded 2D objects rotated, scaled and translated in 3D space. The proposed feature extraction method are implemented and systematically tested using several parametric and nonparametric classifiers. The obtained results clearly justify the use of the 2D shape feature extraction method based on the moment invariants.

FACE DETECTION FROM COMPLEX SCENES IN COLOR IMAGES

Doina Petrescu and Marc Gelgon, Nokia Corporation Research Center

Automatic detection of faces in color images is a task of increasing importance for different multimedia applications. In this paper a 3-stage method for detecting human faces from images using color, shape and the gradient of luminance is presented. The aim of the method is to detect faces for a great variety of face sizes, positions and 3D orientations, including frontal and non-frontal views and situations when more than one face is present in the image. A measure of confidence that a region is a face is defined using parameters computed from the color, shape and local gradient of luminance features. This measure allows tuning the face detector to application requirements.

WedPmOR1
Small Audiotorium
Image and Video Coding
Chair: M. Kunt ***Ecole Polytechnique Federale de Lausanne, Switzerland***

14:00

TREE-STRUCTURED METHOD FOR IMPROVED LUT INVERSE HALFTONING

Murat Mese and P.P. Vaidyanathan, California Institute of Technology, USA

Recently we have proposed a Look Up Table (LUT) based method for inverse halftoning of images. The LUT for inverse halftoning is obtained from the histogram gathered from a few sample halftone images and corresponding original images. The method is extremely fast (no filtering is required) and the image quality achieved is comparable to the best methods known for inverse halftoning. The size of the LUT can become bigger even though most of the elements in the table are not used in practice. We propose a tree structure which will reduce the storage requirements of an LUT by avoiding nonexistent patterns. Tree-structure inverse halftoning will need only a fraction of its LUT equivalent for storage. We demonstrate the performance on error diffused images.

14:20

PACKET LOSS RESISILENT H.263+ COMPLIANT VIDEO CODING

Fabrice Le Léannec and Christine Guillemot, Institut National de Recherche en Informatique et en Automatique, France

This paper considers the problem of real-time point to point video transmissionover the public Internet. A set of techniques in the direction of packet loss resilience of video compressed streams is presented. Aiming at a best trade-off between compression efficiency and packet loss resilience, a procedure for adapting the video coding modes and quantizer parameters to varying network characteristics is introduced. The network adaptive coding modes and quantizers selection is based on a rate-distortion procedure with global distortion metrics incorporating channel characteristics under the form of a packet loss model. The method has been implemented in a H.263+ software video encoder. As a second step, the coding mode selection strategy is refined by exploiting feedback information which consists in indicating lost video packets. We show that exploiting this knowledge of the channel state allows to significantly improve the robustness of H.263+ video streams in a packet loss prone environment.

14:40

H.263 PICTURE HEADER RECOVERY IN H.324 VIDEOPHONE

Miska M. Hannuksela, Nokia Mobile Phones

The scope of the H.324 videophone standard includes mobile networks which can in bad radio conditions be susceptible to higher bit error rates than most fixed networks. Different parts of the H.263 video bit-stream have unequal importance in video signal reconstruction. For example, the contents of a video picture cannot be decoded unless a so-called picture header has been correctly received or successfully recovered. In this paper we review methods for recovering corrupted H.263 picture headers. Moreover, we experimented how well two selected recovery methods operate in H.324 terminals connected to a simulated mobile network. The results showed that both tested algorithms play an important role in picture header recovery. In addition, we found that it is beneficial to set a certain parameter in the picture header differently than suggested in the H.263 recommendation.

15:00

NONLINEAR BANK FILTERS ADAPTED FOR PROGRESSIVE AND EXACT IMAGE RECONSTRUCTION

A. Benazza-Benyahia and A. Amara, Ecole Sup'erieure des Communications de Tunis, Tunisia
J.-C. Pesquet, LSS (CNRS/UPS) Supélec, France

In this paper, an extension of the recently proposed nonlinear subband decomposition schemes with perfect reconstruction is investigated. Such generalized scheme has the advantage of exploiting efficiently the redundancies contained in the images. Simulation tests performed on a great number of natural images show that the proposed method is suitable for lossless and progressive transmission.

15:50

VARIABLE TO FIXED ENTROPY CODERS: WHY AND HOW? (And their application to H.263)

Charles Boncelet, University of Delaware, USA

Entropy coders fall into several general categories: Huffman and Huffman-like coders that parse the input into fixed length pieces and encode each with a variable length output, arithmetic coders that take an arbitrarily long string as an input and encode with a single output string, and Tunstall-like coders that parse the input into variable length strings and encode each with a fixed length output. This paper is about a Tunstall-like coder called BAC (for Block Arithmetic Coding). We argue that this class of coders is very appropriate for many situations, especially when the probabiliities vary, when channel errors may occur, or when fast operation is needed.

16:10

IMAGE COMPRESSION USING DERIVATIVE INFORMATION WITH DISTANCE TRANSFORMS

Leena Ikonen, Ville Kyrki, Pekka J. Toivanen and Heikki Kälviäinen, Lappeenranta University of Technology, Finland

In this paper, a new image compression method is presented using the Distance Transform on Curved Space (DTOCS) and derivative information in finding positions for control points. In previous work it has been shown that the control points are not in exactly optimal positions. This paper presents theoretical considerations according to which the new method enhances the decompressed image quality particularly in the areas of rapid changes. The obtained results shown verify the correctness of the theoretical considerations. The reconstructed image quality is clearly better measured by error criteria. Also visually the difference is significant.

16:30

SPATIO-TEMPORAL SCALABILITY USING MODIFIED MPEG-2 PREDICTIVE VIDEO CODING

Adam Luczak, Slawomir Mackowiak and Marek Domanski, Poznan University of Technology, Poland

The paper describes spatio-temporally scalable MPEG video coders proposed. Such an encoder produces two bitstreams: base layer bitstream which represents a video sequence with low spatial and temporal resolution and an enhancement layer bitstream which provides additional data needed for reproduction of pictures with full resolution and full temporal frequency. The bitrate overhead measured relative to the single layer MPEG-2 bitstream varies between 2% and 22% for progressive television test sequences. The base layer bitstream constitutes 34-40% of the overall bitstream. The base layer encoder is fully compatible with the MPEG-2 video coding standard. The enhancement layer encoder is a modified version of that used by MPEG-2 for spatial scalability.

16:50

BIT RATE AND LOCAL QUALITY CONTROL USING RATE/DISTORSION CRITERION FOR ON-BOARD SATELLITE IMAGES COMPRESSION

Delphine Le Guen, Stéphane Pateux and Claude Labit, IRISA /INRIA
Gilles Moury, CNES
Dimitri Lebedeff, Alcatel Space Industries

The method we present is developed as part of an on-board satellite images compression with target ratios greater than 6. Such a compression is required because of the weight and memory constraints of on-board equipment and must be integrated in a bit rate control scheme. In order to guarantee a high local quality on some critical areas, we have developed an adapted encoder which integrates a local quality constraint within a bit rate control. This encoder, based on a

progressive transmission of information by using an EZW algorithm, realizes coding and bit rate regulation in the same time. Actually, our encoder is a quite different from Shapiro's one due to the integration of Regions Of Interest (ROI) by setting up a classification map. Such a technique allows to limit artifacts on ROI without creating too much degradation on the rest of the image.

17:10

SCHEDULING STRATEGIES FOR 2D WAVELET CODING IMPLEMENTATIONS

M. Ravasi, M. Mattavelli and D. J. Mlynek, Swiss Federal Institute of Technology, Switzerland

Wavelet image compression adopted in the JPEG-2000 and MPEG-4 standards offers several advantages over existing methods based on DCT. This paper presents some wavelet codec scheduling strategies obtained by the joint optimization of both the algorithmic part and the architectural features, according to the target system implementation. Results are presented allowing optimization of system performance either for dedicated ASIC design or for embedded software implementations based on software/hardware system resources partitioning. The optimization can target different features such as execution speed, external and internal cache memory performance, power dissipation, number of parallel wavelet filters.

WedPmOR2
Hall A
Adaptive Filtering
Chair: M. Najim ***ENSERB, France***

14:00

FAST ALGORITHMS FOR IDENTIFICATION OF RAPIDLY FADING CHANNELS

Maciej Niedzwiecki and Tomasz Klaput, Technical University of Gdansk, Poland

The problem of identification/tracking of rapidly fading communication channels is considered. When the channel coefficients vary rapidly with time the most frequently used weighted least squares (WLS) and least mean squares (LMS) algorithms are not capable of tracking the changes satisfactorily. To obtain good estimation results one has to use more specialized adaptive filters, such as the basis function (BF) algorithms which are based on explicit models of parameter changes. Unfortunately, estimators of this kind are numerically very demanding. The paper introduces a new class of recursive algorithms which combine low computational requirements, typical of WLS and LMS filters, with very good tracking capabilities, typical of BF filters.

14:20

CROSS CORRELATION P-VECTOR INFLUENCE ON LMS CONVERGENCE

Luis Vicente and Enrique Masgrau, University of Zaragoza, Spain

The principal weakness of Least Mean Squares (LMS) algorithm is that adaptation can be sometimes slow. Convergence is known to depend mainly on eigenvalue spread of the input signal, through the time constants of the various convergence modes. However, most LMS convergence analysis do not consider the influence of cross correlation between input and desired output signals, which plays also a significant role on convergence and is the main topic of this paper. The extreme cases of high and low statistical similarity between input and desired output are analysed in detail. Furthermore, an LMS-based adaptive system that seizes the convergence properties explored is also introduced. This system is shown to achieve better performance (that is, faster convergence while maintaining the steady-state error level) than LMS when input and desired output present low or moderately low statistical similarity.

14:40

THE OPTIMUM ERROR NONLINEARITY IN LMS ADAPTATION WITH AN INDEPENDENT AND IDENTICALLY DISTRIBUTED INPUT

Tareq Y. Al-Naffouri, Stanford University, USA
Azzedine Zerguine and Maamar Bettayeb, King Fahd University of Petroleum and Minerals, Saudi Arabia

The class of LMS algorithms employing a general error nonlinearity is considered. The calculus of variations is employed to obtain the optimum error nonlinearity for an independent and identically distributed input. The nonlinearity represents a unifying view of error nonlinearities in LMS adaptation. In particular, it subsumes two recently developed optimum nonlinearities for arbitrary and Gaussian inputs. Moreover, several more familiar algorithms such as the LMS algorithm, the least-mean fourth (LMF) algorithm and its family, and the mixed norm algorithm employ (non)linearities that are actually approximations of the optimum nonlinearity.

15:00

TRACKING PERFORMANCE OF MOMENTUM LMS ALGORITHM FOR A CHIRPED SINUSOIDAL SIGNAL

LK Ting, CFN Cowan and RF Woods, The Queen's University of Belfast, UK

In this paper we study the tracking performance of the momentum LMS (MLMS) algorithm in adaptive prediction for a time-varying chirped sinusoidal signal. The momentum term of the algorithm not only helps to speed up the convergence rate, but also improves the tracking capability for a nonstationary signal. We compare the simulation results of the MLMS with the conventional LMS to highlight the tracking performance. The simulation results show that the MLMS algorithm with an additional momentum term has a better tracking capability in an 8-tap adaptive predictor under noise-free conditions, especially when the filter tracks a fast time-variant signal. However, the MLMS does not have significant improvement when tracking a noise corrupted chirp signal. The normalised MLMS (NMLMS) algorithm has similar simulation results as the ordinary MLMS algorithm for the tracking performance.

15:50

FAST RLS ADAPTIVE ALGORITHMS OF QUADRATIC VOLTERRA ADF

Jinhui Chao, Tomomnori Kubota and Shinpei Uno, Chuo University, Japan

It is shown by these authors that quadratic Volterra adaptive filters (ADF) have error surfaces which are always extremely steep on only one particular direction but relatively flat on the other directions. This explains the instability in the learning processing of Volterra ADF and implies unavoidable slow convergence of traditional gradient adaptive algorithms. On the other hand, the RLS algorithm for Volterra ADF costs O(N^4) multiplications where N is the number of linear terms in Volterra ADF. This paper shows a new algorithm for Gaussian input signals which converges in the same rate as RLS but costs only O(N^2) multiplications which is the same as

the LMS algorithm. This algorithm is based on a complete analysis of the intrinsic geometry of the error surface for white input signals and whitening operation of any colored input signals. Simulations shown that this algorithm works well even in non-Gaussian input cases.

16:10

NOVEL IMPLEMENTATION TECHNIQUE OF RLS ALGORITHM FOR IMPROVING THROUGHPUT OF ADAPTIVE FILTERS

Kiyoshi Nishikawa and Hitoshi Kiya, Tokyo Metropolitan University, Japan

This paper proposes a method for increasing the throughput of recursive least squares (RLS) adaptive filters when only a few processing elements (PEs) are available. Throughput can be traded against the rate of convergence by varying a parameter. Because the conventional methods for improving the throughput of an RLS filter require the use of more or faster PEs, the proposed technique will expand the area of applications for the RLS algorithm.

16:30

PERFORMANCE ANALYSIS OF AN ADAPTIVE FILTER STRUCTURE EMPLOYING WAVELETS AND SPARSE SUBFILTERS

Mariane R. Petraglia and Julio c. B. Torres, Federal University of Rio de Janeiro, Brazil

The properties of an adaptive filter structure which employs a wavelet to decompose the input signal and sparse adaptive filters in the subbands are investigated in this paper. The necessary conditions on the structure parameters for the exact modeling of an arbitrary linear system with finite impulse response (FIR) are derived. An analysis of the convergence speed of the adaptation algorithm is presented, showing that a significant improvement in the convergence rate can be obtained using very simple wavelets. Computer simulations are presented to illustrate the convergence behavior of the adaptive structure investigated in the paper.

16:50

DOES WHITE INPUT GIVE BETTER CONVERGENCE SPEED IN ADAPTIVE FILTERING?

Y.Ben Jemâa, L.S.S, SUPELEC, France
H.Besbes, SUPCOM, Tunisia
M.Jaidane, L.S.Télécoms, ENIT, Tunisia

In adaptive filtering developed by the signal processing community, systems have been extensively designed and implemented in the area of digital communication. A rigorous performances analysis was difficult to find by classical approaches. The mathematical analysis developed in this paper, turned to be simpler in the finite alphabet set. So, exact performance analysis can be deduced without any constraining or unrealistic hypothesis. Instead of classical approaches where the convergence speed depends only on the second order moment of the input,

we demonstrate that convergence rate depends on all high order statistics of the input. Consequently, in some cases, we prove that adaptive algorithms with decorrelating properties do not speed up the convergence as expected. Finally, we give cases where the convergence speed is more important for correlated input than for white one.

17:10

ADAPTIVE GENERAL PARAMETER EXTENSION TO FIR PREDICTORS

Jarno M. A. Tanskanen and Seppo J. Ovaska, Helsinki University of Technology, Finland
Olli Vainio, Tampere University of Technology, Finland

In this paper, a general parameter (GP) extension to polynomial and sinusoidal FIR predictors is proposed for extended sinusoidal and Rayleigh fading signal prediction. With a single adaptive general parameter, it is shown possible to extend prediction capabilities of polynomial and sinusoidal FIR predictors beyond polynomial input signals, or beyond the nominal design frequencies. This allows for more accurate prediction of input signals with unknown time varying statistics. The form of adaptation proposed can be regarded as the simplest possible adaptive filtering, and is shown to yield improved filtering performance at low computational cost. It is demonstrated that the proposed method provides for improved prediction of sinusoidal and Rayleigh distributed signals. Here, Rayleigh distributed signal models the received transmission power in a mobile communications system, and the sinusoid is a model of the power line frequency.

WedPmOR3
Room 200
Time-Frequency Analysis
Chair: H. Sakai *Kyoto University, Japan*

14:00

ELIMINATION OF INTERFERENCE TERMS OF THE DISCRETE WIGNER DISTRIBUTION USING NONLINEAR FILTERING

Gonzalo R. Arce and Syed R. Hasan, University of Delaware

Methods for interference reduction in the Wigner distribution have traditionally relied on linear filtering. This paper introduces a new nonlinear filtering approach for the removal of cross terms in the discrete Wigner distribution. Realizing that linear smoothing kernels are unable to completely cancel the cross-terms without compromising time-frequency concentration and resolution of the auto-terms, a nonlinear filtering algorithm is devised where the filter automatically adapts to the rapidly changing nature of the Wigner distribution plane. Varying the filter behavior from an identity operation at one extreme to a low pass linear filter at the other, a near optimal removal of cross terms is achieved. Experimental results are reported and a comparison with some popular contemporary techniques is performed to demonstrate its superior performance. Unlike traditional smoothing and optimal kernel design techniques, this algorithm does not reduce the time-frequency resolution and concentration of the auto-terms and performs equally well for a very large variety of signals; furthermore this algorithm can be implemented in real time due to its low computational complexity.

14:20

CHIP SIGNAL ANALYSIS BY USING ADAPTIVE SHORT-TIME FRACTIONAL FOURIER TRANSFORM

Feng Zhang, Guoan Bi and Yan Qiu Chen, NTU, Singapore

This paper proposes an adaptive short-time fractional Fourier transform. The proposed transform improves the resolution of the short-time Fourier transform for chirp signals, and avoids the cross-terms. The proposed adaptive short-time Fourier transform is based on the de-chirp effect of the fractional Fourier transform. It adaptively de-chirps the signal with time to achieve a high signal concentration in the time-frequency domain. An effective measure for local signal concentration is used to optimize the selection of the parameter of the fractional Fourier transform.

14:40

BIORTHOGONAL FILTER BANKS WITH NEARLY OPTIMAL TIME-FREQUENCY LOCALIZATION

Cyrille Siclet and Pierre Siohan, France Telecom, France
Tanja Karp, University of Mannheim, Germany

In this paper we study the time-frequency localization properties of biorthogonal modulated filter banks. The localization of the prototype filter is computed using two different measures. Design examples are presented showing that the lower bound of the uncertainty relation can be nearly reached. We also show that biorthogonal filter banks offer attractive trade-offs between reconstruction delay and time-frequency localization, not allowed when using orthogonal ones.

15:00

USING ARCAP TIME-FREQUENCY REPRESENTATIONS FOR DECISION

H. Cottereau, M. Davy and C. Doncarli, CNRS, France
N. Martin, LIS, France

We address the problem of estimating and classifying multicomponent narrow band signals using ARCAP Time-Frequency Representations (TFRs). ARCAP TFRs are build using firstly, an instantaneous frequency estimator and, secondly, an instantaneous power estimator, and results in a set of points without any time links. The spectral trajectories have then to be extracted, and we apply a Kalman filter. The trajectories parameters (chirps slopes, beginning and ending instants and frequencies...) enable the implementation of a supervised classification procedure. Results are given for mono and multicomponent chirp signals.

WedPmSS1
Hall B
Signal Processing for Multimedia Surveillance Systems
Chair: I. Pitas ***University of Thessaloniki, Greece***

14:00

ADAPTIVE CHANGE DETECTION APPROACH FOR OBJECT DETECTION IN OUTDOOR SCENES UNDER VARIABLE SPEED ILLUMINATION CHANGES

Lucio Marcenaro, Gianluca Gera and Carlo Regazzoni, University of Genoa, Italy

In this paper an adaptive algorithm is presented for the detection of changes in outdoor video-surveillance images. The proposed method can be considered as basis for a low level image processing stage in a advanced video-surveillance system. The main feature of the algorithm is the robustness to the illumination changes in the scene: this robustness is achieved by using a background updating module working in cooperation with the change detection algorithm. The background updating method works in a different way according to speed of illumination changes. If a sudden illumination change is detected the background is heavily updated while if a slow lightning variation occurs, a continuous soft updating module is used to take into account the long-term slow illumination changes. Experimental results of an automatic people counting system including the proposed low level image processing stage indicate that the method detects changes accurately in case of time-varying illumination.

14:40

SCALABLE MULTIMEDIA SYSTEM FOR INTERACTIVE SURVEILLANCE AND VIDEO COMMUNICATION APPLICATIONS

Marco Raggio, Ivano Barbieri and Gianluca Bailo, University of Genoa, Italy

Recent advances in embedded computer electronics and integration in multimedia standard specifications are enabling cost-effective terminals for video communication and conferencing with potential capabilities in video surveillance application, both on wireless and wired links. Following a tutorial approach, the paper describes the practical implementation of a multimedia system and terminals implemented at our labs. The system (Real Time Scalable Video Communication Unit – VCU) is based on latest extension to Standard Recommendation ITU-T H.324 and H.323, it performs real time audio, video and data communication with the today available network technology and consumer electronics components. The system was provided with enhanced video and remote control setting capabilities, leading to multimedia terminals suitable for both interactive video conferencing applications and remote video surveillance, in particular, when variable and/or low bitrate links are available, exploiting video scalability.

15:00

AUTOMATIC SOUND DETECTION AND RECOGNITION FOR NOISY ENVIRONMENT

Alain Dufaux, Michael Ansorge, and Fausto Pellandini, University of Neuchâtel, Switzerland
Laurent Besacier, University Joseph Fourier, Grenoble

This paper addresses the problem of automatic detection and recognition of impulsive sounds, such as glass breaks, human screams, gunshots, explosions or door slams. A complete detection and recognition system is described and evaluated on a sound database containing more than 800 signals distributed among six different classes. Emphasis is set on robust techniques, allowing the use of this system in a noisy environment. The detection algorithm, based on a median filter, features a highly ro-bust performance even under important background noise conditions. In the recognition stage, two statistical classifiers are compared, using Gaussian Mixture Models (GMM) and Hidden Markov Models (HMM), respectively. It can be shown that a rather good recognition rate (98% at 70dB and above 80% for 0dB signal-to-noise ratios) can be reached, even under severe gaussian white noise degradations.

15:50

DIGITAL WATERMARKING FOR THE AUTHENTICATION OF AVS DATA

M. Barni, Università di Siena, Italy
F. Bartolini and A. Piva, Università di Firenze, Italy
J. Fridrich and M. Goljan, SUNY Binghamton

In Advanced Video-based Surveillance (AVS) applications, if AVS video sequences have to be used as proofs on a legal basis it is important to verify the identity of the content originator, and whether the content has been modified or falsified since its distribution. These requirements have been until now satisfied by using digital signatures and cryptographic algorithms. However, digital watermarking techniques can be used as an alternative for the content verification. It is the purpose of this work to discuss the possibility of using watermarking to ensure the integrity of the image data produced by AVS systems. The main requirements watermarking algorithms must satisfy for the correct implementation of an AVS authentication scheme are outlined and some possible solutions based on the state of the art in the field are described.

16:10

ROBUST AND ILLUMINATION INVARIANT CHANGE DETECTION BASED ON LINEAR DEPENDENCE FOR SURVEILLANCE APPLICATION

Emrullah Durucan and Touradj Ebrahimi, Swiss Federal Institute of Technology, Switzerland

The subject of this paper is to provide an illumination invariant moving object detector for indoor surveillance applications. Furthermore we want to treat the illumination change as a mathematical/physical transformation procedure on images. Therefore the intention to this paper could also be defined as to provide an operator, which is invariant to transformations. The

proposed detector relies on a model assigning a vector to every pixel location of the reference and the current image. The vector represents information on the neighborhood region of that pixel. Based on the above definition, the theorem of linear dependence of vectors is used to describe an operator for the detection of objects. For the purpose of an objective evaluation, the proposed technique is compared to the state-of-the-art Statistical Change Detection method. The proposed operator proved to be robust to noise as well as global illumination changes and local shadows and reflections.

16:30

EVALUATION OF VIDEO SEGMENTATION METHODS FOR SURVEILLANCE APPLICATIONS

Kevin McKoen and Raquel Navarro-Prieto, Motorola Labs
Benoit Duc, Motorola SPS
Emrullah Durucan, Francesco Ziliani and Touradj Ebrahimi, EPFL

Current research on performance evaluation of video segmentation methods is primarily focussed on the development of objective figures of merit. There is no standardised methodology for subjective evaluations of segmentation performance and these are currently perceived as too onerous. Using an experimental design and data analysis method derived from current practice in experimental psychology, we have explored a performance evaluation procedure that is largely based on subjective assessments. We report on statistically significant differences in perceived performance between three multi-object video segmentation and tracking methods developed for surveillance applications. The assessments were performed by a group each of experts and novices on a wide range of video content.

16:50

REAL-TIME ROBUST DETECTION OF MOVING OBJECTS IN CLUTTERED SCENES

Franco Oberti and Carlo S. Regazzoni, University of Genoa, Italy

Object recognition is a very important task in computer vision and different techniques have been presented to solve it. In this paper a Hough-type low-computational algorithm for detection of objects in cluttered scenes is presented. The approach is based on the detection of the shape of an object, modeled by means of a set of corners. An automatically model learning method is introduced. The method is used in an existing video-surveillance system in order to increase its detection performances. Results show that the proposed approach provides good performances with low processing times.

17:10

A 3D MODEL BASED VISUAL SURVEILLANCE SYSTEM

Isabel Martins and Luís Corte-Real, ISEP/INESC Porto and FEUP/INESC Porto

This paper presents a visual surveillance system tailored for remote surveillance of indoor environments. It uses a two layers video codec based on a 3D model of the background \cite{martins}. The objective is to achieve high data compression rates and to enable the support for applications that require the 3D knowledge of the scene. Examples of such applications are statistical applications (e.g. counting persons) and security applications (e.g. control of entrances in restricted areas). The system supports the ability to localize moving objects in the 3D space assuming that all objects are placed on the ground plane. This feature allows the localization of people on a top view map of the site. Results of some tests are presented in order to show the system ability to deal efficiently with the motion of the camera and to illustrate the two layers separation based on a background/foreground segmentation.

WedPmSS2
Studio
Recent Developments in Transforms and Filterbanks
Chair: K. Egiazarian *Tampere University of Technology, Finland*

14:00

UPPER BOUNDS OF WAVELETS ON 2-D LIPSCHITZ CLASS AND ZEROTREES

Arthur Petrosian, Texas Tech University, USA

The wavelet transform method has become one of the most powerful tools in image compression applications. It has a number of advantages with respect to other spectral transforms. With lossy compression, after a suitable wavelet transform is chosen two basic procedures - zonal and threshold coding - are usually applied to the spectral image. In zonal coding only a fixed small "zone" of transformed image is encoded and the optimal zonal coding method ensures the best selection of spectral zones at which a minimum mean-square error of reconstruction is achieved. In order to determine optimal zonal coding method for the chosen transform one has to obtain the estimates of its spectra on a given class of input images. We suggest in this paper a unified approach to obtain exact upper bounds of a given wavelet transform on the class of discrete images with bounded first order finite differences. The presented estimates allow to apriori identify the spectral "zones" that are likely to be of significance in image reconstruction for a given wavelet transform.

14:20

DECOMPOSITION OF 2D HYPERCOMPLEX FOURIER TRANSFORMS INTO PAIRS OF COMPLEX FOURIER TRANSFORMS

Todd A. Ell, Minnesota, USA
Stephen J. Sangwine, The University of Reading, UK

Hypercomplex 2D Fourier transforms have been proposed by several authors with applications in image processing of both greyscale and colour images. Previously published works on hypercomplex Fourier transforms have utilized direct evaluation of a Fast Fourier transform using hypercomplex arithmetic. This paper shows that such transforms may be implemented by decomposition into two independent complex Fourier transforms and may thus be implemented by building upon existing complex code. This is a significant step because it makes available to researchers using hypercomplex Fourier transforms all the investment made by others in efficient complex FFT implementations, and requires substantially less effort than coding hypercomplex versions of existing code.

14:40

DFT, DCT, MDCT, DST AND SIGNAL FOURIER SPECTRUM ANALYSIS

L. Yaroslavsky, Tel Aviv University, Israel
Ye Wang, Nokia Research Center, Finland

DFT, DCT, DST and MDCT are compared in terms of their resolution power for signal Fourier spectrum analysis and energy compaction properties. For test sinusoidal signals with random frequency it was shown by computer simulation that the resolution power of the transforms is not uniform within the frequency band and that on average over the frequency range, DFT, DCT and DST have almost the same resolution power. while MDCT slightly remises them in this respect.

15:00

SIGNAL TRANSFORMS FOR THE DETECTION AND IDENTIFICATION OF SIGNALS

Gianpaolo Evangelista, École Polytechnique Fédérale de Lausanne, Switzerland
Sergio Cavaliere, University of Naples, Italy

In this paper the authors address the problem of detection and identification of signals buried in high level noise. Frequency domain techniques with long integration intervals are particularly well suited to perform this task if the signal is composed of a mixture of stationary sinusoidal terms. One can achieve reliable detection even in the very low SNR case. However when the signal itself exhibits time-varying features, even when these are known in advance, detection and identification is reliable only over time intervals where the signal is approximately stationary. The limited integration time puts a lower bound to the allowable SNR. In this paper we propose the use of an adaptive signal transformation previously introduced by the authors, which reverts the time-varying signal to a simpler stationary one. Constant features over time allow longer integration times -- up to the duration of the total event -- thus granting proper detection even in the extremely low SNR case. This is for example the case of a simple monochromatic or narrow-band signal, whose frequency varies over time, with a known frequency law over time, such as a chirp signal. In the paper we review the relevant features of the proposed transformation and detail our method providing significant bounds for its numerical performance and examples.

15:50

NEW ORTHOGONAL EXTENSION METHODS FOR TREE-STRUCTURED FILTER BANKS

Maria Elena Dominguez Jimenez, Universidad Politecnica de Madrid, Spain
Nuria Gonzalez Prelcic, Universidade de Vigo, Spain

When processing finite length sequences via paraunitary filter banks, a commonly used technique consists of artificially extending the signal before the analysis stage. The use of the extension methods avoids the border effects, and perfect reconstruction, even orthogonal, size-limited filter banks can be defined. In this paper we first characterize and generate all signal

extension methods which yield orthogonal subband transforms. Secondly, the particular case of non-circular orthogonal extension methods is investigated, and the first general design method of non-circular orthogonal extensions is derived. Finally, the computational cost of the design algorithm is evaluated and compared to that of orthogonal boundary filter generation methods.

16:10

ON FAST HADAMARD TRANSFORMS OF WILLIAMSON TYPE

Hakob Sarukhanyan, Institute of Informatics and Automation Problems of NAS of Armenia
Sos Agaian, University of Texas at San Antonio, USA
Karen Egiazarian and Jaakko Astola, Tampere University of Technology, Finland

The Hadamard transform of Sylvester's type, which is also known as the Walsh-Hadamard transform, is widely used in signal processing and communication. Note that the Walsh-Hadamard transform operates only with vectors whose length N is a power of 2. If N is not a power of two, then in order to compute the Walsh-Hadamard spectrum of the vector one has to either discard components or pad zeros up to the next power of two. In the first case we have an information loss and in the second case extra computations are needed. Thus, construction of fast Hadamard transforms of different orders is important problem. In this paper we develop fast Hadamard transforms based on special classes of Hadamard matrices, namely, the Williamson type Hadamard matrices.

16:30

DESIGN OF FILTER BANKS FOR DATA FUSION BY SPECTRUM SUBSTITUTION

Fabrizio Argenti and Luciano Alparone, University of Florence, Italy

In this work, we investigate on the design of filter banks that allow to substitute part of the spectrum of one signal with that of another signal. One application of this technique is fusion of data collected by sensors having different resolutions, that is a a typical problem encountered in remote sensing, when data from low-resolution multi-spectral sensors and high-resolution panchromatic sensors are to be merged, either to alleviate visual identification tasks, or to expedite automatic detection and recognition. The approach that is proposed here is based on the use of cosine-modulated uniform filter banks. We assume that the ratio of the sampling periods of the input data is not integer and show how to design the filter banks so that spectra from different signals can be integrated with minimum distortion.

16:50

TRANSFORM-BASED DENOISING AND ENHANCEMENT IN MEDICAL X-RAY IMAGING

Til Aach, Medical University of Luebeck, Germany

Physical and clinical constraints of the imaging process often degrade the quality of medical images. A typical case is low-dose X-ray image acquisition, where the signal-to-noise ratio is limited due to X-ray quantum noise (photon-limited imaging). This paper discusses and quantitatively evaluates spectral transform-based methods for denoising and enhancement of such images, where we will focus on estimation of spectral amplitude. Estimation of short time amplitude spectra is well-established in speech restoration, since it is well adapted to the short time stationarity of speech. Motivations for transferring this approach to restoration of images is their short-space stationarity, and the fact that the observed X-ray quantum noise observed in the acquired images is coloured, i.e. spatially correlated, what is easily taken into account in the spectral domain. We examine algorithms based on the DFT, the Modulated Lapped Transform (MLT) and a new lapped transform termed the Lapped Directional Transform (LDT).

17:10

A TRANSFORMATION DOMAIN DESCRIPTION FOR NONLINEAR PARTIAL DIFFERENTIAL EQUATIONS

Rudolf Rabenstein and Lutz Trautmann, University of Erlangen-Nuremberg, Germany

Transfer function models for the description of physical systems with distributed parameters have recently been introduced to the field of multidimensional digital signal processing. They are an extension of the transfer functions description for one-dimensional systems and provide an alternative to the conventional representation by partial differential equations. Transfer function models are the starting point for the development of efficient discrete algorithms that are suitable for computer implementation. This paper extends the presented transformation domain description to nonlinear systems. A suitable choice of the integral transformations for the independent variables allows to treat certain types of nonlinearities in a similar way as for linear multidimensional systems.

WedPmPO1
Exhibition Hall
DSP Applications
Chair: S. Ovaska *Helsinki University of Technology, Finland*

MODELING RADAR DATA WITH TIME SERIES MODELS

S. de Waele and P.M.T. Broersen, Delft University of Technology, The Netherlands

In Frequency Modulated Continuous Wave (FMCW) radar, the Fast Fourier Transform (FFT) is very efficient in separating reflections from various range cells. For the subsequent determination of the Doppler spectrum, the FFT is less suitable. The raw FFT is very erratic, while the smoothed FFT is limited in the spectral shapes it can accurately describe For determination of the Doppler spectrum, ARMAsel time series analysis is better than the FFT. With ARMAsel time series analysis, an ARMA model is estimated from the data. The Doppler spectrum can be calculated from the ARMA-parameters. The model order and type are determined automatically from the data. Improved accuracy in comparison to other time series techniques is obtained by using robust estimation algorithms and order selection criteria, and by selecting automatically between several model types.

THE APPLICATION OF THE ROBUST DISCRETE WAVELET TRANSFORM TO UNDERWATER SOUND

Shun-Hsyung Chang, Fu-Tai Wang, National Taiwan Ocean University, Taiwan, R.O.C

This paper proposes the use of robust discrete wavelet transform (RDWT) as a solution to a signal processing problem of to detect transient features in the sound recordings that contain interference as well as multipaths. The discrete wavelet transform (DWT) was used to detect signal in the underwater. In underwater and a multipaths environment where numbers signal components arrive with arbitrary delays, we use the RDWT in detection. We adopt the RDWT instead of using DWT for two reasons. The first one is that the spectrum of the shifted signal is the same as the unshifted version when using the RDWT. The second one is that while the RDWT provides a solution to the translation invariance problem, it maintains the DWT's attractive attributes, such as efficiency and suited for transients. Since each subband should preserves its energy when using the RDWT in a multipaths environment, summary features of its robust wavelet decomposition are the same as the unshifted version. By an adaptive model of the background interference, using recursive density estimation of the joint distribution of certain summary features of its robust wavelet decomposition, we may detect the transient in a multipaths and a background underwater sound environment.

MINIMAL GEODESICS BUNDLES BY ACTIVE CONTOURS : RADAR APPLICATION FOR COMPUTATION OF MOST THREATHENING TRAJECTORIES AREAS & CORRIDORS

Frédéric Barbaresco, Bernard Monnier, THOMSON-CSF AIRSYS , France

We propose to use shortest path computation method based on Front propagation by Level Set for a radar application. This new radar function consists in computing most threathening trajectories & corridors in the radar coverage in order to adapt radar modes for detection optimization. This Radar problem may be declined as a variationnal problem solved by calculus of variations and geodesic active contours. A partial differential equation PDE drives the temporal evolution of contours of iso-energy (level lines of the manifold defined by the minimal potential surface given by the integration of local detection probability along every potential trajectories). We proove that this PDE approach could be also extended to others radar applications (clutter segmentation, spectral estimation, ...).

A SIMULATION MODEL FOR IMPROVING A NON-LINEAR SENSOR RESPONSE USING LINEAR SOURCE SEPARATION

Y. Naudet, M. Haritopoulos and A. Billat, LAM URCA, France

This paper presents an empirical simulation model for the frequency response of an eddy-current sensor to a distance measurement disturbed by target temperature variations. In previous work, we have shown that source separation could be a method to obtain a good distance measurement independently from temperature variations. The linearisation of the non-linear frequency response of the sensor using a model based on a calibration curve seems not to be sufficient to always obtain good and precise enough results. The aim of the simulation model is then to represent the mean behavior of the sensor, in order to test the applicability of linear separation techniques for this problem. Hence, exploitation of experimental results permits to obtain mathematically a linear mixing of distance and temperature. We show that despite a mixing matrix close to singularity, good results can be obtained using linear BSS with simulated signals, and comparison is given between separation of real signals and simulated ones.

SEPARATION AND FUSION OF OVERLAPPING UNDERWATER SOUND STREAMS

Ivars P. Kirsteins, Naval Undersea Warfare Center

Passive classification of underwater acoustic signatures is an important sonar problem. Sources of signals include commercial shipping, naval vessels, active sonars, geological exploration vessels, and biologics. However, reliable automatic classification has remained an elusive goal. One of the fundamental difficulties has been interference from non-target sound sources and mutual interference from the target itself, which arises from simultaneously radiated narrowband and broadband components that overlap in both spectrum and time. There is considerable evidence suggesting that harmonicity and common fate cues such as correlated amplitude and frequency micromodulations (AM and FM) and onset times of the constituent sound components play an important role in the human auditory system for fusing and segregating the components by source and for recognition of sounds. In this paper we show that the instantaneous frequency

(IF) of multicomponent whale acoustic signals can be estimated using a new non-parametric AM-FM signal representation proposed by Kumaresan, from which the presence of synchronous IF micromodulations across multiple components can be observed. This IF information can then be exploited for grouping components by physical source, thus eliminating the interference problem.

EVOLUTIONARY SPECTRAL BASED CLASSIFICATION OF MACHINE VIBRATIONS

Nihat Kabaoglu and Aydin Akan, University of Istanbul, Turkey

In this paper, we present a method for time-frequency analysis and classification of electrical machine vibrations based on evolutionary spectrum. Time-frequency analysis provides a means for identifying changes in the vibrations produced by machines. In this work, vibrations generated by different electrical machines are recorded by using accelerometers during operation. Three categories of these vibrations are then defined based on their evolutionary spectra: 1) the normal operation, 2) small level of failure, and 3) high level of failure. A method is presented for automatic detection and classification of abnormality in these machine vibrations using their joint time-frequency moments and neural networks. Simulation results are given to illustrate our algorithm.

THE USE OF STOCHASTIC MATCHED FILTER IN ACTIVE SONAR

J.-L. Mori, P.Gounon, Domaine universitaire, France

In this present study we propose to evaluate a kind of matched filter, called "stochastic matched filter", first described by J-F Cavassilas and originated to detect short stationary stochastic "pulses" in stationary noise. We can show in this paper that this filter, with minor changes in its definition, can be applied on frequency time-varying signals, such as wide band modulated sonar signal propagated in shallow water. The stochastic matched filter is able to take into account uncertainties and variations of the propagation channel, thus enhancing the detection efficiency. The main characteristics of this treatment is shown up with simulated data.

ML ESTIMATION OF SSR SIGNALS, IDENTIFIABILITY, AND CRAMER-RAO BOUNDS

Nicolas Petrochilos and Pierre Comon, Delft University of Technology, The Netherlands

In this article, a Maximum Likelihood method is applied to Secondary Surveillance Radar (SSR) signals impinging on an M-element antenna. At base-band, a SSR signal consists of a binary signal with alphabet {0,1} multiplied by a complex exponential. Identifiability is proved in this binary case, and specific Cramer-Rao Bounds are derived and assessed.

BLIND ANGLE AMBIGUITY RESOLUTION IN PARAMETER ESTIMATION OF MOVING TARGETS USING A SINGLE SAR SENSOR

Paulo A C Marques, ISEL, Portugal
Jose M B Dias, IST, Portugal

Detection, imaging, and trajectory parameter estimation of moving targets in Synthetic Aperture Radar (SAR) data is an active area of research. Several methods have been proposed to detect, image and estimate moving targets trajectory parameters in recent literature. These works have not taken into account the antenna radiation pattern, which is equivalent to assuming an omnidirectional antenna. In this case, it has been shown that it is not possible to determine the complete target velocity, but only its magnitude. This limitation is termed the blind angle ambiguity. The main contribution of this paper consists in showing that it is possible to infer the complete velocity vector of a moving target using data from a single sensor, if the antenna radiation pattern is known. We show that the returned echo from a moving target, in the slow-time frequency domain, is a scaled and shifted replica of the antenna radiation pattern, immersed in Gaussian noise of known correlation. The scale is proportional to the target cross-range velocity, while the shift depends on target range speed. Simulation results illustrating the validity of the proposed method are presented.

ON EXTENDED SOURCE LOCALIZATION IN MULTIBASELINE AND MULTIFREQUENCY SAR INTERFEROMETRY

Fabrizio Lombardini, University of Pisa, Italy

Baseline or frequency diversity have been recently proposed to reduce problems of interferometric phase ambiguity and data noise in synthetic aperture radar interferometry (InSAR) for topographic mapping. This paper presents a general model-based multibaseline multifrequency (MBMF) InSAR framework. A statistical model for the MBMF SAR-processed echoes from extended natural targets is presented, then the Cramer-Rao lower bound on the MBMF-InSAR phase accuracy is derived. The maximum likelihood estimator for MBMF-Insar is introduced, and its simulated performance is compared to the bound for the common cases of dual baseline InSAR, dual frequency InSAR, and for a novel joint dual baseline dual frequency InSAR configuration. Performance of these three configurations, and improvement over conventional interferometry is analyzed.

WedPmPO2
Exhibition Hall
Image and Video Coding
Chair: E. Delp *Purdue University, USA*

EFFICIENT ENCODING AND RECONSTRUCTION OF REGIONS OF INTEREST IN JPEG2000

Charilaos Christopoulos, Joel Askelöf and Mathias Larsson, Ericsson Research, Ericsson Radio Systems, Sweden

The upcoming still image compression standard, called JPEG2000, has defined modes for encoding and decoding parts of the image that are considered of particular importance (Regions of Interest - ROI's), with better quality than the rest of the image. The ROI coding modes are based in the "scaling based method" and the generation of the ROI mask. In this paper we describe a fast method for generating the ROI mask for rectangular-shaped ROI's. A revised method of the general scaling based method that does not require any shape information to be transmitted at the decoder is then described. This makes possible to have arbitrary shaped region coding while the decoder is as simple as a non-ROI capable decoder is.

AN ONLINE IMAGE COMPRESSION ALGORITHM USING SINGULAR VALUE DECOMPOSITION AND ADAPTIVE VECTOR QUANTIZATION

Adriana Dapena, Universidad de La Coruna, Spain
Xun Du, Stanley Ahalt, Ohio State University, USA

The Singular Value Decomposition (SVD) is a powerful transform for image compression because it provides optimal energy compaction but, unfortunately, its usefulness for real applications is severely limited because both the computational and the transmission cost are quite high. In order to mitigate the limitations of the SVD, in this paper we propose a new algorithm, called Adaptive Singular Value Decomposition (ASVD), which computes and transmits a reduced set of eigenvectors. In ASVD - as in other SVD-based algorithms- the eigenvectors are coded using Vector Quantization (VQ). However, ASVD differs from previous work in that the eigenvectors's codebooks are generated for each image by potentially sending eigenvectors obtained from the current image. As a consequence, a suitable set of initial codebooks need not be generated a priori and the codebooks are dynamically adapted to changes in the eigenvectors.

ADAPTIVE PREDICTOR COMBINATION FOR LOSSLESS IMAGE CODING

Guang Deng, La Trobe University Bundoora, Australia

This paper is concerned with adaptive prediction for lossless image coding. A new predictor which is an adaptive combination of a set of fixed predictors with a transform domain LMS based predictor is proposed. When a context-based arithmetic encoder is used to encode the

prediction error, the compression performance of the proposed algorithm is better than that of the state-of-the-art algorithms and is close to TMW at a fraction of its computational complexity.

COMPRESSION OF IMAGE OVERHEAD BY HIERARCHICAL ENUMERATIVE CODING

Levent Öktem and Jaakko Astola, Tampere University of Technology, Finland

Hierarchical Enumerative Coding (HENUC) is proposed for the entropy coding of overhead information in image and video compression. Experimental results of two case studies, compression of quadtree representation and compression of switching information, suggest that HENUC has competitive performance. Its advantages over Huffman coding are that it does not require a priori statistics, and it has better compression performance. Its advantages over arithmetic coding are that it adapts faster, thus yielding better compression performance, and it has less computational complexity.

SELECTION OF OPTIMAL WAVELET PACKET TRANSFORMS FOR OBJECT-BASED CODING

Anastasios D. Doulamis, Nikolaos D. Doulamis, Ioannis Stephanakis and Stefanos D. Kollias, National Technical University of Athens, Greece

An algorithm for optimal selection of wavelet packet transforms for object-based coding is presented in this paper. The method exploits the object statistics at each GOP or scene period and estimate an optimal wavelet packet tree and an optimal quantizer for each object so that a) objects of importance (e.g., foreground) are transmitted with higher bit rate (quality) than the background objects, while b) the total transmitted rate does not violate the constraints defined by the network requirements. Furthermore, since the proposed approach estimates object statistics over a GOP or a scene period (an optimal wavelet packet is assigned to each object), the algorithm does not result in a simple bit re-allocation from the background to foreground object but also increases object quality for the same bit rate.

FORTE-VLC: A FORWARD TRACING SELF-ERROR CORRECTION VARIABLE LENGTH CODE FOR IMAGE CODING IN WIRELESS APPLICATION

Yew-San Lee, Keng-Khai Ong, Wei-Shin Chang and Chen-Yi Lee, National Chiao Tung University, Taiwan, R.O.C

Variable length code (VLC), is the most popular data compression technique used for solving transmission channel bandwidth bottleneck in image compression. But, it is vulnerable to loss of synchronization if they are transmitted consecutively through a noisy wireless channel. It will result in large drops in transmission quality. We propose a novel VLC coding scheme, which has high error resilient and correction capability. It uses prefix-suffix coding structure and redundant bits for error tracing and correction. We called it as Forward Tracing Self-Error Correction VLC (FORTE-VLC). It provides high tolerances to random and burst errors in worsening channel conditions. In addition, it exhibits high synchronization capability. It can achieve 82% of error

correction and 91% of synchronization capability with MPEG B.15 VLC table. It also achieves higher picture quality (PSNR=29.5dB) compared to existing VLC schemes at bit error rate (BER) of 10-3 environment. Finally, we present related codec architecture. It is very suitable and efficient for VLSI implementation because of low memory requirement and high throughput rate.

VIDEO SCALABILITY RATE CONTROL METHOD BY DYNAMIC CONTROLLING OF BITS ALLOCATION RATIO

Hiroyuki Kasai, Mei Kodama and Hideyoshi Tominaga, Waseda University

This paper proposes the new coding rate control method in MPEG-2 video scalability. First of all, we investigate the efficiency of SNR scalability from the relations between hierarchical allocated bits and picture quality. Next, base on the the efficiency, we propose the new coding rate control method which has the dynamic controlling of bits allocation ratio. From the simulation experiments, we show the effectiveness in comparison to MPEG-2 TM5 rate control method.

A NEW VERTEX CODING SCHEME USING THE CENTER OF THE GRAVITY CENTER OF A TRIANGLE

Byoung-Ju Yun, Jong-Won Yi and Seong-Dae Kim, Korea Advanced Institute of Science and Technology, Korea

This paper presents an efficient contour-based method for the lossy shape coding. An object is approximated by a polygon because of its inherent quality control and the high-level shape features it intrinsically carries. For encoding the vertices of a polygon, we use the fixed length code because it has more merits than variable length code. The most important factor to determine the amount of bits for a polygon with FLC is the dynamic range of the relative addresses between consecutive vertices of a polygon. To reduce the amount of bits for a polygon, we focus on reducing the dynamic range of vertices using the center of the gravity of a triangle. Simulation results show outstanding performances of the proposed method in the rate-distortion sense.

3D OBJECT DATABASE : MULTI-VIEWPOINT IMAGE CODING \& VIEW SYNTHESIS

JongWon Yi, KwangYeon Rhee, ByoungJu Yun, and SeongDae Kim, Korea Advanced Institute of Science and Technology, Korea

Image-based rendering is a powerful and promising approach for 3D object representation. This approach considers 3D object or scene as a collection of images called key frames taken from the reference viewpoints and generates arbitrary views of the object using these key frames. In this paper, we present an object-based encoding scheme for the multi-view sequences(key frames) of 3D object and a view synthesis algorithm for predicting new views.

SIMPLIFICATION OF IMAGE PARTITIONS FOR REGION-BASED IMAGE CODING

Armando J. Pinho, University of Aveiro, Portugal

In this paper we address the problem of lossy image partition encoding, and compare the compression efficiency of two contour coding techniques associated with appropriate contour simplification methods: the majority filter for the case of chain coding; the reduction of the number of transition points for the encoding technique based on transition points. The experimental results show that chain coding only perform better when unconstrained simplification is allowed, i.e., when the reconstruction error is unlimited. In all other cases, for which the error is bounded, the encoding technique based on transition points shows superior compression efficiency.

THREE-DIMENSIONAL TRANSFORM CODING OF PARTITIONED VIDEO SEQUENCES WITH MATCHING PURSUITS

B. Marusic, P. Skocir and J. Tasic, University of Ljubljana

This work introduces an alternative video coding approach that exploits the temporal correlation present in video signals by applying a combined wavelet/DCT three-dimensional transform to the input video sequence. As the correlation along the temporal axis is likely lowered when motion is present we first partitioning the input sequence into partitions or groups of similar pictures and then applying a transform to highly correlated groups of pictures. Although we apply the transform to correlated pictures, a fraction of the signal energy that is due to motion is scattered in a small number of non-zero "high-frequency" coefficients and is not captured by the compacted "low-frequency" coefficients. The remaining coefficients that are due to motion are then coded using the matching pursuit technique. The majority of the non-zero coefficients are coded by a well-known image coding technique - the LZC coder.

A LIST DIRECTED APPROACH TO FRACTAL IMAGE CODING IN THE WAVELET TRANSFORM DOMAIN

I. Messing and D. Malah, Technion, Israel

This paper describes a technique for Fractal Image Coding in the Discrete Wavelet Transform (DWT) domain employing variable size range and domain subtrees. The DWT domain is partitioned by means of a top-down quad-tree algorithm. The splitting decision function is implemented by a list ordered with respect to a rate-distortion based partitioning gain. The partitioning and encoding are done at the same pass. Each range subtree has three optional coding modes: zero-tree, fractal prediction by a best fit domain subtree from a pool of the same size and 'Intra' scalar quantization of the sub-tree coefficients. The choice between fractal coding and Intra coding is rate-distortion based.

LOSSLESS COMPRESSION OF NATURAL IMAGES USING REGRESSIVE ANALYSIS

Konstantin Balashov and Jaakko Astola, Tampere University of Technology, Finland
David Akopian, Nokia Mobile Phones

This paper develops new lossless coding algorithms belonging to the class of context modeling methods, with direct application to lossless coding of gray level images. The prediction stage and the context modeling stage are performed using one or more linear predictors. Predictors are designed based on the regressive analysis.

REPRESENTATION OF REGIONS FOR ACCURATE MOTION COMPENSATION

R. Rambaruth, W. Christmas and J. Kittler, University of Surrey, United Kingdom

We propose a novel motion compensation technique for the precise reconstruction of regions over several frames within a region-based coding scheme. This is achieved by using a more accurate internal representation of arbitrarily shaped regions than the standard grid structure, thus avoiding repeated approximations for a region at each frame.

IMPLEMENTATION OF DATA COMPRESSION S/W ON A SPACE QUALIFIED DSP BOARD

Wahida Gasti, Thomas Lefort, European Space Agency
Mireille Louys, Université Louis Pasteur de Strasbourg & Observatoire de Strasbourg

Progress in digital imaging sensors such as high resolution CCDs allows space instruments to perform daily observations producing up to tens of gigabytes of data. In contrast with this technology boost, the increase of downlink capability remains insufficient. In the particular case of science missions with long spacecraft-ground distances, it is typically small (0.1 to 2 Mbps). The communication or data storage bottleneck is then a major factor limiting the coverage and/or resolution of science instruments. Considering the ratio between the data volume and the telemetry rate, on-board compression is mandatory. Considering the high cost and the scarce nature of astronomy data, compression impacts have to be analysed. The work presented in this paper was to select a set of compression techniques compliant to astronomy mission objectives and to implement them on a flight representative DSP board taking into account its specific hardware architecture.

UNIFIED IMAGE/BITSTREAM ANALYSIS FOR THE ROBUST DECODING OF COMPRESSED VIDEO SEQUENCES FOR WIRELESS NETWORKS

L. Mac Manus and A. Kokaram, University of Dublin, Ireland

This paper is concerned with the problem of error detection and correction of MPEG-4 video streams transmitted over lossy networks. The problem is first defined and some relevant detail of the MPEG-4 syntax is presented. The difficulties encountered in articulating this problem within

a unified Bayesian framework are explored and two separate frameworks for dealing with error detection and error correction are then presented, along with some results. The paper concludes with some comments on the techniques used and pointers to future plans.

ANCIENT DOCUMENT COMPRESSION AND ARCHIVING VIA PAGE DECOMPOSITION

G. Calvagno, G.A. Mian, R. Rinaldo and L. Zimolo, Universita` di Padova, Italy

In this work we address the problem of document image compression, which has recently received increasing attention in the literature. To guarantee a high compression ratio and a satisfactory visual quality, the problem at hand requires sophisticated compression techniques, together with careful image preprocessing to remove noise, paper degradation artifacts, and ghost images from the back page. The strategy of the proposed technique is to segment the text from the background and apply the most convenient compression technique to the two resulting images. In particular, a pattern matching based compression procedure is applied to the binarized text image, while a lossy compression method is used for the background. The resulting image is perceptually almost lossless in its text component, while the background quality can be tailored to the desired compression ratio.

QUALITY ORIENTED JPEG CODING FOR COLOR STILL PICTURE

Yuukou Horita, Tomoe Nakase, Reda Ragab Gharieb, and Tadakuni Murai, Toyama University, Japan

In the JPEG coding, the compression rate can easily be decided as the coding parameter. When we employ the same coding parameter for some pictures, it is difficult to obtain the same picture qualities for human observer. In this paper, we propose a quality oriented JPEG coding method for color still picture. This coding method can automatically provide the user with the optimal coding parameter corresponding to the previously decided picture quality. By executing the proposed method on the personal computer, it has been found that this method is practically useful for quality oriented communication system.

DESIGN OF A MINIMUM-RATE PREDICTOR AND ITS APPLICATION TO LOSSLESS IMAGE CODING

I. Matsuda, H. Mori and S. Itoh, Science University of Tokyo, Japan

In this paper, we propose a novel method for designing linear predictors suitable for lossless image coding. In general, the Minimum Mean Square Error (MMSE) is an important concept in image coding. However, predictors which are optimized on the basis of the MSE criterion are not necessarily optimum from a viewpoint of coding efficiency. Thereupon our method optimizes a predictor so that a cost function which represents an amount of information on prediction errors can have a minimum. Moreover, we develop an adaptive lossless coding scheme for still images to demonstrate effectiveness of the proposed method. Simulation results indicate that the proposed coding scheme is superior in terms of coding efficiency to the conventional scheme

which utilizes the MMSE predictors, and that a coding rate of the proposed scheme is 0.14-0.40 bits/pel lower than that of the JPEG-LS standard coding scheme.

VIDEO CODING WITH QUAD-TREES AND ADAPTIVE VECTOR QUANTIZATION

Marcel Wagner, Freiburg University, Germany
Dietmar Saupe, Leipzig University, Germany

This paper presents recent progress of our ongoing research evaluating the prospects of adaptive vector quantization (AVQ) for very-low-bitrate video coding. In contrast to conventional state-of-the-art video coding based on entropy coding of motion compensated residual frames in the frequency domain, adaptive vector quantization offers the potential to adapt its codebooks to the changing statistics of image sequences. The basic building blocks of our current AVQ video codec are (1) block-based coding in the wavelet domain where wavelet coefficients correspond to (overlapping) spatial regions, (2) hierarchical organization of the wavelet coefficients using quad-tree structures, (3) three way coding mode decision for each block (block replenishment, product code vector quantization, new VQ block with codebook update), and (4) rigorous rate/distortion optimization for all coding choices (image partition and block coding mode). This video codec does not apply motion compensation, however. A comparison with standard transform coding (H.263) shows that inspite of the improvements of our coder over previously published AVQ video coders it still shows a performance gap of about 1 dB for some test sequences. We conclude that motion compensation is essential also for codecs based on AVQ. First preliminary tests show that AVQ coders that incorporate motion compensation can become competitive with standard transform coding.

HIGH-COMPRESSION OF CHROMINANCE DATA BY USE OF SEGMENTATION OF LUMINANCE

Maciej Bartkowiak and Marek Domanski, Poznan University of Technology, Poland

The paper describes a very simple technique for compression of chrominance data in colour images and video sequences coded at low bit rates. The technique is based on coding of arbitrarily shaped regions, whereby each region is efficiently represented by one value of chrominance which is averaged over the whole region. As the regions are determined on the basis of reconstructed luminance component, no information upon the segment shapes has to be transmitted. In order to obtain high compression, the set of assigned pairs of chromi-nance is processed by vector quantization. Experimental results prove high efficiency of the proposed technique.

A LOW-COMPLEXITY VIDEO CODER BASED ON THE DISCRETE WALSH HADAMARD TRANSFORM

R. Costantini, J. Bracamonte, G. Ramponi, J-L. Nagel, M. Ansorge, and F. Pellandini,
University of Trieste, Italy & University of Neuchatel, Switzerland

This paper reports the results obtained of applying the Discrete Walsh-Hadamard Transform

(DWHT) in a video coding scheme. It will be shown that for low bitrate applications the DWHT can reach performances analogous to those obtained with the Discrete Cosine Transform (DCT) in terms of compression efficiency, PSNR, and visual quality. This suggests the use of the DWHT in video coding systems where the computational cost is a fundamental issue, given the far less computational complexity of the DWHT with respect to the DCT.

WedPmOR4
Room 200
Spectral Analysis
Chair: H. Sakai *Kyoto University, Japan*

15:50

ON THE EFFICIENT IMPLEMENTATION OF THE CAPON SPECTRAL ESTIMATOR

Torbjörn Ekman, Andreas Jakobsson and Petre Stoica, Uppsala University, Sweden

We present an efficient implementation of the Amplitude Spectrum Capon (ASC) estimator. The implementation is based on the FFT and an efficient computation of the Cholesky-factor for the inverse covariance matrix. The Cholesky-factor is obtained from the linear prediction coefficients as computed by the modified covariance method. The implementation is significantly simpler than previous implementations, and it will yield spectral estimates of a similar quality as these. A short discussion on the differences between different Capon estimators is also included.

16:10

TIME-FREQUENCY SPACE CHARACTERIZATION BASED ON STATISTICAL CRITERIONS

C. Hory, N. Martin, A. Chehikian and L.E. Sölberg, Laboratoire des Image et des Signaux, France

In this paper is proposed a method to characterize a Time-Frequency Representation (TFR). This characterization takes place in a process which aim is to extract spectral patterns of an uni-dimensional signal from its TFR. The originality of this method is that the Time-Frequency plane is considered from a local point of view. Statistical features of set of TFR coefficients are defined in order to translate this plane to a new space representation called Features Space (FS). A description of the method is proposed when the TFR is the spectrogram and the features are estimators of the first and second moments of the local sets of coefficients. Statistical properties of those features are described when the analysed signal is a white gaussian noise and when it is a deterministic signal embedded in a white gaussian noise. The obtained theoretical results allow one to discriminate in the Features Space regions of the TFR dedicated to signal spectral patterns from those dedicated to the additive noise. An example is finally presented with seismic data recording of an avalanche.

16:30

FAST ARMA MODELLING OF POWER SPECTRAL DENSITY FUNCTIONS

A.C. den Brinker and A.W.J., Oomen Philips Research Laboratories, The Netherlands

A method for real-time estimation of ARMA models from power spectral density functions is proposed. It is based upon the combination of LPC methods to estimate both numerator and denominator polynomial of the transfer function and an iterative scheme to attribute the appropriate parts of the power spectral density function to pole or zero modelling of the transfer function. This approach yields an estimate to the data which approximately corresponds to a quadratic fit on a logarithmic scale. As further distinctive features we mention that the estimation is fast compared to more traditional methods and that it allows for frequency-warped modelling.

16:50

AN EXACT PARAMETRIC EQUIVALENT OF THE PERIODOGRAM

Einar Wensink, Hollandse Signaalapparaten, The Netherlands

A method is presented that provides the exact equivalent representation of the periodogram by means of an MA(n-1) model. This method will be compared to inferring an MA(q) model directly from the data by estimating the parameters and selecting the optimal model order. Representing the periodogram by means of an MA(n-1) model, enables the use of The Prediction Error to make an exact quantitative comparison of the periodogram to parametric spectral estimation methods. It follows that parametric methods provide better spectral estimates than the periodogram. This is explained by the fact that the periodogram is a non-selective transformation of all data, while the parametric methods discriminate between statistically significant information and noise by means of order selection.

17:10

A DATA-ADAPTIVE REGULARIZATION METHOD FOR LINE SPECTRUM ESTIMATION

I. Santamaria, A. Artes, R. Gonzalez and C. Pantaleon, University of Cantabria, Spain

We present a data-adaptive regularization method for estimation of line spectra. A Gaussian mixture is used as a suitable prior distribution and a different regularization term is associated to each component of the mixture. The regularized functional is minimized by applying an iterative procedure. The parameters of the mixture as well as the noise variance are updated at each iteration. In this way, the method can be applied even if accurate estimates of these parameters are not available. We apply the method to 1-D and 2-D problems showing its high-resolution ability and good performance.

VOLUME III

ThuAmOR1
Room 200
Parameter Estimation
Chair: P. Stoica *Uppsala University, Sweden*

09:20

SEQUENTIAL ESTIMATION OF PROBABILITY OF EVENTS

Petar M. Djuric, State University of New York, USA

In many signal processing problems, it is important to estimate the probability that a signal is present in observed data. As opposed to standard Bernoulli experiments where the outcomes of the experiments clearly show when the event occurred, there are many situations where only probabilistic claims can be made about the occurrence of events. Examples of the latter include a variety of problems related to detection of signals in noise. A processing scheme for estimating the posterior probability density function of the probability of occurrence of an event is proposed. It is based on a sequential importance sampling method, which approximates the desired posterior density with a probability measure composed of particles and their associated weights. With the arrival of new experimental data, the weights of the particles are updated, and as a result, the overall posterior modified. A simulation result is provided that shows the performance of the proposed method.

09:40

CLOSED-LOOP IDENTIFICATION VIA TIME AND FREQUENCY DOMAIN APPROACHES

Lianming Sun and Akira Sano, Keio University

The direct closed-loop identification problem is considered in the paper. In the closed-loop, the unknown plant is managed by a digital feedback controller, so the ordinary identifiability conditions depend on the external test signal or the controller structure. We will develop both the time and frequency domain approaches by applying the output inter-sampling scheme and illustrate that the ordinary identifiability conditions. The effectiveness of the proposed approaches is demonstrated through some numerical examples.

10:00

AN EFFICIENT METHOD FOR CHAOTIC SIGNAL PARAMETER ESTIMATION

C. Pantaleon, D. Luengo and I. Santamaria, University of Cantabria, Spain

Chaotic signals represent a new class of signals that can be applied in a wide range of signal processing applications. This paper describes an efficient method for parameter estimation of chaotic sequences generated by iterating tent maps with a known initial condition, and observed in white noise. The method is based on the connection between the symbolic sequence associated to the chaotic signal and the parameter of the tent map. The proposed method is asymptotically unbiased, and is found to attain the Cramer-Rao Lower Bound (CRLB) for a high Signal to Noise Ratio (SNR).

NONPARAMETRIC STATISTICS FOR SUBSPACE BASED FREQUENCY ESTIMATION

Samuli Visuri, Hannu Oja and Visa Koivunen, Helsinki University of Technology and University of Jyväskylä, Finland

The paper introduces new subspace based frequency estimation methods. The techniques are based on estimating the noise or signal subspace from the sample spatial sign autocovariance matrix. The theoretical motivation for the techniques is shown under the white Gaussian noise assumption. A simulation study is performed to demonstrate the robust performance of the algorithms both in Gaussian and non-Gaussian noise. The results imply that when the noise is Gaussian, the proposed methods have similar good performance as the standard subspace methods (MUSIC, ESPRIT). When the noise is heavy-tailed, the proposed methods outperform the standard subspace techniques.

11:10

A CUMULANT BASED ALGORITHM FOR THE IDENTIFICATION OF INPUT OUTPUT QUADRATIC SYSTEMS

P. Koukoulas, V. Tsoulkas and N. Kalouptsidis, University of Athens, Greece

A parameter estimation algorithm is developed for the identification of an input output quadratic model. The excitation is a zero mean white Gaussian input and the output is corrupted by additive measurement noise. Input output crosscumulants up to fifth order are employed and the identification problem of the unknown model parameters is reduced to the solution of succesive linear systems of equations that are solved iteratively. Simulation results are provided for different SNR's illustrating the performance of the algorithm and confirming the theoretical set up.

11:30

ON THE APPLICATION OF MINIMUM PHASE SPACE VOLUME PARAMETER ESTIMATION

Dejan Djonin and Ljiljana Stanimirovic, University of Victoria, Canada, Institute Mihajlo Pupin, Belgrade

In this paper, we consider the influence of high dimensional noise process on the accuracy of signal parameter estimation in low dimensional chaotic noise. Because of the inherently deterministic nature of the chaotic signal, instead of conventional probabilistic methods, a complexity measure based on phase space volume (PSV) of the reconstructed attractor is used to identify unknown system parameters. It was shown that through minimization of PSV a very effective system identification procedure could be achieved. This procedure however relies upon the fact that PSV of the chaotic process is negligible for embedding dimensions higher than the true dimension of the chaotic attractor, therefore any additional high dimensional noise degrades the estimation accuracy. Monte Carlo simulations are carried out to illustrate the efficiency of the minimum PSV method for parameter estimation in the presence of high dimensional noise. To reduce the optimization complexity a kd-tree search algorithm was used which takes only order Nlog(N) operations.

11:50

FREQUENCY ESTIMATION OF RADAR/SONAR SIGNALS AGAINST CORRELATED NON-GAUSSIAN NOISE

Enzo Dalle Mese, Fulvio Gini, Monica Montanari and Lucio Verrazzani, Univerisità di Pisa, Italy
Salah Bourennane, LSI/ENSPM Domaine Universitaire de Saint Jérôme

The contribution of this work is the derivation of the joint maximum likelihood (ML) estimator of complex amplitude and Doppler frequency of a radar/sonar target signal embedded in correlated non-Gaussian noise modeled as a compound-Gaussian process. The estimation accuracy of the ML frequency estimator is investigated and compared with that of the well-known periodogram and ESPRIT estimators under various operational scenarios. The hybrid Cramér-Rao lower bound (HCRLB) and a large sample closed-form expression for the mean square estimation error are also derived for Swerling I target signal. Finally, numerical results obtained by Monte Carlo simulation are checked by means of measured sea clutter data.

12:10

SMALL SAMPLE STATISTICS OF THE YULE-WALKER METHOD FOR AUTOREGRESSIVE PARAMETER ESTIMATION

Wilbert Dijkhof, University of Twente, The Netherlands
Einar Wensink, Hollandse Signaalapparaten, The Netherlands

In this paper we will give the expectation of (the square of) the reflection coefficient, residual variance and prediction error in small sample statistics in white noise. We will construct approximations of these expectations which are more accurate than the known first order Taylor approximations. We need these better approximations because in some applications (radar applications for example) the number of observations is small.

ThuAmOR2
Hall A
Speech Recognition
Chair: B. Kleijn ***Royal Institute of Technology, Sweden***

09:20

LOW BIT RATE SPEECH COMPRESSION FOR PLAYBACK IN SPEECH RECOGNITION SYSTEMS

Dan Chazan, Gilad Cohen, Ron Hoory and Meir Zibulski Affiliation, IBM Research

In this paper we describe a novel, low complexity, low bit rate speech compression and decompression methods for usage in systems where automatic speech recognition is performed. The coding scheme, referred to as the "Recognition Compatible Voice Coder" (RECOVC), is based on encoding the mel-frequency cepstral coefficients (MFCC), commonly used in large vocabulary continuous speech recognition systems, and the pitch period. The decoder reproduces natural sounding, good quality, intelligible speech for playback purposes. Implementation of a RECOVC scheme in a speech recognition system may simplify the playback procedure by reconstructing speech from feature vectors already extracted and used for recognition. Reduction in storage space or transmission bandwidth may be achieved in distributed speech recognition systems, by eliminating the need to store or transmit two separate bit streams, one for recognition and the other for playback.

09:40

NONLINEAR SIGNAL DECOMPOSITION INTO FUNCTIONAL SERIES FOR SPEECH RECOGNITION: A NEW APPROACH

Alexander M. Krot, Polina P. Tkachova and Boris A. Goncharov, Institute of Engineering Cybernetics of the National Academy of Sciences of Belarus, Belarus

The nonlinear speech signal decomposition based on Volterra-Wiener functional series is described. The solution of phoneme recognition problem by means of measuring Wiener kernels is proposed.

10:00

SPEAKER INDEPENDENT SPEECH RECOGNITION WITH ROBUST OUT-OF-VOCABULARY REJECTION

Charles C. Broun and William M. Campbell, Motorola Human Interface Laboratory Tempe, USA

With the increased use of speech recognition outside of the lab environment, the need for better out-of-vocabulary (OOV) rejection techniques is critical for the continued success of this user

interface. Not only must future speech recognition systems accurately reject OOV utterances, but they must also maintain their performance in mismatched (i.e., noisy) conditions. In this paper we extend our work on low-complexity, high-accuracy speaker independent speech recognition. We present a novel rejection criterion that is shown to be robust in mismatched conditions. This technique continues our emphasis on speech recognition for resource limited applications, by providing a solution that is highly scalable, requiring no additional memory and no significant increase in computation. The technique is based on the use of multiple garbage models (on the order of 100 or more) and a novel ranking method to achieve robust performance. Results on a large, 166-speaker database are provided for different design parameters, and they are compared to traditional threshold methods. Performance is shown to be superior to the approximated optimal Bayes reject rule.

10:50

ROBUST CONNECTED DIGIT RECOGNITION IN A CAR NOISE ENVIRONMENT

Lin Cong and Saf Asghar, Advanced Micro Devices, USA

This paper proposes a robust speaker-independent, connected digit recognition system for mobile applications. The system requires a small amount of ROM and low computational cost with high recognition accuracy. In addition, the system can be efficiently implemented on most currently available 32-bit fixed-point DSP chips. To reach these goals, we combined robust speech parameter processing technologies with dual MQ and VQ pairs, which supply discrete gender-dependent HMM to increase the performance of HMMs. The dual MQ/VQ pairs exploit the "evolution" of the speech short-term spectral envelopes with one pair providing error compensation using LSP mean compensated coefficients. Correspondingly, we proposed the dual MQ/HMM and VQ/HMM decoding pair algorithm. In a car noise environment, the system attains an 80% average connected digit recognition accuracy at around 10 dB SNR. A digit accuracy of 93% is obtained at 5 dB SNR.

11:10

SPEECH RECOGNITION COMPLEXITY REDUCTION USING DECIMATION OF CEPSTRAL TIME TRAJECTORIES

Juha Iso-Sipilä, Nokia Research Center, Finland

The usage of speech recognition technology has become common in a variety of applications ranging from desktop computers with dictation engines to mobile devices with speaker-dependent name dialing. While dictation software is solely run on powerful desktop PCs with huge amounts of memory available mobile devices have limited memory and computational resources. In order to implement speech recognition algorithms into mobile devices, the complexity of the algorithms has to meet the capabilities of the device. This paper addresses the problem of complexity and memory constraints in mobile devices. A specific approach called time domain decimation of feature vectors is presented. This general signal processing technique can be applied to speech recognition due to the band-limited modulation spectrum of the feature vector time trajectories. By decimating the feature vector stream of 100 frames per second by

factors of 2 to 5, the complexity of the speech recognizer can be reduced proportionally to the decimation factor. Experiments with name dialing task show that decimation factor of 4 can be used without any significant degradation in the performance of the speech recognizer. With the proposed method, the computational complexity can be reduced by 70% and over 60% save in RAM usage can be obtained.

11:30

A PHONETIC VOCODER FOR FINNISH

Jukka Kivimäki, Tommi Lahti and Konsta Koppinen, Tampere University of Technology, Finland

A phonetic vocoding system for Finnish is described. This very low bit rate speech coding method is a subclass of segmental vocoding. The proposed system utilizes speech recognition based on a HMM system to phonetically segment and label the input speech. The fundamental frequency is estimated using a robust pitch tracking algorithm. Speech reconstruction is carried out with a conventional LPC speech synthesis algorithm. This approach makes possible a higher-level description of the speech than usually obtained in coders, thus enabling e.g. speaker transformation relatively easily. The coder has a high computational complexity and requires a delay of several hundreds of milliseconds, but the obtainable bit rate is under 500 bits/s. The speech intelligibility of the current implementation is high but the overall speech quality is relatively poor.

11:50

SERBIAN SPOKEN DIALOGUE SYSTEM FOR BUS TRAVEL INFORMATION RETRIEVAL

Ljiljana Stanimirovic, Zoran Cirovic and Dejan Đonin, Mihajlo Pupin Institute, Serbia
Milan Savic, University of Belgrade, Serbia

In this paper we present our recent work in implementing Serbian spoken dialogue system for the bus information retrieval at the main Belgrade bus station. Dialogue is organized into several levels. At each level, system has to recognize a limited number of keywords in continuous speech of Serbian. The keywords were modeled by HMMs (Hidden Markov Models) in such a way that each syllable is three-state HMM. In order to obtain optimal thresholds for integral confidence measure in decoding procedure.

12:10

FUSING LENGTH AND VOICING INFORMATION, AND HMM DECISION USING A BAYESIAN CAUSAL TREE AGAINST INSUFFICIENT TRAINING DATA

Mubeccel Demirekler, Fahri Karahan and Tolga Ciloglu, Middle East Technical University, Turkey

This paper presents the work done to improve the recognition rate in an isolated word recognition problem with single utterance training. The negative effect of errors (due to insufficient training data) in estimated model parameters is compensated by fusing the information obtained from HMM evaluation and those generated for the word length and voicing at the beginning and end of the word. A Bayesian Causal Tree structure is developed to accomplish the fusion. The final decision is made on one of the three candidates which are most likely according to HMM evaluation. The reliability of the HMM ordering is improved by applying variance flooring.

ThuAmSS1
Small Audiotorium
Speech Coding
Chair: P. Haavisto *Nokia Research Center, Finland*

09:20

STANDARDISATION OF THE ADAPTIVE MULTI-RATE CODEC

Kari Järvinen, Nokia Research Center

European Telecommunication Standards Institute (ETSI) initiated a standardisation program in October 1997 to develop an Adaptive Multi-Rate (AMR) codec for GSM. After two competitive selection phases, ETSI chose in October 1998 a codec developed in collaboration between Ericsson, Nokia, and Siemens. The codec standard was finalised, characterised, and formally approved in ETSI during early 1999. The AMR codec provides the next step of speech quality improvement in GSM after the introduction of Enhanced Full-Rate (EFR) codec in 1996. AMR offers substantial improvement in error robustness by adapting speech and channel coding depending on channel conditions. By switching to operate in the GSM half-rate channel during good channel conditions, AMR provides also channel capacity gain over the EFR codec. In April 1999, the Third Generation Partnership Project (3GPP) adopted the AMR codec as the mandatory speech codec for the third generation WCDMA system.

10:00

ADAPTIVE WINDOW EXCITATION CODING IN LOW-BIT-RATE CELP CODERS

Vladimir Cuperman, Allen Gersho, Jan Linden, Ajit Rao and Tung-Chiang Yang, SignalCom Inc.
Sassan Ahmadi, Ryan Heidari and Fenghua Liu, Nokia Mobile Phones

Low bit-rate CELP, adaptive windows, variable-rate speech coding A new paradigm for efficient coding of the fixed-codebook excitation in CELP coders at low-bit rates is presented. In this scheme, the non-zero components of the fixed-codebook excitation are localized to a set of windows whose locations are dependent on the pitch frequency and on the energy contour of a modified residual signal. Highly efficient coding is thus achieved by allocating most of the fixed excitation bits to capture the essential excitation events. The paradigm is validated by computer simulation of a variable-rate speech codec. The performance of the codec is evaluated by informal subjective tests comparing it with TIA standard variable rate speech codecs. The results confirm that the proposed scheme can be used to reproduce speech at average bit rates from 2.3 to 3.4 kbps with very high quality and intelligibility.

10:50

COMBINING PARAMETRIC AND WAVEFORM-MATCHING CODERS FOR LOW BIT-RATE SPEECH CODING

Jacek Stachurski and Alan McCree, Texas Instruments

Waveform-matching coders preserve the shape of the target waveform and the time synchrony between the original and the synthesized signal. In parametric coders the shape of the encoded waveform is often changed, and the time-synchrony between the input and synthesized speech is not preserved. These two issues, time synchrony and waveform shape, are major obstacles in representing different speech regions with parametric and waveform coders, as arbitrary switching between the two results in annoying artifacts in transition regions. We describe a hybrid parametric/waveform coder with MELP used for strongly voiced regions and CELP employed for weakly voiced and unvoiced speech segments. To limit switching artifacts between the coders, alignment phase is estimated and transmitted in MELP making the original and synthesized speech time-synchronous. Additionally, in zero-phase equalization, the phase component of the CELP target signal is removed making the target waveform more similar to the MELP-synthesized speech. These two techniques, alignment-phase encoding and zero-phase equalization, greatly reduce switching artifacts in transition regions between the parametric and waveform coders. Formal listening tests of the 4 kb/s hybrid coder show that it can achieve speech quality equivalent to 32 kb/s ADPCM.

11:10

A 2.4/1.2KB/S SPEECH CODER WITH NOISE PRE-PROCESSOR

M. Stefanovic, Y. D. Cho, S. Villette and A. M .Kondoz, Research University of Surrey, UK

Speech coding at very low bit rates has many applications such as answering machines, IP telephony, mobile communications, military communications etc.. In this paper we describe a speech coder capable of operating at both 2.4 and 1.2kb/s, and produces good quality synthesised speech. The basic principle of the coder is based on frequency domain vocoding method where the LP excitation is classified into voiced and unvoiced parts. The rate of the coder can be switched from 2.4kb/s, which operates on 20ms frames, to 1.2kb/s by having 60ms frames. Both rates use the same analysis and synthesis building blocks over 20ms. Reliable pitch estimation and very elaborate voiced/unvoiced mixture determination algorithms render the algorithm robust to background noise. However in order to communicate in very severe noisy conditions a noise pre-processor has been integrated within the speech encoder. Owing to robust quantisation methods and careful encoding/index assignment algorithms the coder maintains its good quality with 1% random bit errors and 3% random frame (20ms) erasures.

11:30

A HARMONIC+NOISE CODER WITH IMPROVED TRANSIENT SPEECH PERFORMANCE

Eric W. M. Yu and Cheung-Fat Chan, City University of Hong Kong, Hong Kong

This paper presents a 2.4 kbps harmonic+noise coder with improved transient speech performance. The coder operates in either the steady mode or the transient mode. A predictive phase model is proposed for the efficient encoding of phase information in steady mode. In this model, the normalized frequency deviations are used to minimize the phase prediction errors. A polynomial is fitted to the set of normalized frequency deviations and the polynomial coefficients are vector quantized in closed-loop. For the transient speech, the voiced part is modeled by the low-frequency harmonic components while the unvoiced part is modeled by an all-pole model. The transitional behavior is represented by the onset time and growth rate of each harmonic component. An analysis-by-synthesis procedure is proposed to determine the onset times and growth rates of the harmonic components in the time domain. The proposed coder retains the transitional characteristics of speech signal. Subjective testing shows that the speech quality of the proposed coder is sufficient for communication purposes.

11:50

A SUBJECTIVE PERFORMANCE STUDY OF A SINUSOIDAL SPEECH CODING MODEL

Ari Heikkinen, Nokia Research Center

Recently, there has been an increasing interest in developing wireline quality speech coders at rates of 4 kbps and below. Good quality at low bit rates is often claimed to be achieved with spectral domain coders which are based on a sinusoidal representation of the speech signal. However, based for example on the recent efforts to standardize a 4 kbps speech coder in International Telecommunications Union (ITU) the current technology has proved to be immature to meet the wireline quality requirements. In this paper, the quality of a speech coder implementation which is based on a commonly used sinusoidal model is verified by a listening test. Three coder versions are tested to evaluate the performance of the underlying speech coding model itself as well as the performance of the widely used parameter models which make sinusoidal coding amenable for low bit rates. Based on the test results, the quality of the implemented sinusoidal speech coding model is not quite comparable to that of the ITU Recommendation G.726 (ADPCM) at 32 kbps, which is commonly used as a reference for wireline quality. Additionally, it was observed that the implemented phase model degrades clearly the performance of the underlying sinusoidal coding model.

12:10

FRACTAL PREDICTION IN WAVEFORM CODING OF SPEECH SIGNALS

V.Almenar and A. Abiol, Universidad Politecnica de Valencia, Spain

This paper describes how fractal predictive coding can be used in speech coding. First the IFS theory, on which fractal coding is based, is presented. Then, the coding algorithm is described and its performance is compared with the ADPCM coder G.726. Next, some perceptual considerations are introduced, which allow better performance at lower bit rates, obtaining good results at 16~kb/s. Finally, the use of this algorithm is evaluated in wideband speech coding, giving good performance at 32~kb/s.

ThuAmSS2
Studio
Color Image Processing
Chair: C. Christopoulos *Ericsson Radio Systems AB, Sweden*

09:20

COLOR IMAGE RESTORATION USING COMPOUND GAUSS-MARKOV RANDOM FIELDS

Javier Mateos and Rafael Molina, University of Granada, Spain
Aggelos K. Katsaggelos, Northwestern University, USA

In this work we extend the use of Compound Gauss Markov Random Fields to the restoration of color images. While most of the work in color image restoration is concentrated on enforcing similarity between the intensity values of the pixels in the image bands, we propose combining information by means of the line process. In order to find the multichannel restoration modified versions of ICM and SA are proposed. The methods are finally tested on real images.

09:40

MULTICHANNEL IMAGE PROCESSING USING FUZZY VECTOR MEDIAN-RATIONAL HYBRID FILTERS

Lazhar Khriji, E.N.I.M., Tunisia
Moncef Gabbouj, Tampere University of technology, Finland

A new multichannel filtering approach is introduced and analyzed in this paper. These filters are based on rational functions (RF) using fuzzy transformations of the Euclidean distances among the different vectors to adapt to local data in the color image. The output is the result of vector rational operation taking into account three sub-functions, such as two Fuzzy Vector Median (FVM) sub-filters and one Fuzzy Center Weighted Vector Median Filter (FCWVMF). Simulation studies indicate that the filters are computationally attractive and have excellent performance such as edge and details preservation and accurate chromaticity estimation.

10:00

NEW MULTISPECTRAL EDGE DETECTION METHODS BASED ON ROUGHNESS AND DIFFERENTIAL PREWITT

Jarkko Ansamäki, Kymenlaakso Polytechnic, Finland
Pekka J. Toivanen, Linköping University, Sweden
Jussi Parkkinen, University of Joensuu, Finland

In this paper, two new edge detection methods for multispectral images are presented. The first method is based on calculating Euclidean distances between pixel vectors inside a suitable mask.

This method approximates the roughness of the pixel neighborhood. The result is a scalar edge image. This method is tested using an airborne remote sensing image and two artificial images. One artificial image contains metametric colors and the other have group of colors arranged using the Self Organizing Map. In all the cases, the edges are clearly visible without further filtering. The second method is a differential Prewitt method using vectors for calculating the gradient value. This case the edge image is a vector image. This method is tested using the same test images as in the first case.

10:50

USING COLOUR FEATURES TO BLOCK DUBIOUS IMAGES

Yi Chan, Richard Harvey and J. Andrew Bangham, University of East Anglia

This paper describes a vision system that classifies web-images containing people. It works by identifying skin-coloured regions, extracting very simple features from these regions and making a classification decision. A two-stage skin filtering algorithm using likelihood matrices in HSV space followed by some local clustering works well. Our conclusion is that a simple approach using low-level features can work as well as much more complicated methods.

11:10

ENHANCEMENT AND REGISTRATION OF MULTICHANNEL IMAGES BASED ON VECTOR ORDER FILTER

Patrick Lambert and Sébastien Gaspard, Université de Savoie, France

This paper proposes a new structure for the classical median vector filter. It is based on a vector ordering relation obtained by associating a cumulative distance to each vector. The specifity of the proposed filter relies in the definition of the neighboring window used to calculate this distance. The filter presents both an enhancement and a registration effect, combined with a smoothing effect. His properties are studied and some applications are proposed. The filter is also compared with other classical filters. It can be used for any multichannel images, like color images, multi spectral images or multi-temporal images.

11:30

COLOR TRANSFORMATIONS FOR LOSSLESS IMAGE COMPRESSION

Marek Domañski and Krzysztof Rakowski, Poznañ University of Technology, Poland

The paper defines the conditions for the coefficients of a linear color transformation that guarantee that after one cycle of transformation and inverse transformation all further cycles of transformation do not introduce rounding errors to sample values. Moreover, the upper limits of the rounding error can be calculated according to the formulas given in the paper. The results are very important for nearly lossless compression of color images. Color transformations that decorrelate color components allow a significant increase in compression ratio achieved by use of standard lossles compression techniques applied independently to the components. The results of the paper prove that further cycles of compression and decompression together with color transformations do not increase the rounding errors. The experimental results for the RGB ® YCRCB transformation are also included.

11:50

FAST k-NN CLASSIFICATION WITH AN OPTIMAL k-DISTANCE TRANSFORMATION ALGORITHM

Olivier Cuisenaire, Swiss Federal Institute of Technology, Switzerland
Benoit Macq, Universite catholique de Louvain, Belgium

The k-NN classification rule uses information from the k nearest prototypes in order to classify a pattern. In this paper, we improve Warfield's lookup table approach, where the classification problem is reformulated in terms of distance transformations. We propose a new k-distance transformation algorithm using ordered propagation. We show that - using this algorithm - the k-NN classification of F possible patterns in a D-dimensional space has a O(k.D.F) complexity.

12:10

THE COLOUR IN THE UPCOMING MPEG-7 STANDARD

Charilaos Christopoulos and Daniel Berg, Ericsson Radio Systems, Sweden
Athanassios Skodras, University of Patras, Greece

The colour spaces supported by the different image and video coding standards are presented in the present communication. Extensive reference to the colour spaces and the relevant colour descriptors supported by the MPEG-7 standard is given, and experiments on colour-based image retrieval efficiency are illustrated.

ThuAmSS3
Hall B
Multimedia Indexing, Browsing and Retrieval
Chair: M. Gabbouj ***Tampere University of Technology, Finland***

09:20

VIDEO CLASSIFICATION BASED ON HMM USING TEXT AND FACES

Nevenka Dimitrova and Lalitha Agnihotri, Philips Research
Gang Wei, Wayne State University, USA

Video content classification and retrieval is a necessary tool in the current merging of entertainment and information media. With the advent of broadband networking, every consumer will have video programs available on-line as well as in the traditional distribution channels. Systems that help in content management have to discern between different categories of video in order to provide for fast retrieval. In this paper we present a novel method for video classification based on face and text trajectories. This is based on the observation that in different TV categories there are different face and text trajectory patterns. Face and text tracking is applied to arbitrary video clips to extract faces and text trajectories. We used Hidden Markov Models (HMM) to classify a given video clip into predefined categories, e.g., commercial, news, sitcom and soap. Our preliminary experimental results show classification accuracy of over 80% for HMM method on short video clips. This paper describes continuity-based face and text detection and tracking in video for the above HMM classification method.

10:00

INDEXING SPATIAL CONTENT OF STILL TEXTURE IN MPEG-4

Jayank Bhalod, Mohammed Zubair V and S. Panchanathan, Arizona State University, USA

There has been a rapid increase in audio-visual information in digital form with the ever-increasing popularity of the Internet. The Moving Pictures Expert Group has recently presented a video compression and content manipulation standard, called MPEG-4 [1]. This was developed in response to the growing need for a coding method that can facilitate access to visual objects in natural and synthetic moving pictures in a wide variety of applications. In this paper, we propose a method for indexing the spatial content of Still Texture in MPEG-4 Video. There has been a number of approaches proposed recently [1],[3],[5],[6], which address content based indexing using features such as, color, shape and/or texture. We note that the still texture information is encoded using Wavelet Transform in MPEG 4. Since still texture can have an arbitrary shape, we propose to employ a combination of shape, color and texture features. The shape features are derived from the alpha plane and represented using Cubic B-spline approximation. The color and texture features are extracted directly from the wavelet compressed domain. An important attribute of the proposed approach is also the introduction of the novel concept of “containment” of spatial features for content based indexing.

10:50

AN EFFECTIVE IMAGE REPRESENTATION FOR VISUAL INFORMATION RETRIEVAL

Jane You, Griffith University, Australia

This paper presents an agent-oriented approach to visual information retrieval. A new data model is introduced to represent image content in a top-down fashion and a deductive agent-oriented database structure is proposed to facilitate visual information representation, indexing, query, searching and maintenance. In addition, the integration of techniques in image understanding and information retrieval is adopted for effective and efficient visual information retrieval.

11:10

CONTENT-BASED IMAGE RETRIEVAL USING FUZZY VISUAL REPRESENTATION

Anastasios D. Doulamis, Nikolaos D. Doulamis and Stefanos D. Kollias, National Technical University of Athens, Greece

In this paper, a fuzzy approach is used for efficiently representing the visual content. Initially, the image is partitioned into several segments (objects) and for each segment appropriate features, such as the segment color and texture, are extracted. In the following, all features are classified in a fuzzy framework, resulting in a content interpretation closer to the human perception. Furthermore, such fuzzy representation of visual content gives to the users the capability of expressing video queries using statements drawn from the natural language. An objective criterion has been also proposed to evaluate the system reliability, which indicates the advantages of the proposed fuzzy approach compared to other traditional techniques for image retrieval.

11:30

A FRAMEWORK FOR ORDINAL-BASED IMAGE CORRESPONDENCE

Ilya Shmulevich, Bogdan Cramariuc, Moncef Gabbouj, Tampere University of Technology, Finland

We propose a general framework for ordinal-based image correspondence. This framework not only contains Kendall's tau and Spearman's rho correlation measures as special cases, but allows one to design other correspondence measures that can potentially incorporate region-based spatial information. We consider one such possible correspondence measure and evaluate its performance on a set of test images.

11:50

COLOR FRAGMENTATION-WEIGHTED HISTOGRAM FOR SKETCH BASED IMAGE QUERIES

Eugenio Di Sciascio, Cataldo Guaragnella and Marina Mongiello, Politecnico di Bari, Italy

In this work we present a low-dimensional color fragmentation-weighted histogram technique specifically designed for the query by sketch approach. It keeps into account the fragmentation colors have in an image and the assumption that the number of colors typically present in a sketch are almost always just a few. The algorithm has been implemented in DrawSearch, our prototype CBIR system. According to our experiments, the algorithm outperforms classic histograms and compares favorably with much more complex techniques.

12:10

COLOR-BASED RETRIEVAL OF FACIAL IMAGES

Yannis Avrithis, Nicolas Tsapatsoulis and Stefanos Kollias, National Technical University of Athens, Greece

Content-based retrieval from image databases attracts increasing interest the last few years. On the other hand several recent works on face detection based on the chrominance components of the color space have been presented in the literature showing promising results. In this work we combine color segmentation techniques and color based face detection in an efficient way for the purpose of facial image retrieving. In particular, images stored in a multimedia database are analyzed using the M-RSST segmentation algorithm and segment features including average color components, size, location, shape and texture are extracted for several image resolutions. An adaptive two-dimensional Gaussian density function is then employed for modeling skin-tone chrominance color component distribution and detecting image segments that probably correspond to human faces. This information is combined with object shape characteristics so that robust face detection is achieved. Based on the above, a query by example framework is proposed, supporting a highly interactive, configurable and flexible content-based retrieval system for human faces. Experimental results have shown that the proposed implementation combines efficiency, robustness and speed, and could be extended to generic visual information retrieval or video databases.

ThuAmPO1
Exhibition Hall
Array Signal Processing
Chair: B. Champagne ***McGill University, Canada***

MINIMAX ROBUST M-BEAMFORMING FOR MOVING SOURCES AND IMPULSE NOISE ENVIRONMENT

Vladimir Katkovnik, Tampere University of Technology, Finland

The minimax robust M - beamforming is developed for complex-valued array observations (snapshots) contaminated by impulse random errors having an unknown heavy-tailed error distribution. The beamformer as a robust estimator of the time-varying direction of arrival (DOA) and the waveform signal (envelope) is defined by minimizing a nonquadratic loss function of residuals. A tracking ability of the estimates is assured by using the local polynomial approximation model of a source movement and a sliding window of observations. A proposed new beamformer has a two-dimensional power function. Maximum peaks of this power function are used for source separation and estimation of DOA and their first derivatives. The asymptotic variance and bias of these estimates are obtained for wide classes of the loss functions and probability distributions of the noise. These results justify using the minimax Huber's estimation theory for a selection of the loss function of the M - beamforming loss functions.

STRUCTURED SPATIAL INTERFERENCE REJECTION COMBINING

George Jöngren and Björn Ottersten, Royal Institute of Technology
David Astély, Nokia Networks

Modern communication systems are often interference limited. By modeling the co-channel interference as spatially colored, temporally white Gaussian noise, it is straightforward to incorporate interference rejection in the metric of a sequence estimator. In general, estimates of both the channels and the spatial color of the co-channel interference and the noise are needed. In this work, a structured model for the spatial noise covariance matrix is proposed and maximum likelihood estimates of the parameters are derived. The choice of model order is also addressed. Simulation results show large gains due to the use of these structured estimates compared with the conventional, unstructured, approach.

ROBUST SPEAKER LOCALIZATION USING A MICROPHONE ARRAY

Norbert Strobel and Rudolf Rabenstein, University of Erlangen-Nuremberg, Germany

This paper presents a speaker localization system using a microphone array. The array is operated as a steered filter-and-sum beamformer implemented as a summed correlator. In particular, we emphasize the use of a speech pause detector to improve the robustness of the speaker localization system by avoiding erroneous position estimates when no speech signal is present. Simulation results and measurements show that speech pause detection improves the

overall system performance considerably.

ROBUST PHASE SHIFT ESTIMATION IN NOISE FOR MICROPHONE ARRAYS WITH VIRTUAL SENSORS

Mijail Arcienega, Andrzej Drygajlo and Joseph Maisano, Swiss Federal Institute of Technology, Switzerland

Virtualization is a new method which allows to build highly directive microphone arrays even if a small number of sensors is available. Using this method in the case of electronic hearing aids, non-physical (virtual) sensors can be created by having only a linear end-fire array of two physical microphones. Then the global array processing with more than two sensors can be implemented in the time or frequency domain. In this paper, frequency domain techniques, which include the application of the discrete Fourier transform (DFT) and phase shift beamforming are considered. Our prime interest is in the phase shift estimation for virtual microphones in the presence of acoustic background noise. A careful consideration of the non-stationary nature of the signals and the noise influence allows the design of robust frequency domain algorithms to be easily integrated into a complete and efficient hearing aid system.

ARRAY CALIBRATION IN THE PRESENCE OF UNKNOWN SENSOR CHARACTERISTICS AND MUTUAL COUPLING

Konstantinos V. Stavropoulos and Athanassios Manikas, University of London, UK

In this paper, a method that is capable of handling simultaneously gain, phase and location as well as mutual coupling uncertainties is proposed for calibrating a planar array of general geometry. The proposed technique assumes that the array gain and phase errors are non-directional and that three time-disjoint pilot sources are available. The efficiency and the potential benefits of the proposed pilot calibration method are illustrated by a number of representative examples. Keywords array calibration gain, phase, location and mutual coupling uncertainties

DETECTION-ESTIMATION OF MORE UNCORRELATED GAUSSIAN SOURCES THAN SENSORS USING PARTIALLY AUGMENTABLE SPARSE ANTENNA ARRAYS

Yuri Abramovich and Nick Spencer, Cooperative Research Centre for Sensor Signal and Information Processing

We introduce a new approach for the detection-estimation problem for "partially augmentable" M-sensor nonuniform linear antenna arrays, whose set of intersensor differences is not complete. We propose a positive-definite completion method of the partially specified M_α-variate Toeplitz covariance matrix via direct augmentation, followed by its transformation into a sequence of positive-definite Toeplitz matrices T_{μ}, each having their $(M_\alpha-\mu)$ smallest eigenvalues equalised, and so being an appropriate candidate for the number of sources μ. The method adopts convex and linear programming

algorithms. Likelihood ratio information criteria (AIC, MDL and MAP) are then used to select the best candidate model, simultaneously providing DOA estimates for the indicated number of sources. Comparison of the detection performance of this method with the traditional AIC/MDL techniques applied to the corresponding $\Ma$-sensor ULA with $m.

DOWNLINK SPACE-TIME PROCESSING FOR FDD SYSTEMS

P. Forster, L. Fety and M. Le Bot, Conservatoire National des Arts et Metiers, France

The focus of this paper is downlink space-time processing for FDD (Frequency Division Duplex) radiocommunication systems whose base stations are equipped with antenna arrays. It shows how space-time filters can be appropriately designed for downlink in order to both combat co-cohannel interferences and reduce fading effects. These filters, which can be estimated from uplink data only, are an extension of previous schemes which considered only space processing for downlink.

MULTI-SOURCE LOCALIZATION IN REVERBERANT ENVIRONMENTS

Elio D. Di Claudio, Raffaele Parisi and Gianni Orlandi, University of Rome "La Sapienza", Italy

The very large relative bandwidth of acoustic sources, coupled with the high number of reflections of a typical listening room, makes localization a challenging task, since all basic assumptions of classical array processing algorithms constitute at the best viable approximations in real-world environments. In this work, a novel decentralized approach for acoustic localization in reverberant environment is presented. It is based on a two-stage strategy. First, candidate source positions are found by a Time-Delay-Of-Arrivals (TDOA) analysis of signals received by colocated pairs of microphones. Differential delays are estimated by a robust ROOT-MUSIC based technique, applied to the sample cross-spectrum of whitened signals recorded from each microphone pair. A subsequent clustering stage in the spatial coordinates validates the raw TDOA estimates, eliminating most of false detections. The new algorithm is capable of tracking multiple speakers at the same time, exhibits a very good consistency of location estimates, and compares favourably with previous approaches.

ROBUST AND CONSTRAINED DOWNLINK BEAMFORMING

Mats Bengtsson, Royal Institute of Technology, Sweden

When antenna arrays are introduced in a cellular system, one critical aspect is the design of beamformers for downlink transmission. We extend a recently proposed strategy for optimal downlink beamforming to give increased robustness to channel uncertainties and to incorporate constraints on the dynamic range. The solution can be efficiently calculated using semidefinite optimization. In a few cases, the optimal solution is given by a time-varying beamformer. We explore the performance of these modified algorithms.

A DECONVOLUTION METHOD FOR THE CHARACTERIZATION OF DISTRIBUTED SOURCES VIA LINEAR PREDICTION

T. Abdellatif, P. Larzabal and H. Clergeot, LESiR - ENS Cachan, France

This paper deals with high resolution bearing estimation in urban radiocommunication scenarii. Indeed, in such environments, scatterers local to the emitter engender diffuse paths that deteriorate the performances of conventional subspace-based algorithms. A deconvolution technique, involving Linear Prediction methods, is designed to characterize so called distributed sources by returning the mean angle and the angular spreading of the signal angular power density. Two ways of implementation are proposed in two extreme cases of diffuse paths correlations. Simulation results show that this originally proposed method provides satisfaying results compared to the Cramer Rao-Bound and moreover outperform more famous subspace-based algorithms.

USING PHASE CONJUGATION FOR UNDERWATER ACOUSTIC COMMUNICATION: DESIGN GUIDELINES

Joao Gomes and Victor Barroso, Instituto Superior Tecnico - Instituto de Sistemas e Robotica

Broadband phase conjugation using a time-reversal mirror may be used to focus acoustic waves in inhomogeneous media. This technique has been successfully applied in the ocean over distances of a few kilometers, stimulating research into possible applications. In this paper, a time-reversal mirror is considered as a device for mitigating intersymbol interference in coherent communication, hence reducing the complexity of underwater receivers. Approximate expressions are obtained relating its focusing power with key array and environmental parameters. These provide guidelines for designing actual phase-conjugate arrays. Results obtained for a monochromatic source are then extended to broadband signals used in digital communication in order to predict the evolution of intersymbol interference in the vicinity of the focal point.

OPTIMUM SELF-REFERENCE SPATIAL DIVERSITY PROCESSING FOR FDSS AND FH COMMUNICATION SYSTEMS

Daniel Pérez Palomar and Miguel Angel Lagunas, Universitat Politecnica de Calalunya, Spain

In 1996, a new spread spectrum communication system, called Frequency Diversity Spread Spectrum (FDSS), was proposed as a system to mitigate the problems of the common spread spectrum techniques. In this paper, two blind self-reference optimum beamforming methods tailored for the FDSS signalling scheme using the trivial repetition code are derived based on the inherent cross-correlation properties present among all the frequency bands. The method is also applicable to fast Frequency Hopping (FH) systems. The proposed methods use the information of the all the frequency bins jointly to yield optimum beamforming in the sense of maximum Signal to Interference-plus-Noise Ratio (SINR). One method computes all the beamvectors jointly performing a single generalized eigenvector computation. The other calculates each beamvector independently aiming at a parallel implementation. The two methods are analytically studied and tested via Monte-Carlo simulations. They are compared to other existing methods in

terms of bit error rate, outperforming them and showing a performance close to the optimum value.

ThuAmPO2
Exhibition Hall
Motion Estimation and Video Compression
Chair: T. Ebrahimi ***Ecole Polytechnique Federale de Lausanne, Switzerland***

ADAPTIVE MULTIRESOLUTION IMAGER BASED ON FPGAs

Pelegrín Camacho, Francisco Coslado, Martín González and Francisco Sandoval, E.T.S. Ingenieros de Telecomunicación Málaga, Spain

This paper describes the architecture of adaptive space-variant imagers with Cartesian structure. Besides their multiresolution outputs, reconfigurable imagers can be adapted for contextual image preprocessing and upgraded for generation of specific data to be processed at the higher level modules of vision systems, making it possible to unload those stages of certain tasks and improving their time response without significant addition of hardware. A synthesizable implementation of these imagers, based on last generation FPGAs adapted to CCD cameras or Active-Pixel digital image CMOS sensors, is also described.

ROTATIONAL MOTION ESTIMATION AND ANALYSIS IN DIGITAL IMAGE SEQUENCES

Mingqi Kong and Bijoy Ghosh, Washington University

In this paper we consider the problem of motion estimation, analysis and selective reconstruction of objects undergoing rotational motion. There are multiple objects rotating with different angular velocities in the image sequences. The goal is to estimate their distinct motion parameters(in this case is the angular velocities), and also identify their locations at each time instance by selective reconstruction. These parameters and locations can be used for various purpose such as motion compensation in video coding, military target detection, recognition and tracking, robot's focus/shift attention, road navigation, etc.The new algorithm we have developed is based on angular velocity tuned 2D+T filters. One of the important fact about our algorithm is that it is effective for both spinning motion and orbiting motion, thus unifies the treatment of the two kinds of rotational motion. Furthermore, it can also derives the angular velocities seperatly by choosing differnt scale values once the objects have different sizes. The algorithm is robust against noise and occlusion. It is simulated and tested on several synthesized image sequences corrupted by noise and shows to be accurate.

REAL-TIME IMPLEMENTATION OF H.263 VIDEO ENCODER ON TMS320C6201 FIXED POINT DSP

Olli Lehtoranta, Timo Hämäläinen and Jukka Saarinen, Tampere University of Technology, Finland

The real-time implementation of H.263 video encoder following ITU-T H.263 recommendation is described. The current implementation shows that QCIF-size images can be encoded in real-

time by the TMS320C6201 when computationally light motion estimation algorithm and basic H.263 coding mode is used. The real-time performance can be achieved using C-code and optimizing some key functions with assembler. In addition, careful memory management design for program code and application data is required. With presented coding scheme, up to 31 fps can be achieved.

AN ACTIVE MESH BASED TRACKER FOR FEATURE CORRESPONDENCES

A. Griffin and J. Kittler, University of Surrey, UK

We address the problem of obtaining feature correspondences from a set of image frames that comprise a video sequence. These correspondences are to be used subsequently within a 3-d estimation stage, based for example on the Fundamental Matrix or Trifocal Tensor. When working with video sequences, two major problems present themselves. Firstly adjacent video frames correspond to negligible camera motion leading to instability of the 3-d metric. The solution to this would be track features until the camera motion provides stability. This however leads directly to the second problem. Traditionally feature trackers are based upon finding the optic flow of an image patch located around an image feature. The problem is that image features usually correspond to say a junction of objects or the junction of object and background. The assumption that the composition of that patch remains constant is obviously violated under the 3-d motion that we seek to estimate. Our solution is to treat the problem as one of fitting an active mesh to the sequence. The optic flow of each mesh element then votes for the overall motion of its corner points to achieve the best fit of the mesh to each subsequent frame. We find that this method allows us to fit 3-d metrics to a high degree of accuracy.

ROBUST MOTION ESTIMATION USING SPATIAL GABOR FILTERS

E. Bruno and D. Pellerin, Laboratoire des Images et des Signaux LIS-INPG

This paper presents a new algorithm for motion estimation. It combines Gabor filter decomposition and robust least squares estimation in a multiresolution framework. Spatial Gabor filter bank provides a multichannel decomposition of frame sequence. Then, applying the brightness constancy constraint on each channel between two consecutive frames, we obtain an overdetermined system of velocity equations at each pixel. In order to be robust to outliers, this overdetermined system is solved using a robust least squares technique. We have used a multiresolution framework in order to manage large and small displacements. Performances of our algorithm are tested on synthetic and real sequences, and are compared with other techniques.

JOINT SOURCE AND CHANNEL RATE ALLOCATION FOR VIDEO TRANSMISSION OVER ERASURE CHANNELS

X. Henocq, C. Guillemot and F. Le Leannec, INRIA/IRISA Campus, France

This paper presents a joint source and channel rate allocation algorithm for video transmission over erasure channels. The approach relies on unequal error protection (UEP) of the video

stream, in order to minimize the received signal distortion. It takes into account both the frames and the channel characteristics. The loss behavior is approximated by a two states Markov model. The algorithm is incorporated in a H.263 version 2 compliant encoder. The encoder supports in addition a TCP-compatible (TCPC) congestion control mechanism and adapts the coding mode of each macro-block (MB) to the network loss behavior. Rate-distortion models are introduced in order to reduce the computational cost. A new payload format is also described for the transport of the streams over RTP. The overall approach, compared against equal error protection and against FEC -forward error correction- adapted to the channel only, leads to improved PSNR performance, with a more stable video quality.

MOTION FIELD ESTIMATION BY COMBINED VECTOR RATIONAL AND BILINEAR INTERPOLATION FOR MPEG-2 ERROR CONCEALMENT

S. Tsekeridou and I. Pitas, Aristotle University of Thessaloniki, Greece
F. Alaya Cheikh and M. Gabbouj, Tampere University of Technology, Finland

The combined use of bilinear and vector rational interpolation for motion field estimation of erroneously received MPEG-2 video bitstreams is reported. Motion vector rational interpolation is capable of adapting its behaviour with respect to neighbouring motion information. Bilinear interpolation, operating on a finer interpolation grid, takes neighbouring spatial correlations into account. Thus, the combined method is expected to lead to improved motion estimates of the lost motion data and thus satisfactory concealment. It additionally proves to be adequately fast for real-time applications (MPEG-2 decoder). Simulation results prove the efficiency of the method compared to similar approaches.

VELOCITY ESTIMATION AND MOVING OBJECT'S DETECTION BY USING WAVELET TRANSFORM AND PARALLEL KALMAN FILTER

Katsuya Kondo, Yasuo Konishi and Hiroyuki Ishigaki, Himeji Institute of Technology

We propose a novel method for velocity estimation and detection of a moving object in an image sequence. This purpose is achieved by using discrete wavelet transform and parallel bank of extended complex Kalman filter in the transform/spatio-temporal mixed-domain. In the mixed-domain, image sequence processing is replaced by 1-dimensional(1-D) complex signal processing. Then, trajectory signal with approximately constant speed and orientation is considered as a bunch of 1-D complex sinusoidal waves. By applying extended complex Kalman filter to each 1-D complex signal, object's velocity can be estimated. In addition, these estimates can be combined with depending on the energy of the subband images. Through some simulation results, it is shown that moving object's velocity is accurately estimated and the object is detected effectively.

IMPROVED COLOR CODING FOR SCALABLE 3D WAVELET VIDEO COMPRESSION

Béatrice Pesquet-Popescu, Marion Bénetière, Vincent Bottreau and Boris Felts, Laboratoires d'Electronique Philips

With the recent expansion of multimedia applications, video coding systems are expected to become highly scalable, that is to allow partial decoding of the compressed bit-stream, in order to adapt to heterogeneous networks and users capabilities. Encoding techniques based on subband/wavelet decompositions offer a natural hierarchical representation for still pictures and their high efficiency in progressively encoding images yields a scalable representation. Progressive encoding of video data represented by a 3D-subband decomposition was recently proposed as an extension of image coding techniques exploiting hierarchical dependencies between wavelet coefficients. In this paper we propose a new coding technique for the chrominance coefficients, which not only delivers a bit-stream with a higher degree of embedding, but also takes advantage of the dependencies between luminance and chrominance components to provide a more effective compression.

A DATA FUSION METHOD FOR IMPROVED MOTION ESTIMATION

A.M.Peacock, D.Renshaw, J.Hannah and P.Grant, the University of Edinburgh, UK

Previous work has shown that information from different artificial vision approaches to the same problem can be combined using Data Fusion methods to produce more robust results. Different techniques may also have different computational requirements, and there is often a trade off between algorithm complexity and accuracy. This paper reports the use of state and error predictors combined with data fusion methods to create vision systems with improved performance. Results for both synthetic and real examples are reported. These results show that substantial improvements can be made using this temporal-data fusion approach.

MULTI-RESOLUTION MESH-BASED MOTION ESTIMATION USING A ``BACKWARD IN FORWARD'' TRACKING METHOD

Gwenaëlle Marquant, Stéphane Pateux and Claude Labit affiliation, IRISA / INRIA, France

This paper introduces a mesh-based motion estimation scheme for image sequences and nodal motion vectors optimization by using a multi-resolution differential method. Because our final aim is mesh tracking throughout a video sequence, neither backward tracking nor forward tracking is well suited. The backward tracking provides good results when simply applied to two successive images, but it is damaged by the initial mesh which can not generally be considered as known on the last frame. Likewise, forward tracking suffers from an analysis/synthesis problem but it allows long time trackings. The motivation of our work is to take advantage of both forward tracking (which enables tracking) and backward tracking (for its efficiency) in a ``backward in forward" method. To illustrate the proposed method, first results concerning the motion compensation of a mesh are shown and compared to the usual forward approach for one single mesh level, with or without multiresolution.

ThuAmPO3
Exhibition Hall
DSP for Communications
Chair: M. Renfors ***Tampere University of Technology, Finland***

AN ADAPTIVE METHOD FOR CLOSED-LOOP TAD IN FDD MODE

Jinho Choi, Konkuk University, Korea

For better performance, various diversity techniques can be utilized in wireless communications. Recently, transmit antenna diversity (TAD) has been employed for 3rd generation (3G) code division multiple access (CDMA) systems. Performance can be greatly improved by using more transmit antennas. In this paper, we present an adaptive method for closed-loop TAD. It is shown that the adaptive method can efficiently increase the number of transmit antennas without signifcantly increasing the feedback bit rate.

DIGITAL FILTER DESIGN FOR I/Q IMBALANCE COMPENSATION

Markku Renfors and Mikko Valkama, Tampere University of Technology, Finland

All communication receiver structures utilizing I/Q signal processing share the problem of matching the amplitudes and phases of the I- and Q-branches. In practise, the imbalances are unavoidable and this results in finite and in many cases insufficient attenuation of the image frequency band. Without any additional image rejection, this causes interference and needs to be compensated. We carry out the general signal and imbalance analysis of an I/Q processing based receiver structure and discuss the possibilities of digital imbalance compensation in cases where the dominating part of the imbalance is known in advance. A novel method for fixed imbalance compensation in a wideband approximative quadrature sampling receiver is derived and the performance of the proposed solution is illustrated through an example design.

ADAPTIVE TRAINED NEURAL NETWORKS FOR TRAFFIC PREDICTION OF VBR MPEG-2 VIDEO SOURCES

Nikolaos D. Doulamis, Anastasios D. Doulamis, and Stefanos D. Kollias, National Technical University of Athens, Greece

In this paper, a novel adaptively trained neural network structure is proposed for traffic rate prediction in MPEG-2 video sequences. The algorithm automatically adapts the network weights in case of highly rate changes taking into account both the current and the previous traffic samples. Application of the proposed scheme to real life sequences has shown a superior performance compared to other traditional methods.

ADAPTIVE CHANNEL EQUALIZATION USING CLASSIFICATION TREES

Taneli Haverinen, Arto Kantsila, Mikko Lehtokangas and Jukka Saarinen, Tampere University of Technology, Finland

This paper focuses on adaptive equalization of binary signals in a baseband digital telecommunication system. Equalization and detection are considered as a classification problem. Fixed-length sequences of received observations form a multidimensional signal space, which can be partitioned using the proposed classification tree algorithm. Top-down approach is used in tree induction, and splitting is done based on information gain criterion. Overfitting is avoided by utilizing a pruning algorithm. The advantages of this method are its simplicity and straightforwardness and thereby the reduction of computational complexity compared to other well performing equalizers. Experimental results and comparison with a cascade-correlation trained multilayer perceptron neural network equalizer are given.

MULTIUSER DETECTION IN MULTIPATH NON-GAUSSIAN CHANNELS

H. Vincent Poor, Princeton University, USA
Mario Tanda, University of Naples, Italy

This paper addresses the problem of multiuser detection in multipath fading code-division multiple-access (CDMA) channels with non-Gaussian noise. A robust multipath decorrelator is proposed to mitigate the multiple-access interference and the multipath interference, and to ameliorate non-Gaussian ambient noise. The performance of the proposed multiuser detector is assessed via computer simulations and compared with that of the linear multipath decorrelator and with that of the maximum-likelihood detector designed under the assumption of perfect knowledge of the channel fading coefficients.

HIGH PERFORMANCE ESTIMATION ALGORITHM FOR PSK FREQUENCY RECOVERY

Gabriella Olmo, Davide Bosetto and Letizia Lo Presti, Politecnico di Torino, Italy

In this paper, a novel algorithm for carrier frequency estimation is described, whose basic principle is to exploit an iterative mechanism to refine the estimate by progressively reduce the noise subspace. This is accomplished by means of an external denoising system, fed by the master estimation algorithm, in order to feed the basic estimation block with enhanced data at next iteration. In this paper, a well known algorithm for carrier frequency estimation, the Luise and Reggiannini scheme, is used as a master estimator within an iterative scheme, and a proper external denoising system is designed. The obtained results show that the scheme outperforms the master estimator as for both the threshold level and the estimation range, with comparable computational complexity.

DIGITAL DOWN CONVERSION IN SOFTWARE RADIO TERMINALS

Michael Loehning, Tim Hentschel, and Gerhard Fettweis, Dresden University of Technology, Germany

The idea of software radio requires an expansion of digital signal processing towards the antenna. Hence, for converting the received signal to baseband, the need of efficient high speed digital down converters arises. In [1] digital down conversion was identified as one of the 'critical functionalities' because it has to run at a relative high sample rate, and has to provide high resolution. The common approach for digital down conversion (DDC) is the so called ROM table approach where the samples of the input signal are multiplied with amplitude values of the sine- and cosine-function stored in ROM. To achieve high resolution this technique requires a large look-up table which means large chip area, high power consumption, lower speed, and increased costs. In this paper a CORDIC-based digital down converter is described. It enables to reduce the size of the look-up table considerably. Additionally to previous publications, this paper provides an overall worst case quantization error estimation that facilitates the dimensioning of the CORDIC-DDC.

ON-LINE EVALUATION OF EQUALIZER PERFORMANCE BASED ON HIGH-ORDER STATISTICS

J.B. Destro Filho and J. M. Travassos Romano, State University of Campinas, Brazil
J.P. Breda Destro, CTA/IEAv/EIN-A S.J. Campos, Brazil

In this paper, a simple recursive method is developed in order to estimate the bit-error-rate (BER) at the output of an adaptive equalizer. This is accomplished in two steps. Firstly, a theoretical procedure for detecting equalization errors is derived. This test is then used with an adaptive blind channel identification scheme, based on high-order statistics theory, in order to derive the final adaptive BER estimator. Simulation results point out that our method provides on-line and reliable BER estimation with a low computational burden for several linear channels models.

MAJORITY SELECTION AND BLOCK-BASED SELECTION DIVERSITY RECEPTION METHODS FOR 8K DVB-T IN A MOBILE ENVIRONMENT

Jukka Rinne, Tampere University of Technology, Finland

Some comparative results on simple diversity combining schemes are presented. These combining schemes are intended to be used with Orthogonal Frequency Division Multiplexing (OFDM) systems in mobile channels, i.e., multipath channels with time selective fading. The methods are typically simple to implement in the existing Digital Video Broadcasting - Terrestrial (DVB-T) receivers. Both analytical and numerical results of the performance of these methods are given.

A RECURSIVE ALGORITHM FOR MAXIMUM LIKELIHOOD TREND ESTIMATION IN EXPONENTIAL NOISE AND ITS APPLICATION TO IP TELEPHONY

Tõnu Trump, Ericsson Radio Systems AB

A recursive algorithm for maximum likelihood estimation of clock skew i.e. difference between the transmitter and receiver clock frequencies in telephony over asynchronous packet switched networks e.g. Internet telephony, is derived. The underlying maximum likelihood method is based on modeling the clock skew as linear trend and the network delay jitter as an i.i.d. random process with exponential distribution. The memory requirements of the proposed algorithm are data dependent and are investigated by simulations using both computer generated and recorded data sets.

STATISTICAL ANALYSIS OF MULTIDIMENSIONAL FILTERS DRIVEN NON/GAUSSIAN NON/WHITE NOISES

Vladimir A. Kazakov and Ruben Hernandez P. Department of telecommunications, High School of Mechanics and Electric Of the National Polytechnical Institute of Mexico

The problem of the statistical analysis of the multidimensional filters is considered on the base of kinetic equations for the non-markovian stochastic processes. The kinetic coefficients are calculated using the convolution integral instead of the system of differential equations. The advantage of this method is a reduction of the dimension of the problem. Two examples given and some results of a statistical simulation are presented.

COMPLEX DIGITAL OSCILLATOR WITH ABSOLUTE PERIODICITY

Florean Curticapean and Jarkko Niittylahti, Tampere University of Technology, Finland

In this paper, a novel control method for the coupled form complex oscillator is proposed. In this recursive structure, the roundoff errors will accumulate in time and degrade the generated sinusoids due to the finite word-length effects. The introduced control method reduces the accumulated error and forces the output sequence to be periodic. By resetting the oscillator after each discrete period, the generated sequence is made periodic and the accumulated error is canceled. Using one or two extra pair of coefficients periodically, the samples are precisely computed, and as result the accumulated error over one discrete period increases much more slowly. The presented control method allows us to design a complex digital oscillator featuring absolute periodicity, low spectral impurities, and high number of samples per cycle.

WAST3G: A COMPREHENSIVE WCDMA SIMULATION TOOL FOR THIRD-GENERATION MOBILE COMMUNICATIONS

C.B. Ribeiro and A.D. Santana Jr., University Federal do Rio de Janeiro, Brazil

In this paper we describe a software tool (Wideband CDMA Simulation Tool for Third Generation Mobile Communications, WAST3G) that simulates a mobile communication system

based on UMTS. It can be used as a test-bed for new algorithms and methods for deconvolution, demodulation, despreading, equalization and detection, aiming on 3G systems performance and technics. It can also be used as a learning tool for the operation of wireless communications systems in general, and 3G systems in particular, because it implements all necessary steps for a mobile station that was ``turned off" to achieve communication, including synchronization and system registration procedures. All these features make WAST3G a very powerful tool for academic and industrial purposes.

TEAGER-KAISER OPERATOR BASED FILTERING

Ridha Hamila, Markku Renfors and Taneli Haverinen, Tampere University of Technology, Finland
Guðni Gunnarsson, Nokia Networks, Finland

In this contribution, we propose a new filtering and denoising technique for one-dimentional signals based on the nonlinear quadratic Teager-Kaiser operator. This technique is a threshold 'energy' based approach where outliers are first detected and then replaced by their estimated values. The proposed technique performs better compared to the alpha-trimmed and running mean filter, also with moderate complexity. In particular, we present an application of the proposed real-time filtering approach for postprocessing the measurements of time delay signals in a mobile positioning system highly affected by the propagation environment.

ThuPmSS1
Small Audiotorium
Blind Methods in Communications
Chair: A.-J. van der Veen ***Technical University of Delft, The Netherlands***

14:00

LOW COST ADAPTIVE ALGORITHM FOR BLIND CHANNEL IDENTIFICATION AND SYMBOL ESTIMATION

Florence Alberge, Pierre Duhamel and Mila Nikolova, ENST/TSI, France

A recursive/adaptive least squares algorithm (MLRA) is proposed to solve the joint blind channel identification and blind symbol estimation problem. It is based on Deterministic Maximum Likelihood methods (DML) which are pertinent in this field. We prove that when the MLRA converges then it converges towards the global minimum. The MLRA is able to track variations of the system by the introduction of an exponential forgetting factor in the DML criterion. The link between the adaptive algorithm and a Soft Decision Feedback Equalizer is emphasized. Update strategies of the filters can be either of a least squares type or of a stochastic gradient type. Both of them are derived in the paper. Numerical simulations show that the simplifications involved result in small degradations on the performances.

14:20

A FAST ALGORITHM FOR CONDITIONAL MAXIMUM LIKELIHOOD BLIND IDENTIFICATION OF SIMO/MIMO FIR SYSTEMS

K. Abed-Meraim and P. Duhamel, ENST (Telecom Paris), France
Y, Hua, University of Melbourne, Australia
M. Z. Ikram, Georgia Institute of Technology, USA

Blind system identification is important for a wide range of applications. The conditional maximum likelihood (CML) method is one of the most effective ones recently developed for blind system identification. In particular, the CML method is statistically most efficient at relatively high signal-to-noise ratios (SNR). Unfortunately, the original implementation of the CML method via the two-step maximum likelihood (TSML) algorithm is computationally too expensive. In this paper, a computationally attractive implementation of the TSML algorithm based on the Cholesky decomposition is proposed. This leads to a new fast TSML (FTSML) algorithm that has a linear complexity, i.e., O(N) flops as compared to O(N^3) flops of the original implementation, N being the data size. In a second part of the paper, we generalize the FTSML algorithm from the single-source case to the multiple-sources case.

14:40

ON AR EQUALIZATION WITH THE CONSTANT MODULUS CRITERION

Azz'edine Touzni, Lang Tong, Ra'ul Casas and C. Richard Johnson, Cornell University

This work studies the CM criterion applied to equalization with AR channel receivers motivated by recently proposed blind IIR algorithms, but may also be used for initializing a blind adaptive DFE. Blind IIR equalization is of practical interest for two main reasons: the IIR structure not only provides a parsimonious representation of a linear receiver, but may also be used for switching to DFE mode. We begin by showing that, unlike the FIR case, AR-CM receivers are equivalent to Wiener receivers for Gaussian sources. For sub-Gaussian input signals, however, due to the requirements of causality and stability of the receiver, characterization of CM solutions appears to be a complex problem. Thus, for sub-Gaussian sources, analysis is restricted to the special case of a MA(1) channel and AR(1) equalizer. Nevertheless, this study provides insight into the properties of AR-CM receivers by making the connection to the popular IIR-MMSE receivers.

15:00

GLOBALLY CONVERGENT ALGORITHMS FOR BLIND SOURCE SEPARATION

Constantinos B. Papadias, Bell Labs Research, USA

We present a novel class of adaptive algorithms for the blind separation of non-Gaussian mutually independent source signals that can be modeled as independent identically distributed (i.i.d.) discrete random processes. The signals are assumed to be transmitted through a m x p narrow-band (instantaneous linear mixture) channel. The original algorithm, called the Multi-User Kurtosis (MUK) algorithm was first presented in [1] and was derived from a set of conditions that were previously found to be necessary and sufficient for the recovery of all the sources [2]. The analysis presented in [1], [3] has shown that the MUK algorithm is globally convergent to a zero-forcing - ZF (decorrelating) solution both in the absence of noise and in the presence of additive white Gaussian noise (AWGN), provided that the received signals are perfectly pre-whitened. In this paper, we propose other constant-modulus (CM) type variants of the MUK algorithm. These inherit the global convergence behavior of the MUK due to its deflation structure and moreover, they allow for increased convergence speed. These variants of the MUK are particularly useful in cases where the number of received signal snapshots is limited, such as in wireless communication applications.

15:50

A "UNIMODAL" BLIND EQUALIZATION CRITERION

Phillip A. Regalia and Eleftherios Kofidis, Institut National des Telecommunications, France

By reexamining some initialization strategies for blind equalizers, we develop an approximation to a common contrast function for equalization which enjoys a ``unimodal" property: the solution obtained from the modified criterion is unique to within standard scale factor ambiguity, and

moreover yields a perfect equalizer whenever such is attainable. In the more realistic situation in which perfect equalization is unattainable, the modified criterion yields a good approximation which is quantified herein, and is therefore justified as an initialization strategy. An on-line interpretation leads to an adaptive Volterra filter followed by a tensor product approximation which furnishes the coefficients of a linear equalizer, and ties in with some earlier work on equalizer design restricted to finite data sets and Volterra kernel approximations.

16:10

IDENTIFIABILITY CONDITIONS FOR DETERMINISTIC BLIND BEAMFORMING IN THE PRESENCE OF INCOHERENT MULTIPATH WITH SMALL DELAY SPREAD

Nicholas D. Sidiropoulos and Xiangqian Liu, University of Minnesota, USA

In a recent paper, van der Veen has developed a baseband-equivalent data model for antenna array reception of multiple sources subject to incoherent multipath with small delay spread. Capitalizing on the structure of this model, van der Veen proposed a SVD/EVD-based blind beamforming algorithm. The starting point of our work is the realization that the above model can be viewed as a low-rank (trilinear) decomposition of a three-way received data array. Using a basic uniqueness result for low-rank decomposition of three-way arrays, data smoothing, and two new Lemmas pertaining to the so-called k-rank of certain structured matrices, we derive herein several interesting identifiability results for deterministic blind beamforming in the presence of incoherent multipath with small delay spread. The utility of the core results also extends beyond the application scope of this paper.

16:30

BLIND EQUALIZATION OF SPARSE CHANNELS USING ANTENNAS DIVERSITY

Amir Leshem and Alle-Jan van der Veen, Delft University of Technology, The Netherlands

In this paper we consider the blind identifiability of multichannel systems when we know a-priori a bound on the number of non-zero coefficients in each channel. This is interesting for specular multi-path channels. We show that in this case, previously derived identifiability conditions are too strong. We demonstrate how to use the a-priori knowledge to weaken these conditions. We also propose a method to estimate the channel responses as well as the signals. Our method is based on a frequency domain least squares estimation of the channel parameters combined with conditional maximum likelihood estimator for the signals. We also present simulation results.

16:50

ON BLIND FRACTIONALLY-SPACED EQUALIZATION USING ANALYTICAL CONSTANT MODULUS

Gianpiero Panci, Gaetano Scarano and Giovanni Jacovitti, Università di Roma ``La Sapienza'', Italy

In this contribution we discuss the applicability of the so-called analytical constant modulus approach to the case of blind fractionally-spaced equalization of communication channels. We demonstrate that such an approach cannot be pursued when the equalizer is straightforwardly parameterized by the samples of its FIR impulse response. Then, we obtain a new parameterization of the equalizer, exploiting the modulation redundancy associated to the excess bandwidth usually available in fractionally sampled PAM and QAM signals. It is pointed out that the FIR equalizer belongs to properly defined linear subspace, related to the geometrical structure of a suitable matrix build up using time-varying correlations of the received signal. Exploiting this new parameterization, we finally show that the analytical constant modulus approach is successfully applicable in blind fractionally-spaced equalization.

17:10

ASYMPTOTIC ANALYSIS OF BLIND CYCLIC CORRELATION BASED SYMBOL RATE ESTIMATION

P. Ciblat and P. Loubaton, University of Marne-la-Vallee
E. Serpedin, Texas A.M. University
G.B. Giannakis, University of Minnesota

We consider symbol rate estimation of an unknown signal linearly modulated by a sequence of symbols. We rely on the received signal is cyclostationarity, and consider an existing estimator obtained by maximizing in the cyclic domain a (possibly weighted) sum of modulus squares of cyclic correlation estimates. Although widely used, this estimate seems not to have been studied rigorously when the number of samples N is large. In this paper, we study rigorously the asymptotic behavior of this estimate. We establish consistency and asymptotic normality of the estimate, prove that its convergence rate is $N^{3/2}$, and calculate in closed form its asymptotic variance. The obtained formula allows us to discuss in relevant way on the influence of the number of estimated cyclic correlation coefficients to take into account in the cost function to maximize.

ThuPmOR1
Hall A
Statistical Signal Processing and DOA
Chair: J. Joutsensalo ***Tampere University of Technology, Finland***

14:00

WIDE-BAND SIGNAL PARAMETER ESTIMATION BASED ON HIGHER-ORDER STATISTICS

S. Bourennane, B. Costa, SDEM Quartier, France
M. Montanari, University of Pisa, Italy
F. Gini and E. Dalle Mese, Institut Fresnel /E.N.S.P.M D., France

In this paper we develop an algorithm to improve the accuracy of the wideband signal parameter estimation. It is well known that in the presence of an unknown noise, these estimates may be grossly inaccurate. The proposed algorithm uses both the fourth order cumulant for the suppression of the gaussian noise, the transformation matrices for estimating the coherent cumulant matrix and a noneigenvector algorithm for the characterization of the sources. We show that the performances of bearing estimation algorithm improve substantially when the proposed algorithm is used. This method is tested on simulated data and its performances are clearly pointed out.

14:20

A DESIGN METHOD FOR SMALL SENSOR ARRAYS IN ANGLE OF ARRIVAL ESTIMATION

T. Saarelainen and J. Yli-Hietanen, Tampere University of Technology, Finland

The use of small sensor arrays in modern signal processing systems has recently become more common due to the increase in computational processing power and interest in intelligent sensing and surveillance. However, not much information is available on the design of small sensor arrays having arbitrary geometry, that effectively can accomplish these tasks. In this paper we address the problem of designing such small sensor array systems for angle of arrival (AOA) estimation algorithms. Two different cost functions are derived and their applicability is demonstrated in simulation. The accuracy of the AOA estimates is also studied for two different array configurations.

14:40

RESOLUTION ENHANCEMENT OF SPATIAL SPECTRUM BY A VIRTUALLY EXPANDED ARRAY

Young-Su Kim and Hung-Ryong Kang, ETRI, Korea
Young-Soo Kim, KyungHee University, Korea
Han-Kyu Park., Yonsei University, Korea

In this paper, we propose a resolution enhancement method for estimating direction-of-arrival (DOA) of narrowband incoherent signals incident on a general array. The resolution of DOA algorithm is dependent on the aperture size of antenna array. But it is very impractical to increase the physical size of antenna array in real environment. Therefore we propose the method that increases the aperture size by virtually expanding the sensor spacing of original antenna array and then construct the steering matrix of the virtual array using the proper transformation matrix. Superior resolution capabilities achieved with this method are shown by simulation results in comparison with the standard MUSIC for incoherent signals incident on a uniform circular array.

15:00

DETECTION OF THE SOURCE NUMBER BY THE GERSCHGORIN DISKS

Olivier Caspary and Thierry Cecchin, CRAN, Nancy

The inclusion regions of the eigenvalues enable us to work out efficient criteria for the estimation of the number of sinusoids. To exploit those regions, it is necessary first to transform the covariance matrix. That is why we put forward a transformation based on an approximation of the eigenvalues and eigenvectors, so as to obtain the radii and the centers of the Gerschgorin disks, the disks being the studied inclusion regions. We show that the introduction of information concerning the radii and the centers in the detection criteria improves their performances tremendously. A new criterion using the Euclidean distance (and called GDEdist) is also suggested.

15:50

A NOVEL APPROACH TO ARRAY STEERING VECTOR ESTIMATE IMPROVEMENT

Mehrzad Biguesh, Sharif University of Technology, Iran
Benoit Champagne, McGill University, Canada
Shahrokh Valaee, Tarbiat Modares University, Iran

In this paper, we present a method for estimating the signal sources steering vector using an arbitrary planar array with omnidirectional elements. The proposed method improves the initial estimation of the signal steering vector in two steps. In the first step of this algorithm we minimize of the distance between the steering vector and the signal subspace. The second step improves the estimation of the first step using a defined cost function which is based on a structural criterion for signal steering vector. Simulation results show the capability of the

proposed signal steering vector estimate improvement.

16:10

COMPUTATIONALLY EFFICIENT DOA ESTIMATION OF A SCATTERED SOURCE

Petre Stoica, Uppsala University, Sweden
Olivier Besson, Ensica

The problem of Direction-Of-Arrival (DOA) estimation in the presence of local scatterers using a uniform linear array (ULA) of sensors is addressed. We consider two models depending on whether the form of the azimuthal power distribution is explicitly known or not. For both models, the block-diagonal structure of the associated Fisher Information Matrix (FIM) is exploited to decouple the estimation of the DOA from that of the other model parameters. An asymptotically efficient Maximum Likelihood (ML) DOA estimator is derived which entails solving a 1-D minimization problem only. Furthermore, the 1-D criterion can be expressed as a simple Fourier Transform. Numerical simulations illustrate the fact that our computationally very simple DOA estimators have an accuracy close to the Cram\'{e}r-Rao bound in a wide range of scenarios.

16:30

A NEW APPROACH FOR TRACKING MOBILES WITH LOCAL SCATTERING MODELING

Hiroyuki Tsuji, Ministry of Posts and Telecommunications, Japan
Magnus Jansson, Royal Institute of Technology, Sweden
Akira Sano, Keio University, Japan
Mostafa Kaveh, University of Minnesota, USA

A new approach for tracking mobiles in a multipath environment is presented. In this paper, we introduce the local scattering model into the tracking algorithm, which utilizes the linear approximation model, to improve the performance of DOA estimate in a multipath environment. The point of our method is to estimate the DOAs and the angular velocities of the mobiles simultaneously by minimizing a cost function, which is consist of the observed data. Moreover, the cost function takes into account of the scattering factor to cope with the multipath signals. The proposed method can estimate the DOA precisely without increasing much computation load. The efficiency of the method is shown through simulation results.

16:50

USING LINEAR AND NONLINEAR DECOMPOSITIONS IN STATE- SPACE FOR CALCULATING MINIMAL DIMENSION EMBEDDING OF CHAOTIC ATTRACTOR

Alexander M. Krot and Helena B. Minervina, National Academy of Sciences of Belarus, Belarus

The theoretic substantiation of a locally topological method for defining a minimum attractor embedding dimension on the basis of linear and nonlinear decompositions in state-space of a dynamic system is supposed. The computer confirmation of the theoretical results is presented.

17:10

GENERALIZED ALMOST-CYCLOSTATIONARY SIGNALS ON MULTIPATH DOPPLER CHANNELS

Luciano Izzo and Antonio Napolitano, Universita` di Napoli Federico II, Italy

In this paper, the effects of multipath Doppler channels on the very recently introduced generalized almost-cyclostationary signals are analyzed. Multipath Doppler channels generate several replicas of the input signal, each characterized by a different complex amplitude, delay, time-scaling factor, and Doppler shift. It is shown that such channels can be modeled as linear almost-periodically time-variant or as linear time-variant systems, depending on the length of the collect time adopted in the measurements.

ThuPmOR2
Room 200
Channel Estimation
Chair: V. Koivunen *Helsinki University of Technology, Finland*

14:00

JOINT CARRIER PHASE AND FREQUENCY OFFSET TRACKING IN OFDM SYSTEMS

Catharina Carlemalm, H.Vincent Poor, Princeton University, USA

In this paper, we study an OFDM system subject to carrier phase noise and frequency offset. Contrary to many previous approaches, we assume that the carrier phase and frequency offset are drifting. A scheme for joint estimation of the transmitted sequence and tracking of the carrier phase and frequency offset is prsented. The proposed iterative scheme is based on the ECM algorithm - an extension of the EM algorithm - and it performs the joint estimation in a unified manner by, as it turns out, interconnecting an Extended Kalman Smoother with Yule-Walker type equations.

14:20

CO-CHANNEL INTERFERENCE REJECTION USING CLUSTERING TECHNIQUES AND FUZZY INFERENCE

Yannis Kopsinis and Sergios Theodoridis, University of Athens, Greece

This paper presents two novel techniques for channel equalization, in the presence of Co-Channel Interference. Both techniques belong to the class of Cluster Based Sequence Equalizers. The first proposes a new distance metric, and the second incorporates a Fuzzy System philosophy.

14:40

INTERFERENCE-RESISTANT LPTV-MMSE EQUALIZATION

Giacinto Gelli and Francesco Verde, Università degli Studi Federico II di Napoli, Italy

The problem of recovering a digital communication signal distorted by a linear time-invariant channel and contaminated by severe co-channel or adjacent-channel digital interference is addressed in this paper. The proposed linear periodically time-varying (LPTV) receiver jointly performs channel equalization and interference suppression, without requiring explicit knowledge or estimation of the interfering channel. Simulation results confirm the effectiveness of the proposed technique, whose performance exhibit a remarkable robustness with respect to varying interference power level.

15:00

OPTIMAL TRAINING SEQUENCES FOR TDMA SYSTEMS

Markus Rupp Bell-Labs, Lucent Technologies, The Netherlands

This paper deals with the problem of finding optimal training sequences for adaptive equalizers in TDMA systems. Such sequences give the equalizer a good initial value, are required for symbol-timing recovery and can provide a small but robust amount of information by utilizing a small set of codes with strong discrimination power. While every single task has been dealt with in literature, the combination of all three of them is a new problem that requires smart search strategies to explore a huge space of possibilities. On the example of a local wireless loop design, it is shown how the search problem can be reduced.

15:50

TWO-DIMENSIONAL PILOT-SYMBOL-AIDED FREQUENCY OFFSET TRACKING IN ORTHOGONAL MULTICARRIER WIRELESS SYSTEMS

M. Julia Fernandez-Getino Garcia and Jose M. Paez-Borrallo, Universidad Politecnica de Madrid, Spain
Ove Edfors, Lund University, Sweden

Pilot-Symbol Assisted-Modulation (PSAM) is employed in coherent OFDM for channel estimation and it is based on inserting known symbols in the time-frequency grid. We show how this two-dimensional embedded signalling can be used for frequency offset tracking. The ML estimator has been derived, but though optimal, may not be practical due to its complexity. Hence, a frequency-domain low-complex estimator has been proposed, based on only the samples at pilot positions. Also, the stability of the generated estimate is improved significantly by averaging over a few number of OFDM symbols. This estimator is efficient against AWGN and also suitable for use in time dispersive channels.

16:10

MONTE CARLO SAMPLING METHODS FOR ADAPTIVE CHANNEL ESTIMATION AND DETECTION OVER A RAYLEIGH FADING CHANNEL

Jayesh Kotecha and Petar Djuric, State University of New York at Stony Brook, USA

A joint detection and adaptive estimation scheme is presented for a Rayleigh fading channel. The channel is modeled as an autoregressive process and the communication system is modeled as a dynamic state space model. A new filtering method called hybrid Monte Carlo-recursive identification is used, which combines sampling based filters with classical recursive identification methods to estimate the transmitted symbol sequence and the channel fading.

16:30

NEURAL SEQUENCE DETECTOR FOR DIGITAL EQUALIZATION

Elio D. Di Claudio, Raffaele Parisi and Gianni Orlandi, University of Rome "La Sapienza", Italy

In this paper a new approach to the equalization of digital transmission channels is introduced and described. The proposed solution makes use of a fast neural architecture, coupled with a innovative error functional, and is able to perform the equalization task in a Viterbi-like fashion applied to a Decision Feedback architecture for the purpose of improving the resitance to imperfect knowledge of the channel and interference. Performance comparisons with standard techniques for different channels demonstrate the validity of the proposed approach, expecially when the data model departs from assumptions and the computational cost is a critical issue.

16:50

PERFORMANCE STUDY OF DD ML PHASE ESTIMATORS FOR DS-CDMA COMMUNICATIONS SYSTEMS

Schumacher L., Aalborg University, Denmark
Vandendorpe L., Universite catholique de Louvain, Belgium

In the case of Data-Aided (DA) Maximum-Likelihood (ML) phase estimators operating in Direct Sequence-Code Division Multiple Access (DS-CDMA) communication systems, it has been shown that a multiuser (MU) design of the estimator helped to define parameter estimators exhibiting a lower variance than those designed in the conventional, single-user (SU) way. The present paper applies the same MU design strategy in the more realistic case of Decision-Directed (DD) estimators. The performance of these DD structures are derived from the DA variance expressions, assuming decisions to be correct and limiting the mitigation of the Multiple Access Interference (MAI) to its causal contribution.

17:10

EFFECT OF INTERLEAVING ON A GILBERT CHANNEL

T. Rancurel, D. Roviras and F. Castanié, Institut National Polytechnique de Toulouse, France
J. Conan, Ecole Polytechnique de Montréal, Canada

Many models based on Hidden Markov Models were developped to modelise errors burst in communication channels. In this paper, we employ the widely used Gilbert model to modelise error bursts and evaluate the effects of interleaving. We derive an expression for the transition matrix in the presence of an interleaver. Starting from an identified error model, it is possible to simulate easily the effects of any interleaving depth. This allows in a simply way the choice of an interleaving degree and an forward error correction (FEC) code.

ThuPmSS2
Hall B
Multimedia Watermarking
Chair: S. Suthaharan ***Tennesee State University, USA***

14:00

DIGITAL WATERMARKING OF VISUAL DATA: STATE OF THE ART AND NEW TRENDS

M.Barni, Universita' di Siena, Italy
F.Bartolini and A.Piva, Universita' di Firenze, Italy

The state of the art in digital watermarking of visual data is briefly reviewed. A communication perspective is adopted to identify the main issues in digital watermarking and to present the most common solutions adopted by the research community. We first consider the various approaches to watermark embedding and hiding. The communication channel is then taken into account, and the main research trends in attack modeling are overviewed. Particular attention is paid to watermark recovery due to the impact it has on the final reliability of the whole watermarking system. In the last part of the work the most crucial issues that still need to be solved are outlined, and some highlights on current and future research trends given.

14:40

OPTIMAL DETECTION OF MULTIPLICATIVE WATERMARKS

Job Oostveen, Ton Kalker and Jean-Paul Linnartz, Philips Research

We derive a watermark detector for images which are watermarked in a multiplicative way. Under the assumptions that the watermark coefficients are a known, binary valued sequence and that the original image coefficients are an i.i.d.\ random sequence from a Weibull distribution, we show that this watermark should be detected by raising the observations to the power beta before correlating them with the watermark (here beta is the parameter in the exponent of the Weibull probability density function). The approach is based on maximum-likelihood estimation of the embedding strength of the watermark. The result is illustrated by experiments and an extension to Gaussian distributed data is discussed.

15:00

A FRAMEWORK FOR OPTIMAL ADAPTIVE DCT WATERMARKS

Shelby Pereira and Thierry Pun, University of Geneva, Italy

In this paper we address the problem of robustly embedding 64 bits into an image while taking into account the HVS. The proposed method is general in that any mask can be adopted. The main advantage of the framework we present is that we demonstrate how to optimally embed a

watermark given the constraints imposed by the mask in the spatial domain. This is in sharp contrast with the bulk of publications which embed a watermark in the DCT domain and then truncate or modulate in the spatial domain in order to satisfy masking constraints. The problem with these approaches is that spatial domain truncation or modulation leads inevitably to the degradation of the watermark in the DCT domain. Results indicate that our proposed approach is robust against JPEG compression at a quality factor of 30% even for small images of size 64 by 64.

15:50

A ROBUST DIGITAL WATERMARK PROCEDURE FOR STILL IMAGES USING DCT PHASE MODULATION

Faisal Alurki and Russell Mersereau, Georgia Institute of Technology, USA

A digital watermark is a short sequence of information containing an owner identity or copyright information embedded in a way that is difficult to erase. We present a new oblivious digital watermarking method for copyright protection of still images. The technique is based on modifying the sign of a subset of low frequency DCT magnitude coefficients. The embedding process is adaptive and maintains a compromise between robustness and imperceptibility. A major advantage of the technique is its complete suppression of the noise due to the host image. The robustness of the technique to a number of standard image processing attacks is demonstrated using the criteria of the latest StirMark benchmark test.

16:10

WATERMARKING USING COMPLEX WAVELETS WITH RESISTANCE TO GEOMETRIC DISTORTION

Patrick Loo and Nick Kingsbury, University of Cambridge, UK

We present a watermarking algorithm in the complex wavelet domain and show why complex wavelets are better than real wavelets. Being an oversampled transform, the Complex Wavelet Transform requires special precautions during watermark embedding. We then model the watermarking process as a communcation channel and our results show that the complex wavelets domain has relatively higher capacity than both the spatial and the real wavelets domains. We will also outline a motion-based algorithm for image registration, which can help recovering watermarks from images suffering from geometric distortion.

16:30

A REGION-BASED TECHNIQUE FOR CHAOTIC IMAGE WATERMARKING

Athanasios Nikolaidis and Ioannis Pitas, Aristotle University of Thessaloniki

A novel method for embedding and detecting a chaotic watermark in the digital spatial image domain, based on segmenting the image and locating regions that are robust to several image manipulations, is presented in this paper. Each selected region is approximated by an ellipse. The watermark is embedded on its bounding rectangle. This representation proves robust under geometric attacks. The controlled lowpass nature of the chaotic watermark ensures its immunity to lowpass filtering and JPEG compression. Experimental results display the robustness of the method under several kinds of attacks, such as JPEG compression, mean and median filtering, scaling, cropping and rotation.

16:50

PUBLIC KEY WATERMARKING BY EIGENVECTORS OF LINEAR TRANSFORMS

Joachim Eggers, Jonathan Su and Bernd Girod, University of Erlangen-Nuremberg, Germany

Digital watermarks are signals embedded in multimedia data to allow copyright enforcement. In most watermarking schemes the embedded signal must be known for watermark detection, which leads to severe security risks. Van Schyndel et al. proposed a public watermark detection principle that works without explicit reference to the embedded signal. In this paper, extensions of this scheme are considered, and the applicability in practice is investigated. The new approaches are significantly less complex than the previously proposed scheme. Further, they are more robust against attacks via exhaustive search for the embedded watermark and against attacks that intend to confuse the public watermark detector. However, one drawback of all discussed schemes is the large signal length that is necessary for robust detection.

17:10

ACHIEVING IMPERCEPTIBILITY IN ALL-PASS WATERMARKING OF SPEECH SIGNALS

S. Utku Karaaslan and Tolga Ciloglu, Middle East Technical University,Turkey

A watermark is an ideally imperceptible signal embedded into another signal for carrying extra information. In this paper, we investigate the imperceptibility problem of a watermarking system developed by Yardimci, et. al. [1] and suggest two approaches in order to eliminate it. The system in question uses all-pass filters to embed data into a speech signal and introduces some artifacts in doing so. The two approaches we suggest in this paper remove these artifacts while keeping the watermark detectable and all the advantages of the method.

ThuPmSS3
Studio
Efficient Algorithms for Hardware Implementation
Chair: A. Willson ***University of California Los Angeles, USA***

14:00

DESIGN OF DIGITAL FILTERS WITH LOW POWER CONSUMPTION

Lars Wanhammar, Linkšping University Linkšping and The Norwegian University of Science and Technology

In this paper we discuss different methods to design digital filters with low power consumption. One efficient method is based on power supply voltage scaling where the excess speed that may be obtained using maximally fast recursive filter structures or non-recursive structures are exploited. We also discuss several techniques to increase and obtain the maximal sample rate for recursive structures based on frequency masking techniques and wave digital filters. The implementation approach is applicable to bit- parallel, digit-serial, and bit-serial arithmetic in standard CMOS processes.

14:40

ON HIGH-SPEED RECURSIVE DIGITAL FILTERS

Håkan Johansson, Linköping University, Sweden

High-speed recursive digital filters are of interest for applications focusing on high-speed as well as low power consumption because excess speed can be traded for low power consumption through the use of power supply voltage scaling techniques. This paper gives an overview of high-speed recursive digital filters. Two different techniques are mainly considered. The first one makes use of interconnected identical allpass subfilters whereas the second one employs frequency masking techniques.

15:00

VHDL-BASED IMPLEMENTATIONS OF AREA AND POWER EFFICIENT FILTER ARCHITECTURES

Ilkka Saastamoinen, Tapio Saramäki, and Olli Vainio, Tampere University of Technology, Finland

Digital signal processing operations, e.g., digital filters, are one important class of application in communication devices. A digital filtering algorithm can be implemented in various ways by selecting one architecture from the set of possible realizations. By choosing an advanced architecture notable advantages in both the silicon area and power dissipation can be achieved compared to the conventional direct-form realization. This paper focuses on interpolated finite

impulse response (interpolated FIR) filter and recursive running-sum (RRS) filter based architectures. The VHDL-based implementations prove that these advanced architectures are efficient when low-power or low-area characteristics are desired. Over 55 percent savings in the area and in the power dissipation were achieved when an FIR filter with a narrow transition band was implemented using these architectures.

15:50

JOINT MODULE SELECTION AND RETIMING WITH CARRY-SAVE REPRESENTAITON

Zhan Yu, Kei-Yong Khoo and Alan N. Willson, Jr. University of California, USA

Joint module selection and retiming is a powerful technique to optimize the implementation cost and the speed of a digital circuit. The use of carry-save signal representation is also a powerful technique in the high-speed implementation of arithmetic circuits. This work combines these two techniques to solve the joint module selection and retiming problem while allowing the use of carry-save representation. We formulate the problem as a mixed-integer linear programming (MILP) problem. Our algorithm, by allowing carry-save representation, can produce a wider range of solutions. In our experiments, our fastest implementation is 28% faster and our smallest implementation is 47% smaller, in comparison to solutions obtained using the previously known joint module selection and retiming technique.

16:10

ON STRUCTURE AND IMPLEMENTATION OF ALGORITHMS FOR CARRIER AND SYMBOL SYNCHRONIZATION IN SOFTWARE DEFINED RADIOS

Fred Harris, San Diego State University, USA
Chris Dick, Xilinx Inc., USA

Synchronization techniques based on DSP implementations are often digital emulations of their analog prototypes. Such solutions do not include structures and algorithms responsive to DSP system considerations and implementation strengths and weaknesses. We present a number of unconventional algorithms and structures used in carrier and timing recovery schemes. Multirate signal processing, polyphase filter structures, and CORDIC subsystems are at the heart of efficient first principle DSP based solutions to carrier recovery, matched filtering, timing recovery, and phase detection tasks required for synchronization.

16:30

ALGORITHMIC NOISE-TOLERANCE FOR LOW-POWER SIGNAL PROCESSING IN THE DEEP SUBMICRON ERA

Rajamohana Hegde and Naresh R. Shanbhag, University of Illinois, USA

In deep submicron (DSM) VLSI technology, deviations in node voltages due to DSM noise can lead to erroneous system outputs in VLSI implementations of DSP and communication algorithms degrading their performance in terms of signal-to-noise ratio (SNR) or bit-error-rate (BER). We present algorithmic noise-tolerance schemes for digital filtering to detect such errors in system output and mitigate their effect on the system performance. The errors in the system output are detected by employing a low-complexity prediction scheme. It is shown that, the proposed scheme improves the performance of the filtering algorithm by up to 10dB with less than 10% hardware overhead. It is also shown that the proposed scheme can be employed to achieve substantial energy savings with marginal degradation in performance by deliberately introducing errors in DSP hardware by {\em overscaling} the supply voltage.

16:50

H.263 VIDEO ENCODER IMPLEMENTATION ON A SCALABLE DSP SYSTEM

Pasi Kolinummi, Juha Särkijärvi, Timo Hämäläinen and Jukka Saarinen, Tampere University of Technology, Finland

A mapping and realization of H.263 video encoder for video conferencing applications are given using our DSP based multiprocessor system, called PARNEU. PARNEU system is built for computationally intensive applications like image and video processing as well as soft computing applications. PARNEU has flexible communication architecture and thus it allows different mapping possibilities. The presented data parallel mapping has low communication and memory requirements, which allows encoding of any of the five standard H.263 picture formats. With a prototype system using four ADSP-21062 DSPs, a real-time encoding is achieved with QCIF sized picture. Phase times for each step of H.263 encoder are presented.

17:10

FAST IMPLEMENTATION OF THE LMS ALGORITHM

Markus Rupp, Bell-Labs, Wireless Research Lab., The Netherlands

LMS algorithms are next to their numerically robust structure well suited for hardware implementation. In high data rate systems, however, their recursive formulation limits their performance. An architecture with more parallelism is desirable in which speed can be traded against chip area. This paper shows under which conditions such a parallel LMS structure is possible by means of particular training sequences. A new design method is proposed to generate training sequences with very distinct properties.

ThuPmPO1
Exhibition Hall
Wavelets and Filterbanks
Chair: U. Laine *Helsinki University of Technology, Finland*

REAL-TIME LINEAR TIME-FREQUENCY TRANSFORM

Osama Ahmed, King Fahd University of Petroleum and Minerals, Saudi Arabia

A real-time linear time-frequency transform is proposed based on SLTF transform [2]. The proposed transform has a small time delay and a much lower computational cost than other transforms. A fast algorithm to compute the biorthogonal function and the transform coefficients is presented. Finally, the characteristics of the proposed transform is compared with the original SLTF.

A REALIZATION OF WAVELET FILTER BANK WITH ADAPTIVE FILTER PARAMETERS

Damir Sersic, Faculty of EE and CS, Zagreb

In this paper, an efficient realization of the two-channel wavelet filter bank with adaptive filter parameters is proposed. Described time variant wavelet filter bank is more suitable for analysis of non-stationary signals then fixed banks. Basic convergence and regularity properties of the limit wavelet functions and scales are provided by fixed part of the filter bank. Variable part adapts to the analyzed signal. Proposed filter bank combines sub-band decomposition and parametric modeling. Realization is based on the lifting scheme, derived from a method of fixed wavelet filter bank design. Original Lagrange interpolation of samples in the time domain is modified to an approximation scheme that is recomputed at each step of decomposition. Adaptation criterion is calculated from wavelet coefficients, which is under some restrictions reproducible on the reconstruction side. Wavelet filter banks with adaptive filter parameters can outperform fixed banks in a number of applications.

A WAVELET-BASED TECHNIQUE FOR IMAGE REFINEMENT

M.A. Shcherbakov and W.E.Schegolev IVS, State University of Penza, Russia

In the paper we concentrate on a Wavelet-based technique of image refinement. Specifically, we are aimed to obtain the extended version of an original image while preserving its quality. The image to be expanded is given in a non-compressed gray-scale format. According to the method we propose, original image is, heuristically, supposed to be a smoothed version of an unknown sharp image we try to reach for. The former will be referred to as the superimage, in the sense that it is not just a resized version of the original, but is rendered in more details than the original. This is the case if we derive the details out of the given image. We have found it reasonable to look at the problem in the Wavelet background, as it comprises the image rescaling and feature extraction operations in their natural combination. We offer a number of approaches to the problem, which are supported by the relevant examples. Applications are, at least,

expected to arise in the remote sensing image refinement.

DESIGN OF CAUSAL STABLE IIR FILTER BANKS WITH POWERS-OF-TWO COEFFICIENTS

See-May Phoong and Bor-Ting Lin, National Taiwan University, Taiwan, R.O.C
Yuan-Pei Lin, National Chiao Tung University, Taiwan, R.O.C.

We introduce an efficient method for the design of causal stable IIR filter bank (FB) with Powers-of-two coefficients. Ladder structure is used to construct the IIR FBs with perfect reconstruction (PR). In the proposed method, FBs with real coefficients are first designed and an iterative procedure is then employed to discretize the coefficients. A sensitivity measure is introduced to determine the order of coefficients to be discretized. To ensure the stability of the IIR filters, the stability triangle is used. Even though the method is suboptimal, it avoids integer programing and yields satisfactory results.

MULTIPLIER-LESS LOW-DELAY FIR AND IIR WAVELET FILTER BANKS WITH SOPOT COEFFICIENTS

W. Liu, S. C. Chan and K. L. Ho, The University of Hong Kong, Hong Kong

In this paper, a new family of multiplier-less two-channel low-delay wavelet filter banks using the PR structure in [3] and the SOPOT(sum-of-powers-of-two) representation is proposed. In particular, the functions and in the structure are chosen as nonlinear-phase FIR and IIR filters, and the design of such multiplier-less filter banks is performed using the genetic algorithm. The proposed design method is very simple to use, and is sufficiently general to construct low-delay wavelet bases with flexible length, delay, and number of zero at (or 0) in their analysis filters. Several design examples are given to demonstrate the usefulness of the proposed method.

COEFFICIENT QUANTIZATION IN PERFECT-RECONSTRUCTION COSINE-MODULATED FILTER BANKS

Juuso Alhava and Ari Viholainen, Tampere University of Technology, Finland

This paper analyzes perfect-reconstruction (PR) cosine-modulated filter banks (CMFBs) from the quantization point of view. We study how the straightforward coefficient quantization affects on the performance of the transmultiplexer (TMUX) systems. In this paper, TMUX implementation is based on the structure of fast extended lapped transform (ELT). The quantized TMUX systems are analyzed in terms of stopband attenuation, intercarrier interference (ICI), and intersymbol interference (ISI).

PLANAR SHAPE REPRESENTATION BASED ON MULTIWAVELETS

Fernando Pérez Nava, Universidad de La Laguna, Tenerife, Spain
Antonio Falcón Martel, Universidad de Las Palmas de Gran Canaria, Spain

A technique is presented to construct a multiscale representation of planar contours based on the multiwavelet transform (MWT). To generate this multiwavelet description, a partial 1-D discrete multiwavelet transform (DMWT) is applied to the vertical and horizontal components of a length-parametrized planar curve. This multiscale representation descomposes the curve into different levels of resolution and allows to reconstruct it to a desired degree of aproximation. A comparison between the multiwavelet, wavelet and the elliptic fourier transform (EFT) is presented. The results show that typical objects are well represented by a small number of multiwavelet coefficients allowing for a compact object shape representation.

FILTER DESIGN FOR ADAPTIVE LIFTING SCHEMES

Tomasz P. Zielinski, Jacek Stepien and Krzysztof Duda, Technical University AGH, Poland

The paper addresses the problem of optimum design of predict and update filters for lifting scheme of the wavelet transform. It extends previous works of Baraniuk, Claypoole and Nowak, and presents general principles for building matrix equations for computation of arbitrary order lifting filters. Exemplary designs are presented in the paper. (2,2) Cohen-Daubechies-Feauveau and (4,4) Deslauriers-Debuc wavelet transforms can be obtained as special cases from the presented approach. Both predict-first and update-first schemes are taken into account.

2-D DIAMOND/QUADRANT FIR DIGITAL FILTER BANKS WITH SHARP RESPONSES

Min-Chi Kao and Sau-Gee Chen, National Chiao Tung University, Taiwan, R.O.C

This paper presents a novel design scheme, involving response sharpening process, for generating two- dimensional diamond/quadrant perfect-reconstruction (PR) linear-phase FIR filter banks with considerably reduced arithmetic operations, narrow transition bandwidth and good frequency selectivity. The scheme uses short 2-D Nyquist(M) subfilters, preferably multiplier-free ones, as basic building elements. Starting from a diamond/quadrant PR FIR filter bank based on the subfilters, the proposed scheme algebraically composes the building elements such that it refines the 2-D filter bank into a new 2-D filter bank with better frequency responses. The scheme can be successively applied till a satisfactory diamond/quadrant filter bank is obtained. The proposed scheme results in a tree-like multi-stage cascaded structure. It is composed of shared building elements with trivial coefficients and short wordlength lengths, which is suitable for finite-precision realization. The structure is highly modular, repetitive in all stages, and thus very suitable for VLSI implementation.

INTEGER-MODULATED FILTER BANKS PROVIDING PERFECT RECONSTRUCTION

Alfred Mertins, University of Wollongong
Tanja Karp, University of Mannheim
Joerg Kliewer, University of Kiel

In this paper, we extend the perfect reconstruction conditions known for cosine modulation to other, more general modulation schemes. The modified PR conditions provide additional degrees of freedom which can be utilized to design integer-modulated filter banks. Techniques for the design of prototypes and modulation sequences are presented.

DEGENERATE EIGENVALUES - A METHOD TO DESIGN ADAPTIVE DISCRETE TIME WAVELETS

Dan Stefanoiu and Ioan Tabus, Tampere University of Technology, Finland

This paper introduces a new method for the maximization of the coding gain of dyadic filter banks. We conjecture that in the generic case, the optimum impulse response is the maximum eigenvector of a certain Toeplitz matrix which is maximally degenerated, which ensures that the maximum eigenvector of the Toeplitz matrix has maximum possible multiplicity. We derive analyticaly the optimum FIR filter of order N=6, whereas closed form solutions have been available only up to order N=4. In an overhelming majority of the extensive simulations with AR input processes we found our conjecture holding true. What is therefore needed to add to our conjecture formulation in order to transform it into a theorem is the precise description of the ``non-generic'' conditions when it fails to hold.

COMPARISON OF OFDM AND WPM FOR FOURTH GENERATION BROADBAND WLAN

Imed Ben Dhaou and Hannu Tenhunen, Electronic System Design Laboratory, Sweden

In this paper, we propose a qualitative comparison between OFDM (Orthogonal Frequency Division Multiplexing), and WPM (Wavelet Packet Modulation). The comparison is done for two separate cases. Firstly, the efficiency of the two signaling systems will be compared. Secondly, the requirements for hardware implementation will be performed. From the performance and the VLSI implementation viewpoint, we found that WPM outperforms OFDM. However, the out-of-band radiation and peak-to-average-power for the case of OFDM is better compared to the WPM. The extensive simulation results, show that the average increase of peak-to-average-power is approximately 0.98dB compared to the OFDM. The increase of the adjacent channel power ratio channel is approximately 12.41dBc compared to the OFDM.

ON BLOCK-RECURSIVE LOGARITHMIC FILTERBANKS

Unto K. Laine, Helsinki University of Technology, Finland

Filterbanks with arbitrary time-frequency plane tilings can be efficiently realized by block-recursive structures which are based on vector ARMA models. The block-recursive realization leads to some approximation errors which depend among others on the block size and the block order of the model. A group of logarithmically spaced gammatone filters is used as a target system. The dependence of the design error on the choices of different design parameters are analysed. The resuls show that by proper choices of parameters the block-recursive structures are computationally efficient and of high quality.

A LINE SPECTRUM APPROACH FOR THE DESIGN OF OPTIMUM COMPACTION FIR FILTERS

Ioan Tabus, Riitta Niemisto and Jaakko Astola, Tampere University of Technology, Finland

In this paper we propose a new design method for optimal compaction gain FIR filters of a finite order. Starting from the odd polyphase component r(1),r(3),...,r(2K-1) of the input correlation sequence we find a suitable extension to an infinite sequence $r_e(k)$, such that the corresponding spectrum $S_e(\omega)$ is a line spectrum with K_1 lines. In quite general conditions it is possible to design the optimal compaction filter H(z) of order 2K-1 such that it has zeros at the line frequencies of $S_e(\omega)$ and $|H(e^{j\omega})|^2$ obeys the Nyquist(2) condition. The analytical method presented in [Kirac98] can be easily shown to be a particular case in this framework, but our method finds the optimal solution in some of the cases when [Kirac98] fails.

ThuPmPO2
Exhibition Hall
Parameter Estimation
Chair: A. Abdul Salam ***American University of Sharjah, UAE***

ROBUST ESTIMATION OF AN AR MULTI-CHANNEL MODEL BY T-DISTRIBUTION ASSUMPTION

Junibakti Sanubari, Satya Wacana University, Indonesia
Keiichi Tokuda, Nagaoya Institute of Technology, Japan

In this paper, we propose a new error criteria for determining the optimal multi-channel model system. The error criteria is based on assuming that the error is t-distributed with α degree of freedom. A small weighting factor is assigned for large amplitude signal portionparts and large weighting factor is used for small amplitude signal portion sections. By doing so, the effect of large amplitude signal to the estimated parameter is reduced. The simulation results show that the average of the obtained parameter by uisng small α t distribution is close to the ideal parameter than that when the conventional Gaussian assumption is applied. Furthermore, the standard deviation of the estimation result by using small α is smaller than that when $\alpha = \infty$ is utilized.

MEASURING RADAR STATISTICS BY USING KALMAN FILTER

Ahmed O. Abdul Salam, American University of Sharjah, United Arab Emirates

A simple approach for interrogating the residual sequence of Kalman filter is presented in this article. It has been found that the variance parameter of an input noise of radar can be successfully estimated by using a recursive formula. After employing some simulation examples, the suggested formula shows acceptable results and thus can be practically adopted to produce the required statistics.

AN AUGMENTED ITERATIVE METHOD FOR LARGE LINEAR TOEPLITZ SYSTEMS

Jacob Benesty, Mohan Sondhi and Tomas Gaensler,Bell Labs, Lucent Technologies, USA

Efficiently solving a large linear system of equations, $\textbf{A} \textbf{x} = \textbf{b}$, is still a challenging problem. Such a system appears in many applications in signal processing, especially in some problems in acoustics where we deal with very long impulse responses, i.e. $\textbf{x}$ is long. In this paper, we show how to efficiently use the so-called basic iterative algorithms when the matrix $\textbf{A}$ is Toeplitz, symmetric, and positive definite. We also propose an improved version that converges much faster than some other iterative methods. We present some simulations and compare the new method to the conjugate gradient algorithm.

ASYMPTOTIC NORMALITY OF SINUSOIDAL FREQUENCIES ESTIMATED BY SECOND-ORDER ALGORITHMS FOR MIXED SPECTRA TIME SERIES

Jean-Pierre Delmas, Institut National des Télécommunications, France

This paper addresses the asymptotic normal distribution of the sample covariance matrix of mixed spectra time series containing a sum of sinusoids and a linear stationary process. A new central limit theorem is proved for real or complex valued processes whose linear stationary process is possibly noncircular and not necessarily Gaussian. As an application of this result, the asymptotic normal distribution of any sinusoidal frequency estimator of such a time series based on second-order statistics is deduced. The case of the noise whitening is also considered in this general formulation. It is shown, in particular, that under mild assumptions, the asymptotic performance of most covariance-based frequency estimators is independent of the distribution of the noise.

SUPERVISED FREQUENCY CHANGE DETECTION USING MCMC METHODS

Eric Hitti, Christian Doncarli and Marie-Françoise Lucas, Institut de Recherche en Cybernétique de Nantes, France

Classical abrupt change detection needs to fix a threshold often difficult to be tuned. We propose to learn this value on a training set of signals, considering then the detection as a supervised classification problem. We describe here a bayesian approach. As a closed form of this bayesian learning is untractable, a stochastic simulation method (MCMC) is proposed leading to a good approximation of the posterior probability of change. The method is applied first to an academic problem for which an analytical solution exists and allows comparisons. Then a generalization of the composite hypothesis, corresponding to the most of real cases, is proposed and applied to abrupt frequency change detection in noisy multicomponent signals. Presented results show the influence of training set size on the performances of detection.

ESTIMATION AND SMOOTHING OF INSTANTANEOUS FREQUENCY OF NOISY NARROW BAND SIGNALS

Petr Tichavsky, Institute of Information Theory and Automation, Czech Republic
Peter Handel, Royal Institute of Technology, Sweden

Estimation of instantaneous frequency of a narrow-band noise corrupted signal from discrete time phase-only data is considered. By aid of Kalman filter theory, filter and smoothing estimates are derived and their equivalence to estimates obtained by the MFT algorithm is established, MFT being a practical algorithm for multiple frequency tracking and smoothing.

FREQUENCY ESTIMATION BY 1-BIT QUANTIZATION AND TABLE LOOK-UP PROCESSING

Tomas Andersson, Mikael Skoglund and Peter Händel, Royal Institute of Technology, Sweden

A method for fast frequency estimation by table lookup processing (FFETL) is presented. The estimation is based on data that has been quantized at one bit per sample, and all data processing is represented by a single table look-up operation, resulting in O(1)-complexity. The performance of the new method is compared with the proper Cram er-Rao bound for one bit quantized data, and the MLE for unquantized data by aid of Monte Carlo simulations. FFETL is shown to be (almost) sta- tistically e cient over a wide range of SNR, as encoun- tered in practical applications. Practical aspects such as implementational issues, and performance limitations due to quantization are discussed in some detail.

AMBIGUITIES IN STATISTICAL SIGNAL PROCESSING

Vasileios Lefkaditis and Athanassios Manikas, Imperial College of Science, Technology and Medicine, UK

This paper is concerned with the ambiguity problem which is of major concern in a number of statistical signal processing applications. Initially, the concept of 'hyperhelical' parameterisation is presented and then a compact algorithm for the classification and calculation of ambiguities is proposed, in conjunction with two application areas: the array processing and the harmonic retrieval problem. Furthermore, it is demonstrated that in the harmonic retrieval problem, under certain conditions, the sampling rate can even be lower than the Nyquist rate.

STATISTICAL ANALYSIS OF A PARAMETRIC MODEL FOR PHOTOMETRIC SIGNALS

A. Ferrari, Université de Nice Sophia-Antipolis, France
J.Y. Tourneret, ENSEEIHT/TESA, France

This communication studies a new model for photometric signals under high flux assumption. Photometric signals are modeled by Gaussian autoregressive processes having the same mean and variance denoted Constraint Gaussian AR Processes (CGARP's). This model is first derived from the data asymptotic distribution under high flux assumption. The performance of the CGARP parameter estimators is then studied by comparing their mean square errors to the Cramer Rao lower bounds (CRLB's). Asymptotic expressions are derived to approximate the CRLB's for large values of the number of samples. Computer simulations confirm the validity of these expressions. The achievable performance for CGARP parameter estimation is compared to those obtained with the unconstraint model. The purpose of this model is to derive a Neyman Pearson detector for the change-point detection problem that arises in the extrasolar planets detection problem.

MAP AND LS ESTIMATION OF ABRUPT CHANGES IN MULTIPLICATIVE OF ADDITIVE NOISE USING DYNAMIC PROGRAMMING

Martial Coulon and Jean-Yves Tourneret, ENSEEIHT, France

This paper addresses the problem of estimating abrupt changes corrupted by multiplicative noise. Two estimators are considered depending on the noise assumptions: the Maximum A Posteriori (MAP) estimator and the Least-Squares (LS) estimator. Both estimators are computed using dynamic programming. This study is then applied to edge detection in SAR images.

ORDER SELECTION OF 2-D AR MODEL USING A LATTICE REPRESENTATION AND INFORMATION CRITERIA FOR TEXTURE ANALYSIS

Olivier Alata and Christian Olivier, Université de Poitiers, France

In the context of parametric modeling for image processing, we use a lattice representation, i.e. based on reflection coefficients, to derive an estimation method for both the parameters and the order of the 2-D Quarter Plane Autoregressive model. The method is based on the combination of an Information Criterion (IC) and the prediction errors of models computed from a lattice parameter estimation algorithm. In this paper, we propose the use of two criteria which are consistent conversely the Akaike one: the Kashyap and Chelappa criterion is a 2-D extension of Bayesian Information Criterion (BIC); the second criteria, called phi_beta criterion, which is extended here to the 2-D case, is a generalization drawn on Rissanen's works. Simulations are provided on synthetic and natural textures. The results show the interest of using lattice estimation algorithm and phi_beta criterion to a characterization of textures.

CHOOSING PRIORS FOR AN IMPORTANT CLASS OF SIGNAL PROCESSING PROBLEMS

Yufei Huang and Petar M. Djuric, State University of New York, USA

Proper choice of prior distributions is a very important issue in Bayesian methodology. It is particularly important when the number of available data for processing is rather small. When little is known a priori, noninformative priors are usually employed. A well known approach for determining noninformative priors is Jeffreys' rule, which practically provides meaningful and locally uniform priors of the unknowns. In this paper, we carefully follow Jeffreys' rule to determine noninformative priors for an important class of signal processing problems that involve frequency estimation and DOA estimation. Cases of one and two signals are discussed in detail. Their analysis is also extended to include more general scenarios.

TIME-DELAY ESTIMATION OF THE LINE-OF-SIGHT SIGNAL IN A MULTIPATH ENVIRONMENT

Gonzalo Seco and Juan Fernandez-Rubio, Universitat Politecnica de Catalunya, Spain

The problem of estimating the time-delay of the line-of-sight (LOSS) or direct signal received by an antenna array in a multipath scenario is addressed. This problem is essential in many communication, radar, sonar and navigation systems, since the signal that propagates through the direct path may be the only one that bears useful information for the operation of the receiver. In many of these systems, it is possible to have an approximate a priori knowledge about the direction-of-arrival (DOA) of the direct signal. We analyze several time-delay estimators that exploit this information. In order that they can operate in the presence of co-channel interference (CCI), it is assumed that the noise field has an arbitrary and unknown spatial correlation. We show that the maximum likelihood (ML) estimator is the only one that presents a sufficient robustness against calibration or modeling errors. Starting from the ML estimator, we derive a new method that takes the uncertainty about the steering vector of the LOSS into account.

ThuPmPO3
Exhibition Hall
Acoustic, Echo and Noise Control
Chair: A. Carini ***Telit Mobile Terminals S.p.a, Italy***

COMPARISON OF DIFFERENT ADAPTIVE ALGORITHMS FOR STEREOPHONIC ACOUSTIC ECHO CANCELLATION

Peter Eneroth, Lund University, Sweden
Jacob Benesty, Tomas Gänsler and Steven Gay, Bell Laboratories, USA

In this paper, different adaptive algorithms for stereophonic acoustic echo cancellation are compared. The algorithms include the simple LMS algorithm and two specialized two-channel adaptive algorithms. Due to the high calculation complexity needed in stereophonic acoustic echo cancellation applications, the time domain algorithms are applied in a subband structure. The comparison include aspects as convergence rate, calculation complexity, signal path delay and memory usage. Real-life recordings are used in the evaluation.

FREQUENCY DOMAIN ACTIVE NOISE CONTROL SYSTEM WITHOUT SECONDARY PATH MODEL

Yoshinobu Kajikawa, Yasuo Nomura, Kansai University

In this paper, we propose a frequency domain active noise control (ANC) system without a secondary path model. The proposed system is based on the frequency domain simulta-neous perturbation (FDSP) method we have proposed. In this system, the coefficients of the adaptive filter are updated only by error signals. The conventional ANC system using the filtered-x algorithm becomes unstable due to the error between the secondary path, from secondary source to error sensor, and its model. In contrast, the proposed ANC system has the advantage that it does not use a model. In this paper, we show the principle of the proposed ANC system, and examine its efficiency by means of computer simulations.

GSAEC - ACOUSTIC ECHO CANCELLATION EMBEDDED INTO THE GENERALIZED SIDELOBE CANCELLER

Wolfgang Herbordt and Walter Kellermann, University of Erlangen-Nuremberg, Germany

In this paper, we examine the integration of acoustic echo cancellation (AEC) with a robust generalized sidelobe canceller (GSC) after O. Hoshuyama and A. Sugiyama with regard to exploitable synergies. Since the AEC is located behind the fixed beamformer (FBF) of the GSC only one AEC is required for an arbitrary number of array elements. System analysis shows that ideal suppression of local interferers and of acoustic echoes cannot be achieved simultaneously. However, experiments confirm that the interference reduction (IR) is degraded by less than 3 dB compared to a cascade of AEC and GSC and that the echo reduction is improved by more than \$13\$ dB relative to the GSC in echoic environments.

STEP-SIZE CONTROL FOR ACOUSTIC ECHO CANCELLERS APPLYING AN RBF NETWORK

Andreas Mader, Darmstadt University of Technology, Germany

The control of an acoustic echo canceller is an essential part of hands-free telephone sets. Due to the fact that no single estimator is yet known to reliably control the AEC, various estimators should be implemented. Nevertheless, the combination of several estimators is quite difficult and usually determined heuristically. In this paper, an approach for automatic combination of estimators, based on a radial-basis function (RBF) network, is presented. Expert knowledge of estimators is no longer required and the heuristics involved are clearly reduced. Results of the proposed method in comparison to a conventional method are shown at the end of the paper.

IMPROVED NOISE REDUCTION FOR HANDS-FREE CAR PHONES UTILIZING INFORMATION ON VEHICLE AND ENGINE SPEEDS

Henning Puder and Frank Steffens, Darmstadt University of Technology, Germany

In this paper we present an improved method for the spectral estimation of car noise in order to enhance the performance of noise reduction systems. An algorithm is developed that allows us to track changes in the noise spectrum during speech activity. For this tracking, we use the knowledge of the speed of the car and the revolutions of the engine. The paper starts with a detailed analysis of the car noise. The proposed algorithm based on this analysis first removes the harmonic components of the engine noise by selective filtering in time. The remaining wind and tyre noise is predicted during speech activity, based on the last available estimate and the vehicle speed.

FREQUENCY SELECTIVE STEP-SIZE CONTROL FOR ACOUSTIC ECHO CANCELLATION

Martin Heckmann, Julia Vogel and Kristian Kroschel, University of Karlsruhe, Germany

A new algorithm to control the step-size of a frequency domain echo cancellation system is presented. The step-size is controlled via the estimate of the coherence between the microphone signal and the output of the echo cancellation filter. The use of coherence allows an independent step-size control for each frequency bin. Additive local noise correlated with the echo signal and thus corrupting the adaption is monitored by the coherence and the step-size for the adaption is set correspondingly. The algorithm allows better convergence than conventional algorithms due to its frequency selectivity and does not require any a-priori knowledge.

A KALMAN FILTER APPROACH TO ACTIVE NOISE CONTROL

Paulo A. C. Lopes and Moisés S. Piedade, Instituto Superior Técnico, IST / INESC

Most Active Noise Control (ANC) systems use some form of the LMS algorithm due to its reduced computational complexity. However, the problems associated with it are well-known, namely slow convergence and high sensitivity to the eigenvalue spread. To overcome these problems the RLS algorithm is often used, but it is now widely known, that the RLS loses many of its good properties for a forgetting factor lower than one. Namely, it has been shown that in some applications the LMS algorithm is actually better in tracking non-stationary systems than the RLS algorithm. One approach, which works well with non-stationary systems, is to use some specialized form of the Kalman filter, which can be interpreted as a generalization of the RLS algorithm. The Kalman filter has a high computational complexity, similar to that of the RLS algorithm, which can make it costly for some applications. Nevertheless, for narrow-band ANC, the number of taps is not very large, and the application of the Kalman filter in ANC may be easily handled by today DSP's. In this paper, a specialized version of the Kalman filter fitted to ANC is developed; both control filter adaptation and secondary path modeling. It is shown, throw computer experiments, that a large reduction in the residual noise can be achieved in non-stationary environments, compared with the LMS and RLS based algorithms, especially when online secondary path modeling is used.

FLEXIBLE COCHLEAR SYSTEM BASED ON DIGITAL MODEL OF COCHLEA: STRUCTURE, ALGORITHMS AND TESTING

Jaroslaw Baszun and Alexander Petrovsky, Bialystok University of Technology, Poland

This paper presents complete speech processing system for cochlear prosthesis. The main elements of the system are speech enhancement system, speech processor and system for speech reconstruction. Speech enhancement system exploits properties of modulation spectrum of the human speech. Speech signal is split into 64 equal bands using maximally decimated polyphase DFT filter bank. Power spectrum of the signal for each channel is computed and filtered with tunable bandpass filter. Characteristic of the bandpass filter is computed independently for each channel base on the properties of power spectrum of the signal in the channel. Output signal is reconstructed from filtered power spectrum and original phase of the input signal. Tests proved that the system is suitable for cochlear implants and as a preprocessing procedure in Automatic Speech Recognition (ASR) systems. Speech processor dedicated for CIS stimulation was design base on digital, two dimensional, nonlinear model of the human cochlea model. The analysis filter bank was built with use tunable bandpass filters.

RLS-ADAPTED POLYNOMIAL FOR NONLINEAR ACOUSTIC ECHO CANCELLING

Alexander Stenger and Walter Kellermann, University of Erlangen-Nuremberg, Germany

Low-cost audio components in hands-free telephone applications call for nonlinear adaptive echo cancellation. It has been demonstrated that a cascade of a polynomial and an FIR filter can cancel echoes due to nonlinearities of typical low-cost applications [1]. Overcoming the slow convergence of an NLMS-based adaptation [5], we derive an RLS-type adaptation for the

polynomial preprocessor. Furthermore, we present a robust control of the RLS algorithm. Experiments with double-talk scenarios and real audio hardware show more than 8 dB overall echo reduction gain and fast and robust convergence of the polynomial. In addition, an implementation with reduced computational complexity and similar robustness is presented.

SPARSE-BEFAP: A FAST IMPLEMENTATION OF FAST AFFINE PROJECTION AVOIDING EXPLICIT REGULARISATION

Geert Rombouts and Marc Moonen, KULeuven/ESAT-SISTA, Belgium

It is well known that regularisation of the covariance matrix is necessary in Affine Projection Algorithm (APA) based adaptive filters. A technique to achieve regularisation is using non--successive equations in the system of equations that is solved in the APA--algorithm, leading to the Sparse--APA algorithm . In this paper, a fast exact frequency domain implementation of Sparse--APA is derived which does not rely on any of the approximations which are found in the classical fast affine projection algorithms, while only a very small number of extra computations is needed.

A NONLINEAR ACOUSTIC ECHO CANCELLER FOR HANDS-FREE TELEPHONY

Abdellatif Ben Rabaa and Rached Tourki, Electronic and Micro-Electronic Laboratory, Tunisia

In this paper, we propose an Acoustic Echo Canceller (AEC) composed by a cascade of a Pipelined Recurrent Neural Network (PRNN) and a linear filter. The PRNN is so called because of its recurrent and its modular processing. The Recurrent processing is used in order to more accurately model the room/speakerphone transfer function at high volumes. The linear subsection is a linear filter using a Fast Affine Projection Algorithm (FAP). FAP's key features include LMS like complexity and memory requirement (low), and RLS like convergence (fast) for the important case where the excitation signal is speech. Simulations results show, at high volume, an improvement in the echo return loss enhancement (ERLE) of 9 dB over a conventional linear adaptive filter with little additional computation.

VOLUME IV

FriAmOR1
Studio
Filter Design
Chair: I. Hartimo *Helsinki University of Technology, Finland*

09:20

SIMPLE DESIGN OF FRACTIONAL DELAY ALLPASS FILTERS

Vesa Välimäki, Helsinki University of Technology, Finland

A novel closed-form method for designing fractional delay allpass filters is proposed. The design uses closed-form formulas and is based on truncating the coefficient vector of a Thiran allpass filter. While the resulting filters are non-optimal, they allow a wider approximation bandwidth than the Thiran allpass filter, which yields a maximally flat delay approximation at the zero frequency. Formulas have been derived to assist in choosing the two parameters, order and prototype order, for the new design. There is practically no upper limit for the filter order, since the method is not prone to numerical problems.

09:40

AN FIR DIGITAL FILTER USING ONE-HOT CODED RESIDUE REPRESENTATION

Jimson Mathew and D. Radhakrishnan, Nanyang Technological University

Several Digital Signal Processing (DSP) structures based on Residue number systems (RNS) have been proposed in the technical literature. Most of them employ a look up table approach, that is highly space inefficient and slow. In this regard, we propose a memoryless high-speed error detecting FIR filter architecture using 1-out-of-n code for representing the residue numbers. With the use of 1-out-of-n code, error detection is achieved without any redundant moduli. The proposed design exhibits VLSI efficient layout, operand independent delay and low power consumption. We also propose a technique to achieve direct analog output from the system.

10:00

BAYESIAN SWITCHING ALGORITHM FOR THE OPTIMAL INCREASING BINARY FILTER

Nina S. T. Hirata and Junior Barrera, University of Sao Paulo, Brazil
Edward R. Dougherty, Texas A & M University, USA

The optimal windowed, translation-invariant binary image operator depends on conditional probabilities p(Y|x), where Y is a pixel value in a window about the pixel. The switching algorithm derives an optimal increasing filter from the optimal operator by switching observation vectors in or out of the kernel in such a way as to obtain an increasing filter with minimal increase in error over the optimal operator. These operators are usually designed by estimating the conditional probabilities from observed-ideal pairs of images. However, samples are typically too small to obtain good estimates of these probabilities. This paper discusses the design of increasing optimal filters by the switching algorithm using prior distributions for the conditional probabilities.

10:20

HOMOGENEOUS NONLINEAR DIGITAL FILTERS

Ronald K. Pearson, Instutut fuer Automatik ETH Zuerich, Switzerland

Linearity may be defined behaviorally via the well-known principle of superposition. This paper describes three useful relaxations of the principle of superposition, all of which lead to interesting classes of nonlinear digital filters: homogeneity, positive-homogeneity, and static-linearity. In fact, many popular digital filters fall into these classes and this paper explores the use of these characterizations in the analysis and design of nonlinear filters. Complete characterizations of the FIR subclass are given for two of these nonlinear filter classes, and useful closure properties are presented which provide a basis for constructing more general (e.g., recursive) members of all three of these filter classes. These ideas are used to develop a recursive modification of the standard median filter.

11:00

A CLASS OF TWO-CHANNEL IIR/FIR FILTER BANKS

Per Löwenborg, Håkan Johansson, and Lars Wanhammar, Linköping University, Sweden

A class of digital filter banks is introduced, in which the analysis filters are half band IIR filters and the synthesis filters are FIR filters. The proposed filter bank satisfy approximately perfect reconstruction and has an exact linear phase. Further, it is suitable for applications where a very low complexity analysis filter bank is required.

11:20

LINEAR PHASE FIR FILTER DESIGN FOR MULTIRATE SYSTEMS BASED ON FAST CONVOLUTION

Alexandra Groth, Frank Budke, Heinz G. Goeckler and Gennaro Evangelista, Ruhr-Universitaet Bochum, Germany

Fast convolution is well known for efficient FIR filtering. In a recently proposed approach [2] it is extended to multirate signal processing. Thereby additional savings in computational expenditure can be obtained by suitable frequency domain filter coefficients. Hence, new design methods for linear phase interpolation and decimation filters in multirate systems with fast convolution are suggested.

11:40

DESIGN OF GRAY-SCALE NONLINEAR FILTERS VIA MULTIRESOLUTION APERTURES

Roberto Hirata Jr. and Junior Barrera, University of Sao Paulo, Brazil
Edward R. Dougherty, Texas A & M University, USA

This paper applies pyramidal multiresolution design, which has been used to design binary filters, to design aperture filters, which are nonlinear gray-scale filters that operate on data in a range-domain window (aperture). The designed filters are used to find markers for the region of the eyes in an image database of faces.

12:00

EFFECT OF ZEROS ON IIR FILTERS WITH SYMMETRIC IMPULSE RESPONSE

Mladen Vucic and Hrvoje Babic, Faculty of Electrical Engineering and Computing, Croatia

The properties of the symmetric impulse response filters with real and complex zeros are considered. The pole-zero locations of the rational transfer functions are found, by minimization of the symmetry error. The contribution of zeros to the symmetry error is shown and compared to the systems with all zeros at the origin. The design procedure of such filters based on time or frequency domain requirements is outlined.

12:20

FILTER CATEGORIES PRESERVING DESIRABLE SIGNAL CHARACTERISTICS

Ronald K. Pearson, Instutut fuer Automatik ETH Zuerich, Switzerland

This paper considers the problem of characterizing nonlinear digital filters via their qualitative input-output behavior. For example, it is known that (non-causal) symmetric median filters

preserve arbitrary monotone sequences. Behavioral characterizations ask converse questions like: what can be said about the class of all filters F that preserve monotonicity? The primary point of this paper is to illustrate that category theory provides a natural framework for examining such questions.

FriAmOR2
Hall B
Wavelets and Filterbanks
Chair: T. Saramäki *Tampere University of Technology, Finland*

09:20

ORDER SELECTION FOR THE MULTIRATE BURG ALGORITHM

S. de Waele and P.M.T. Broersen, Delft University of Technology, The Netherlands

In many applications one is interested in the behavior of a process at various time scales or intervals T. The key issue in accurate time series analysis at interval T is order selection, rather than parameter estimation. Order selection at interval T uses knowledge of the statistal behavior of estimated AR parameters at interval T. A finite sample expression for this behavior is derived and verified in simulations. Spectral details which are not considered significant with standard modeling are accurately described with modeling at interval T. As a result, order selection at interval T yields better models than application of standard order selection.

09:40

DESIGN OF BIORTHOGONAL MULTI-BAND FIR FILTER BANKS GIVEN SEVERAL OF THE ANALYSIS AND SYNTHESIS FILTERS

Eleftherios Kofidis and Phillip A. Regalia, Institut National des Telecommunications, France

The problem of designing an N-band biorthogonal FIR analysis/synthesis system, given K of its analysis and synthesis filters, is addressed in this paper. A necessary and sufficient condition on the K filters, that guides their selection, is derived and the solution set is completely parameterized via ladder-type structures that guarantee the perfect reconstruction property under both coefficient quantization and roundoff errors. Results are presented for both nonlinear- and linear-phase systems. A design example is provided to illustrate the theory.

10:00

ON LOSSY IMAGE COMPRESSION USING ADAPTIVE WAVELET FILTERS

Subhasis Saha and Rao Vemuri, University of California Davis, USA

Adaptive Wavelet-based Coding is a new technique where instead of using a fixed wavelet filter for coding and decoding all images, different filters are used for different images. Since there is no single filter that gives the maximum PSNR for all images, adaptive selection and usage of wavelet filters improves the coding performance. In this paper, we discuss a methodology for adaptive wavelet filter selection, and analyze the performance of such adaptive wavelet-based lossy image coders. rather than using a fixed filter for all images, the most appropriate wavelet filter should be chosen adaptively depending on the statistical nature of the image being coded.

The block diagram of such an adaptive wavelet-based image coder is shown in Figure 2. Adaptive wavelet coders like this would be most useful for asymmetric applications where the image need to be coded only once and decoded many times. In this paper, we discuss a methodology for adaptive wavelet filter selection for lossy image coders, and present some promising results.

10:20

POLYNOMIAL METHODS FOR MULTI-DIMENSIONAL FILTER BANKS' DESIGN

Mikhail K. Tchobanou, Moscow Power Engineering Institute, Russia

The problem of the design of multi-dimensional filter banks with prescribed properties is considered. The most important requirements are perfect reconstruction property, linear phase property, optimization for a given class of signals, maximum number of vanishing moments and others. A survey of different approaches to this problem proves that the most promising are polynomial approaches that allow to find the result directly in analytical form. The results of filter banks' optimization are presented (so that the stop-band energy of the filter is minimum in the least-squares sense).

11:00

CLASSIFICATION OF VOICELESS PLOSIVES USING WAVELET PACKET BASED APPROACHES

Ewa £ukasik, Poznañ University of Technology, Poland

There are contradictory reports on the usefulness of the Wavelet packet Transform for feature extraction. In this paper we concentrate on the influence of the feature reduction method on the classification. The data set for experiments consists of three classes of non-stationary speech signals, namely unvoiced plosive consonants /p/, /t/, /k/. Two strategies have been applied for features reduction: feature selection, performed using the Local Discrimination Basis and feature projection performed using Primary Components Analysis (Singular Value Decomposition). Classification has been performed by cluster analysis and neural network. The classification results obtained for PCA outperform those for LDB and other methods examined earlier.

11:20

IMPULSE NOISE REDUCTION BASED ON LIFTING WAVELET TRANSFORM

Koichi Kuzume, Yuge National College of Technology, Japan
Shigeru Takano and Koichi Niijima, Kyushu University, Japan

This paper presents a novel method to restore signal corrupted with impulse noise by using time-varying lifting wavelet filters (TVLWF), which are designed based on Sweldens' lifting wavelet theory. In general, it is difficult to remove such a noise by conventional linear filters and the Donoho's wavelet shrinkage method. We design the TVLWF so as to vanish all the wavelet

coefficients by tuning free parameters contained in the TVLWF. Using the designed TVLWF, we can detect the locations of impulse noise and estimate their amplitude. To remove the noise from a corrupted signal without any degradation of signal details, the detected impulse noise are subtracted from the noisy signal by using the estimated ones. Proposed noise reduction method is tested on an ECG signal corrupted with mixed Gaussian and random-valued impulse noise. Major improvement is obtained in comparison with conventional median filter.

11:40

MULTIRESOLUTION BASED IMAGE ADAPTIVE WATERMARKING SCHEME

Kaewkamnerd, Chulachomklao Royal Military Academy
K.R.Rao, University of Texas at Arlington

This paper proposes an image adaptive watermark embedding method based on discrete wavelet transform (DWT). To increase the robustness and perceptual invisibility of watermark, the algorithm is combined with the quantization model based on human visual system (HVS). Number of factors that affect the noise sensitivity of human eye, such as background luminance, proximity to an edge, frequency band and texture masking are taken into consideration. To extract the embedded watermark, both of blind (the uncorrupted image is not required) and non-blind methods are introduced. Experimental results show that the proposed method is robust to the image intensity attack such as high compression.

12:00

HIGHLY EFFICIENT FAST ARCHITECTURES FOR DISCRETE WAVELET TRANSFORMS BASED ON THEIR FLOWGRAPH REPRESENTATION

D. Gevorkian, and J. Astola, Tampere University of Technology, Finland
F. Marino, Politecnico di Bari, Italy
S. Agaian, University of Texas at San Antonio, USA

Flowgraph representation of discrete wavelet transforms (DWTs) is presented. It is shown that this representation is useful for developing efficient DWT algorithms and VLSI architectures. As an examples two DWT architectures are proposed with the efficiency (counted as the measur of the average utilization of PEs,) of approximately 100%. Proposed architectures are very fast and provide excellent performance with respect to area-time characteristics. They are scalable, simple, regular, and free of long connections (depending on the length of input signal). Results can be easily extendet to inverse DWTs as well as to Hadamard wavelets and wavelet packets.

12:20

ON THE CONVERGENCE OF MULTIWAVELET THRESHOLDING

Lixin Shen and Hwee Huat Tan, National University of Singapore, Singapore

In this paper we introduce a general thresholding operator based on multiwavelets. It offers the flexibility that multiwavelet coefficients generated by different component multiwavelet functions in a multiwavelet vector can be filtered by different thresholding rules. A convergence result of the proposed operator is given. Numerical results for denoising simulation are also presented for showing its robustness

FriAmOR3
Hall A
Watermarking
Chair: V. Cappellini ***Università di Firenze, Italy***

09:20

CONTENT ADAPTIVE WATERMARKING BASED ON A STOCHASTIC MULTIRESOLUTION IMAGE MODELING

Sviatoslav Voloshynovskiy, Frédéric Deguillaume, and Thierry Pun Cui, University of Geneva, Italy

In this paper, a wavelet domain robust watermarking technique for still images is presented. The proposed watermarking algorithm is based on 3 key aspects. First, message encoding is accomplished based on iterative error correction codes to reach channel capacity with reasonable decoder complexity. Secondly, watermark embedding is performed in the wavelet domain using a stochastically driven perceptual criterion to provide watermark invisibility. This approach considers edge and texture masking properties as well as the multiresolution sensitivity of the human visual system. Thirdly, a new principle of periodic spatial allocation, based on the analysis of the watermark magnitude spectrum, is proposed to recover from general affine geometrical distortions, without need for an additional template. In experiments the watermark was decoded with no error even from higly JPEG-compressed images. Due to the wavelet decomposition approach used, the proposed watermarking technique is also well matched with the recent image compression standard JPEG2000.

09:40

IMAGE AUTHENTIFICATION AND TAMPER PROOFING USING MATHEMATICAL MORPHOLOGY

A. Tefas and I. Pitas, Aristotle University of Thessaloniki, Greece

A novel method for image authentication and tamper proofing is proposed. A binary watermark is embedded in a grayscale or a color host image. The method succeeds in detecting alterations made in a watermarked image. The proposed method is robust against high quality lossy image compression. It provides the user not only with a measure for the authenticity of the test image but also with an image map that highlights the unaltered regions of the image when selective tampering has been made. Mathematical morphology techniques are also developed for the accurate detection of alterations in fine image details.

10:00

A SUBBAND DCT APPROACH TO IMAGE WATERMARKING

V. Fotopoulos and A. Skodras, University of Patras, Greece

A subband-DCT approach for image watermarking is proposed in this communication. The watermark is casted in a selected number of coefficients of all four bands of a one-level decomposition. A great number of coefficients is being used. Each band gives a different detection output. The result is taken as the average detection result of all bands. It is shown that the final result is better than the detection output of each individual band, thus leading to a very robust watermarking scheme.

10:20

USING PERMUTATIONS TO HIDE INFORMATION

Heikki Huttunen and Pauli Kuosmanen, Tampere University of Technology, Finland
Charles Boncelet, University of Delaware, USA

We describe a new image steganography method that uses permutations of selected pixels to hide information. The basic idea is that a subset of pixels is selected by a keyed random number generator. The values of these pixels are altered by as little as possible but such that the sorted ordering of these values corresponds to a specific permutation. N pixels can be ordered in N! ways, corresponding to log2(N!) bits of hidden information. We explore two different methods of selecting the pixels, one that selects them in a scattered fashion and one that clusters them in small neighborhoods. We also examine the tradeoffs associated with large versus small subsets of pixels.

11:00

A ROBUST FRAME-BASED TECHNIQUE FOR VIDEO WATERMARKING

R. Caldelli, F. Bartolini and A. Piva, University of Firenze, Italy
M. Barni, University of Siena, Italy

During the last few years multimedia systems success has been definitely spurred by the development of network-based on line services, like electronic commerce, teleducation, teletraining, media distribution and so on, in which multimedia systems have a major role. In this new scenario the request for privacy, security and authenticity of the documents, undergone to these transactions, is strongly upcoming. Watermarking techniques, deriving from information hiding theory, have been suggested so far as an effective and suitable solution to satisfy this kind of requirements, and in a short time many and disparate algorithms applying different strategies to achieve the same goal have been proposed in scientific literature and by commercial entities. In this paper a watermarking algorithm, originally conceived for still images applications, has been extended to raw video, treating it as a set of single still frames. A good robustness against usual image processing and geometric manipulations has been achieved, moreover experiments with MPEG2 coding/decoding at different bit rates have been carried out giving positive results.

11:20

LOSSLESS MULTIRESOLUTION TRASNFORM FOR AUTHENTICATING WATERMARKING

B. Macq, Universite Catholique de Louvain, Belgium

This paper present a new algorithm able to achieve a one-to-one mapping of a digital image onto another image. This mapping takes into account the limited resolution and the limited quantization of the pixels. The mapping is achieved in a multiresolution way. Performing small modifications on statistics of the details allows to build a lossless, i.e. revertible, watermarking authenticating procedure.

11:40

SELF-SIMILAR RING SHAPED WATERMARK EMBEDDING IN 2-D DFT DOMAIN

V. Solachidis and I.Pitas, Aristotle University of Thessaloniki, Greece

A new watermarking algorithm for still images is presented in this paper. The watermark is embedded in magnitude of the DFT domain and introduces image changes that are invisible to the human eye. It is robust to compression, filtering, cropping, translation, rotation and scaling. The detection algorithm does not require the original image. The watermark has a self-similar structure that accelerates the detection procedure.

12:00

CAPACITY OF DIGITAL WATERMARKS SUBJECTED TO AN OPTIMAL COLLUSION ATTACK

Jonathan K. Su and Joachim J. Eggers, University Erlangen-Nuremberg, Germany
Bernd Girod, Stanford University, USA

One envisioned application of digital watermarking is fingerprinting, in which different information is embedded into several copies of the same original signal. Several attackers may collude by combining their copies to produce an attacked signal. In the case of independent watermarks, a collusion-attack model is presented and shown to be analogous to the Gaussian multiple-access channel. The attack parameters are optimized to minimize the information rate under a constraint on the distortion of the attacked signal. Another fingerprinting method, collusion-secure codes, is then related to the attack. Finally, independent and collusion-secure watermarking are compared for the same attacked-signal distortion and probability of false identification.

12:20

TRIANGULAR MESHES: A SOLUTION TO RESIST TO GEOMETRIC DISTORSIONS BASED WATERMARK-REMOVAL SOFTWARES

Franck Davoine, Laboratory Heudiasyc, University of Technology of Compiegne, France

We present in this paper a compensation technique allowing to retrieve a watermark in an image attacked by random local geometric distortions, with the help of the original image, or its edges. For this purpose, we use a flexible triangular 2-D mesh, in order to compensate parts of geometric distortions that could be introduced by softwares such as Stirmark. In order to illustrate the usefulness of the compensation on attacked images, we use two watermarking techniques: a classical spread spectrum algorithm proposed by Cox et al. (the watermark is embedded into a selected set of DCT coefficients) and a new wavelet-based watermarking technique.

FriAmSS1
Small Audiotorium
Splines in Signal Processing
Chair: M. Unser ***Ecole Polytechnique Federale de Lausanne, Switzerland***

09:20

SPLINE INTERPOLATION IN MEDICAL IMAGING: COMPARISON WITH OTHER CONVOLUTION-BASED APPROACHES

Erik H. W. Meijering, University Medical Center Utrecht, The Netherlands

Interpolation is required in a variety of medical image processing applications. Although many interpolation techniques are known from the literature, evaluations of these techniques for the specific task of applying geometrical transformations to medical images are still lacking. In this paper we present such an evaluation. We consider convolution-based interpolation methods and rigid transformations. The evaluation involves a large number of sinc-approximating kernels, including piecewise polynomial and windowed sinc kernels, and images from a wide variety of medical image modalities. The results show that for all modalities, spline interpolation constitutes the best trade-off between accuracy and computational cost, and therefore is to be preferred over all other methods.

10:00

NON-UNIFORM TO UNIFORM GRID CONVERSION USING LEAST-SQUARES SPLINES

Arrate Munoz, Thierry Blu and Michael Unser, Swiss Federal Institute of Technology, Switzerland

We propose a new technique to perform nonuniform to uniform grid conversion: first, interpolate using nonuniform splines, then project the resulting function onto a uniform spline space and finally, resample. We derive a closed form solution to the least-squares approximation problem. Our implementation is computationally exact and works for arbitrary sampling rates. We present examples that illustrate advantages of our projection technique over direct interpolation and resampling. The main benefit is the suppression of aliasing.

10:20

HIERARCHICAL CODING OF 3-D SURFACES USING BOX SPLINES

Michael G. Strintzis and Nikolaos V. Boulgouris, Aristotle University of Thessaloniki, Greece
Sotiris Malassiotis, Informatics and Telematics Institute, Greece

Optimal mechanisms are determined for the hierarchical decomposition of wire-frame surfaces. A family of box-splines with compact support, suitable for the approximation of wire-frames is

defined, generated by arbitrary sampling matrices with integer eigenvalues. Criterion of optimality is the minimization of the variance of the error difference between the original surface and its representation at each resolution level. This is needed so as to ensure that the wire mesh produces at each resolution as close a replica of the original surface as possible.

11:00

ON THE OPTIMALITY OF SPLINES FOR EXPRESSING THE HAND-WRITING PROCESS

Masaru Kamada, Ibaraki University, Hitachi

For the model of writing letters with a pen steered by hand so as to minimize the mean-square jerk under the constraints of passing through several via points and pausing at some of them, the optimal function is proven to be quintic B-splines with the knots placed at the via points and especially double knots at the pause points. A steeple like the left bottom of `h' is a pause point where the pen must pause at a moment to alter its direction. Application to automatic generation of words in script style from individual letters and feature extraction of hand-written letters are presented.

11:20

A GENERAL FRAMEWORK FOR MULTISCALE IMAGE REPRESENTATIONS USING B-SPLINE APPROACHE

Yu-Ping Wang and S. L. Lee, National University of Singapore, Singapore

Scale is a basic aspect of image modeling. The Gaussian kernel has long been used in multiscale image analysis. This paper presents a general framework for mutiscale geometric representations using B-splines. In particular, the following computational and statistical issues will be discussed. (a) Construction of wavelets using B-splines leads to computationally efficient algorithms. We show that for a class of wavelets, their masks can be factored into simple B-spline factors. As a result, these wavelets can be implemented using fast filter bank algorithms. (b) The analysis of statistical properties of wavelet transforms can be facilitated using B-splines. We show how the statistical properties of the class of wavelet transforms that are derived from B-splines can be easily deduced from that of the Gaussian function since it approximates the B-splines.

11:40

ANALYSIS OF fMRI DATA USING SPLINE WAVELETS

Manuela Feilner, Thierry Blu and Michael Unser, Swiss Federal Institute of Technology, Switzerland

Our goal is to detect and localize areas of activation in the brain from sequences of fMRI images. The standard approach for reducing the noise contained in the fMRI images is to apply a spatial

Gaussian filter which entails some loss of details. Here instead, we consider a wavelet solution to the problem, which has the advantage of retaining high-frequency information. We use fractional-spline orthogonal wavelets with a continuously-varying order parameter alpha; by adjusting alpha, we can balance spatial resolution against frequency localization. The activation pattern is detected by performing multiple (Bonferroni-corrected) t-tests in the wavelet domain. This pattern is then localized by inverse wavelet transform of a thresholded coefficient map. In order to compare transforms and to select the best alpha, we devise a simulation study for the detection of a known activation pattern. We also apply our methodology to the analysis of acquired fMRI data for a motor task.

12:00

FRACTIONAL DERIVATIVES, SPLINES AND TOMOGRAPHY

Michael Unser, Stefan Horbelt and Thierry Blu Short, Swiss Federal Institute of Technology, Switzerland

We develop a spline calculus for dealing with fractional derivatives. After a brief review of fractional splines, we present the main formulas for computing the fractional derivatives of the underlying basis functions. In particular, we show that the γth fractional derivative of a B-spline of degree α (not necessarily integer) is given by the γth fractional difference of a B-spline of degree $\alpha-\gamma$. We use these results to derive an improved version the filtered backprojection algorithm for tomographic reconstruction. The projection data is first interpolated with splines; the continuous model is then used explicitly for an exact implementation of the filtering and backprojection steps.

FriAmOR4
Room 200
Signal Reconstruction
Chair: E. Coyle *Motorolla, USA*

09:20

NEW ALGORITHMS FOR BAND-LIMITED INTERPOLATION AND EXTRAPOLATION: A SYNTHETIC VIEW

Paulo J. S. G. Ferreira, IEETA Universidade de Aveiro, Portugal

We propose a classification of several algorithms for solving the discrete-discrete band-limited interpolation and extrapolation problems. The classification is based on the dimension of certain underlying vector spaces, and distinguishes between time-domain and frequency-domain methods. This perspective allows a synthetic view of the problem, clarifies the connections between some of the existing algorithms, and establishes the existence of ``missing variants". We introduce the missing variants, and analyze the relative performance of the complete set of methods. It is shown that no method can be ``best overall". Instead, there are classes of problems for which one set of methods clearly outperforms the others. These classes are characterized, and the main issues that arise when designing efficient algorithms for the solution of a specific problem are discussed.

09:40

IMPLEMENTATION OF AUTOMATIC DIGITAL COMPENSATION IN IQ MODULATORS

J. Tuthill and A. Cantoni, University of Western Australia

We consider Digital Signal Processor (DSP) based IQ modulators generating Continuous Phase Frequency Shift Keying (CPFSK) signals. Departures from a flat-magnitude, linear phase frequency response in the pass bands of signal reconstruction filters in the I and Q channels cause ripple in the modulator output signal envelope. Amplitude Modulation (AM) in the signal envelope function produces undesirable sidelobes in the FSK signal spectrum when the signal passes through nonlinear elements in the transmission path. A structure is developed for digitally compensating for the magnitude and phase characteristics of the signal reconstruction filters. Optimum digital compensation filters are found using least squares (LS) techniques. We propose a method by which the optimum compensation filters can be estimated using test input signals thus allowing the compensation process to be automated. We describe the implementation of the automatic digital compensation technique on a DSP platform and present experimental results.

10:00

SIGNAL RECONSTRUCTION FROM THE MAGNITUDE OR PHASE OF A GENERALISED WAVELET TRANSFORM

D.M. Lopes, 20/20 Speech Limited, UK
P.R. White, University of Southampton, UK

This paper considers signal reconstruction from a class of linear Time Frequency Representations (TFRs) referred to as the Generalised Wavelet Transform (GWT). This generalisation incorporates a number of linear TFRs, including the Short-Time Fourier Transform (STFT) and Wavelet Transform (WT). The problem addressed is that of constructing a time series from partial information regarding a GWT. Following the methodology previously employed for the STFT, a Least Squares (LS) approach for signal reconstruction from the magnitude of the GWT is presented. Along with a signal reconstruction algorithm using only the phase of the GWT. Examples are given illustrating the validity of these two algorithms.

10:20

QUANTIZATION NOISE COMPENSATION BY ANALYSIS FILTER BANK OPTIMISATION

Onoriu Bradeanu, Military Technical Academ, Romania
Ulrich Appel, Bundeswehr University, Germany

A method to improve and to enlarge the set of filter banks applications consists in overcoming the effects of developing the perfect reconstruction condition without considering distortions inducted, inevitable, by quantization and transmission over additive noise disturbed channels. In this paper is presented an analysis filter design algorithm that, based on the knowledge of the input - AR signal model, optimally compensates the quantization noise effects. The adjusting of the analysis filter bank using criteria based on the properties of the input signal remove the necessity of providing the synthesis part of the system with additional information and pull the processing complexity towards the transmitter end. The employed optimisation criteria consists in minimising the noise gain factor of the equivalent state-space representations of the equivalent input signal - analysis filter bank system. Simulation results, worked out on speech signal, prove significant performance improvements related to those obtained by conventional approach.

FriAmPO1
Exhibition Hall
Audio Compression and Electroacoustics
Chair: C. Martins *INESC/ENIDH, Portugal*

TIME-VARYING AUTOREGRESSIVE MODELING OF AUDIO AND SPEECH SIGNALS

Aki Härmä, Helsinki University of Technology, Finland
Marko Juntunen, Elektrobit Ltd.
Jari P. Kaipio, University of Kuopio, Finland

Conventional linear predictive techniques for modeling of speech and audio signals are based on an assumption that a signal is stationary within each analysis frame. However, natural signals are often continuously time-varying, i.e., nonstationary. Therefore this assumption might not be well justified. In this paper, we study a time-varying autoregressive (TVAR) modeling technique in which this restriction is relaxed. A frequency-warped formulation of the Subba Rao-Liporace TVAR algorithm is introduced in the article. The applicability of the presented methodology to various speech and audio signal processing tasks is illustrated and discussed. It is also shown that the TVAR scheme yields an efficient parametrization for time-varying sounds.

A DOUBLE-THRESHOLD-BASED APPROACH TO IMPULSIVE NOISE DETECTION IN AUDIO SIGNALS

Paulo A. A. Esquef, Luiz W. P. Biscainho, Paulo S. R. Diniz, Fabio P. Freeland, COPPE/PEE & EE/DEL, Rio de Janeiro, Brazil

In this paper we review threshold-based algorithms for detection of impulsive noise in audio signals, and propose a new detection strategy that can increase their capability to detect impulsive disturbances without any remarkable increase in computational complexity. The main features addressed by this paper are the use of two thresholds, one for detection and the other for estimation of impulse durations; the merging of disturbances detected as adjacent, under certain circumstances; and the alteration of the threshold level during the re-processing procedure of a given signal block. Experimental results assessed by human subjects as well as objective measurements confirm the improvements attained by the proposed detection scheme.

CROSSTALK ROBUST STRUCTURES BASED ON SUB-BAND NOISE ESTIMATION

Carlos R. Martins, Escola Náutica Infante D. Henrique, Portugal

This paper will focus on a new approach, which was derived from the multi-band ANC concept. The proposed structures referred as Fast Multi-Band ANC's (FMB-ANC) are immune to crosstalk, allowing higher robustness to instability and faster convergence than the CrossTalk Resistant ANC approach (CTRANC). The method is efficient for speech enhancement because it removes reasonably the correlation between the primary and reference estimated speech

components. A comparison between FMB-ANC's based on the transversal filter (FIR-LMS) and on the linear combined gradient adaptive lattice filter (GAL) is presented. Noisy speech acquired in real environments and coloured synthetic input signals, a single sinusoid and a chirped sinusoid in white noise were used to evaluate and compare the two adaptive structures. The adaptive lattice filters are less sensitive to eigenvalue spread and present faster convergence than the conventional ANC. However, in the FMB-ANC the sub-band noise estimation facilitates the convergence and increase the level of sinusoid cancellation. The experimental results obtained with real time implementations on a DSP, proved the superiority of the simpler solution based on transversal filters, while the computational burden is kept low.

A NEW CONVOLUTIVE BLIND SIGNAL SEPARATION ALGORITHM BASED ON SECOND ORDER STATISTICS USING A SIMPLIFIED MIXING MODEL

Bin Yin and Piet Sommen, Eindhoven University of Technology, The Netherlands

In this paper a new convolutive blind signal separation (BSS) algorithm is proposed, which is based on a simplified mixing model and second order statistics. The simplified mixing model uses the fact that the acoustic transfer functions from a source to two closely spaced microphones are very similar. Only one difference between these two transfer functions is needed. Over other BSS algorithms, the main advantage of this algorithm is the significant reduction of parameters to be estimated, which makes it less time and power consuming and more easily implemented in real time. The algorithm is efficiently realized in frequency domain and its effectiveness is shown by the experimental results using speech signals recorded in a real acoustic environment.

LOW DELAY AUDIO CODER USING ADAPTIVE VECTOR QUANTIZATION

D. Martinez Munoz, M.Rosa Zurera, F.Lopez Ferreras and N.Ruiz Reyes, Universidad de Jaen, Spain

Nowadays, there is not a low delay audio coding standard that achieves nearly-transparent quality. The ISO-MPEG standards are profusely used in audio for high quality coding. They use perceptual models that require high frequency resolution. In applications where the delay is a critical parameter, i.e. when the use of a feedback channel is important, the ISO-MPEG algorithms are not suitable. In this paper we present a new coder that responds to a subband ADPCM structure and incorporates adaptive vector quantization, requiring only 2 bits/sample. Each vector component (prediction error) is weighted according to its estimated standard deviation. The introduced coding delay is less than 2 milliseconds, and is only due to the perfect reconstruction filter bank. In order to obtain the best configuration, results have been obtained varying the number of bands, the number of bits in the quantizer and the analysis filters.

A NEW PERCEPTUAL ENTROPY-BASED METHOD TO ACHIEVE A SIGNAL ADAPTED WAVELET TREE IN A LOW BIT RATE PERCEPTUAL AUDIO CODER

N. Ruiz Reyes and D. Martínez Muñoz , Universidad de Jaén, Escuela Universitaria Politénica, Spain
M. Rosa Zurera and F. López Ferreras, Universidad de Alcalá, Escuela Politécnica, Spain

This paper outlines a wavelet based audio coding scheme that uses a perceptual entropy criterion to achieve a signal adapted wavelet tree. We are interested in decompositions adapted to the nature of audio signals, but attending to the characteristics of the human hearing system. Our audio coder is compared with two other wavelet based coding schemes: the first one implements a fixed wavelet decomposition which closely mimics the critical band division at the inner ear, while the second one uses an adaptive wavelet tree controlled by a classical entropy criterion, such as Shannon entropy. The results confirm that perceptual entropy-based decomposition is the way you must take if you are looking for maximum compression rates and transparent coding. To achieve a further bit rate reduction ensuring transparent coding, our audio coder performs a filter order optimization, which is possible by using a novel algorithm to translate the psycho-acoustic information from the frequency to the wavelet domain. Experimental results indicate that the proposed coder ensures transparent coding of monophonic CD quality audio signals at bit rates lower than 64 kb/s (1.45 bit/sample) for most audio signal. Subjective listening tests tell us that our coding method behaves as well as ISO/MPEG 1 - layer III and much better than MPEG 1 - layer II at that bit rate.

WAVELET SHRINKAGE DENOISING APPLIED TO REAL AUDIO SIGNALS UNDER PERCEPTUAL EVALUATION

Luiz W. P. Biscainho, Fabio P. Freeland, Paulo A. A. Esquef and Paulo S. R. Diniz,
COPPE/PEE & EE/DEL, Rio de Janeiro, Brazil

This paper addreses wavelet denoising of audio recordings from a practical viewpoint. First, a set of real high-quality audio signals with distinct characteristics is artificially corrupted by pseudo-white noise. Then, the classical wavelet shrinkage denoising method is applied to them under varied settings, including different wavelets, numbers of scales, threshold application and computation methods etc. At last, an adapted version of the Perceptual Audio Quality Measure (PAQM) is used as an objective quality index to compare the obtained results. This way, the work provides some insight on the performance of wavelet shrinkage applied to corrupted high-quality signals, the performance of the PAQM in this application and the relative importance of different parameters in the final quality determination.

SUBJECTIVE EVALUATION OF LSF QUANTIZATION IN CONVENTIONAL AND WARPED LP BASED AUDIO CODING

Markus Vaalgamaa, Nokia Mobile Phoneys
Aki Härmä, Helsinki University of Technology, Tampere

The objective of this study is to evaluate subjective quality of spectral envelope quantization in audio coding. In this paper the spectral envelope is modeled using Linear Prediction (LP) and

Warped Linear Prediction (WLP). The advantage of WLP compared to LP is that the frequency resolution of analysis can be modified so that it follows perceptual frequency scale, like the Bark- or ERB-scale. The coefficients in both cases are transformed into Line Spectral Frequency (LSF) parameters and quantized. The quantization of these parameters is studied and the subjective and objective qualities are compared.

QUALITATIVE AND QUANTITATIVE ASPECTS IN THE DESIGN OF PERIODICITY ESTIMATION ALGORITHMS

Anssi Klapuri, Tampere University of Technology, Finland

Several approaches have been taken to estimate the fundamental frequency of periodic or pseudoperiodic acoustic signals. However, since the algorithms operate in different transform domains, it has proven difficult to combine the desirable features of different algorithms, and to utilize simulation results across domains. In this paper, we set out to define a common ground on which to investigate the underlying models of different periodicity estimation algorithms. Implications of the analysis are applied in a new algorithm which aims at combining several advantages. Validation experiments were made using a database of musical instrument sounds, both in noise and in harmonic interference.

FriAmPO2
Exhibition Hall
Image and M-D Signal Processing
Chair: S. Boussakta *University of Teesside, UK*

REAL-TIME RECONSTRUCTION OF 2D SIGNALS WITH MISSING OBSERVATIONS

Khaled Asswad, Elisabeth Lahalle and Jacques Oksman, SUPELEC, France

This paper describe real-time methods to restore two dimensional (2D) uniformly sampled signals with randomly missing observations. It is based upon autoregressive (AR) modelling. The model parameters are estimated by means of an LMS-like algorithm adapted to the case of the non uniform context. The algorithm is proposed to proceed as follows: if sample is available, it updates the model parameters by minimizing a quadratic criterion defined at available samples. Otherwise, it replaces missing sample by its estimated value. Two types of algorithms: with and without delay, can be obtained by considering the different causality cases of AR-2D signals. Because of missing samples, calculating of the gradient of the optimized criterion becomes complicated. So, it is approximated. It is shown by an example that this approximation does not damage the reconstruction results in the non stationary case(image). Some reconstruction results of 'Lena' image, as function of missed samples rate are done, where the case of constant probability of missing is considered.

INVERSE SCATTERING VIA LINEAR AND NONLINEAR TOMOGRAPHY - A REVIEW

George A. Tsihrintzis, University of Piraeus

Nonlinear tomographic reconstruction algorithms are developed for inversion of data measured in scattering experiments in which the complex phase of the wavefields is modeled by an arbitrarily large (possibly infinite) number of terms in the Rytov series. The algorithms attain the form of a Volterra series of nonlinear operators, with the usual filtered backpropagation algorithm of Diffraction Tomography as the leading linear term.

BAYESIAN SPATIO-TEMPORAL MOTION DETECTION UNDER VARYING ILLUMINATION

Daniel Toth, Til Aach and Volker Metzler, Medical University of Lübeck, Germany

The detection of moving objects in image sequences acquired by a static camera is a demanding task, especially in the presence of noise. With fast illumination variations detection becomes even more difficult since simple motion detection will always be subject to illumination artifacts. To solve these problems, we propose a Bayesian spatio--temporal motion detection algorithm combined with homomorphic filtering. The image sequence is modelled as being generated by an illumination and a reflectance component that are approximately separated by the filter.

Detection of changes in the reflectance component is directly related to scene changes, i.e. object motion. Experimental results show that the presented method is considerably less sensitive to noise and time--varying scene illumination.

3-D VECTOR RADIX ALGORITHM FOR THE 3-D DISCRETE HARTLEY TRANSFORM

O. Alshibami and S. Boussakta, The University of Leeds, UK

The three-dimensional discrete Hartley transform (3-D DHT) has been proposed as an alternative tool to the 3-D discrete Fourier transform (3-D DFT) for 3-D applications when the data is real. The 3-D DHT has been applied in many three-dimensional image and multidimensional signal processing applications. This paper presents a fast three-dimensional algorithm for computing the 3-D DHT. The mathematical development of this algorithm is introduced and the arithmetic complexity is analysed and compared to related algorithms. Based on a single butterfly implementation, this algorithm is found to offer substantial savings in the total number of multiplications and additions over the familiar row-column approach.

ACCURATE SURFACE RECONSTRUCTION FROM APPARENT CONTOURS

Federico Pedersini, Augusto Sarti and Stefano Tubaro, Dip. Politecnico di Milano, Italy

In this paper we propose a 3D reconstruction technique based on the analysis of the extremal boundaries of the imaged surface. The method correctly determines the correspondences between apparent (viewer-dependent) contours and accurately determines location, orientation and curvature of the surface in the proximity of the apparent rims.

MULTI-RESOLUTION 3D RECONSTRUCTION THROUGH TEXTURE MATCHING

Federico Pedersini, Augusto Sarti and Stefano Tubaro, Politecnico di Milano, Italy

In this paper we present a general and robust approach to the problem of close-range partial 3D reconstruction of objects from multi-resolution texture matching. The method is based on the progressive refininement of a parametric surface, which is described using an increasing number of radial functions.

EFFICIENT ALGORITHMS FOR THREE-DIMENSIONAL FILTERING AND LINEAR PREDICTION

George-Othon Glentis, Technological Education Institute of Krete, Greece

In this paper optimal MSE three-dimensional (3-D) filtering and linear prediction is considered. 3-D filter masks of general shape are allowed. An efficient multichannel embedding is utilized that transforms the 3-D model to an equivalent multichannel formulation. Efficient recursions are developed for updating of lower order filter parameters towards neighboring points. Fast algorithms are applied for the solution of the 3-D normal equations in an order recursive way.

DETECTION OF BIDIMENSIONAL SHAPES UNDER GLOBAL DEFORMATIONS

J.M. Gonzalez-Linares and N. Guil, E.L. Zapata, University of Malaga

In this work a regularization of a bidimensional shape detection algorithm is presented. The algorithm is based in the Generalized Hough Transform (GHT). New and more robust coefficients to measure the accuracy of the algorithm are developed. These new coefficients are used to extend the algorithm in order to detect small global deformations. This approach can be used to obtain an initial template placement for active contours methods. Another option is to use the GHT as an objective function that must be minimized in order to fit the template to the deformed object.

AUTOMATIC SHAPE RECONSTRUCTION OF RIGID 3-D OBJECTS FROM MULTIPLE CALIBRATED IMAGES

Gerald Eckert, University of Hannover, Germany

We present a method for reconstructing the shape of a rigid 3-D object based on multiple calibrated images from an arbitrary camera setup. The algorithm is robust and able to reconstruct concavities. It is based on the idea to reconstruct a photo-consistent shape by carving and coloring voxels from a volume description of the object. Coloring and carving steps are carried out with respect to a color verification from the input images for each visible voxel. As compared to other approaches, the threshold used for the color verification is adapted automatically. Information included in the input images is processed and applied simultaneously without any explicit integration or assignment to the volume. The approach is evaluated using camera images of a real scene.

HYBRID METHOD FOR STILL IMAGE CODING

A. Zergaïnoh and J.-P. Astruc, Université de Paris Nord, France

This paper proposes an image coding method combining two transformation techniques. The first transformation consists in extracting a fraction of pixels using an adaptive sub-sampling algorithm. The second transformation encodes efficiently the map of positions, covariance model and gray level values of sub-sampled image. Simulation results present the rate-distortion performance of the image compression method.

A ROBUST IMAGE VECTOR QUANTIZATION TRANSMISSION SYSTEM WITH AN EMBEDDED DEGRADATION CONTROL MECHANISM

R. Iordache, Tampere University of Technology, Finland
A. Beghdadi, Universite Paris XIII, France

This paper proposes a vector quantization transmission system that both has a mechanism for detection and correction quantization artifacts, and is channel-robust, in the sense that provides some degree of immunity against channel noise. The basic idea of the source coding scheme is to treat separately the blocks whose direct quantization generates a high level of degradation. They are reconstructed via an interpolating procedure that exploits the spatial correlation in the image. The protection against the channel noise is provided by an appropriate index assignment procedure, based on Hadamard transform.

A NOVEL APPROACH FOR AUTOMATIC AIRCRAFT DETECTION

Wei Yi Stephen, Marshall University of Strathclyde, UK

The automatic recognition of planes in aerial images is an important application in the image analysis field. However, it remains a problem despite many years of work due to the arbitrary original poses and the variation in the shapes of planes. This paper proposes a novel approach for automatic aircraft detection based on statistical theory and common features of different kinds of planes. Experiments show its advantage in both speed and accuracy. It is another application of statistical theory applied to image processing.

FriAmPO3
Exhibition Hall
Multimedia System Designs and Applications
Chair: C. Regazzoni ***University of Genova, Italy***

ARCHITECTURES FOR ARITHMETIC CODING IN IMAGE COMPRESSION

Roberto R. Osorio and Javier D. Bruguera, University of Santiago de Compostela, Spain

In this work we present and evaluate new architectures for the arithmetic encoding and decoding of multilevel images. Arithmetic coding is of great interest due to the excellent results that it gives. On the other hand, the complexity of its implementation has always gone against it and its different applications usually suffer from a high computational cost, slowness or both. By introducing a new memory scheme, based on a cache memory, we solve the classic inconveniences of multilevel arithmetic codification hardware, obtaining architectures that are simpler and faster than the previous ones.

LOW BIT-RATE CODEC BASED ON LAR METHOD FOR VIDEO SURVEILLANCE VIA INTERNET

L. Bedat, O. Deforges, T. Landais and J.L. Corre, UMR IRER, INSA, France

This paper presents a full codec for a distributed video surveillance system. It is based on a new method called LAR, mixing both spatial and spectral approaches. The spatial coder provides a low resolution image but preserving objects boundaries, whereas the spectral coder can add the local texture information. The encoding scheme is progressive, providing large flexibility for rate/quality trade-off.

SELECTION OF NATURAL SCALE IN DISCRETE WAVELET DOMAIN USING EIGENVALUES

Azhar Quddus and Moncef Gabbouj, Tampere University of Technology, Finland

In this paper we present a novel technique for the selection of global natural scale from discrete wavelet transform. Here we define natural scale as the level associated with most prominent (dominant) eigenvalue. This technique is iterative and does not require full decomposition before finding the optimal wavelet level.

HIGH-RESOLUTION COLOR IMAGE RECONSTRUCTION FROM COMPRESSED VIDEO SEQUENCES

Javier Mateos, Rafael Molina, University of Granada, Spain
Aggelos K. Katsaggelos, Northwestern University, USA

In this work we propose an algorithm for the estimation of high resolution color frames from a low resolution compressed color video sequence. The algorithm exploits the existing correlation between the high and low resolution frames to obtain a high resolution frame reducing the artifacts introduced by the compression process. The performance of the proposed algorithm is demonstrated experimentally.

A FUZZY SYSTEM FOR EMOTION CLASSIFICATION BASED ON THE MPEG-4 FACIAL DEFINITION PARAMETER SET

Nicolas Tsapatsoulis Kostas Karpouzis, George Stamou, Frederic Piat and Stefanos Kollias
Affiliation: Department of Electrical and Computer Engineering National Technical University of Athens

The human face is, in essence, an advanced expression apparatus; despite its adverse complexity and variety of distinct expressions, researchers have concluded that at least six emotions, conveyed by human faces, are universally associated with distinct expressions. In particular, sadness, anger, joy, fear, disgust and surprise form categories of facial expressions that are recognizable across different cultures. In this work we form a description of the six universal facial expressions, using the MPEG-4 Facial Definition Parameter Set (FDP). According to the MPEG-4 Standard, this is a set of tokens that describe minimal perceptible actions in the facial area. Groups of such actions in different magnitudes produce the perception of expression. A systematic approach towards the recognition and classification of such an expression is based on characteristic points in the facial area that can be automatically detected and tracked. Metrics obtained from these points feed a fuzzy inference system whose output is a vector of parameters that depicts the systems' degree of belief with respect to the observed emotion. Apart from modeling the archetypal expressions we go a step further: by modifying the membership functions of the involved features according to the activation parameter we provide an efficient way for recognizing a broader range of emotions than that related with the archetypal expressions.

CONTOUR-BASED OBJECT RECOGNITION USING WAVELET-TRANSFORM

Faouzi Alaya Cheikh, Azhar Quddus and Moncef Gabbouj, Tampere University of Technology, Finland

In this paper we propose a new approach for shape recognition using the wavelet transform modulus maxima. And we apply it to the problem of content-based indexing and retrieval of fish contours. The description scheme and the similarity measure proposed here are simple and take into consideration the way our visual system perceives objects and compares them. The proposed scheme is invariant to translation, rotation, scale change and to noise corruption. Moreover, this description scheme allows accurate reconstruction of the shape boundary from the feature vector

used to describe it. The experimental results and comparisons show the good performance of the proposed technique.

LOW BIT-RATE VIDEO CODING USING SPACE-FREQUENCY ADAPTIVE WAVELET TRANSFORM

Kai Yang, NK-EXA Corporation
Tsuneo Saito, University of Tsukuba

In the still picture coding, we proposed a Space-Frequency Adaptive Coding (SFAC) approach, which is a zerotree based coder with a new optimization algorithm in the rate-distortion sense. In this paper, we extend the SFAC to video coding, in which the zerotree quantization is applied for both intraframe and interframe in order to trace energy compaction in motion residual image. To remove the temporal correlation efficiently, motion compensation is performed in image domain rather than in wavelet domain. The experimental results with the standard video image sequences, such as Football, Salesman and Claire, demonstrate that higher performances both for intraframe and interframe are achieved with zero-tree optimization. Compared to the conventional techniques, MPEG 2 and other wavelet video coders, 2-3 dB improvement in Football(SIF 760 kbps, 30 fps), 2-5 dB improvement in Salesman(QCIF 30 kbps, 10 fps) and 3-6 dB improvement in Claire(QCIF 30 kbps, 10 fps) can be obtained. The improvement is remarkable in lower bit-rate coding and the improved subjective quality can be observed in decoded image sequences.

FLEXIBLE SOFTWARE DECODING OF DIGITAL VIDEO

Irek Defee, Tampere University of Technology, Finland

Modern microprocessors and desktop systems are increasingly being designed for multimedia performance. Video, audio, and 3-D graphics processing require architecture which takes into account synchronous parallel processing of massive amounts of data in realtime. To deal with this type of processing multimedia insitruction sets are used. Multimedia processing poses also many problems on a system level. These problems are related to real time operation, task coordination and synchronization necessary for good perceptual effect. Typical example of multimedia processing is the decoding of digital video compressed in the MPEG-2 standard. In this paper we overview the problems of MPEG-2 decoding on standard microprocessors with multimedia instructions sets and different operating systems. We indicate that in a near future software-only digital television will be possible leading to deeper integration of broadcast and interactive services in the PC and set-top boxes.

THE VISUAL GOODNESS EVALUATION OF COLOR-BASED RETRIEVAL PROCESSES

Petteri Kerminen, Tampere University of Technology, Pori, Finland
Moncef Gabbouj Tampere University of Technology, Finland

A content-based image retrieval system helps users to find suitable images or videos from large databases for their purposes. In this paper we present the results of analyses of visual outputs from our image retrieval system, based on color matching. These analyses are based on human evaluation of color content similarity. Computer-based simulation results have been analyzed visually with the opinion scores method, i.e. how good the results are according to the human visual system. Tests have shown that there is no superior retrieval method that is best in every case. The goodness results of each retrieval method will be presented in this paper. The usability and usefulness of each system will be presented as well as a possible utilization domain.

EVALUATING CENTRALIZED AND DISTRIBUTED MULTI-AGENT ARCHITECTURES FOR VISUAL SURVEILLANCE APPLICATIONS

Luca Marchesotti, DIBE, Universita` degli studi di Genova, Italy
Paolo Remagnino and Graeme A. Jones, Kingston University, UK

The paper proposes the use of agent technology to implement the basic framework for distributed visual surveillance applications. A complex environment is usually monitored by a set of cameras with partially overlapping fields of view. A parametrically varied simulation of the camera data flow is used to evaluate implementations of distributed and centralized architectures. The performance evaluation on the basis of the synthetic data generated by the simulator was carried out.

VEHICLE EXTRACTION BASED ON FOCUS OF ATTENTION, MULTI FEATURE SEGMENTATION AND TRACKING

Andrea Cavallaro, Francesco Ziliani and Touradj Ebrahimi, Swiss Federal Institute of Technology, Switzerland
Roberto Castagno, Nokia Mobile Phones

In this paper, we propose an automatic object tracking method for traffic video surveillance. The method is designed as a hybrid between a region based and a feature based technique. In a first stage, a motion detection algorithm identifies the objects from the background providing binary masks of the moving objects. In a second stage, a segmentation tool based on a multi-feature analysis further segments the areas corresponding to moving objects into homogenous regions. For each region, the method provides a set of characteristic feature values, which are used to track the regions (and thus the objects) along time. The results of this low-level analysis can be exploited by the content understanding module of an advanced video surveillance system for the detection of potentially dangerous situations, for law enforcement purposes, and for statistical traffic analysis.

REGIONAL WATERMARKING IN SPATIAL DOMAIN

Dinu Coltuc and Philippe Bolon, University of Savoie, France

Histogram specification based marking considers as watermarks positive integer functions defined on the graylevel range. Image watermarking consists in exact histogram specification. Robustness against attacks is achieved when watermarks are almost zero on compact intervals, i.e., groups of graylevels are very little represented in the marked image. Two types of such watermarks are considered: sawtooth shaped and histogram with notches. This paper addresses histogram based watermarking in a regional context: instead of the whole image, only a region is marked. The histogram of the other region is exactly specified in order to preserve the image original histogram. The marked region is selected to be simply connex and it is assumed to contain essential details of the image. Two strategies for region extraction are discussed: the inner quarter of the image or a true region surrounded by edges. While the former is computationally very simple border effects do appears at marking, effects which are completely avoided by the latter strategy. Besides the statistical invisibility of the method, the results obtained so far exhibits robustness properties.

TERRAIN FEATURE CLASSIFICATION IN SAR IMAGERY

Adrian G. Bors, Edwin R. Hancock and Richard C. Wilson, University of York, UK

This paper describes a maximum likelihood feature detector for extracting terrain features from Synthetic Aperture Radar (SAR) images. We commence by deriving the probability distribution for SAR image amplitude. According to our model, the SAR image amplitude follows a product of Rayleigh and Bessel functions distribution. We show how the two parameters of this model can be estimated using robust statistics. With the model to hand we develop a maximum likelihood feature detector. Eventually, we classify the detected terrain features in ridges or ravines.

ILLUMINATION INVARIANT OBJECT RECOGNITION USING THE MNS METHOD

D. Koubaroulis, J.Matas and J.Kittler, University of Surrey, UK, CTU Prague, Czech Republic

The suitability of the Multimodal Neighbourhood Signature (MNS) method for illumination invariant recognition is investigated. The MNS algorithm directly formulates the problem of extracting illumination invariants from local colour appearance of an object. The invariants are the channel-wise ratio and the cross-ratio computed from modes (pairs of modes respectively) of colour density function in neighbourhoods with multimodal density function. The MNS algorithm is tested on a colour object recognition task designed to test the effectiveness of algorithms claiming illumination invariance properties. The image set used is publicly available from the Simon Fraser University. Results previously reported using colour constancy and histogram matching were comparable to the performance of the presented method that achieved recognition rate of 60%. When the pose of the objects was fixed recognition performance was 84%.

INDEXING SPOKEN AUDIO BY LSA AND SOMS

Mikko Kurimo, Helsinki University of Technology, Finland

This paper presents an indexing system for spoken audio documents. The framework is indexing and retrieval of broadcast news. The proposed indexing system applies latent semantic analysis (LSA) and self-organizing maps (SOM) to map the documents into a semantic vector space and to display the semantic structures of the document collection. The SOM is also used to enhance the indexing of the documents that are difficult to decode. Relevant index terms and suitable index weights are computed by smoothing the document vectors with other documents which are close to it in the semantic space. Experimental results are provided using the test data of the TREC's spoken document retrieval track.

FriAmOR5
Room 200
Image Quality Measures
Chair: E. Coyle *Motorolla, USA*

11:00

STATISTICAL ANALYSIS OF IMAGE QUALITY MEASURES

Ismail Avcibas and Bülent Sankur, Bogaziçi University, Turkey

In this paper we conduct a statistical analysis of the sensitivity and consistency behavior of objective image quality measures. We categorize the quality measures and compare them for still image compression applications. The measures have been categorized into pixel difference-based, correlation-based, edge-based, spectral-based, context-based and HVS-based (Human Visual System-based) measures. The mutual relationships between the measures are visualized by plotting their Kohonen maps. Their consistency and sensitivity to coding as well as additive noise and blur are investigated via ANOVA analysis of their scores. It has been found that measures based on HVS, on phase spectrum and on multiresolution mean square error are most discriminative to coding artifacts.

11:20

A PERCEPTUAL PSNR BASED ON THE UTILIZATION OF A LINEAR MODEL OF HVS, MOTION VECTORS AND DFT-3D

M. Caramma, R. Lancini and M. Marconi, CEFRIEL-Politecnico di Milano, Italy

In this paper we propose two different methods to obtain a video sequence PSNR taking into account of the subjective quality evaluation of the final users. The proposed "subjective" PSNRs are based on the spatio-temporal human sensitivity method suggested by Z. L. Budrikis and on the experimental results given by Kelly and Robson. The first metric (WPSNR1) consists in a PSNR weighed by the Motion Vectors typical of MPEG 1-2 algorithms; the second one (WPSNR2) is obtained by mean of the Discrete Fourier Transform applied to a three dimensional domain. Experimental results show that these two WPSNRs are more related to the subjective quality perceived by the final user than the usual PSNR.

11:40

AN EFFICIENT METHOD FOR OBJECTIVE AUDIO AND VIDEO QUALITY ASSESSMENT IN DIGITAL TELEVISION APPLICATIONS

Nathalie Montard, Alexandre Joly, Pierre Britillon and Jamal Baona, TDF-C2R, CPE-Lyon/LISA

The quality of service that is provided to the end-user is a very important performance criterion in television. Traditional automatic mesasurement techniques used on analogue signals are not applicable to digital ones. This paper presents a method to monitor audio and video quality, applicable in the context of a digital broadcasting network. Because of bandwidth limitations, most existing methods cannot be applied. A method based on the evaluation of several of the impairments typically encountered in digital MPEG audio and video signals is proposed. The method has also been implemented, and works in real time. Performance results in various conditions using the prototype device are presented, as well as various applications.

12:00

OBJECTIVE EVALUATION OF SEGMENTATION MASKS IN VIDEO SEQUENCES

Xavier Marichal, Universite catholique de Louvain, Belgium
Paulo Villegas, Telefonica I+D

In order to avoid the charge of visually evaluating the quality of segmentation masks, the present paper introduces a procedure for evaluating this quality by means of algorithmically computed figures of merit. Assuming the existence of a perfect (reference) mask, generated manually or with a reliable procedure over a test set, these figures of merit take into account visually desirable properties of a segmentation mask in order to provide the user with outputs that best qualify the spatial and temporal accuracy of the segmentation masks. For the sake of easy interpretation, results are presented on a PSNR-like logarithmic scale.

12:20

BLOCKWISE DISTORTION MEASURE FOR LOSSY COMPRESSION OF MULTISPECTRAL IMAGES

Arto Kaarna, Lappeenranta University of Technology, Finland
Jussi Parkkinen, University of Joensuu, Finland

A new blockwise distortion measure is proposed to evaluate the quality of lossy compressed multispectral images. The measure is based on blockwise distortions which are calculated between the original multispectral image and the compressed/reconstructed multispectral image. The calculated measures with various compression ratios are matched to a visual scale which is based on the mean opinion score (MOS). The nonlinear weighting using neural network is used for matching. The results from the new measure are compared to mean-square-error (MSE) based measures, to two-dimensional blockwise distortion measure, and to picture quality scale for grayscale images. The multispectral images are compressed using multiwavelets, principal

component analysis, and DCT/JPEG.

FriPmSS1
Hall A
DSP for Virtual Acoustics
Chair: M. Karjalainen *Helsinki University of Technology, Finland*

14:00

VIRTUAL ACOUSTICS - APPLICATIONS AND TECHNOLOGY TRENDS

Jyri Huopaniemi, Nokia Research Center
Riitta Väänänen, Lauri Savioja and Tapio Lokki, Helsinki University of Technology, Finland

Virtual acoustics is a general term for the modeling of acoustical phenomena and systems with the aid of a computer. It comprises many different fields of acoustics, but in the context of this paper the term is restricted to describe a system ranging from sound source and acoustics modeling in rooms to spatial auditory perception simulation in humans. In other words, virtual acoustics concept covers three major subsystems in acoustical communication: source, transmission, and the listener. This paper discusses virtual acoustics from the point of view of applications and technology trends.

14:40

SPATIAL PROCESSING OF SOUND IN MPEG-4 VIRTUAL WORLDS

Riitta Väänänen, Helsinki University of Technology, Finland
Jyri Huopaniemi, Nokia Research Center

MPEG-4 is a standard for interactive multimedia applications, where tools are specified for presenting sound in 3-D virtual scenes. This includes parametric representations of 3-D sound source models and the environment where the sound is heard. This article gives an overview on how interactive sound environments are parametrized in the scene description language of MPEG-4. Furthermore, DSP issues for the implementation of the acoustic environment are discussed.

15:00

EFFICIENT HRTF SYNTHESIS USING AN INTERAURAL TRANSFER FUNCTION MODEL

Gaëtan Lorho, Jyri Huopaniemi, Nick Zacharov and David Isherwood, Nokia Research Center, Finland

We propose an alternative method for binaural synthesis using the ratio of the HRTFs at the two ears, the interaural transfer function (ITF), for deriving the response at the contralateral ear from the ipsilateral one. This technique allows an optimal binaural synthesis if an accurate ITF model is used but a computationally more efficient binaural synthesis can also be obtained by

simplifying the ITF filter used for the contralateral channel processing, which is perceptually less critical than the ipsilateral response. Different filter designs and filter orders are investigated for both speech and wide-band signals. Subjective tests are carried out to investigate the perceptual differences introduced by this ITF model. The results indicate that an ITF filter can be of considerably lower complexity when compared to the ipsilateral filter, thus allowing for computational savings without significant perceptual degradation.

15:50

TRANSFER FUNCTION MODELS WITH NONLINEAR EXCITATIONS FOR DIGITAL SOUND SYNTHESIS

Lutz Trautmann and Rudolf Rabenstein, Telecommunications Laboratory University of Erlangen-Nuremberg, Germany

Transfer functions for digital sound synthesis based on physical models have recently been presented. The method transforms a continuous model for the vibrating body, given by a partial differential equation (PDE), into a multidimensional transfer function model (TFM). The TFM not only takes initial and boundary conditions, as well as excitation functions into account, but also treats the physical effects modeled by the PDE exactly. The algorithms obtained after discretization of the TFM preserve the inherent physical stability and are suitable for real-time implementations on digital signal processors. A recently presented example of a linear transversal oscillating tightened string with frequency dependent loss terms is extended here by nonlinear excitations. They are modeled from real bow-string and hammer-string interactions in violins and pianos.

16:10

SOUND SIGNAL PROCESSING FOR A VIRTUAL ROOM

Jarmo Hiipakka, Tommi Ilmonen, Tapio Lokki and Lauri Savioja, Helsinki University of Technology, Finland

This paper presents the audio system built for the virtual room at Helsinki University of Technology. First we discuss the general problems for multichannel sound reproduction caused by the construction of, and the equipment in virtual rooms. We also describe the acoustics of the room in question, and the effect of the back-projected screens and reverberation to the sound. Compensation of the spectral deficiencies and the problems with the large listening area and high frequency attenuation are introduced. The hardware configuration used for sound reproduction is shortly described. We also report the software applications and libraries built for sound signal processing and 3-D sound reproduction.

16:11

EFFICIENT PHYSICS-BASED SOUND SYNTHESIS OF THE PIANO USING DSP METHODS

Balazs Bank, Vesa Valimaki and Matti Karjalainen,Helsinki University of Technology, Finland
Laszlo Sujbert, Budapest University of Technology and Economics, Hungary

In the recent years, digital waveguide modeling of musical instruments has proven to be an effective tool for sound synthesis purposes, but some practical questions still have remained unanswered. In this paper a new equivalent structure of the digital waveguide for string synthesis is presented. This structure can be used for highly efficient modeling of beating and two-stage decay, an important characteristic of the piano sound. The complexity of the traditional structure can be reduced by replacing one of the string models with a resonator bank.

16:12

MODELING AND MODIFICATION OF VIOLIN BODY MODES FOR SOUND SYNTHESIS

Jan-Markus Holm and Vesa Välimäki, Helsinki University of Technology, Finland

This paper discusses a new technique for obtaining efficient computational models of violins that are classified as being of different quality. The motivation of the work is the desire for parametric adjustment of the transfer characteristics of the violin body changing the instrument quality, and giving realistic and efficient body resonance models by signal processing methods. Model-based signal processing methods for estimating the model parameters and implementing an efficient violin body model for use with physical modeling synthesis is described. The main approach is to extract the lowest modes of the violin body for fully parametrized implementation of these resonances using digital filtering. The advantage of this approach is the capability of adjusting the parameters of body modes, such as the resonance frequencies affecting the sound quality in the violin sound synthesis.

16:13

A DIGITAL FILTERING APPROACH TO OBTAIN A MORE ACOUSTIC TIMBRE FOR AN ELECTRIC GUITAR

Henri Penttinen, Vesa Välimäki and Matti Karjalainen, Helsinki University of Technology, Finland

In this paper we propose signal processing methods to transform electric guitar tones to sound more acoustic. This is achieved by applying digital filtering to the signal obtained from a pickup of an electric guitar. The electric guitar differs from the acoustic one structurally and in the way the final acoustic radiation is produced. Hence, the timbres of the two guitars are distinguishably different. We use two digital filter designs to transform the electric guitar tones to resemble an acoustic guitar. The first one relies on impulse response measurements of the body of an acoustic guitar and the second method builds on deconvolving two spectrally rich signals. We also

accomplish an improved controllability over the final timbre by extracting and replacing the lowest resonances with IIR resonators. The presented methods achieve to give the magnetic pickup tone an acoustic guitar-like timbre. The perceived responses have a distinct soundbox effect and simulate the important lowest resonances between 80 and 200 Hz of the acoustic guitar and the reverberant response of the body

16:14

IMPLEMENTATION OF A GENERAL-PURPOSE AURALIZER ON A FLOATING-POINT SIGNAL PROCESSOR

Juha Merimaa and Matti Karjalainen, Helsinki University of Technology, Finland

The design and implementation of a real-time general-purpose auralization device is presented in this paper. The goal has been to make a stand-alone device that can also be controlled from a PC. Several auralization and spatialization methods are included, such as binaural headphone and loudspeaker auralization, and vector base amplitude panning for multi-channel loudspeaker sound reproduction. Combining these to a single device allows easy comparison and choosing the best method for any purpose. The different auralization principles and their implementations using the SHARC digital signal processor are described.

16:15

COMPUTATIONAL QUALITY ASSESSMENT OF HRTFS

Klaus A J Riederer, Helsinki University of Technology, Finland

The verification of quality in head-related transfer function (HRTF) measurement data holds a complex problem field, involving acoustic (background noise), physical (head movement, different head shape) and electrical (DC-offset) deviations. The disrupted measurements may be difficult to separate from the correct ones, because HRTFs are inherently highly dependent on the person, direction, and frequency. Since there exists no "absolute truth" what a HRTF should be like, deviated measurements can be assessed from the typical HRTF shape, i.e., the common trend of the measured HRTFs. This paper presents an approach to calculate an objective deviant measure of HRTFs by DSP means. The perceptually justified deviance index is applied for the evaluation of a HRTF measurement system and a HRTF database.

FriPmSS2
Room 200
Positioning and Navigation
Chair: J. Syrjärinne ***Nokia Mobile Phones, Finland***

14:00

DESIGN CONSIDERATIONS FOR POSITIONING SYSTEMS IN URBAN ENVIRONMENTS

Paul W. McBurney, eRide, Inc., San Francisco, USA

Ranging signals in the urban environment suffer from significant attenuation and additive errors caused by reflections. Special design considerations for ranging-based positioning systems that operate in the Urban Canyon are presented. The tradeoff between acquisition sensitivity and acquisition time is presented. The integrity of the measurements is improved by verifying that the signals exhibit the characteristics of the signal structure. The concept of self-aiding using ranging or position fix information to acquire further signals is discussed. Measurement and position fix integrity schemes are discussed which either reject the lower quality measurements due to reflections when possible, or properly adopt them when no other redundant information is present.

14:40

RAPID ZERO-KNOWLEDGE GPS SIGNAL ACQUISITION

Ville Eerola, Janne Takala, and Tapani Ritoniemi, VLSI Solution Oy

A GPS acquisition system which achieves extremely short search times is presented. The system requires no prior knowledge of position or time. It operates autonomously, and is based on a parallel matched filter structure. Using 50 MHz clock frequency, 25 satellites can be searched in parallel. The acquisition system has been implemented in 0.35 um CMOS process and will be used as a building block for a GPS based sensor application. The matched filter based implementation provides orders of magnitude improvement over traditional serial search based on correlator implementations. Furthermore, the parallel architecture offers a significant acquisition speed improvement over a single satellite search approach. Under nominal signal conditions, 2 second total search times have been demonstrated. The acquisition unit performance has been verified both with live satellites as well as with a GPS simulator. The acquisition threshold with a RF noise figure of 4 dB was found to be -134 dBm. The paper reviews the design and used algorithms.

15:00

BAYESIAN APPROACH TO SATELLITE SELECTION

Toni Vartiainen, Jukka Saarinen and Jari Syrjärinne, Tampere University of Technology, Finland

In this paper, Bayes' theorem is used in selection of global positioning system (GPS) satellites. By selecting the most reliable satellites among all visible satellites, system performance can be improved by fusing together information from multiple sources. This requires reliable modeling of the satellites by means of probabilities and probability distributions. Reliable satellite modeling requires a comprehensive set of parameters that gives the best possible satellite state representation. A measure of satellite appropriateness can then be calculated by applying Bayesian probabilistic reasoning to a parameter combination. Performance of the suggested method is verified by simulations.

15:50

ON RELATIVE POSITIONING

Paula Korvenoja and Jari Syrjärinne, Research & Technology Access Nokia Mobile Phones

The problem of relative positioning is introduced and some solutions are proposed. The topic is studied as applied to the Global Positioning System (GPS). Unlike usual GPS positioning, in relative positioning, the navigator's location in some Earth-bound coordinates is not the desired result, but instead, the position of one GPS user relative to another. The relative position is given in user-centered coordinates. Rather than determining the relative position vector through explicit calculation of both navigator locations, alternative methods are developed to serve specifically relative positioning. The purpose is to reduce the needed amount of computations in this particular problem and to enhance accuracy. The former objective is met by algebraic manipulations of the positioning equations and the latter one is achieved by making use of the typical properties of GPS measurement errors.

16:10

PEDESTRIAN WALKING DISTANCE MEASUREMENTS USING HAND HELD GPS RECEIVER

Pasi Viitanen and Pentti Mattila, VTT Automation, Finland

This paper presents proposed walking distance measurement method for hand-help GPS receivers. The method is based purely on GPS receiver output. The following requirements can be set for hand held GPS receiver distance measurement: * The operation of the distance measurement must be autonomous. * The distance measurement must work under any condition, where the receiver is operable * The distance measurement is based purely on receiver output * The distance measurement must be accurate enough, so that the user feels convenient using it The preliminary test results are presented.

16:30

TERMINAL POSITIONING IN WCDMA

Kari Kalliojärvi, Nokia Research Center, Finland

This paper provides an overview of the terminal positioning capabilities in Wideband CDMA (WCDMA). The current situation in standardization is covered and the basic terminal positioning methods applicable in WCDMA are discussed. Performance evaluations by simulations of the selected solutions are provided at the end of the paper.

16:50

WLAN EVALUATION OF RSSI-BASED HUMAN TRACKING

Juha Latvala, Hannu Ikonen, and Jarkko Niittylahti, Tampere University of Technology, Finland
Jari Syrjärinne, Nokia Mobile Phones, Finland

This paper evaluates a location and tracking system built on an existing wireless local area network (WLAN). The tracking system is implemented without any additional hardware and is implementable in any WLAN environment with minor verification of basestation positioning. The tracking system utilizes the easily extractable signal strength values of the WLAN in order to calculate a position estimate for the observed target. This tracking system is designed to work within predefined surroundings, such as offices, hospitals, prisons etc., with an accuracy of one room.

17:10

A POSITION AWARE INFORMATION APPLIANCE

Giuliano Benelli, Alberto Bianchi and Michelangelo Diligenti, Universita' degli Studi di Siena, Italy

The information related to the user actual and previous position is a very important issue for research activities on new interaction paradigms navigating physical spaces. This information is obtained from a coherent use of infrared TXs and a developed software which computes the probability that the user is located and oriented in a specific way. A number of IR TXs are fixed to walls around the rooms, and an IR receiver is attached to the user handing a PDA. The received signals are then processed, on the PDA, to derive the user's location and orientation. Combinations of IR TXs can be used to discover location, distance and orientation. A standalone C++ application on the PDA, called Barriera, reads IR codes and processes the stream of codes to remove spurious readings. Hence Barriera does a probabilistic calculation on the readings to infer the orientation and location of the user.

FriPmSS3
Studio
Multimedia Systems: Integration and Implementation
Chair: I. Defee ***Tampere University of Technology, Finland***

14:00

A DYNAMICALLY RECONFIGURABLE SYSTEM-ON-A-CHIP ARCHITECTURE FOR FUTURE MOBILE DIGITAL SIGNAL PROCESSING

Jürgen Becker and Manfred Glesner, Darmstadt University of Technology, Germany
Ahmad Alsolaim and Janusz Starzyk, Ohio University, USA

The evolving of current and future broadband access techniques into the wireless domain introduces new and flexible network architectures with difficult and interesting challenges. The system designers are faced with a challenging set of problems that stem from access mechanisms, energy conservation, error rate, transmission speed characteristics of the wireless links and mobility aspects. This paper presents first the major challenges in realizing flexible microelectronic system solutions for digital baseband signal processing in future mobile communication applications. Based thereupon, the architecture design of flexible system-on-a-chip solutions is discussed. The focus of the paper is the introduction of a new parallel and dynamically reconfigurable hardware architecture tailored to this application area. Its performance issues and potential are discussed by the implementation of a flexible and computation-intensive component of future mobile terminals.

14:20

A SYSTEM-ON-A-CHIP FOR MULTIMEDIA STREAM PROCESSING AND COMMUNICATION

E. Juárez, M. Mattavelli and D. Mlynek, Swiss Federal Institute of Technology, Switzerland

In this paper, an architecture for multimedia stream processing and communication is presented. The system is aimed to communicate MPEG4 multimedia applications over generic networks. Main features of the architecture are scalability in the number of multimedia streams managed, bandwidth sharing, capacity to control the offered service quality and possibility to implement mobile applications. A chip area network (CIAN) is used to interconnect the different architectural elements. Four main units are distinguished in the design: cell communication, QoS control, protocol processing and DMA (Direct Memory Access). Preliminary results of an implementation of the cell communication unit as an ATM-cell-based multiplexing one show the suitability of the architecture for STS-12/STM-4/OC-12 throughputs (622.08 Mb/s).

14:40

HARDWARE IMPLEMENTATION OF THE IMPROVED WEP AND RC4 ENCRYPTION ALGORITHMS FOR WIRELESS TERMINALS

Panu Hämäläinen, Marko Hännikäinen, Timo Hämäläinen, and Jukka Saarinen, Tampere University of Technology, Finland

This paper presents hardware implementations for Improved Wired Equivalent Privacy (IWEP) and RC4 ("Ron's Cipher #4") encryption algorithms. IWEP is a block algorithm providing light-strength encryption. The algorithm has been designed for a new Wireless Local Area Network (WLAN), called TUTWLAN (Tampere University of Technology Wireless Local Area Network). On the contrary RC4, developed by RSA Data Security, Inc., is a powerful stream algorithm used in many commercial products. It is also utilized in the Wired Equivalent Privacy (WEP) standard algorithm for WLANs. The objective of this work has been to study the suitability of hardware implementation for these previously software-implemented ciphers. Hardware is needed to replace software especially in wireless multimedia terminals, in which real-time data processing and limited on-chip memory sizes are key elements. The implementations are made in Very high-speed integrated circuit Hardware Description Language (VHDL) on Xilinx Field Programmable Gate Array (FPGA) chips.

15:00

A HARDWARE ORIENTED ANALYSIS OF CRYPTOGRAPHIC SYSTEMS FOR MULTIMEDIA APPLICATIONS

A. Romeo and M. Mattavelli, Swiss Federal Institute of Technology Integrated Systems Laboratory, Switzerland

This paper presents a hardware-oriented complexity analysis of some of the most popular cryptographic algorithms. A sample algorithm has been chosen from each group in which cryptographic algorithms are usually classified: factorization problem, discrete logarithm problem, stream cipher, block cipher and hybrid systems. The analysis yields measures of the types and number of operations necessary to perform each algorithm providing a comparison among the different classes. Such analysis is particularly oriented toward the encryption of large volumes of data such as the ones found in multimedia applications. Moreover, these results suggest how significant increase in performance can be provided by specific hardware implementations.

15:50

MULTITHREADING FOR VIDEO PROCESSING APPLICATIONS RUNNING ON PC WORKSTATIONS

Eric Debes and Fulvio Moschetti, Swiss Federal Institute of Technology, Switzerland

The aim of this paper is to show how an efficient exploitation of the multithreading technique can speed up video applications running on PC Workstations. We show how to use

multithreading for pure processing in order to exploit the full computational power of multiprocessor machines. Different multithreading techniques are compared and the efficiency of multithreading is discussed. It is explained why multithreading can be a very helpful technique to avoid CPU idle state while reading and writing video to disk or displaying video on the screen. Disk reading is chosen as an example to give the gain that can be obtained using multithreading for input/output operations. Finally it is shown how multithreading can be used to take advantage of the external processing power available in graphic cards or acquisition boards. Indeed, very often in video processing applications, some routines can be done in hardware while another thread is running in parallel in order to keep the main CPU busy.

16:10

REUSEABLE INTERFACE IN MULTIMEDIA HARDWARE ENVIRONMENT

Vesa Lahtinen, Kimmo Kuusilinna, Timo Hämäläinen, Tero Kangas, and Jukka Saarinen,
Tampere University of Technology, Finland

DSP and particularly multimedia systems are among the most rapidly developing areas of digital IC (Integrated Circuit) design. Multimedia hardware designers face the same time-to-market problems as the rest of the IC designers but, in addition, a large number of data transfers is required. In this paper we utilize a novel IP (Intellectual Property) interface solution based on our Heterogeneous IP-Block Interconnection (HIBI) scheme. The purpose of this scheme is to simplify design reuse in multimedia system development by acting as an efficient interconnection between heterogeneous IP-blocks. As an example, a multimedia hardware environment based on the HIBI scheme is presented. This paper shows how designing the environment out of reuseable blocks complying with the HIBI scheme is straightforward, and the result is still flexible and efficient. The performance metrics of the environment are studied with the aid of a video encoder example. With the encoder, real-time operation for QCIF (Quarter Common Intermediate Format) video can be achieved.

16:30

ADAPTIVE AND INTEGRATED VIDEO COMMUNICATION FOR WIRELESS ATM

Jozsef Vass, Eyeball.com Network Inc.
Xinhua Zhuang, University of Missouri-Columbia

Highly efficient and robust source coding, channel coding, and packetization techniques are proposed for video streaming over wireless ATM. At the base station, which is located at the boundary of the wireline and wireless networks, video received from the wireline source is dynamically transformed to match both the hardware capabilities of mobile hosts and the time-varying wireless channel conditions. For wireless transmission, source coding, channel coding, and packetization are jointly implemented as part of the application. For source coding, we propose to use our three-dimensional significance-linked connected component analysis video codec. For channel coding and packetization, both intracell and interlaced (intercell) forward error correction are applied. Furthermore, the time-varying channel characteristics is exploited by adaptively allocating the total bit budget between source coding and channel coding.

Extensive performance evaluation demonstrates the effectiveness of the proposed wireless video streaming technique.

16:50

ADVANCED PROTOTYPE PLATFORM FOR A WIRELESS MULTIMEDIA LOCAL AREA NETWORK

Kimmo Tikkanen, Marko Hännikäinen, Timo Hämäläinen, and Jukka Saarinen, Tampere University of Technology, Finland

This paper presents an advanced version of a configurable demonstrator platform developed for a new wireless local area network called TUTWLAN. TUTWLAN is targeted for limited service areas with stationary or portable terminals. Applications range from simple wireless sensors to multimedia laptops. The network supports the different Quality of Service (QoS) requirements of these applications. The improved development platform has been designed because of the restrictions discovered in the first prototype. The new platform provides better testing environment for developing Medium Access Control (MAC) protocols for TUTWLAN and for designing embedded stand-alone applications. Furthermore, various other designs can be tested, for example hardware implementations of encryption algorithms. Both the new and the old prototypes consist of a Digital Signal Processor (DSP), external memory modules for the DSP, and a Field Programmable Gate Array (FPGA) circuit. The platform is connected to a radio module and can be attached to a host computer using Peripheral Component Interconnect (PCI) bus. Compared to the original platform, the new prototype contains more memory, a faster and larger FPGA, and a higher bit-rate radio.

17:10

CONTENT DELIVERY AND MANAGEMENT IN NETWORKED MPEG-4 SYSTEM

Florin Lohan and Irek Defee, Digital Media Institute, Finland
Roberto Castagno and Serkan Kiranyaz, Nokia Mobile Phones

Networking of MPEG-4 content is the topic of standardization efforts in the ISO MPEG and IETF. DMIF (Delivery Multimedia Integration Framework) was developed in the ISO MPEG as a generic networking platform for MPEG-4 content. In this paper we are describing the implementation of a MPEG-4 client-server architecture, uses DMIF with Internet RTSP (Real Time Streaming Protocol) for signaling. Mapping of DMIF functionalities onto RTSP messages is shown. The implemented system runs on PC platform and its operation is explained in details.

FriPmOR1
Small Audiotorium
Signal Processing for Communications
Chair: D. Slock *Eurecom Institute, France*

14:00

SEMI-BLIND SPATIO-TEMPORAL CHANNEL IDENTIFICATION AND INTERFERENCE CANCELLATION IN TDMA CELLULAR SYSTEMS

Hafedh Trigui, Telecom MODUS Ltd.
Dirk T.M. Slock, Institut EURECOM

In wireless communications, spatial (via antenna arrays) and temporal (excess bandwidth) diversity may be exploited to simultaneously equalize a user of interest while canceling or reducing (cochannel) interfering users. This can be done using the classical Viterbi algorithm after a noise-plus-interference whitening operation. Cochannel interference cancellation for TDMA cellular systems remains an important issue in the migration from second to third generation systems such as EDGE. In fact, interference cancellation becomes even more important because denser symbol constellations get used and performance requirements increase as data services get introduced. The receiver depends on the channel for the user of interest, to be estimated with a training sequence, and contains a blind interference cancellation part. The critical part is the channel estimation. The usual least-squares method may lead to poor estimates in high interference environments. Significant improvements result from the Maximum-Likelihood and suboptimal techniques investigated here.

14:20

OPTIMIZING THE CAPACITY OF ORTHOGONAL AND BIORTHOGONAL DMT CHANNELS

P. P. Vaidyanathan, Caltech, USA
Yuan-Pei Lin, National Chiao Tung University, Taiwan
Sony Akkarakaran, Caltech, USA
See-May Phoong, National Taiwan University, Taiwan

The uniform DFT filter bank has been used routinely in discrete multitone modulation (DMT) systems because of implementation efficiency. It has recently been shown that principal component filter banks (PCFB) which are known to be optimal for data compression and denoising applications, are also optimal for a number of criteria in DMT communication. In this paper we show that such filter banks are optimal even when scalar prefilters and postfilters are used around the channel. We show that the theoretically optimum scalar prefilter is the half-whitening solution, well known in data compression theory. We conclude with the observation that the PCFB continues to be optimal for the maximization of theoretical capacity as well.

14:40

ADAPTIVE INTERFERENCE SUPPRESSION IN CDMA SYSTEMS BY LCL-PTV FILTERING

Giacinto Gelli, Luigi Paura, and Luisa Verdoliva, Università degli Studi Federico II di Napoli, Italy

This paper deals with narrowband interference suppression in direct-sequence code-division multiple-access (CDMA) systems. In order to take advantage of the cyclostationary nature of the interfering signal, we propose here to adopt a linear conjugate-linear (LCL) polyperiodically time-varying (PTV) filter in the suppression stage, whose adaptive implementation is based on the RLS algorithm. Unlike the conventional linear time-invariant filter, the LCL-PTV structure exploits also the spectral correlation properties of the interference, achieving therefore a better suppression performance. The numerical results confirm the superiority of the proposed LCL-PTV scheme with respect to the conventional one, especially when operating in the presence of strong interfering signals.

15:00

IMPROVED AND UNIFIED APPROXIMATION FOR THE BER OF LINEAR BLOCK CODES

Claude Desset, Fabrice Labeau, Luc Vandendorpe and Benoît Macq, Université catholique de Louvain, Belgium

In this paper, we provide an approximation for the BER achieved by using linear block error-correcting codes. Our work unifies and improves two anterior approximations, in the case of hard-decoding limited to the minimum distance of the code. We compare them to exact values for codes whose weight distribution is known. Our first improvement takes into account undecoded words, enabling the use of our approximation for all linear block codes, while the original ones were only accurate for perfect codes. A second improvement allows us to work in the large channel BER domain, by considering the binomial approximation of weight distributions. Previous approximations were not accurate, as asymptotic expressions based on low-weight error patterns do not hold when dealing with high BER values.

15:50

RLS-BASED INITIALIZATION FOR PER TONE EQUALIZERS IN DMT-RECEIVERS

Katleen Van Acker, Geert Leus, Marc Moonen, Katholieke Universiteit Leuven, Belgium
Thierry Pollet, Access to Networks Corporate Research Centre, Belgium

Per tone equalization has been proposed as an alternative to time domain equalization for DMT-based systems. In this paper, an iterative initialization scheme based on so-called RLS with inverse updating is presented for these equalizers. Simulation results show convergence with an acceptably small number of training symbols. Complexity calculations are made for per tone equalization and for the case where tones are grouped. It is demonstrated with an example that in

the latter case, initialization complexity becomes sufficiently low and comparable to complexity during data transmission.

16:10

DESIGN OF FIR FILTERBANK TRANSCEIVERS WITH EFFECTIVE BAND SEPARATION

Yuan-Pei Lin and See-May Phoong, National Chiao Tung University, National Taiwan University

The DMT (discrete multitone modulation) transceivers have been shown to be a very useful technique for data transmission over frequency selective channels. The DMT scheme is realized by a transceiver that divides the channel into subbands. The efficiency of the scheme depends on the frequency selectivity of the tranceiving filters. The filterbank transceiver or DWMT (discrete wavelet multitone) system, has been proposed as an implementation of DMT transceiver that has better frequency band separation. In this paper, we show how to design filterbank transceivers that have good frequency selectivity and at the same time cancel ISI completely.

16:30

SPECTRAL INTERPOLATION CODER FOR IMPULSE NOISE CANCELLATION OVER A BINARY SYMETRIC CHANNEL

Abraham Gabay, Pierre Duhamel and Olivier Rioul, Ecole Nationale Superieure des Telecommunications, France

In this paper, a spectral interpolation coder (SIC) and decoder are investigated for simultaneous source coding and impulse noise cancellation. For simplicity of the analysis, we restrict ourselves to the framework of scalar quantization of a memoryless gaussian source to be transmitted over binary symmetric channel (BSC). Our approach is to make a carefully designed interpolation of the data in the spectral domain prior to quantization and transmission. The SIC decoder then exploits the properties of SIC codes in order to analyse, detect and correct (or reduce) erroneous data. A nice feature of this procedure is that the decoder deals simultaneously with the quantization noise and impulse channel noise; therefore it is able to reduce distortion introduced not only by the transmission channels errors but also by the quantizer. A comparaison study is also investigated in this paper: Simulations show that we obtain a 3dB improvement in SNR over the classical TSC scheme for a global rate of 8.2 transmitted bits per source sample and small BSC crossover probability.

16:50

CODEBOOK INDEX ASSIGNMENT BY AN APPROXIMATE SOLUTION OF THE TRAVELING SALESMAN PROBLEM

A. Spira, R. Mayrench and D. Malah, Technion, Israel

Index Assignment (IA) is a process of indexing the vectors in a codebook (vector quantizer) for the purpose of reducing the distortion caused by transmission over a channel with errors. Achieving an optimal IA is difficult since it is an NP-complete problem. Common approximate solutions to the IA problem consist of iterative algorithms which gradually reduce a distortion measure, till reaching a local minimum. In this paper we propose a new method for IA which is based on an approximate solution of the Traveling Salesman Problem (TSP). The proposed method has a low complexity dependent only on the number of vectors in the codebook. It results in a distortion not much larger than that achieved by the "natural" ordering obtained from the LBG-splitting codebook design algorithm, thus enabling a fast and simple IA when the "natural" IA is not given.

17:10

JOINT SELF AND MULTIUSER INTERFERENCE REDUCTION FOR RELIABLE SYNCHRONIZATION OF DS-CDMA SYSTEMS OVER FREQUENCY-SELECTIVE FADING CHANNELS

S. Zazo, A. Montagut-Gastaminza and J.M. P‡ez-Borrallo, Universidad PolitŽcnica de Madrid, Spain

This paper deals with the reduction of both self interference (SI) due to multipath, and multiuser interference (MUI) due to multiple access for synchronous DS-CDMA communications over frequency-selective fading channels. We are concerned with the down-link where the base station transmits all the users information synchronously and any mobile receiver must perform a reliable demodulation. In this scenario, the user receiver's tasks involving synchronization and single detection are severely degraded by the high interference level. Our proposal intends to reduce this error source by using subspace blind detectors with a structure similar to the Generalized Side Lobe Canceller (GSLC) exploiting the canonical representation for the desired user linear detector. The orthogonality condition between both components is imposed by the proper design of the blocking matrix in the unknown multipath environment. Working at chip rate rather than baud rate, a multirate structure at the lower branch provides an almost MUI/SI free signal for time synchronization and multipath optimum combining.

FriPmOR2
Hall B
Transforms and Filterbanks
Chair: R. Creutzburg ***University of Applied Sciences, Brandenburg, Germany***

14:00

FAST N-D FOURIER-HEISENBERG TRANSFORMS

E.Rundblad-Labunets and V.Labunets, State Technical University, Russia

We develop fast Fourier-Heisenberg number theoretical transforms on discrete 1-D and n-D Heisenberg groups H(3,GF(p),GF(p)) and H(n+2,GF^n(p),GF^n(p)) and over an arbitrary commutative ring K, where GF(p) is the Galois field, p=q^m and q is a prime.

14:20

FACTORS INFLUENCING IMAGE CODING PERFORMANCE OF THE SHAPE-ADAPTIVE DCT

Ryszard Stasinski, Hoegskolen i Narvik, Norway

In the paper two factors influencing image compression efficiency of SA DCT are examined. Firstly, it is shown that the transform performance is strongly dependent on the smoothness of coded shape border. It is shown that even miniscule corrections of the contour may result in important improvement of restored images, both in terms of measured error, as well as when evaluated subjectively. Secondly, relation between important for coding efficiency orthogonality and DC preserving property of the SA DCT is discussed. The general condition for a transform to be DC preserving is formulated. It implies that there is no easy way to guarantee that a separable shape adaptive transform is both DC preserving and orthogonal. The DC corrected SA DCT is a good compromise solution to this problem.

14:40

DATA DECORRELATION BY WAVELET TRANSFORM

Daniela Coltuc, Polytechnic University of Bucharest, Romania
Jean-Marie Becker, CPE France

The Wavelet Transform is not optimal for image compression. The coefficients on the same level of decomposition preserve a residual correlation which may harm the efficiency of the encoding algorithm. In the case of entropic codes, this effect can be reduced by using an appropriate coefficients scanning and long enough contexts for the conditional probabilities. Such an approach needs the knowledge of the residual correlation. This paper proposes a set of formulas for the evaluation of the coefficients correlation, based on the image and wavelet autocorrelations. The wavelet type and its length is shown to be unimportant for data decorrelation, as proven by the similarity of various experiments. According to the proposed formulas, in the case of images with separable autocorrelation, the transformation of a coordinate preserves the other coordinate correlation. For classes of signals like images, possessing mathematical models for their autocorrelation, general encoding procedures could be provided using these formulas.

15:00

WAVELET AND PYRAMID FILTERING OF SIGNAL-DEPENDENT NOISE

Bruno Aiazzi and Stefano Baronti, IROE-CNR, Italy
Luciano Alparone, DET-University Firenze, Italy

In this paper after reviewing a general model to deal with signal-dependent image noise, the well known Local Linear Minimum Mean Squared Error (LLMMSE) Kuan's filter is derived for the most general case. Signal-dependent noise filtering is approached in a multiresolution framework either by LLMMSE processing ratios of combinations of low-pass images, which are tailored to the noise model in order to mitigate its signal-dependence, or by thresholding a normalized non-redundant wavelet transform designed to yield signal-independent noisy coefficients as well. Experimental results demonstrate that the Laplacian pyramid approach largely outperform LLMMSE filtering on a unique scale and is still superior to wavelet de-noising by thresholding.

15:50

JOINT DENOISING AND LOSSLESS/LOSSY COMPRESSION OF SAR IMAGES

Marco Grangetto, Enrico Magli and Gabriella Olmo, Politecnico di Torino, Italy

It is known that the performance of the Discrete Wavelet Transform (DWT) decreases when attempting to compress noisy images, and that it can be improved if a suitable denoising algorithm is used as a preprocessing stage. In this paper we show that this cannot be straightforwardly applied to the Integer Wavelet Transform (IWT) due to the non-linear operations involved. We propose an algorithm based on the IWT, which is capable of achieving performance nearly equal to that of the DWT, and can be satisfactorily employed for joint denoising and lossless/lossy compression. Simulation results are given in the case of SAR images.

16:10

POWER-LAW TIME-SCALE CONTRAST FOR ABRUPT CHANGE DETECTION IN MULTIPLICATIVE NOISE

Marie Chabert and Jean-Yves Tourneret, ENSEEIHT/GAPSE 2, France

This paper addresses the problem of AC detection in multiplicative noise using the Continuous Wavelet Transform (CWT) and possible non-linear pre-processing. The quadratic pre-processing is shown to improve the performance of signature-based time-scale detectors under specific

conditions on noise and signal parameters.

16:30

TWO METHODS FOR NONPARAMETRIC SPECTRUM PEAK DISCRIMINATION

Miroslav Zivanovic and Alfonso Carlosena, Universidad Pública de Navarra, Spain

The conventional DFT-oriented nonparametric interpolation methods, based on time windowing and the DTFT envelope curve resampling (zero padding, Chirp-z, frequency scale distortion, etc.), can improve the spectrum computational resolution. Two methods proposed herein involve some modification of the frequency domain representation and apparently improve the spectrum physical resolution. This is done by intervening in the frequency domain and changing the properties of the corresponding interpolated DFT spectrum, rather than applying a particular time window to the input sequence. That is, we intent to change the form of the Discrete Time Fourier Transform (DTFT) envelope curve, without altering significantly the physical reality. It could be seen as applying a "frequency window" to the input sequence N-point DFT spectrum, by interpolating it at only N points in two different ways. However, an interpretation in terms of time window is also possible.

16:50

EVALUATION OF POTENTIALS FOR LOCAL TRANSFORM DOMAIN FILTERS WITH VARYING PARAMETERS

Rusen Oktem and Karen Egiazarian, Tampere University of Technology, Finland
Leonid Yaroslavsky, Tel Aviv University, Israel

We discuss the performance of a set of local transform domain filters, with respect to varying parameters. We perform Monte Carlo simulations of the filters over five different test images at different noise levels, and evaluate how much the selection of filter parameters affect the filter performance and how far their performances are from that of the ideal Wiener filter.

17:10

CONTOUR RECOGNITION USING NEURAL NETWORK APPLICATION

Andrzej Dziech and Ali Amuri, AGH Cracow University, Poland
Wiera Dziech, Kielce University of Technology, Kielce, Poland

In this paper a new approach for feature extraction based on calculation of eigenvalues from a contour is proposed. In the first part different steps of feature extraction are introduced. The tested contours for recognition have been extracted from their original images using OCE method (object oriented contour extraction). Then the applied neural network for the recognition of noisy contours is described. The recognition is carried out by a feedforward neural network which is well known as a robust and reliable recogniser. In the last part the recognition results of seven tested contours for different SNR (signal to noise ratio) are shown followed by brief discussion about these results and the introduced approach.

FriPmPO1
Exhibition Hall
Nonlinear Signal and Image Processing
Chair: O. Yli-Harja ***Tampere University of Technology, Finland***

COLOUR IMAGE SKELETONISATION

Ioannis Andreadis, Maria I. Vardavoulia, Gerasimos Louverdis and Nikolaos Papamarkos, Democritus University of Thrace, Greece

In this paper a new morphological technique suitable for colour image skeleton extraction is presented. Vector morphological operations are defined by means of a new ordering of vectors of the HSV colour space, which is a combination of conditional and partial sub-ordering. Then these are used to extract skeletons of colour images in terms of erosions and openings. The proposed method was tested with a variety of images and such experimental results are provided. Its applications include image compression and recognition problems.

FIBONACCI THRESHOLDING, SIGNAL REPRESENTATION AND MORPHOLOGICAL FILTERS

A. Grigoryan, S. Agaian and E. Dougherty, Texas A&M University, USA

This paper presents a new weighted thresholding concept for the set-theoretical representation of signals and design of new morphological filters. Such representation maps many operations of nonbinary signal and image processing into simple operations over the binary signals and images. The weighted thresholding is invariant relative to the morphological transforms, including the basic ones, erosion and dilation. The main idea of using the weighted thresholding is the right choice of special levels of thresholding for signal and image processing. Fibonacci thresholding is defined and decomposition of the median filter in terms of such thresholding is described.

NONDETERMINISTIC KINETICS BASED FEATURE CLUSTERING FOR FRACTAL PATTERN CODING

Kohji Kamejima, Osaka Institute of Technology, Japan

Newton potential is reformulated in terms of the Hausdorff distance to design reduced affine mappings associated fractal attractors. By applying maximum entropy analysis to observed patterns, stochastic features are extracted as well as boundary points where the fixed points of the mappings should be located. To linear segments of potential fixed points, feature points are nondeterministically attracted following the Hausdorff potential. Guided by this feature clusters, random patterns are partitioned to estimate mapping parameter.

IDENTIFICATION AND ELIMINATION OF SECOND-ORDER NONLINEAR DISTORTION OF LOUDSPEAKER SYSTEMS USING DIGITAL VOLTERRA FILTER

M. Tsujikawa, T. Shiozaki and Y. Kajikawa, Kansai University, Japan

Modeling loudspeaker system with the Volterra series expansion is essential to eliminate the nonlinear distortion. We have proposed a method measuring the Volterra kernel of loudspeaker systems by multi sinusoidal waves. This method, however, has problems not to consider the phase property of the nonlinear element and the third-order distortion. Therefore, we propose a novel method measuring the second-order Volterra kernel by multi sinusoidal waves. In this method, the phase property of the nonlinear element and the third-order distortion are considered. Moreover, we eliminate the nonlinear distortion of loudspeaker systems with the Volterra kernel measured by the proposed method and demonstrate the effectiveness through the results.

ON THE THEORETICAL PROPERTIES OF LP MEAN COMPARATORS

Constantine Kotropoulos and Ioannis Pitas, Aristotle University of Thessaloniki, Greece

Digital implementations of sorting networks that rely on a Digital Signal Processor core are not as efficient as their analog counterparts. This paper deals with the Lp comparators for which simple analog implementations exist. From a statistical point of view, the Lp comparators are based on the nonlinear means. Their probability density function and the first and second-order moments are derived for independent uniformly distributed inputs. Lp comparators provide estimates of the minimum and maximum of their inputs. Therefore, they introduce errors. A proper approach to compensate for the estimation errors is proposed.

CAN STOCHASTIC RESONANCE BE USED IN DETECTION?

S. Zozor and P.O. Amblard LIS, Groupe Non Linéaire, France

This paper deals with stochastic resonance and its application in sine detection. The nonlinear physical phenomenon of stochastic resonance generally occurs in bistable systems excited by a random noise plus a sine. Such systems force cooperation between the input noise and the input sine: Provided a fine tuning between the power noise and the dynamics, the system reacts periodically. The interesting fact is that the local output signal-to-noise ratio presents a maximum when plotted against the input power noise. In this paper we recall the main results for the discrete-time nonlinear AR(1) systems. We then show how stochastic resonance can be used to detect small noisy sine and that the classical incoherent detector can be improved in some non-gaussian.

SIMPLIFIED VOLTERRA FILTERS FOR ACOUSTIC ECHO CANCELLATION IN GSM RECEIVERS

Andrea Fermo and Giovanni Sicuranza, DEEI University of Trieste, Italy
Alberto Carini, Telit Mobile Terminals S.p.A.

Acoustic echo cancellers commonly implement linear filters that have to identify as close as possible the impulse response of the acoustic echo path system. However such a system is highly nonlinear, and thus it is reasonable to suppose that a better system identification could be achieved by a nonlinear filter. Volterra filters are well suited for modelling that system but they need in general too many computational resources for a real time implementation. Here we present a novel nonlinear structure which exploits the echo path characteristics and thus is more efficient than conventional Volterra filters. For this structure we have considered two kinds of adaptive algorithms: standard LMS and Affine Projection (AP) algorithms that we adapted to this structure.

THE FIRST ABSOLUTE CENTRAL MOMENT IN IMAGE ANALYSIS

M. Paterni, V. Gemignani and A. Benassi, CNR Institute of Clinical Physiology, Italy
M. Demi, Esaote SpA, Italy

In this paper we show how the generalization of the first absolute central moment gives rise to a class of nonlinear filters and how they can be used in image analysis to enhance lines, edges, corners and intersections between different discontinuities. Since the filters are nonlinear the recovered edge information can be also combined to obtain information that would not be obtained by varying the parameters of the original filter. Furthermore, we show how a mass center of the first absolute central moment can be defined and how this can be used to develop a new contour tracking procedure. The mass centers computed at the points of a given approximate starting contour are closer to the "true" contour than the points of the starting contour. Therefore, the final contour can be localized by iteratively computing the mass centers of the first absolute central moment.

FUZZY ADAPTIVE SIGNAL PREDISTORTER FOR OFDM SYSTEMS

J. Bas and A. Pérez-Neira, Universitat Politècnica de Catalunya, Spain

This work develops an adaptive High Power Amplifier (HPA) predistorter that applies the fuzzy logic and fuzzy set theory. As Volterra and Fourier series models or Neural Networks, a fuzzy logic system (FLS) is a nonlinear function approximator and we demonstrate its ability to compensate the nonlinear distortions for orthogonal multicarrier transmitters and relay stations. When compared with Volterra or Fourier Series predistorters, the main features of the fuzzy predistorter are low computational complexity and simple design. The good performance of the proposed predistorter is compared with that of a Fourier Series predistorter via simulations applied to the DVB (Digital Video Broadcasting) standard.

FAST COMPUTATION OF RANK ORDER STATISTICS

Dinu Coltuc and Philippe Bolon, University of Savoie, France

This paper proposes an algorithm for the computation of 1D rank order statistics. For a window filter of size n and a rank r the computation takes place on groups of 2n samples. Two ordered strings of r samples are constructed by straight insertion and their partial results are combined to cover n+1 consecutive window positions. The filter output is found either directly, taking the rank taking the r-th sample in rank from ordered sequences (2 results) or by selecting it from two ordered sub-strings (n-1) results. For ranks far apart from the median, the behavior of the algorithm is outstanding. Thus, for max/min the computational complexity, regardless the window size, is less than 3 comparisons/sample. For the second in rank, one gets less than 7 comparisons/sample, etc. When the rank approches the median, the computational complexity increases to O(log_2 n).

ODIF FOR L-FILTERS

Sari Peltonen and Pauli Kuosmanen Tampere University of Technology, Finland

In this paper we study robustness of L-filters by using a recently introduced method called output distributional influence function (ODIF). Unlike the traditionally used methods, such as the influence function and the change-of-variance function, the ODIF provides information about the robustness of finite length filters. So the ODIF is not only a good theoretical analysis tool but it can also be used in real filtering situations for selecting filters behaving as desired in the presence of contamination. The usefulness of the ODIF in the analysis of the robustness of different L-filters is demonstrated in several illustrative examples by using the ODIFs for the expectation and the variance giving local robustness of the value and the variance, respectively.

ANALYSIS OF OUTLIER INFLUENCE ON NONLINEAR FILTER OUTPUT FOR LINEARLY INCREASING/DECREASING SIGNALS WITH NOISE

Sergey K. Abramov and Vladimir V. Lukin, State Aerospace University
Sari Peltonen, Pauli Kuosmanen and Jaakko Astola, Tampere University of Technology, Finland

The influence of an outlier present in a scanning window on nonlinear filter output characteristics like bias and variance is analyzed using output distributional influence functions and estimates of statistical parameters by computer simulation. The standard median, Wilcoxon and a-trimmed filters with different scanning window sizes are considered. The peculiarity of signal model is that besides constant signals we exploit the linearly increasing/decreasing ones. It is shown that there exist specific peculiarities of filter output behavior depending upon noise variance, outlier and signal derivative signs and values. Filter output bias and variance also depend on filter type and scanning window size. All these effects should be taken into account in many practical applications. nonlinear filters, LIDS, output bias, output variance, ODIF.

RELIABLE LEARNING USING POST CLASSES

Ilya Shmulevich and Moncef Gabbouj, Tampere University of Technology, Finland

The complexity of the consistency problem for several important classes of Boolean functions is analyzed. The classes of functions under investigation are those which are closed under function composition or superposition. Several of these so called Post classes are considered within the context of machine learning with an application to breast cancer diagnosis. The considered Post classes furnish a user-selectable measure of reliability. It is shown that for realistic situations which may arise in practice, the consistency problem for these classes of functions is polynomial-time solvable.

REDUCTION FACTORS IN FINITE AUTOMATA WITH APPLICATION TO NONLINEAR FILTERS

A. Niemistö, O. Yli-Harja, A. Valmari, P. Koivisto and I. Shmulevich, Tampere University of Technology, Finland

We examine the reduction factors achieved in finite automata describing several important classes of stack filters. It has been shown that finite automata and Markov Chain theory can be used to compute certain statistical properties of stack filters. However, if the automaton is not state-minimal, the solution may be intractable, which is the motivation for this work. Formulae for the number of states in state-minimal automata are presented.

A RATIONAL Nx IMAGE INTERPOLATOR

Livio Tenze and Sergio Carrato, University of Trieste, Italy

In this paper we present an innovative interpolator which performs high quality $N \times$ interpolation on both synthetic and real world images. Its structure, which is based on a rational operator, provides edge sensitive data interpolation, so that sharp and artifact free images are obtained.

A HYBRID TRANSFORM METHOD FOR IMAGE DENOISING

Ben-Zion Shaick, Leonid Ridel and Leonid Yaroslavsky, Tel Aviv University, Israel

Image filtering in moving window in DCT domain has proven its capability in image edge preserving denosing. For the highest noise suppression capability, such a filtering requires, in principle, optimal selection of the size of the moving window for each particular image. In order to overcome this drawback, a combination of signal wavelet expansion (sub band decomposition) and moving window filtering in DCT domain is suggested and experimentally tested on a number on synthetic and real life images.

FriPmPO2
Exhibition Hall
Digital Signal Processing
Chair: O. Simula ***Helsinki University of Technology, Finland***

THE DECOMPOSITION OF VECTOR FUNCTIONS IN VECTOR-MATRIX SERIES INTO STATE-SPACE OF NONLINEAR DYNAMIC SYSTEM

Alexander M. Krot, Institute of Engineering Cybernetics of the National Academy of Sciences of Belarus, Balarus

Decomposition relationships for nonlinear dynamic system operators into state-space based on vector-matrix series are proposed. The representation for a shift operator on trajectories of nonlinear dynamic system through shift operators on trajectories of multidimensional linear dynamic systems was obtained.

GENERALIZED IIR POLYNOMIAL PREDICTIVE FILTERS

Konsta Koppinen and Jaakko Astola, Tampere University of Technology, Finland

Polynomial predictive filters refer to linear filters capable of unbiased extrapolation of polynomial signals, which are useful in applications where the signals of interest are slowly varying and the processing delay should be minimized. Several methods have been proposed to design FIR and IIR predictive filters. In this paper, we derive the general expression for the transfer function of any polynomial predictor and show that the previous structures are obtained as special cases. This general class is likely to include filters with better performance in situations of interest than those considered previously.

A WAVE THEORY OF LONG LMS ADAPTIVE FILTERS

H.J. Butterweck, Eindhoven University of Technology, The Netherlands

Long LMS filters of the tapped-delay line type are in widespread use, particularly in acoustic applications. For the limiting case of an infinite line length the behaviour of such filters is shown to be governed by remarkably simple laws. This is true for the steady state, where for small stepsizes the weight-error correlations become independent of the input signal, but also for the transient behaviour, where the spatial Fourier transform of the weight-error distribution decays exponentially. Moreover, a necessary and (probably) sufficient stability bound for the stepsize is derived. The "wave theory" developed for the infinite line length also predicts the behaviour of rather short filters with sufficient accura cy, particularly for a moderately coloured input signal. No independence assumption is required and no assumption concerning the spectral distribution of the additive noise. Under steady-state conditions, the weight-error correlation between two line taps is solely determined by the noise autocorrelation, with the time delay replaced by the tap distance.

SIMULATED EVOLUTIONARY CODE GENERATION FOR HETEROGENEOUS MEMORY-REGISTER DSP-ARCHITECTURES

Bernhard Wess, Vienna University of Technology, Austria

Efficient algorithms exist that generate optimum straight-line code for expression trees. However, when applied to graphs, the drawback of tree-based straight-line code generation is the fact that there is no joint optimization of the tree code and the data transfers between the trees. The purpose of this paper is to introduce an evolutionary hybrid that combines evolutionary optimization strategies with tree techniques. The goal is to minimize the execution time of the program by jointly optimizing the schedule, selected instructions, and allocated registers. The core of our technique is a linear-time algorithm that translates expression trees into optimal straight-line code segments satisfying a set of boundary conditions for the tree interface variables. Experiments indicate that this technique allows to generate code of such high quality that is extremely difficult to achieve manually.

WORST-CASE EXECUTION TIME ANALYSIS FOR DIGITAL SIGNAL PROCESSORS

Niklas Holsti, Thomas Langbacka and Sami Saarinen, Space Systems Finland Ltd., Finland

We present ongoing work to develop a software tool for estimating worst-case execution times for real-time, embedded programs. The tool applies static analysis to executable machine-code programs. Currently we mainly aim at supporting the TSC-21020E Digital Signal Processor, but the tool is designed to be easy to adapt to other target processors as well. Contrary to most other WCET tools we attempt (whenever possible) automatic estimation of loop bounds. We also provide a rich assertion language, which can be used to set bounds on loops that the tool itself cannot bound.

THE ENDOMORPHIC MODEL: AN APPROACH TO ADAPTIVE PREDICTION

Asoka Korale and Anthony G. Constantinides, Imperial College, London, UK

This paper describes a method by which a set of piecewise constant autoregressive parameters could be obtained when the source signal is subject to a multistage analysis. It also illustrates an FIR prediction scheme with application to a speech process. It describes a method where the prediction is carried out on a sample by sample basis. This prediction scheme demonstrates the possibility of having control over the prediction error by constraining it to be within a predetermined limit.

OPTIMAL POLE CONDITIONS FOR LAGUERRE MODELS THAT SATISFY SOME INTERPOLATION CONSTRAINTS, USING AN p-NORM, 1<P<INFINITY >

Tomás Oliveira e Silva, Universidade de Aveiro, Portugal

The optimal pole conditions for Laguerre models are available in the literature for the 2-norms (for continuous-time or discrete-time systems, with or without an impulsive input signal, and in the time or frequency domains). Recently, the author was able to extend the available results i) to other p-norms, and ii) to models whose responses to known input signals satisfy, in the time and/or frequency domains, some interpolation constraints. In this paper we combine both extensions. It turns out that the optimality conditions for the poles of Laguerre models constrained as stated above, and using an p-norm, have the same functional form as the already available optimality conditions: the last optimal weight of the model vanishes or the last optimal weight of the model of the next higher order vanishes.

TRACKING ANALYSIS OF A GRADIENT-BASED ADAPTIVE IIR NOTCH FILTER WITH CONSTRAINED POLES AND ZEROS

Yegui Xiao, Hiroshima Prefectural Women's University, Japan
Yoshihiro Takeshita, Katsunori Shida Saga University, Japan

Gradient-type adaptive IIR notch filters have many attractive merits in real-life applications, since they require a small amount of computation while demonstrate good performance. However, it is generally quite difficult to assess their performances analytically. In particular, tracking properties of them have not been investigated yet. In this paper, tracking performance is analyzed in detail of a plain gradient algorithm for a second-order adaptive IIR notch filter with constrained poles and zeros, which takes a chirped signal as its input. First, two set of difference equations for frequency tracking error and mean square error (MSE) are established in terms of convergence in the mean and convergence in the mean square, respectively. Closed form expressions for the asymptotic tracking error and MSE are then derived from these difference equations. Optimum step size parameter of the algorithm is also worked out based on the minimization of the asymptotic tracking error and MSE. It is discovered that the asymptotic tracking error can be driven to zero for a positive chirp rate by selecting a proper step size value, which is a rare property for an adaptive filtering algorithm. Extensive simulations are provided to support the analytical findings.

FriPmPO3
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Systems and Blind Identification
Chair: R. Leonardi ***University of Brescia, Italy***

BLIND IDENTIFICATION OF SECOND ORDER HAMMERSTEIN SERIES

Panos Koukoulas and Nicholas Kalouptsidis, University of Athens, Greece

Blind identification of second order Hammerstein series is considered. The output cumulants up to order 5 are used to determine the Volterra kernels, when the input is a stationary zero mean Gaussian white stochastic process. Both infinite and finite extent kernels are considered.

3D BLIND IMAGE RECONSTRUCTION USING COMBINED NONLINEAR AND STATISTICAL TECHNIQUES

M.Razaz and S.Nicholson, University of East Anglia

Blind reconstruction or deconvolution, is the process of restoring an observed image without explicit knowledge of the imaging system's point spread function (PSF). Images produced from an imaging system, for example confocal laser scanning or widefield optical microscope, are noisy and invariably blurred. For robust scientific interpretation and analysis of a typical image obtained in this way, it is essential that the image is further processed to remove these aberrations. Measurement and modelling the PSF is not ideal and the measurements are quite difficult and time consuming to perform. Here we present an algorithm to perform blind reconstruction in three dimensions using a nonlinear reconstruction algorithm called "Iterative Deconvolution algorithm" (IDA). This technique is combined with the use of statistical a priori knowledge in an iterative manner to create a robust and computationally efficient method of blind reconstruction.

POLE PLACEMENT TECHNIQUES FOR A CLASS OF IIR BLIND EQUALIZERS

M. C. Campi, R. Leonardi and L. A. Rossi, University of Brescia, Italy

This work proposes a method for blind equalization of possibly non-minimum phase channels using particular infinite impulse response (IIR) filters. In this context, the transfer function of the equalizer is represented by a linear combination of specific rational basis functions. This approach estimates separately the coefficients of the linear expansion and the poles of the rational basis functions by alternating iteratively between an adaptive (fixed pole) estimation of the coefficients and a pole placement method. The focus of the work is mainly on the issue of good pole placement (initialization and updating).

BLIND IDENTIFICATION OF NONMINIMUM PHASE SYSTEMS USING THE MEAN DIFFERENTIAL CEPSTRUM

J. Antoni, J. Danière and F. Guillet, Laboratoire d'Analyse des Signaux et des Processus Industriels

In this paper we present a new blind identification scheme for nonminimum-phase linear time-invariant systems excited by stochastic transient processes. Conditions of identifiability are rather general and also apply to the multipath case. The theory is based on the definition of the mean differential cepstrum which is closely related to the spectral correlation density. Under assumption of cycloergodicity, we then propose an algorithm which only involves monodimensional sequences and is computationaly fast and effective. Performances are evaluated through computer simulations and application is made to the identification of a real mechanical inertance.

DIRECT BLIND MULTICHANNEL IDENTIFICATION

M. Frikel, W. Utschick and J. Nossek, Technical University of Munich Institute for Circuit Theory and Signal Processing, Germany

In this paper, different techniques for the estimation of the signal parameters and/or the channel coefficients for single-input/multiple-output systems, are presented. These methods are based, either on the use of a small part of observations or on the minimization of a quadratic form with quadratic constraint. Simulations have been established and these techniques are compared to the classical approaches. This work is supported by Alexander von Humboldt-Stiftung.

BLIND EQUALIZATION BASED ON FOURTH-ORDER CUMULANTS

Kamel Abderrahim, Ridha Ben Abdennour and Faouzi Msahli, Ecole Nationale d'Ingénieurs de Gabès, Tunisia
Mekki Ksouri, Ecole Nationale d'Ingénieurs de Tunis, Tunisa
Gerard Favier, Laboratoire I3S, UNSA/CNRS, France

This paper addresses the problem of blind equalization based on Higher-Order statistics. In fact, we propose a new method for blind identification. This method uses only fourth order cumulants and consequently it is insensible to additive Gaussian noise. Moreover, it exploits cumulants whose arguments are near to the origin which improve the estimates accuracy. The Channel impulse response coefficients are obtained by using a least squares method. A recursive method is also developed to claim the uniqueness of the least squares Next, we exploit this method in the context of blind equalization. Numerical examples are also presented in order to illustrate the performances of the proposed method.

FriPmPO4
Exhibition Hall
Filter Design and DSP Computation
Chair: Ari Visa *Tampere University of Technology, Finland*

ANALYSIS OF PASSIVE PLANAR MICROSTRIP CIRCUITS USING THE ITERATIVE TECHNIQUE

Y. Ounejjar, R. Douma, A. Gharsallah and A. Gharbi, Equipe d'Electronique, Tunisia

A general implementation of the iterative method based on the concept of waves has been presented. It consists on establishing a recurrence relation ship between the wave in media (1) and (2) using the reflection operator in the spectral domain and boundary conditions at the dielectric metal discontinuity in spatial domain. It takes the advantage of the simplicity, which does not involve based functions and inversion of matrix. It is capable to analyzes longer bodies. Moreover, the introduction of Modal Fast Fourier Transformations owed simplifying calculations and accelerating the convergence with a reduced CPU time. An algorithm has been developed to offer a rigorous characterisation of microstrip filter. The results were compared and is accuracy verified. Consequently, the present approach will be investigated for further new applications such as in bridges diodes, active element, etc.

IMPLEMENTATION OF REAL-VALUED DISCRETE TRANSFORMS VIA ENCODING ALGEBRAIC INTEGERS

R. Baghaie and V. Dimitrov, Helsinki University of Technology, Finland

In this paper, we propose a novel approach for computing real-valued discrete transforms such as the discrete cosine transform and the discrete Hartley transform. The approach is based on the algebraic integer encoding scheme. With the aid of this scheme, an error-free representation of the cos, sin and cas functions becomes possible. Furthermore, for the implementation of these algorithms a fully pipelined systolic architecture with O(N) throughput is proposed.

SMALL-SAMPLE ESTIMATION OF THE ERROR OF THE OPTIMAL BINARY FILTER

D. Sabbagh and E. Dougherty Texas A&M University, USA

Precise small-sample estimation of the error of an optimal filter is theoretically limited. This paper shows the possibility of obtaining better estimation in a Bayesian context by postulating prior knowledge regarding the probability distribution of the model. Prior knowledge is employed to estimate the estimation error, and thereby obtain a better estimate of filter error. Error estimation is done in a conservative manner in order not to obtain a low-biased estimate of filter error. This key condition is achieved by finding a majorant of the bias in the estimation of estimation error. The quality of our estimate of the error depends upon the precision of the prior knowledge.

LEAST SQUARES FIR FILTER DESIGN USING FREQUENCY DOMAIN PIECEWISE POLYNOMIAL APPROXIMATIONS

Marios S. Pattichis, The University of New Mexico, USA

A general framework for specifying and designing one-dimensional FIR digital filters is presented. The system uses piecewise polynomials to guarantee that the ideal impulse response maintains continuity up to a given number of derivatives (given as parameter p). In this system, it is possible to efficiently design optimal (in the least-squares sense) digital filters that converge pointwise to any (continuous) desired frequency response, while the filters' coefficients decay in the order of 1/n^(p+1). It is shown experimentally that the maximum absolute error also decays as 1/n^(p+1). Results on (i) bandpass filter design and (ii) bandpass differentiating filter design are shown.

FIXED-POINT DSP TIMING OF PULSES BASED ON A HIGH-PRECISION DIVISION TECHNIQUE

Angelo Geraci, Stefano Riboldi and Giancarlo Ripamonti, Politecnico di Milano, Italy

In some applications, it is important to time the occurrence of pulses of variable amplitude arriving randomly with respect to a clock signal with a precision higher than one sample interval by using digital tecniques. The proposed method is based on FIR filtering of the sampled signal and on a high-precision and high-efficiency algorithm to perform division on a fixed-point digital signal processor. It allows to determine the time of occurrence estimation with a precision up to one hundredth of the sampling period, depending on signal-to-noise ratio, and to process up to 100k pulses/second. The method is the cascade of three steps: (i) rough timing of the examined pulse, with tolerance up to ten times the sampling period; (ii) FIR filtering of the pulse which returns the information of the event occurrence time multiplied by the amplitude of the pulse; (iii) normalization of the result of (ii) through its division by a separate estimation of the amplitude value of the pulse. As a high precision is required, the operation of fixed-point division is the crucial point of the processor. We introduce a high-precision method for implementing division of integer numbers, which is also capable of high-speed.

NUMBER THEORETIC TRANSFORM MODULO K.2^n +1, A PRIME

M. Bhattacharya and J. Astola, Tampere University of Technology, Finland

Number Theoretic Transform is attractive for computation of convolution due to its simple and real arithmetic structure. However, there exists a stringent relation between the choice of modulus M i.e., the wordlength and convolution length. Choice of modulus as K.2^n+1, a prime, leads to relaxation of this constraint and wide choices of wordlength, with each of these associated with many choices of convolution length are are obtained. Under these choice of modulus a computational structure when the convolution length is a perfect square is presented.

DIGITAL CODING BY MEANS OF LINEAR PERIODIC TIME-VARYING FILTERS

Alban Duverdier, CNES
Bernard Lacaze, ENSEEIHT/SIC

In modern telecommunications applications, the interest of linear periodic time-varying filters has been often demonstrated. They can be used for scrambling, multi-user access or channel modeling. Recently, the authors have shown that any linear periodic filter can be attractively realized by means of periodic clock changes. Nevertheless, all the solutions based on periodic clock changes correspond to analogue filtering cases that spread the spectrum. The original part of this paper consists of using periodic clock changes for binary signals with a non-spreading spectrum solution. We first recall the definition of linear periodic filters. In particular, it is shown that a stationary process subjected to such a filter becomes cyclostationary. The paper proposes then to use a periodic clock change on a binary signal as a coding. It is followed by a modulation that permits to obtain a spectrally efficient solution. We present the reconstruction method of the initial signal. An example illustrates a binary sequence transmission by means of a periodic clock change.

PRIME FACTOR ALGORITHM OF DISCRETE COSINE TRANSFORM

Guoan Bi and Yonghong Zeng, NTU, Singapore

Prime factor fast algorithms are computationally efficient for various discrete transforms. However, they generally need an index mapping process to convert one-dimensional input sequence into a two-dimensional array, which results in a substantially computational overhead and an irregular computational structure. This letter attempts to minimize the computation overhead by a simple and general mapping procedure.

ROBUST DESIGN OF TRANSMIT PULSE SHAPE

Ba-Ngu Vo, Curtin University of Technology, Australia
Antonio Cantoni, Atmosphere Networks Inc.

This paper considers the design of Transmit Pulse Shapes such that after transmission through a dispersive channel, the distorted pulse at the receiver fits in a prescribed template. The norm of the transmit pulse is minimized so as to reduce the effect of cross-talk at the receiver end. This design problem is formulated as a Quadratic Programming problem with affine functional inequality constraints. In practice, errors in the implementation of the optimal filter cause the received pulse to violate the template constraints. We present a robust formulation which incorporate the uncertainties to ensure that the constraints are satisfied even in the presence of implementation errors. This technique is applied to determine the transmit pulse shape to be programmed on a Siemens FALC54, a T1 Line Interface Unit.

DIRECT DESIGN OF BANDPASS WAVE DIGITAL LADDER FILTERS

Mohamed Yaseen, University of Assiut, Egypt

The paper extends further the results of wave digital filter design problem. It presents a complete direct design method for bandpass wave digital filters having simple ladder structures. These structures are characterized by amplitude characteristics that are monotonic in the two stopbands and equiripple in the passband, i.e., they have the same features of Chebyshev response. The approximation is relying on the proper formulation of the transmission function in the reference domain such that it exhibits all of its zeros at the origin and infinity. Consequently, it is designed by applying interpolation techniques combined with the Remez-exchange algorithm. The transmission function is synthesized by extracting the poles at the origin and infinity from the successive resulting impedance and admittance functions. Finally, the wave digital realization is obtained by applying three-port series and parallel adaptors.

COMPARISON OF TWO METHODS FOR REALIZATION OF MULTIPLIERLESS ELLIPTIC IIR FILTERS

Bojan Radulovic, University of Belgrade, Yugoslavia
Ljiljana Milic, Institut Mihajlo Pupin, Yugoslavia

Two methods for multiplierless elliptic IIR filter implementation are discussed: cascade realization and parallel connection of two allpass networks. Elliptic IIR filters are selected because they fulfill the requirements with lower order, and, usually, they require fewer coefficient multipliers. The comparison is made on the basis of adder costs, amplitude characteristics in pass and stop bands, coefficient wordlength and normalized product round-off noise variance at the output of the filter. This paper concentrates on the fixed-point arithmetic. The comparison between these different design algorithms for digital multiplierless IIR filters has not yet been reported. In this paper that comparison has been made for the first time. It is very important to point out that the same conditions were applied for analysis of these filters.

A NEW APPROACH TO FINITE WORDLENGTH COEFFICIENT FIR DIGITAL FILTER DESIGN USING THE BRANCH AND BOUND TECHNIQUE

Ahmed Nabil Belbachir, Benaoumer Boulerial and Mohamed Faouzi Belbachir, University of Science and Technology of Oran, Algeria

Parks-Mc Clellan method allows the design of linear phase FIR filters. The coefficients h(n) which give the best Chebyshev approximation to the desired frequency response 'HD(ejw)' are obtained. This method however uses an infinite precision optimisation. When these filters are implemented on Digital Signal Processor with a special purpose-hardware, each filter coefficient has to be represented by a finite number of bits (b). The simplest and the most widely used approach to the problem are the rounding of the optimal infinite precision coefficients to its (b) bits representation. However, the filters obtained are degraded and in most case there exists another set of finite word length coefficients which gives the best Chebyshev approximation to the desired frequency response 'HD(ejw)'. To find these coefficients, it is necessary to include the finite word length restriction into the filter design. It has been shown that the branch and

bound technique is effective for the design of finite wordlength optimal digital filters. This technique is however expensive in computing time. In this paper, we present a robust branch and bound branching strategy named Sequential and Progressive Search, improving the design of filters on a large wordlength processor in a reasonable computing cost. The details of the algorithm and many examples are given and compared to the other methods.

Chairman's Message

Dear Colleagues and Friends,

On behalf of the EUSIPCO 2000 Organizing Committee, we wish to extend our warmest invitation to you to join us in Tampere next September in celebrating the tenth European Signal Processing Conference.

Started in Lausanne in 1980, EUSIPCO is now well established as a regular biennial event and as the most prominent European gathering in signal processing, providing an excellent opportunity for researchers from universities and industry to present their latest results and discuss state-of-the-art trends.

EUSIPCO has during its history traveled through most of Central and Southern Europe and comes now for the first time to Scandinavia. The venue for EUSIPCO 2000 is Tampere Hall in the heart of Tampere, the largest inland city in Scandinavia and the second-largest urban region in Finland. Tampere, a dynamic centre of high-tech industry, culture, research and education, is a prime example of a clean, modern and safe Nordic city. Tampere is large enough to provide all the services that a major conference may require, yet small enough to make delegates and their families feel at home during their stay. Hotels, shops and restaurants are all within walking distance of the conference hall.

Even though there are several major conferences around the time of EUSIPCO 2000 we received about the same number of submissions as the previous record-making EUSIPCO ' 98 in Rhodes. The accepted papers were allocated to 36 oral and 28 poster sessions (22 of which are special sessions), reflecting our striving to provide an attractive and balanced program.

I would like to express my deep appreciation to all Organizing Committee members and in particular to Moncef Gabbouj for coordinating the technical programme. I would like to thank the International Scientific Committee Members and Reviewers for offering their time in reviewing the submitted papers. Their efforts made possible the rigorous reviewing process, matching the prestige of the EUSIPCO conferences. Last but not least I want to thank the plenary speakers, the special session organizers, the session chairmen, and all the authors for their efforts towards a successful scientific event.
We are very much looking forward to meeting you in Tampere.

Jaakko Astola
Chairman of the Organizing Committee

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Cowan Colin
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Cramariuc Bogdan
Creutzburg Reiner
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Cusani Roberto
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Davidson Jennifer
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Decenciere Etienne
Defee Irek
Del Re Enrico
Delathauwer Lieven
DeLeon Phillip
Delp Edward
den Brinker Bert
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Depovere Geert
Deravi Farzin
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Gevorkian David
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Gotchev Atanas

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Hooli Kari
Horneman Kari
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Hottinen Ari
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Howard Heys
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Lian Yong
Lilleberg Jorma
Lim YC

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Mandridake Elimberaza
Mannerkoski Jukka
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Marques Ferran
Marra Mike
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Marshall Steve
Martins Isabel
Matero Jorma
Matrouf Driss
Mattone Raffaella
McWhirter John
Mech Roland
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Melvasalo Maarit
Menendez Jose
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Mengali Umberto
Merched Ricardo
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oraintara Soontorn
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torres luis
Touzni Azzedine
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Trindade Antonio
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Tubaro Stefano
Tujkovic Djordje
Tziritas Georges
Unser Michael
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Valimaki Vesa
Valkama Mikko
Valkealahti Kimmo
walker Jim
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Wang Xiaofeng
Wanhammar Lars
VanLandingham Hugh
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