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Biometric Image Processing And Recognition

Authors:
P. J. Phillips, R. M. McCabe, *National Institute of Standards and Technology (USA)*
R. Chellappa, *University of Maryland (USA)*

ABSTRACT

Biometric-based identification and verification systems are poised to become a key technology, with applications including controlling access to buildings and computers, reducing fraudulent transactions in electronic commerce, and discouraging illegal immigration. There are at least eight image-based biometrics that are being actively considered. In image-based biometrics, the biometric signature is acquired as an image and the image is processed using techniques from computer vision, image understanding, and pattern recognition. We consider two promising image-based biometrics, faces and fingerprints. For each, we provide a critical assessment of the state of the art, suggest future research directions, and identify technological challenges.

Digital Watermarking : An Overview

Authors:
G.Voyatzis, N.Nikolaidis, I.Pitas, *University of Thessaloniki (GREECE)*

ABSTRACT

In this paper we describe a general framework for image copyright protection through digital watermarking. In particular we present the main features of an efficient watermarking scheme, discuss robustness issues and describe the three main stages of a watermarking algorithm namely watermark generation, embedding and detection.

On The Reliability Of Detecting Electronic Watermarks In Digital Images

Authors:
T. Kalker, J.P. Linnartz, G. Depovere, M. Maes, *Philips Research (THE NETHERLANDS)*

ABSTRACT

A watermark is a perceptually unobstructive mark embedded in an image, audio or video clip or other multimedia asset. A watermark can carry additional information, for instance about the source and copyright status of a document or its intended recipient, its rights and restrictions. We analyse the reliability of detecting such watermarks, modeling it as a detection problem where the original content acts as noise or interference. Probabilities of incorrect detections are expressed in terms of the watermark-to-image power ratio, showing a significant similarity in the problem of detecting watermarks and that of receiving weak spread-spectrum signals over a radio channel with strong interference. Theoretical results are verified by experiments.

A M.A.P. Identification Criterion For DCT-based watermarking

Authors:
M. Barni, F. Bartolini, V. Cappellini, A. Piva, F. Rigacci, *Universita di Firenze, (ITALY)*

ABSTRACT

For reliably protecting the rights of multimedia data owners, digital watermarking techniques appear to be the best solution. A digital watermark is a code which is hidden into the multimedia data for carrying information regarding the copyright. In this paper the problem of the reliable identification of the watermark, in absence of the original unmarked image, is addressed. An optimum (based on a MAP, Maximum A Posteriori probability, criterion) watermark identificator is designed, for extracting watermarks inserted in the DCT domain. The watermark identificator is based on the knowledge of the probability density function of the DCT coefficients. The results support the optimality of the identificator and prove the robustness of the approach.

A Blind Wavelet Based Digital Signature For Image Authentication

Authors:
L.Xie, G.R.Arce, *University of Delaware (U.S.A)*

ABSTRACT

A content based digital image signature scheme for image authentication is proposed. The signature is implemented by image watermarking in the wavelet domain. We introduce interdependency between the watermark and the image data sequence, so that the detection and verification of the signature are independent of the original image and the knowledge of the original watermark sequence or location are not needed. The signature is embedded and tested within the Entropy Zerotree Wavelet Coding (EZW) model. The information capacity of the algorithm is determined.

Robust Audio Watermarking In The Time Domain

Authors:
P.Bassia, I.Pitas, *University of Thessaloniki (GREECE)*

ABSTRACT

The audio watermarking method presented below offers copyright protection to an audio signal by modifying its temporal characteristics. The amount of modification embedded is limited by the necessity that the output signal must not be perceptually different from the original one. The watermarking method presented here does not require the original signal for watermark detection. The watermark key is simply a seed known only by the copyright owner. This seed creates the watermark signal to be embedded. Watermark embedding depends also on the audio signal amplitude in a way that minimizes the audibility of the watermark signal. The embedded watermark is robust to MPEG audio coding, filtering, resampling and requantization.

Optimal And Sub-Optimal Decoding For Vector Quantization Over Noisy Channels With Memory

Authors:
M. Skoglund, *Royal Institute of Technology (SWEDEN)*

ABSTRACT

This paper considers optimal decoding for vector quantization over a noisy channel with memory. The optimal decoder is soft in the sense that the unquantized channel outputs are utilized directly for decoding, and no decisions are taken. Since the complexity of optimal decoding is high, we also present an approach to sub-optimal decoding, of lower complexity, being based on Hashimoto's generalization of the Viterbi algorithm. We furthermore study optimal encoding and combined source-channel coding. Numerical simulations demonstrate that both optimal and sub-optimal soft decoding give prominent gain over decision-based decoding.

Approximating The Protection Offered By A Channel Code In Terms Of Bit Error Rate

Authors:
F. Labeau, C. Desset, B. Macq, L.Vandendorpe, *UCL Communications and Remote Sensing Laboratory (BELGIUM)*

ABSTRACT

In many joint source-channel coding applications, there is a need to assess the performance of a system on a given channel, with different combinations of channel codes. What is interesting is to know some measure of the overall bit error rate, in order to design the source and/or channel codes accordingly. We propose here an approximation of the bit error rate of the cascade of a linear block code and a memoryless binary channel. With this approximation, both of these components can be considered in this respect as a single binary symmetric channel, with known transition probability, and can be used as such for further processing. We also show some possible applications in joint source-channel coding schemes.

Joint Binary Symmetric Source-Channel Coding With Small Linear Codes

Authors:
F.-X. Bergot, O. Rioul, *URA CNRS (FRANCE)*

ABSTRACT

The strong similarity between binary symmetric source (BSS) coding relative to Hamming distortion and binary symmetric channel (BSC) coding is discussed in this paper. It is further utilized for designing data compression systems with error probability criterion. These are, in turn, used to construct simple joint source and channel coding systems.

Image Compression Utilizing The DTOCS, Topological Raster Patterns And Delaunay Triangulation

Authors:
P. J. Toivanen, *Lappeenranta University of Technology (FINLAND)*

ABSTRACT

In this paper, a new image compresion method is presented. It is based on the use of the Distance Transform on Curved Space (DTOCS), which calculates an integer approximation of the weighted pseudo-Euclidean distance transform along discrete 8-paths, and topological raster patterns to determine coordinates for control points. The decompression is based on triangulating among the control points using Delaunay triangulation. The obtained images are compared to previously published methods and to the JPEG DCT-based method. It is shown that the obtained results are of the same quality as JPEG results and exceed other DTOCS and Delaunay based methods. It is also shown experimentally that the time complexity of the compression method presented in this paper does not depend on the number of control points, i.e. the compression ratio. Because of this and the locality of the decompression scheme this method could well be used in applications where a rapid flashing image is made more accurate over time.

Polyphase Adaptive Filter Banks For Fingerprint Image Compression

Authors:
O. N. Gerek, A. E. Cetin, Bilkent University (TURKEY)

ABSTRACT

Subband decomposition is widely used in signal processing applications including image and speech compression. In this paper, we present Perfect Reconstruction (PR) polyphase filter bank structures in which the filters adapt to the changing input con-ditions. This leads to higher compression results for images containing sharp edges such as fingerprint images. The fingerprint image compression is an important problem due to the high amount of fingerprint images in databases [1]. For example, the FBI database contains 30 million sets of fingerprints. We experimentally observed that our method is successful for binary and gray-valued fingerprint images.

Universal Context Modeling
For Lossy Wavelet Image Compression

Authors:

B. Menser, F. Muller, *Rheinisch-Westfalische Technische Hochschule (RWTH) (GERMANY)*

ABSTRACT

In this paper we present a new wavelet image coding scheme based on universal context modeling. The wavelet coefficients are quantized using a layered quantization approach. The resulting symbol stream is entropy-coded with a context-based arithmetic coder. The efficiency of arithmetic coding depends on how well the probability model used by the encoder fits the data generating source. We apply the universal context modeling algorithm introduced by Rissanen and Weinberger to estimate both structure and parameter of the source model. This context selection strategy avoids the problem of context dilution which arises when using a plain Markov model and leads to higher compression.

Hybrid Coding Of Video With Spatio-Temporal Scalability Using Subband Decomposition

Authors:
M.Domañski, A.Luczak, S.Mackowiak, R.Swierczyñski, *Poznan University of Technology (POLAND)*

ABSTRACT

The paper deals with scalable (hierarchical) coding of video at bitrates in the range of about 2-5 Mbps. A practical goal is to obtain comparable bitrates in a base layer and an enhancement layer. The solution proposed in the paper is based on both temporal and spatial resolution reduction performed for data transmitted in a base layer. Two principal variants are presented: systems with three-dimensional filter banks for spatio-temporal analysis and systems where some B-frames are allocated to an enhancement layer but subband analysis is purely spatial. The assumption is that a base layer is fully MPEG-2 compatible. Bitstreams related to high spatial frequencies are encoded either using DCT-based MPEG-like coding or a technique that exploits mutual dependencies between low- and high-spatial-frequency subchannels.

Fast Embedded Video Compression Using Cache-Based Zerotree Processing

Authors:
C. D. Creusere, *Naval Air Warfare Center (USA)*

ABSTRACT

An embedded coding algorithm creates a compressed bit stream which can be truncated to produce reduced resolution versions of the original image. This property allows such algorithms to precisely achieve fixed bit rates, reducing or eliminating the need for rate control in video transmission applications. Unfortunately, a large amount of scratch memory is required for embedded coding, making it difficult for a cache-based processor to continuously operate at full speed. We address this problem in two steps. First, we partition the wavelet coefficients into separate groups and code them sequentially using the embedded zerotree wavelet algorithm. Next, we use information gathered in the first step to optimize the bit allocations for each partition. The changes to the allocation are transmitted to the decoder and applied to the next video frame.

Facts And Fiction In Spectral Analysis Of Stationary Stochastic Processes

Authors:
S. P.M.T. Broersen, *Delft University of Technology (NETHERLANDS)*

ABSTRACT

New developments in time series analysis can be used to determine a better spectral representation for unknown data. Any stationary process can be modelled accurately with one of the three model types: AR (autoregressive), MA (moving average) or the combined ARMA model. Generally, the best type is unknown. However, if the three models are estimated with suitable methods, a single time series model can be chosen automatically in practice. The accuracy of the spectrum, computed from this single ARMA time series model, is compared with the accuracy of many tapered and windowed periodogram estimates. The time series model typically gives a spectrum that is better than the best of all periodogram estimates.

A Fast Algorithm For Efficient Estimation Of Frequencies

Authors:
T.-H. Li, *University of California (USA)*

ABSTRACT

The paper addresses the problem of fast and accurate estimation of sinusoidal frequencies from noisy measurements. An iterative algorithm of complexity O(n log n) is proposed that produces accurate frequency estimates whose precision can be arbitrarily close to that of maximum likelihood. Convergence and accuracy of the algorithm are ensured by a rigorous mathematical analysis which includes statistical consistency, asymptotic distribution, and requirements on initial values.

The Generalized ACM-Music Without Estimation Of The Number Of Sinusoidal Components

Authors:

O. Caspary, P. Nus, *Université H. Poincaré, Nancy 1 (FRANCE)*

ABSTRACT

The family of MUSIC estimators is often used in the case of the spectral analysis of sinusoidal signals. One of those estimators, called ACM-MUSIC, is based on the direct estimation of the vectors which constitute the signal and noise subspaces thanks to the autocorrelation matrix. In this paper, we suggest using the iterated power method in order to improve the estimation of those vectors. Besides, instead of estimating the number of signal components with criteria such as AIC and MDL, we resort to a judicious weighting which enables us to separate the signal and noise subspaces. The results show frequency spectra close to those obtained by the MUSIC estimator to which has been associated the decomposition of the autocorrelation matrix into eigenvalues. However, the numerical calculation of the proposed method can be lower, and the choice of the number of sinusoidal components can be avoided. That approach can also be applied to the ROOT-ACM-MUSIC estimator, which is a direct extension of the ACM-MUSIC.

On Nonparametric Spectral Estimation

Authors:

T. Sundin, P. Stoica, *Uppsala University (SWEDEN)*

ABSTRACT

In this paper the Cramer-Rao bound (CRB) for a general nonparametric spectral estimation problem is derived under a local smoothness condition (more exactly, the spectrum is assumed to be well approximated by a piecewise constant function). Furthermore it is shown that under the aforementioned condition the Thomson (TM) and Daniell (DM) methods for power spectral density (PSD) estimation can be interpreted as approximations of the maximum likelihood PSD estimator. Finally the statistical efficiency of the TM and DM as nonparametric PSD estimators is examined and also compared to the CRB for ARMA-based PSD estimation. In particular for broadband signals, the TM and DM almost achieve the derived nonparametric performance bound and can therefore be considered to be nearly optimal.

New Criteria Based On Gerschgorin Radii For Source Number Estimation

Authors:
O. Caspary, P. Nus, *CNRS ESA 7039 - Université H. Poincaré (FRANCE)*

ABSTRACT

In the spectral analysis of short data signals composed of sinusoids or exponentials, the source number estimation is a crucial problem for the performances of the high resolution methods (MUSIC, ESPRIT, etc.). Indeed, for those methods, we must truncate the covariance matrix in signal and noise subspaces. That truncation depends on the estimation of the source number. So, we propose new criteria based on the Gerschgorin disks that can be applied in many situations : white noise, colored noise, signals with few samples and sources with different powers. For the last point, we put forward a simple deflation method associated to our criteria that improves the source number estimation. The Gerschgorin radii can be connected to the Least-Squares through the transformed cross-correlation vector to produce new criteria. Different simulations are carried out to compare the performances of those new criteria with other criteria such as AIC and MDL.

Speech Enhancement In Wireless Digital Communication Via Heuristic Rules And Image Relaxation Techniques

Authors:

E. Mumolo, *L. Pivetta, Dipartimento di Elettrotecnica, Elettronica ed Informatica Universita di Trieste (ITALY)*

C. Chiaruttini, *Dipartimento di Architettura Navale, Mare, Ambiente Universita di Trieste (ITALY)*

ABSTRACT

A novel algorithm for the reduction of several types of noise that occur typically in wireless digital speech communications is described in this paper. The algorithm aims at reducing the spectral discontinuities of the signal by analyzing the 2D spectral map and closing the gaps between the frames using heuristic rules. Some experimental evaluations are reported.

Analytical And Iterative Approaches To The Equalisation Of Sub-Band Errors In Speech And Speaker Recognition

Authors:

R. Auckenthaler, *University Graz (AUSTRIA)*
J. S. Mason, *University of Wales (UK)*

ABSTRACT

Recent work in both speech and speaker recognition has shown some interesting apparent benefits of sub-band processing: dividing the acoustic band into sub-units to give multiple stream (sub)-classifiers. In this paper we extend our recent work on sub-band error equalisation by considering 4 separate cases from combinations of male and female speaker sets in the con text of speech and speaker recognition. We show that sub-band error equalisation can be achieved by changing the conventional mel frequency warping function. This can also reduce the overall error rates significantly: in the case of the female speaker set the rate is reduced by over 50%.

Multirate Acoustic Echo Cancellation: Which Adaptive Filters For Which Subbands?

Authors:
Dr. D. K. Phillips, *Loughborough University (UK)*
Prof. C. F. N. Cowan, *Queen's University of Belfast (UK)*

ABSTRACT

Multirate signal processing is one of the best ways to minimise the computational complexity of Acoustic Echo Cancellation (AEC). The subband decomposition of the excitation signal (speech) results in different statistical properties in each subband. As there are a wide range of Adaptive Filters (AFs) with different performances and costs, a challenge is posed in how to match AFs to subbands so as to maximise performance while minimising computational cost. This paper discusses a hypothetical automated benchmarking methodology that addresses some of the issues raised by the attempt to optimise multirate AEC architectures as a function of desired performance. In this context, a performance comparison of the NLMS and RLS algorithms is conducted.

Robust Performance Of The Adaptive Periodic Noise Canceller In A Closed-Loop System

Authors:
J. Timoney, J. B. Foley, *Trinity College (IRELAND)*

ABSTRACT

This paper addresses the issue of robustness of the LMS-driven Adaptive Periodic Noise Canceller (APNC) in a closed-loop system. The concept of robustness provides a framework within which it is possible to make a useful assessment of algorithm performance. By adopting an analysis based on H^{∞}-theory, conditions are shown under which the APNC, driven by the LMS algorithm, will exhibit robust performance properties. Results are presented for the case of a broadband signal input to a one-dimensional closed-loop system. They display the relationship between the algorithm stepsize, the magnitude of the feedback coefficient and their bounds for robust performance. This result can be directly related to the use of the APNC in an echo control application.

A Binaural System For The Suppression Of Late Reverberation

Authors:
K. Lebart, J.M. Boucher, *ENST-Br (FRANCE)*
P.N. Denbigh, *University of Sussex (UK)*

ABSTRACT

The propriety of spatial decorrelation of late reverberation has often been used in binaural systems of dereverberation. However, the use of a small array of two microphones limits the performance of such systems, especially at low frequencies. We present in this paper a new algorithm, derived from the classical method proposed by Bloom et al.. Both methods are assessed in terms of gain in signal to noise ratio (SNR), noise reduction, distortion and improvement in performances of automatic speech recognition. The proposed method leads to better noise reduction and consistent improvement in speech recognition scores.

The All-Pass Filtered-X Algorithm

Authors:
J. Garas, P. C.W. Sommen, *Eindhoven University of Technology (NETHERLANDS)*

ABSTRACT

The convergence speed of the filtered-x LMS algorithm is known to be slow due to the secondary acoustic path in front of the adaptive filter. Two methods are presented to improve the convergence properties of that algorithm. The exact method, referred to as the decorrelated filtered-x algorithm, suffers from 'the division by small number' problem and an alternative approximate method is proposed. The approximate method, referred to as the all-pass filtered-x algorithm, avoids division by using only the phase information of the secondary acoustic path to generate the required filtered-x signal, which is equivalent to performing partial decorrelation. Performance comparison of the diffrent algorithms shows that the all-pass filtered-x, although approximate, performs better than the decorrelated filtered-x.

Multichannel Noise Reduction
- Algorithms And Theoretical Limits -

Authors:

J. Bitzer, K.-D. Kammeyer, *University of Bremen (GERMANY)*
K.U. Simmer, *Aureca GmbH (GERMANY)*

ABSTACT

In this contribution we present a theoretical study on the performance of several multichannel noise reduction algorithms. It is known that the magnitude squared coherence (MSC) determines the performance of a class of adaptive algorithms i.e. active noise control or noise reduction with a reference microphone. However, the MSC is not sufficient for performance evaluation of other noise reduction methods like the Generalized Sidelobe Canceller (GSC) or adaptive post--filter techniques. We propose to measure the spatial complex coherence (CC) or normalized cross power spectrum of the sound field and show that it can be used as a much more general tool for performance analysis. First of all, we summarize the results of previous studies and present new results for the performance of the Generalize Sidelobe Canceller (GSC) as a function of the complex coherence. In the second part we examine different noise fields to show theoretical limits of multichannel noise reduction schemes.

Design Of A Transform Coder For High Quality Audio Signals

Authors:
M. J. Absar, G. Sapna, A.o Alvarez, *SGS-Thomson Microelectronics Pte. Ltd. (SINGAPORE)*

ABSTRACT

The design and implementation of a transform coder which exploits the frequency domain psychoacoustic model of the human auditory system is presented here. The design of an audio coder for consumer applications require trade-offs to be made in terms of MIPS (million instructions per second), memory and algorithms. Algorithms for each processing stage of the encoder have been designed keeping these concerns in view. In particular, for efficient coding of spectral exponents, a Neural Network based solution is presented.

A New Subband Perceptual Audio Coder Using CELP

Authors:

O. Van der Vrecken, *BaBel Technologies SA (BELGIUM)*

L. Hubaut, F. Coulon, *Faculté Polytechnique de Mons (BELGIUM)*

ABSTRACT

This paper presents an audio coding system which uses filter banks to decompose, in the frequency domain, the audio signal into constant width subbands. A specific compression is applied in each subband. This compression is achieved by means of CELP coders. In order to obtain a high audio quality, psychoacoustic models allocate dynamically the number of bits needed in each subband. A particular care has been taken for the elaboration of the filter banks in order to limit the delay and the computational cost of the system. We have implemented several filter banks and tested their influence on the perceptual quality of the output audio signal. Finally, we show that our proposed coder is capable of delivering excellent audio signal quality at bit rates of 50-60 kbit/s.

Auditory Modeling
Via Frequency Warped Wavelet Transform

Authors:
G. Evangelista, S. Cavaliere, *University "Federico II," (ITALY)*

ABSTRACT

In this paper we present an approach to auditory modeling based on frequency warped wavelet transforms. This set of wavelets overcomes the limitations of ordinary wavelets given by octave band resolution or by rational sampling rate filterbanks. The frequency bandwidths of the basis sequences may be arbitrarily designed by selecting a single parameter for each scale level. This parameter controls the pole position of a chain of identical first order all-pass filters, which is the computational structure of the Laguerre transform. The Laguerre parameter determines the amount of warping necessary at each scale in order to match the prescribed cutoff frequencies. Bark, mel or other filterbanks based on perceptual criteria can be exactly implemented by means of our transform, taking full advantage of the othogonality and completeness of the transform.

Wavelet Transform Based Coherence Function For Multi-Channel Speech Enhancement

Authors:
D. Mahmoudi, A. Drygajlo, *Swiss Federal Institute of Technology at Lausanne, (SWITZERLAND)*

ABSTRACT

This paper addresses the problem of enhancing a speech signal acquired by a microphone array used for hands-free voice communication applications. A new algorithm based on the coherence function developed in the wavelet domain and applied to the beamforming output signal is presented. The wavelet coherence function based nonlinear post-filtering provides a further noise suppression. Its performances is comparable to that of the wavelet Wiener filter proposed in our earlier work.

The Auditory Critical Bands Interpreted As A Local Kautz Transformation

Authors:
A. C. den Brinker, *Eindhoven University of Technology (NETHERLANDS*)

ABSTRACT

The Zwicker data describes the relation between the critical bands and the center frequencies of the auditory system. The data can roughly be divided into two parts. At the low center frequencies the critical bandwidth is constant. At the high center frequencies the bandwidth increases in a monotonic way with the center frequency. The following interpretation of this data is proposed. It is assumed that the auditory system performs a running orthogonal transformation, i.e., first the signal is windowed (localized in time) and next an orthogonal transformation is performed on the windowed signal. The constant bandwidth at the low center frequencies is interpreted as stemming from the (constant) window. The varying bandwidth at the high center frequencies can be modelled by a Kautz transformation. It is shown that such an interpretation holds not only qualitatively, but quantitatively as well.

A Classification Of Byzantine Singing Voices Based On Singer's Formant

Authors:
D. S. Delviniotis, *University of Athens (GREECE)*

ABSTRACT

A classification of different singing voices of Greek psaltes(chanters) of the Church, based on the relative positions and shapes of the "singer's formant" (SF) is presented. It was found that there are 8 categories defined on basis of the following criteria: 1) the location of the center of the clustered vowels in the corre-logram (F3-F4), 2) the maximum of F3 and minimum of F4 or F5, 3) the clustering of F3-F4-F5, and, 4) the variability of singer's formant shape among vowels of individual psaltic voice. There are two main singer's formants contrary to that (only one) of famous opera male singers.

Some Thoughts
On Morphological Pyramids And Wavelets

Authors:
H. J.A.M. Heijmans, *CWI (NETHERLANDS)*
J. Goutsias, *The Johns Hopkins University (USA)*

ABSTRACT

The aim of this paper is to show how one can use tools from mathematical morpho-logy in a systematic manner for the development of signal and image pyramids and wavelets. The abstract framework is illustrated by means of simple examples.

New Fast DCT Algorithms
For Composite Sequence Lengths

Authors:
G. Bi, L. W. Yu, *Nanyang Technological University (SINGAPORE)*

ABSTRACT

This paper presents fast algorithms for the type-II discrete cosine transform of composite sequence length. In particular, a radix-q algorithm, where q is an odd integer, is derived for uniform or mixed radix decomposition of the discrete cosine transform. Compared to other reported fast algorithms, the proposed one uses a smaller number of operations and provides a wider range of choices of sequence lengths.

Fast Algorithms For The Recursive Computation Of Two-Dimensional Discrete Cosine Transform

Authors:

W.-H. Fang, N.-C. Hu, S.-K. Shih, *National Taiwan University of Science and Technology (TAIWAN)*

ABSTRACT

This paper presents an efficient algorithm for computing the two-dimensional discrete cosine transform (2-D DCT) of size p^r*p^r, where p is a prime. The algorithm decomposes the 2-D DCT outputs into three parts: the first part contains outputs whose indices are both multiples of p and forms a 2-D DCT of size p^{r-1}*p^{r-1}, whereas the remaining outputs are further decomposed into two parts, depending on the summation of their indices. The latter two parts can be reformulated as a set of circular correlation (CC) or skew-circular correlation (SCC) matrix-vector products. Such a decomposition procedure can be repetitively carried out, resulting in a sequence of CC and SCC matrix-vector products. Employing fast algorithms for these CC/SCC operations, we can thus obtain algorithms with minimum multiplicative complexity.

Fast Actualization Of Moments In Sliding Window Applications

Authors:
J. Martinez, F. Thomas, *Institut de Robotica i Informatica Industrial (CSIC -- UPC) (SPAIN)*

ABSTRACT

This paper describes a fast actualization rule for accumulation moments. Geometric and accumulation moments are related through a one-to-one linear transformation: any set of geometric moments up to a given order is uniquely related to a set of accumulation moments up to the same order. When accumulation moments are used instead of geometric moments as description parameters in sliding window applications, it is shown how the actualization cost drops from O(m^3) to O(m^2), where m refers to the amount of moments. Therefore, any application of this kind that traditionally required geometric moments can benefit from the use of accumulation moments.

The Fast Computation Of DCT In JPEG Algorithm

Authors:
M. Popoviæ, T. Stojiæ, *University of Belgrade (YUGOSLAVIA)*

ABSTRACT

In this paper two new parallel algorithms are proposed for the computation of Discrete cosine transform (DCT) in JPEG algorithm on 32-bit computers. These algorithms are twice as fast as the fastest known DCT algorithm published to date. This is possible because 16-bit wordlength is sufficient for the computation of DCT, so the arithmetic unit is used to perform simultaneously two 16-bit operations. All operations in the DCT algorithm, that are not suitable for parallel processing, are substituted by equivalent operations.

Optimized Frame Design
For A Matching Pursuit Based Compression Scheme

Authors:

K. Engan, J. Haakon Husoy, S. Ole Aase, *Høgskolen i Stavanger (NORWAY)*

ABSTRACT

A technique for designing frames to use with vector selection algorithms, for example Matching Pursuits (MP), is presented along with a novel compression scheme using these optimized frames. The frame design algorithm is iterative and requires a training set. We apply the frame design algorithm and the complete multi-frame compression scheme to electrocardiogram (ECG) signals. Complete coding experiments using the optimized frames in the novel compression scheme are compared with coding experiments from transform based compression schemes like the Discrete Cosine Transform (DCT) with run-length entropy coding. The experiments demonstrate improved rate-distortion performance for the multi-frame compression scheme compared to DCT at low bit-rates.

Transform-Domain Polynomial Filtering

Authors:
G. Peceli, T. Kovacshazy, A. R. Varkonyi-Koczy, *Technical University of Budapest (HUNGARY)*

ABSTRACT

Polynomial filters are nonlinear systems whose output is a truncated Volterra series expansion of the input or it is the solution of a recursive nonlinear difference equation. In this paper, for implementing polynomial filters, the framework of transform-domain digital signal processing is considered. This approach, due to recent advances in time-recursive evaluation of signal transforms, seems to be useful in a large variety of applications.

Design Of Multi-Delay Predictive Filters Using Dynamic Programming

Authors:
K. Koppinen, J. Yli-Hietanen, P. Handel, *Tampere University of Technology (FINLAND)*

ABSTRACT

Multi-delay predictive FIR filters utilizing a small number of multipliers are proposed. These filters are shown to have substantially lower noise gain than the standard minimum-length predictors using the same amount of multipliers. These filters are formulated for both arbitrary-order polynomial and sinusoidal signal prediction. The use of dynamic programming for the efficient optimization of these filters is proposed.

Signal Segmentation Using Time-Scale Signal Analysis

Authors:
A. Prochazka, M. Kolinova, J. Stribrsky, *Prague University of Chemical Technology (CZECH REPUBLIC)*

ABSTRACT

Change-point detection is a fundamental problem in many areas of signal segmentation, feature extraction and classification closely related to signal de-noising and modelling. There are various statistical methods that can be applied to estimate boundaries of signal segments. The paper presents another method based upon the wavelet transform and signal decomposition used to detect signal irregularities. This approach can be very efficient if the initial wavelet function is carefully chosen and a proper level of signal decomposition together with its threshold level is selected. Basic methods presented in the paper are verified for simulated sequences at first and then used for biomedical EEG signal segmentation and the subsequent classification using competitive neural networks. Resulting algorithms are validated in the MATLAB environment.

Anytime Algorithms
In Embedded Signal Processing Systems

Authors:
A. R. Varkonyi-Koczy, T. Kovacshazy, *Technical University of Budapest (HUNGARY)*

ABSTRACT

In this paper a family of so-called anytime signal processing algorithms is introduced to improve the overall performance of larger scale embedded digital signal processing (DSP) systems. In such systems there are cases where due to abrupt changes within the environment and/or the processing system temporal shortage of computational power and/or loss of some data may occur. It is an obvious requirement that even in such situations the actual processing should be continued to insure appropriate performance. This means that signal processing of somewhat simpler complexity should provide outputs of acceptable quality to continue the operation of the complete embedded system. The accuracy of the processing will be temporarily lower but possibly still enough to produce data for qualitative evaluations and supporting decisions. Consequently anytime algorithms should provide short response time and be very flexible with respect to the available input information and computational power. The paper presents such algorithms based on the re-configurable version of recursive signal transformer structures.

Fast Algorithms For Reduction A Modulo Polynomial And Vandermonde Transform Using FFT

Authors:
A. M. Krot, H. B.Minervina, *National Academy of Sciences of Belarus (BELARUS)*

ABSTRACT

This paper shows on how the real algorithms for the reduction a modulo arbitrary polynomial and fast Vandermonde transform (FVT) are realized on computer using fast Fourier transform (FFT). This real-valued FVT algorithm on the developed fast reduction polynomial algorithm is based. The realization of FVT algorithm on computer with real multiplicative complexity $O(2N\log_2^2 N)$ and real additive complexity $O(6N\log_2^2 N)$ is obtained. New FVT algorithm is applied in digital signal, filtering and interpolation problems.

Robust Adaptive Beamforming: Experimental Results

Authors:

A.B.Gershman, E.Nemeth, J.F.Boehme, *Ruhr University (GERMANY)*

ABSTRACT

The performances of adaptive array algorithms are known to degrade in scenarios with moving interfering sources. Recently, several robust approaches have been proposed to overcome this problem. Below, we compare conventional and robust algorithms using shallow sea sonar data with moving co-channel interference originated from shipping noise. This data set was recorded by a towed horizontal Uniform Linear Array (ULA) of 15 hydrophones. Our real data processing results demonstrate drastic performance improvements that can be achieved using robust algorithms relative to conventional adaptive beamforming techniques.

Modified Subspace Smoothing Technique For Multipath Direction Finding

Authors:
M. Feng, K.-D. Kammeyer, *University of Bremen (GERMANY)*

ABSTRACT

Recently a new subspace smoothing algorithm for estimation of directions of arrival (DOA) in coherent signal environment by using antenna arrays is presented. Unlike the existing forward and backward spatial smoothing algorithm, the new subspace smoothing algorithm is not restricted to uniformly linear arrays, and M, and M-1 DOA's of direct path and multipath components can be estimated, where M is the number of the sensors. Unfortunately the steering vectors have to be estimated several times in that algorithm, which is computationally very intensive. In order to reduce the computational cost but keep the advantages of the new idea, a modified version of the subspace smoothing algorithm will be presented in this paper. We will show that there is equivalence between the modified subspace smoothing algorithm and the well-known forward spatial smoothing algorithm. We will also show that the array aperture is extended virtually by using the modified subspace smoothing algorithm. Finally, simulation results will demonstrate the features of our modified algorithm.

Self Calibration In Presence Of Correlated Discrete Sources

Authors:
O. Grellier, *I3S-CNRS (FRANCE)*
P. Comon, *I3S-CNRS - Eurecom Institute (FRANCE)*

ABSTRACT

In this paper, we are concerned by the self-calibration of a linear equispaced array impinged by unknown correlated sources. We propose a new method based on the fact that sources have discrete distribution, which is meaningful in a digital communication context. The assumption of discrete sources allows an estimation, in the LS sense, of the mixing matrix which is then used to compute the sensor gain, phase and location errors.

On Forward-Backward Mode for Direction of Arrival Estimation

Authors:

M. Jansson, *Royal Institute of Technology (KTH) (SWEDEN)*
P. Stoica, *Uppsala University (SWEDEN)*

ABSTRACT

We apply the MODE (method of direction estimation) principle to the forward-backward (FB) covariance of the output vector of a sensor array to obtain what we call the FB-MODE procedure. The derivation of FB-MODE is an interesting exercise in matrix analysis, the outcome of which was somewhat unexpected: FB-MODE simply consists of applying the standard MODE approach to the eigenelements of the FB sample covariance matrix. By using an asymptotic expansion technique we also establish the surprising result that FB-MODE is outperformed, from a statistical standpoint, by the standard MODE applied to the forward-only sample covariance (F-MODE). We believe this to be an important result that shows that the FB approach, which proved quite useful for improving the performance of many suboptimal array processing methods, should not be used with a statistically optimal method such as F-MODE.

MVDR Beamforming with Inverse Updating

Authors:

M. Moonen, *Katholieke Universiteit Leuven (BELGIUM)*

I. K. Proudler, *DERA (UK)*

ABSTRACT

A stable alternative is described for the 'standard' systolic MVDR beamforming algorithm of McWhirter and Shepherd, which is known to suffer from linear round-off error build-up. The algorithm combines a so-called inverse QR-updating algorithm, with (part of) the McWhirter and Shepherd procedure. Similar to the McWhirter and Shepherd algorithm, it is amenable to parallel (pipelined) implementation. Unlike the McWhirter and Shepherd algorithm, it is stable numerically, and hence does not need repeated re-initializations.

High Resolution Array Processing For Non Circular Signals

Authors:

C. Adnet, *THOMSON CSF AIRSYS RD/RAN (FRANCE)*

P. Gounon, *Domaine Universitaire (FRANCE)*

J. Galy, *ENSICA (FRANCE)*

ABSTRACT

We present in this article a direction of arrival estimation algorithm for non circular sources. We show how to take into account non circularity of signals in array processing and develop extensions of classical algorithms. The main improvement linked to the non circularity concerns the resolution, the variance of estimation and the

number of resolvable sources. These characteristics are illustrated by simulations and theoretical analysis.

Spatio-Temporal Coding
For Radar Array Processing

Authors:
P. Calvary, D. Janer, *THOMSON-CSF AIRSYS (FRANCE)*

ABSTRACT

The aim of this paper is to present a new method allowing radar digital beamforming (DBF) with only one receiver channel. The key point is the use of a particular spatio-temporal waveform transmitted by an active phased array antenna. By properly choosing the temporal modulation applied on each of the transmission elements, one can transmit different signals in different directions, allowing (under certain hypotheses) angular localisation with only one receiver channel. The major advantage is the design of a low cost system providing DBF capabilities.

A Method Of Direction Finding Operating On An Heterogeneous Array

Authors:
Y. Erhel, *Centre de Recherches des Ecoles de Coetquidan (FRANCE)*
L. Bertel, *Universite deRennes 1 (FRANCE)*

ABSTRACT
In a multipaths context, direction finding must resort to high resolution algorithms. Classically, they operate on arrays of isotropic or identical sensors because they use the geometrical phase that the space diversity induces on the received signals. We propose to derive the standard music algorithm as the array is made up with sensor which are different from each other, assuming that their spatial complex responses are well know. An original computation is then described and some consequence advantages are underlined; for example, a new approach of the array ambiguity is presented, as the geometrical phase is no more the single relevant parameter attached to a direction of arrival. At last, some experimental results are listed in the field of h.f. radio direction finding, aiming to achieve a single site localization.

Adaptive Separation Of Sources And Estimation Of Their Directions Of Arrival

Authors:
O. Macchi, Z. Malouche, *CNRS-Supelec (FRANCE)*

ABSTRACT

We consider array processing when the multiple sources are statistically independent narrowband signals, as in CDMA radio communications with a multisensor receiver. For a uniform linear array, a simple source separation-based criterion is proposed which jointly recovers each source on one output channel. The correponding system has two stages whose respective parameters are the estimates of the source directions of arrival (DOA) and certain gains which constrain the output powers. The criterion involves a coupling parameter á so that all outputs recover different sources. When all

the sources have negative kurtosis, the criterion minima correspond to perfect estimation of the DOA. Adaptive implementation is possible at a very low computational price. After initialization, the coupling parameter á can be cancelled, which greatly simplifies implementation. These results have been confirmed through many computer simulations.

Maximum Likelihood Time-Of-Arrival Estimation Using Antenna Arrays: Application To Global Navigation Satellite Systems

Authors:
G. Seco, J. A. Fernandez-Rubio, *Polytechnic University of Catalonia (SPAIN)*

ABSTRACT

The problem of estimating the time-of-arrival (TOA) of a known signal in the presence of interferences and multipath propagation is addressed. This problem is essential in high precision receivers of the Global Navigation Satellite Systems. This paper presents the maximum likelihood TOA estimator when an antenna array is used in the receiver. The desired signal impinges the array with a known direction-of-arrival (DOA) vector, which allows to model all the undesired signal as unknown and arbitrary spatially correlated noise. This simplified model makes only the desired parameters remain in the formulation explicitly, then avoiding complex maximization schemes needed by other models. The fact that estimator is formulated in the frequency domain permits the introduction of the temporal correlation of the noise. Simulation results illustrate the satisfactory performance of the estimator.

Simultaneously Estimating Frequency And Direction Of Coherent Signal Via An Array Triplet In Motion

Authors:

X. Su, T. Chen, *University of Electronic Science and Technology of China (CHINA)*

ABSTRACT

Combining the merits of moving array and ESPRIT approach [9], we propose a new method for simultaneously estimating frequencies and directions of coherent narrowband signals with an array triplet in motion. This expands the multiple invariance ESPRIT [10] in the coherent signal environment. The computer simulation results have verified the performance of presented method.

Error Resilience And Concealment In Video Coding

Authors:

L. K. Katsaggelos, F. Ishtiaq, L. P. Kondi, M.-C. Hong, *Northwestern University (USA)*
M. Banham, J. Brailean, *Motorola Corporate Systems Research Laboratories (USA)*

ABSTRACT

In this paper we review error resilience and concealment techniques which have been developed both within and outside the various videoconferencing standards. We consider the H.324 videoconferencing standard with its accompanying lower-level H.263 and H.223 video and multiplexing standards, MPEG-2, and MPEG-4. We then describe an error resilience algorithm commonly used with variable length codewords, and finally review error concealment techniques that have appeared in the literature.

Networking Technologies For Future Video Communications

Authors:

K. Sano, S. Ono, *NTT Optical Network Systems Laboratories (JAPAN)*
N Ohta, *NTT Network Service Systems Laboratories (JAPAN)*

ABSTRACT

This paper describes advanced networking technologies that support high quality video and image communications. We focus on maintaining Internet compatibility and extracting the merit of ATM technologies. A concept of Global Mega-media Network (GMN) is introduced as the future multimedia network. As part of work associated with GMN, an overview of AMInet project and some of the promising technologies are described. It is demonstrated that proposed resource reservation protocol virtually supports switched VCs so that we can enjoy the merit of ATM through Internet compatible applications for video communications.

Multiple Description Coding Using Non-Hierarchical Signal Decomposition

Authors:
Y. Wang, *Polytechnic University of Brooklyn (USA)*

ABSTRACT

Multiple description coding (MDC) combined with multiple path transport (MPT) is a new research direction for image and video transmission over unreliable networks. Using the signal decomposition and reconstruction framework that has been proven very effective for image coding, several new approaches to MDC have been developed. Instead of using hierarchical decomposition, which is aimed at maximizing the coding efficiency, non-hierarchical decomposition is used to generate correlated descriptions, to facilitate the estimation of a lost description from a received one. In this paper, we review these recent developments.

Rate Control For Delay-constrained Video Transmission

Authors:
C,-Y. Hsu, A. Ortega, *University of Southern California (USA)*

ABSTRACT

We study the rate control problem for video transmission environments which require real-time encoding and transfer of the video data, and where rate-distortion (RD) optimal rate control approaches requiring a long encoding delay cannot be used. We propose two novel RD optimized rate control approaches based on dynamic programming and Lagrangian optimization.

Compression And Packetization For MPEG-2/4 Video And Audio

Authors:

T. Murakami, K. Asai, H. Ohira, Mitsubishi Electric Corporation (JAPAN)

ABSTRACT

We describe a flexible LSI architectural design concept suitable for various video-oriented multimedia applications. Our proposed LSI design assumes various schemes and picture formats of compression in a single-chip architecture. The hybrid architecture of an high-performed embedded Digital Signal Processor (DSP) core and dedicated hardware processing units, and its scaleable architecture are the key features. DSP core supports the processing that requires the complexity and flexibility and lower processing power. Hardware parts support the processing components that are frequently used and require processing power and lower complexity. In this architecture, evaluated performance shows that a single chip can perform MPEG MP@ML (Main Profile at Main Level) processing including video and audio codec, and media packetization. With its scaleable architecture, MPEG MP@HL (Main Profile at High Level) compression can be carried out with only 6 chips. This single LSI can also have a capability of extending to H.263 or forthcoming MPEG-4 standard algorithm by a single chip on the embedded DSP core basis.

MPEG-4 Transmission Over Wireless Networks

Authors:
R. Schafer, C. Fehn, A. Smolic, *Heirrich-Hertz-Institut fur Nachrichtentechnik Berlin GmbH, (GERMANY)*

ABSTRACT

This paper reviews some application scenarios for transmitting MPEG-4 encoded video over wireless networks. These include side information or interactive services in digital terrestrial video broadcast (DVB-T), audio-visual information over in-home networks and digital video over DAB (digital audio broadcast system). For the first two applications the "Delivery Multimedia Integration Framework (DMIF)" is investigated, while for the latter application encapsulation of MPEG-4 stream in MPEG-2 transport stream (TS) and appropriate preprocessing are considered.

Optimization Of Wireframe Model Adaptation And Motion Estimation In A Rate-distortion Framework

Authors:
D. Tzovaras, S. Vachtsevanos, M. G. Strintzis, *Aristotle University of Thessaloniki (GREECE)*

ABSTRACT

A rate-distortion framework is used to define a very low bit rate coding scheme based on wireframe model adaptation and optimized selection of motion estimators. This technique achieves maximum reconstructed image quality under the constraint of a target bitrate for the coding of the motion vector field, the wireframe representation information and the prediction error. Experimental results illustrating the performance of the proposed techniques in very low bit rate image sequence coding application areas are presented and evaluated.

High-order Motion Compensation
For Low Bit-rate Video

Authors:
B. Girod, T. Wiegand, E. Steinbach, M. Flierl, X. Zhang, *University of Erlangen (GERMANY)*

ABSTRACT

The concept of high-order motion compensation is introduced. It is argued that in hybrid video coding, motion-compensated prediction has to be viewed as a source coding problem with a fidelity criterion. Based on our considerations, various high order motion compensation approaches are presented that achieve significantly improved video coding results. The designed motion-compensated predictors achieve gains by exploiting high-order statistical dependencies in the video signal.

A Contribution To Video Coding Using Layers

Authors:
L. Torres, M. Lecha, *Polytechnic University of Catalonia (SPAIN)*

ABSTRACT

Layered representations of video sequences, originally introduced in [1], are important in several applications. Specifically, they are very appropriate for object tracking, object manipulation, content-based scalability and video coding which are among the main functionalities of the future standard MPEG-4, [2] [3]. In addition they will be also useful for MPEG-7 in the corresponding image analysis part. A robust representation of moving images based on layers has been presented in [4]. The objective of this paper is to provide the first results on video coding using the segmentation scheme presented in [4]. To that end, the shape information (alpha map) has been coded using a Multigrid Chain Code approach and the layers (the texture) have been coded using a Shape Adaptive DCT scheme that is being considered as a texture coding approach in the current definition of MPEG-4. Results are provided for a variety of compression ratios and different image quality that prove the validity of the approach.

Results On Cyclic Signal Processing Systems

Authors:

P. P. Vaidyanathan, *Dept. Electrical Engr., Caltech, Pasadena*

ABSTRACT

We present a state space description for cyclic LTI systems which find applications in cyclic filter banks and wavelets. We also revisit the notions of reachability and observability in the cyclic context, and show a number of important differences from traditional noncyclic case. A number of related problems such as the paraunitary inter-polation problem and the cyclic paraunitary factorizability problem can be understood in a unified way by using the realization matrix defined by the state space description.

Non-Uniformly Downsampled Filter Banks

Authors:

B.E. Sarroukh, A.C. den Brinker, *Eindhoven University of Technology (NETHERLANDS)*

ABSTRACT

The problem of perfect reconstructing non-uniformly down-sampled filter banks is considered using frame theory and filter bank theory. This problem can be reformulated to a uniformly downsampled filter bank thus allowing the usual analysis. Of special interest are those filter banks where the output of the analysis bank has a direct interpretation e.g., the sliding-window Fourier transform or the wavelet transform. The concept of the sliding-window Fourier transform can be extended by replacing the Fourier transformation by an arbitrary unitary transformation. As an example the sliding-window Kautz transformation is considered.

A Design Method
Of 2-D Axial-Symmetric Paraunitary Filter Banks
With A Lattice Structure

Authors:

S. Muramatsu, A. Yamada, H. Kiya, *Tokyo Metropolitan University (JAPAN)*

ABSTRACT

A lattice structure of 2-D axial-symmetric paraunitary filter banks (ASPUFBs) is proposed, which makes it possible to design such filter banks in a systematic manner. ASPUFBs consist of non-separable axial-symmetric (AS) filters, and can be regarded as a subclass of non-separable linear-phase paraunitary (PU) ones. The AS property is desirable for image processing, because it enables us to use the symmetric extension method. Since our proposed system structurally restricts both the PU and AS properties, it can be designed by using an unconstrained optimization process. A design example will be given to show the significance of the lattice structure.

Versatile Building Blocks For Multirate Processing Of Bandpass Signals

Authors:

M. Renfors, T. Kupiainen, *Tampere University of Technology (FINLAND)*

ABSTRACT

In this paper we develop a special down-converting bandpass decimator structure which can be used for reducing the sampling rate by a factor two without causing any disturbing aliasing effects for a bandpass signal located originally around quarter of the sampling rate (pi/2). Based on this special bandpass filter design and usual halfband lowpass and highpass decimation filters, an efficient stepwise decimation and mixing approach is developed for filtering out and down-converting a narrowband signal located in frequency domain anywhere within a wideband digitized signal. This kind of down-conversion and decimation approach finds applications in communications receivers where the role of DSP-based solutions is becoming more and more important.

Fast Filter Bank Synthesis By Periodical Anamorphosis

Authors:

C. Croll, *Laboratoire des Images et des Signaux (LIS) (FRANCE)*
D. Pellerin, J. Hérault, *Laboratoire des Images et des Signaux (LIS) - Université Joseph Fourier (FRANCE)*

ABSTRACT

We present a method for analysing spatial frequency and orientation, based on a space variant (SV) filter bank. Frequency response of each filter depend on its spatial position. Theses filters are inspired by retino-cortical map and describe a model ("Pinwheel") of orientation column of striate cortex. These SV filters result from the combination of two transformations: a periodical projection and a space invariant filtering. This special case of implementation of SV filtering brings a computed time gain in comparison to classical filter bank such as the wavelets Morlet 2D. After description of the method, we explain the different parameters which optimise the frequency cover. It appears that the filter bank constitutes good attributes for texture segmentation.

Constrained Least Squares Design Of FIR Filters Using Iterative Reweighted Least Squares

Authors:
C. S. Burrus, *Rice University, (USA)*

ABSTRACT

This paper presents a new application of the iterative reweighted least squared error algorithm to obtain a constrained least squares approximation for designing a linear phase FIR filter. A new method is given which has good initial convergence coupled with quadratic final convergence. We compare two methods of using the IRLS approach to the CLS problem.

Design Of Polynomial Interpolation Filters Based On Taylor Series

Authors:
J. Vesma, R. Hamila, T. Saramäki, M. Renfors, *Tampere University of Technology (FINLAND)*

ABSTRACT

Interpolation filters are used to interpolate new sample values at arbitrary time instants between the existing discrete-time samples. Interpolation filters utilizing polynomial-based interpolation can be efficiently implemented using the Farrow structure. This paper introduces a new method for designing Farrow-structured interpolation filters. The proposed synthesis method is based on the relationship between the Farrow structure and the Taylor series of the interpolating continuous-time signal formed based on the existing sample values.

A Design Method For Half-Band Filters

Authors:
A. N. Willson, Jr. H. J. Orchard, *UCLA (USA)*

ABSTRACT

Earlier methods of design have used iterative approaches such as the well-known Parks-McClellan algorithm or some variant of linear programming. Here we give a direct method of design, using Chebyshev polynomials, which provides a reduction in design time over previous methods by about a factor of ten. The ideal equal-ripple response in both passband and stopband is not achieved exactly, but the error is extremely small and would normally be undetectable in practical realizations.

Design Of Multiplierless Elliptic IIR Halfband Filters And Hilbert Transformers

Authors:
M. D. Lutovac, *IRITEL Institute (YUGOSLAVIA)*
L. D. Milic, *Mihajlo Pupin Institute (YUGOSLAVIA)*

ABSTRACT

A new design method for multiplierless IIR halfband filters and Hilbert transformers is presented. The coefficients of the halfband filter and the Hilbert transformer are determined using the same procedure. The minimal number of shift-and-add operation design (multiplierless) is based on the sensitivity analysis and the phase tolerance scheme.

New Realizations Of Second-Order All-Pass Transfer Functions

Authors:
A. Simic, M. Lutovac, *IRITEL Institute (YUGOSLAVIA)*

ABSTRACT

This paper presents an implementation of a systematic method for realization of the second-order all-pass transfer functions using the minimum number of multipliers. The presenteded realizations are analyzed with regard to the error due to roundoff of the products. The realizations are compared in order to determine optimum realization for particular pole position. Using these filter sections for realisation of some recursive transfer function as the parallel connection of two all-pass networks, will lead us to the computationally efficient realization with low pass-band sensitivity.

Linear Projection Algorithms And Morphological Dynamic Link Architecture For Frontal Face Verification

Authors:
C. Kotropoulos, I. Pitas, *Aristotle University of Thessaloniki (GREECE)*

ABSTRACT

The combined use of linear projection algorithms with a variant of dynamic link architecture based on the multi-scale morphological dilation-erosion is proposed for face verification. The performance of the combined scheme is evaluated in terms of the receiver operating characteristic (ROC) for several threshold selections on the matching error in the M2VTS database. The experimental results indicate that the incorporation of linear projections in the morphological dynamic link architecture improves significantly the verification capability of the method.

Nonlinear Scale Decomposition Based Features For Visual Speech Recognition

Authors:

I. Matthews, J. A. Bangham, R. Harvey, S. Cox, *University of East Anglia (UK)*

ABSTRACT

A mathematical morphology based filter structure called a sieve is used to process mouth image sequences of a talker's mouth and form visual speech features. The effects of varying the type of filter, the post-processing and hidden Markov model (HMM) parameters on recognition accuracy are investigated using two audio-visual speech databases.

Wire Frame Fitting For Automatic Tracking In Model-Based Video Coding

Authors:

P. M. Antoszczyszyn, J. M. Hannah, P. M. Grant, *The University of Edinburgh (UK)*

ABSTRACT

Model-based video coding requires the application of both image processing and machine vision techniques for proper fitting of the semantic model and its subsequent tracking throughout the rest of the sequence of a certain type (e.g. 'head-and-shoulders' or 'head-only'). In this article a method of automatic semantic wire-frame fitting based on a reference data-base of facial images is presented. The method has been tested on a widely used data-base of images with very good results. It was possible to accurately retrieve the position of the facial features in all cases. The position of the facial features in initial frames can subsequently be used in automatic tracking. Results of automatic fitting are presented as a part of this contribution. Experimental results are also available on-line in the form of compressed movies from our Internet site at **Error! Bookmark not defined.**.

Region-Based Segmentation And Tracking Of Human Faces

Authors:
V. Vilaplana, F. Marques, P. Salembier, L. Garrido, *Universitat Politecnica de Catalunya (SPAIN)*

ABSTRACT

A new algorithm for face segmentation and tracking is presented. The face segmentation step relies on a fine segmentation of the image based on color homogeneity. In order to reduce the number of possible mergings, a merging order using chrominance homogeneity is applied. The sequence of mergings is represented as a binary tree. A set of regions (a node in the tree) is selected maximizing an estimation of the likelihood of being a face. The final face partition is obtained by a region merging process. The face tracking step uses motion information. The face partition in the previous partition is motion compensated (projected). The final face partition is obtained using the previous region merging process, driven by the projected face region.

Facial Feature Extraction In Frontal Views Using Biometric Analogies

Authors:
S. Tsekeridou, I. Pitas, *Aristotle University of Thessaloniki, (GREECE)*

ABSTRACT

Face detection and facial feature extraction are considered to be key requirements in many applications, such as access control systems, model-based video coding, content-based video browsing and retrieval. Thus, accurate face localization and facial feature extraction are most desirable. A face detection and facial feature extraction in frontal views algorithm is described in this paper. The algorithm is based on principles described in [1] but extends the work by considering: (a) the mirror-symmetry of the face in the vertical direction and (b) facial biometric analogies depending on the size of the face estimated by the face localization method. Further improvements have been added to the face localization method to enhance its performance. The proposed algorithm has been applied to frontal views extracted from the European ACTS M2VTS database with very good results.

The Dual-Tree Complex Wavelet Transform: A New Efficient Tool For Image Restoration And Enhancement

Authors:
N. Kingsbury, *University of Cambridge (UK)*

ABSTRACT

A new implementation of the Discrete Wavelet Transform is presented for applications such as image restoration and enhancement. It employs a dual tree of wavelet filters to obtain the real and imaginary parts of the complex wavelet coefficients. This introduces limited redundancy (4 : 1 for 2-dimensional signals) and allows the transform to provide approximate shift invariance and directionally selective filters (properties lacking in the traditional wavelet transform) while preserving the usual properties of perfect reconstruction and computational efficiency. We show how the dual-tree complex wavelet transform can provide a good basis for multi-resolution image denoising and deblurring.

Spectral Amplitude Estimation-Based X-Ray Image Restoration: An Extension Of A Speech Enhancement Approach

Authors:

T. Aach, *Medical University of Lubeck (GERMANY)*

D. Kunz, *Philips Research Laboratories (GERMANY)*

ABSTRACT

This paper describes a class of spectral amplitude estimation-based algorithms for the restoration of low dose X-ray images. Since estimation of spectral amplitude from noisy observations is a widely reported approach to restore noisy speech signals, we discuss our algorithms from the viewpoint of extending a well established speech processing technique to images. We include an analysis of residual noise. Moreover, we provide qualitative and quantitative processing results.

Bayesian Deconvolution Of Poissonian Point Sources

Authors:
G. Stawinski, *LETI CEA-Technologies Avancees CEN Saclay (FRANCE)*
A. Doucet, *LETI CEA-Technologies Avancees CEN Saclay (FRANCE), University of Cambridge (UK)*
P. Duvaut, *ETIS-ENSEA (FRANCE)*

ABSTRACT

In this article, we address the problem of Bayesian deconvolution of point sources with Poisson statistics. A high level Bayesian approach isproposed to solve this problem. The original image is modeled as a list of an unknown number of points sources with unknow parameters. A prior distribution reflecting our degree of belief is introduced on all these unknown parameters, including the number of sources. All Bayesian inference relies on the posterior distribution. This latter admitting no analytical expression, we estimate it using an original Reversible Jump Markov Chain Monte Carlo method. The algorithm developed is tested over real data. It displays satisfactory results compared to traditional low level Bayesian approaches.

Modelling PSF Of Scanning Electron Microscopes For Image Restoration

Authors:
J. Swindells, M. Razaz, K. Tovey, *University of East Anglia (UK)*

ABSTRACT

A procedure to determine the point spread function (PSF) of a two-dimensional SEM imaging system based on experimental data is presented. The specimen required for capture by the imaging system is simple to manufacture, and only requires a sharp edge to be of use. An overview is given of caveats that exist at each stage of the process, in addition to a breakdown of the process itself. Results based on a Scanning Electron Microscope are shown. The use of a PSF estimated in this fashion is shown to result in much improved restorations, when compared to its theoretical equivalent.

Iterative Blind Image Restoration Using Local Constraints

Authors:

K. May, T. Stathaki, *Imperial College of Science(UK)*
A. Katsaggelos, *Northwestern University (USA)*

ABSTRACT

A new method of incorporating local image constraints into blind image restoration is proposed. The local mean and variance of the degraded image are used to obtain an initial estimate of the pixel intensity bounds. As the restoration proceeds, the bounds are updated from the current image estimate. The iterative-bound algorithm shows an improvement over the use of fixed bounds taken from the blurred image, in which case underestimation of the variance occurs at edges and textures. Simulations are presented for both the fixed- and iterative-bound implementations.

On The Integration Of Dialect And Speaker Adaptation In A Multi-Dialect Speech Recognition System

Authors:

V. Digalakis V. Doumpiotis S. Tsakalidis, Technical University of Crete (GREECE)

ABSTRACT

The recognition accuracy in recent Automatic Speech Recognition (ASR) systems has proven to be highly related to the correlation of the training and testing conditions. Several adaptation approaches have been proposed in an effort to improve the speech recognition performance, and have typically been applied to the speaker- and channel-adaptation tasks. We have shown in the past that a mismatch in dialects between the training and testing speakers significantly influences the recognition accuracy, and we have used adaptation to compensate for this mismatch. The dialect of the speaker needs to be identified in a dialect-specific system, and in this paper we present results in this area. To achieve further improvement in recognition performance, we combine dialect- and speaker-adaptation.

Combining Bayesian Learning And Vector Field Smoothing For On-Line Incremental Speaker Adaptation

Authors:

C. Vair, L. Fissore, *CSELT - Centro Studi E Laboratori Telecomunicazioni (ITALY)*

ABSTRACT

In this paper we investigate the combination of Bayesian Learning (also known as Maximum A Posteriori - MAP) and Vector Field Smoothing (VFS) to the on-line incremental speaker adaptation of Continuous Density Hidden Markov Models (CDHMMs). The parameters of the Gaussian mixture output densities are adapted during the MAP step, using the exponential forgetting mechanism and performing the a-priori parameter estimation in a model based outline. The unadapted MAP models are then reestimated using the VFS technique. Several tests compare the error rate reduction as a function of the incremental adaptation step and of the size of adaptation data for each step. The experiments were run on a speaker dependent continuous speech recognition task, for the Italian language, with a test vocabulary size of 247 words, without any language models. The initial speaker independent models were trained on PSTN speech data, while the adaptation data were collected in a quiet environment through a PBX connection. The cepstral mean normalization (CMN) was used to deal with the acoustic mismatch.

Identification Of Spoken European Languages

Authors:

D. Caseiro, I. Trancoso, *INESC/IST (PORTUGAL)*

ABSTRACT

Automatic spoken language identification is the problem of identifying the language being spoken from a sample of speech by an unknown speaker. In this paper we studied the problem of language identification in the context of the European languages, which allowed us to study the effect of language proximity in Indo-European languages. The results reveal a significant impact on the identification of some languages. Current language identification systems vary in their complexity. The systems that use higher level information have the best performance. Nevertheless, that information is hard to collect for each new language. The system presented in this work is easily extendable to new languages because it uses very little linguistic information. In fact, the presented system needs only one language specific phone recogniser (in our case the Portuguese one), and is trained with speech from each of the other languages. With the SpeechDat-M corpus, with 6 European languages (English, French, German, Italian, Portuguese and Spanish) our system achieved an identification rate of about 79% on 5-second utterances.

A Bayesian Triphone Model With Parameter Tying

Authors:

J. Ming, M. Owens, F. J. Smith, *The Queens University of Belfast (UK)*

ABSTRACT

This paper introduces a new statistical framework for constructing triphonic models from models of less context-dependency. The new framework is derived from Bayesian statistics, and represents an alternative to other triphone-by-composition techniques, particularly to the model-interpolation and quasi-triphone approaches. The potential power of this new framework is explored by an implementation based on the hidden Markov modeling technique. It is shown that the new model structure includes the quasi-triphone model as a special case, and leads to more efficient parameter es-timation than the model-interpolation method. Two strategies of state-level tying have been investigated within the new model structure. Phone recognition experiments on the TIMIT database show an increase in the accuracy over that obtained by other systems.

Spectral Subtraction And Missing Feature Modeling For Speaker Verification

Authors:
A. Drygajlo, M. El-Maliki, *Swiss Federal Institute of Technology Lausanne (SWITZERLAND)*

ABSTRACT

This paper addresses the problem of robust text-independent speaker verification when some of the features for the target signal are heavily masked by noise. In the framework of Gaussian mixture models (GMMs), a new approach based on the spectral subtraction technique and the statistical missing feature compensation is presented. The identity of spectral features missing due to noise masking is provided by the spectral subtraction algorithm. Consequently, the statistical missing feature compensation dynamically modifies the probability computations performed in GMM recognizers. The proposed algorithm uses a variation of the general ized spectral subtraction and incorporates in it a criterion based on masking properties of the human auditory system. The originality of the algorithm resides in the fact that instead of using fixed parameters for the noise reduction and missing feature compensation, the noise masking threshold is used to control the enhancement and model compensation processes adaptively, frame-by-frame, hence helping to find the best trade off.

Blind Equalization In The Cepstral Domain For Robust Telephone Based Speech Recognition

Authors:
L. Mauuary, *France Télécom (FRANCE)*

ABSTRACT

An adaptive filter in a blind equalization scheme has recently been proposed to reduce telephone line effects for speech recognizers. The implementation of the blind equalization scheme using a circular-convolution frequency domain adaptive filter has been described in a previous paper. This paper presents a new implementation of this blind equalization scheme in the cepstral domain. The property of a constant long-term cepstrum of speech helps to compute the gradient used for adapting the weights. The performances of the spectral domain and the cepstral domain implementations are then compared. These filters prove to be efficient for the channel equalization task. Furthermore, speech recognition experiments show that the cepstral domain on-line adaptive filter outperforms the cepstral trajectories high-pass filter (on-line approach). This technique offers almost the same performance as the conventional cepstral substraction technique (off-line approach).

Robust Speech Recognition Algorithms In A Car Noise Environment

Authors:
L. Cong, S. Asghar, *Advanced Micro Devices (USA)*

ABSTRACT

In this paper, we present several new robust isolated word speech recognition systems which employ FMQ/MQ as the spectral labelling process, followed by a Hidden Markov Model (HMM), or a HMM and Neural Network (HMM/MLP) classification technique. The ISWR systems provide selective input data to a neural network in response to speech signal to acoustic noise ratios to improve speech recognition system performance. Simply, FMQ/HMM system can exploit error compensation from FVQ/HMM processes. TIDIGITS and NOSEX_92 [7] have been used as the speech and noise databases. These robust algorithms ensure a high recognition accuracy performance even at input SNR as low as 5 and 0 dBs.

Frame Pruning For Automatic Speaker Identification

Authors:
L.Besacier, *LIA/CERI - 339 (FRANCE), IMT Neuchâtel (SWITZERLAND)*
J.F. Bonastre, *LIA/CERI - 339 (FRANCE)*

ABSTRACT

In this paper, we propose a frame selection procedure for text-independent speaker identification. Instead of averaging the frame likelihoods along the whole test utterance, some of these are rejected (pruning) and the final score is computed with a limited number of frames. This pruning stage requires a prior frame level likelihood normalization in order to make comparison between frames meaningful. This normalization procedure alone leads to a significative performance enhancement. As far as pruning is concerned, the optimal number of frames pruned is learned on a tuning data set for normal and telephone speech. Validation of the pruning procedure on 567 speakers leads to a significative improvement on TIMIT and NTIMIT (up to 30% error rate reduction on TIMIT).

Integration Of Parsing And Incremental Speech Recognition

Authors:

S. Wachsmuth , G. A. Fink, G. Sagerer, *University of Bielefeld (GERMANY)*

ABSTRACT

In this paper we propose a new approach to integrating a parser into a statistical speech recognizer. The method is able to incrementally apply grammatical restrictions and robustly combine them with the statistical acoustic and language models. On spontaneous speech data a 11.6% reduction in word error rate could be achieved compared to the baseline system applying statistical models only.

Optimizing Hidden Markov Models for Chinese An-set Syllables

Authors:
Q. H. He, *South China University of Technology (CHINA)*
S. Kwong, *City University of Hong Kong (HK)*

ABSTRACT

Speech recognition for Chinese relied very much on the recognition of Chinese syllables and there are altogether 1345[7] syllables in it. If we take tones into considerations, the number of syllables can reduce to 408 base syllables, with different tones, in which it can further divided into 38 confused set. Among those sets, the Chinese An-set is considered as one of the major confused syllable set. Thus, the recognition of Chinese An-set syllables is very important to the Chinese recognition. In this paper, we proposed a new training approach based on *maximum model distance* (MMD) for HMMs to train the Chinese An-set syllables. Both the speaker-dependent and multi-speaker experiments on the confused Chinese An-set showed that significant error reduction can be achieved through the proposed approach.

A Unified Approach
To Laterally-Connected Neural Nets

Authors:
S. Fiori, A. Uncini, *University of Ancona (ITALY)*

ABSTRACT

The aim of this paper is to present a new unified approach to real- and complex-valued Principal Component Extractor with laterally-connected neural nets as the APEX (Kung-Diamantaras) and the cAPEX (Chen-Hou) based on an optimization theory specialized for such architectures. We firstly propose an optimization formulation of the problem and study how to recursively determine solutions by means of gradient-based algorithms. In this way we find a class of learning rules called ø-cAPEX containing, as a special case, a cAPEX-like one. Through simulations we finally compare the convergence speed and numerical precision at equilibrium of cAPEX and some members of ø-cAPEX.

A Class Of Fast Complex Domain Neural Networks For Signal Processing Applications

Authors:
A. Uncini, F. Piazza, *Università di Ancona Italy (ITALY)*

ABSTRACT

In this paper, we study the properties of a new kind of complex domain artificial neural networks called complex adaptive spline neural networks (CASNN), which are able to adapt their activation functions by varying the control points of a Catmull-Rom cubic spline. This new kind of neural network can be implemented as a very simple structure being able to improve the generalization capabilities using few training epochs. Due to its low architectural complexity this network can be used to cope with several nonlinear DSP problem at high throughput rate.

Efficient Layer-Wise Learning Of Feedforward Neural Networks Using The Backpropagation

Authors:
N. S. Rubanov, *Belarussian State University (BELARUS)*

ABSTRACT

In this paper, we propose the efficient method for learning a feedforward neural network (FNN) that combines the backpropagation (BP) strategy and the layer-wise learning methods (LWMs). More precisely, for updating weights of each layer we use the BP-based iterative procedure. This procedure provides the realistic conditions for fast convergence of the learning process to a global minima. As a result, the new method uses advantages and overcomes the disadvantages associated with both the BP method and LWMs**.**

Recurrent Neural Networks And Filters Adaptation With Stability Control

Authors:
P. Campolucci, F. Piazza, *Università di Ancona (ITALY)*

ABSTRACT

Recurrent Neural Networks and linear Recursive Filters can be adapted on-line but sometimes with instability problems. Stability control techniques exist for the linear case but they are either computationally expensive or non-robust. For the non-linear case, stability control is simply never done. This paper presents a new stability control method for IIR adaptive filters that makes possible to continually adapt the coefficients with no need of stability test or poles projection. This method can be applied to various realizations: direct forms, cascade or parallel of second order sections, lattice form. It can be implemented to adapt a simple IIR adaptive filter or a locally recurrent neural network such as the IIR-MLP with improved performance over other techniques and over not controlling the stability.

A Nonexclusive Classification System Based On Co-Operative Fuzzy Clustering

Authors:

F.M. Frattale Mascioli, G. Risi, A. Rizzi, G. Martinelli, *University of Rome "La Sapienza" (ITALY)*

ABSTRACT

Nonexclusive classification characterizes many real problems in which a hard decision about data labels cannot be taken. In these cases, a decision system capable of fuzzy outputs is desirable, in order to well describe problem's nature and domain. In this paper, an algorithm pursuing this approach is presented. More precisely, a nonexclusive k-class problem is solved by the co-operation of k independent clustering systems. In order to better evaluate and compare the presented neuro-fuzzy classifier in simulation tests, we also propose a fuzzy classification quality (FCQ) measure.

Fuzzy Logic Control Based On Weighted Rules

Authors:

J.-S. Cho, D.-J. Park, *Korea Advanced Institute of Science and Technology(KAIST) (KOREA)*

ABSTRACT

In this paper a method of fuzzy logic control based on possibly inconsistent if-then rules representing uncertain knowledge or imprecise data is studied. When it is hard to obtain consistent rule bases, we propose the fuzzy logic control based on weighted rules depending on output performances using the neural network and we derive the weight updating algorithm. With the final weight change informations, we can make better decisions by taking into consideration conflicting rules. Computer simulations show the proposed method has good performance to deal with the inconsistent rule base in the fuzzy logic control. And real application problems are also discussed.

The Edge Detection From Images Corrupted By Mixed Noise Using Fuzzy Rules

Authors:
H. Ishii, A. Taguchi, M. Sone, *Musashi Institute of Technology (JAPAN)*

ABSTRACT

The edge detection is one of the basic and important process for the image processing. When we detect the edge from noisy images, it is necessary to reduce noises by filtering beforehand. However, even if nonlinear filtering degrade the image edges and details. We have already proposed the method of edge detection from images that corrupted by impulse noise. In this paper, a novel method of edge detection from images that corrupted by mixed noise (i.e., Gaussian noise plus impulse noise) using fuzzy inference is proposed. The proposed method consists of two set of fuzzy rules .The aim of one set of fuzzy rules is the estimating of the number of impulse noise in the window. The aim of the other set of fuzzy rules is the detection of image edges from image corrupted by only Gaussian noise, on the assumption that the number of impulse noise in the window is known. Thus, the edge detection and mixed noise reduction are realized at the same time by the proposed method .

A Combination Of Classical And Fuzzy Classification Techniques An A Self Organized Memories (SOM)-Type Neural Network Computational Platform

Authors:
S. G. Tzafestas, S. N. Raptis, *National Technical University of Athens (GREECE)*

ABSTRACT

In this paper a complex classification scheme including a combination of a cluster generator, based on a neural network and three different types of classifiers, is proposed. The first part of the scheme consists of a self organized memory (S.O.M.)-type unsupervised neural network which converges quickly to the clusters′ vectors. In the second part, which is the main classification component, three classifiers are compared. This comparison is made in a such a way so as to study the improvement of the performance and classification when going from purely classical classification schemes to fuzzy ones, and thus emphasize on the usefulness and reliability of the fuzzy set theory. The overall approach aims at showing that neural networks and fuzzy classifiers can be combined so as to exploit the advantages of both approaches in classification problems. Neural networks, on the one hand, operate very quickly after they are trained, but they are not capable of easily recognizing information not familiar to them. On the other hand, fuzzy systems overcome with success this drawback because they are generalized by their own nature. However they do not provide us with straight decisions. Instead, they give an estimation of the nature of the associated problem. Thus, they give us more than one possible solutions. In the literature, the most common way of putting together the concepts of neural networks and fuzzy systems is the development of neuro-fuzzy schemes which are constructed by fuzzy neurons instead of classical neurons. This paper proposes a two stage classification scheme, and studies three algorithms: a classical algorithm, namely the Nearer Neighbor (N.N.R.), a generalization of it (the Fuzzy-N.N.R.), and a fuzzy classifier, namely the Fuzzy C-Means (F.C.M.). Results show that the fuzzy N.N.R. exhibits significantly better performance expressed by mean classification error, and operational convergence time. However, it is shown that the pure fuzzy schemes are more reliable than the fuzzified classical schemes. The structure of the proposed scheme is simple. There is no need to wait for long convergence delays and apply complex learning procedures since the scheme is of the S.O.M.-type. It is especially designed for optimal and quick convergence to the vectors of clusters.

A New Decision Criterion For Feature Selection Application To The Classification Of Non Destructive Testing Signatures

Authors:

L. Oukhellou, P.e Aknin, *INRETS (FRANCE)*

H. Stoppiglia, G. Dreyfus, *ESPCI (FRANCE)*

ABSTRACT

This paper describes a new decision criterion for feature selection (or descriptor selection) and its application to a classification problem. The choice of representation space is essential in the framework of pattern recognition problems, especially when data is sparse, in which case the well-known *curse of dimensionality* appears inevitably [1]. Our method associates a ranking procedure based on Orthogonal Forward Regression with a new stopping criterion based on the addition of a random descriptor. It is applied to a non destructive rail diagnosis problem that has to assign each measured rail defect to one class among several ones.

A Quantitative Matching Technique Based On Eigenvector Approach

Authors:
S. H. Park, K. M. Lee, S.U. Lee, *Seoul National University (KOREA)*

ABSTRACT

In this paper, we propose a new eigenvector-based linear feature matching algorithm, which is invariant to the rotation, translation and scale. First, in order to reduce the number of possible matches, we use a preliminary correspondence test that generates a set of finite candidate models. Secondly, we employ the modal analysis, in which Gaussian weighted proximity matrices are constructed to record the relative distance and angle information between linear features. Then, the modes of the proximity matrices of the two models are compared to yield the dissimilarity measure. Experimental results on synthetic and real images show that the proposed algorithm performs matching of the linear features fast and efficiently and provides the degree of dissimilarity in a quantitative way.

Direct Evaluation Of Frame-Based Nonstationary Pattern Recognition Methods By Using Bhattacharyya Distance

Authors:

M. Markovi, *Institute of Applied Mathematics and Electronics Kneza Milo (YOGOSLAVIA)*
M. Milosavljevi, *Institute of Applied Mathematics and Electronics - University of Belgrade (YOGOSLAVIA)*
B. Kovaevi, *University of Belgrade (YOGOSLAVIA)*

ABSTRACT

In this paper, a possibility of evaluating frame-based nonstationary pattern recognition methods by using Bhattacharrya distance is considered. Speech signal is used as a nonstationary signal and the comparative analysis is done through analyzing the natural speech, isolately spoken serbian vowels and digits.

Handwritten Farsi Character Recognition Using Evolutionary Fuzzy Clustering

Authors:

M. Dehghan, K. Faez, *AmirKabir University of Technology (IRAN)*

ABSTRACT

In this paper, a fuzzy clustering method based on Fuzzy c-Means clustering (FCM) and Evolutionary Strategies (ES) is proposed for handwritten Farsi character recognition. Experimental result showed that not only this algorithm can represent accurately the ambiguity of handwritten characters but also it outperforms the classical crisp based methods especially when word recognition is the main concern.

Hidden Markov Model With Nonstationary States

Authors:
M. A. Mahjou., *N. Ellouze, LSTS-ENTT (TUNIS)*

ABSTRACT

In this paper we explore the nonstationarity of Markov model and we propose a nonstationary Hidden Markov Model (NSHMM) which is defined with a set of dynamic transitions probability parameters A(t)={aij (t)} that depend on the time t already spent in state i. When compared to traditional models, this NSHMM is defined as generalization of the DHMM. The model was applied to on line recognition of handwritten arabic characters. The characters are represented by a radial sequence which is independent of translation, orientation and homothetie. The complete symbol-generation procedure includes sampling, size normalization and quantization phases. In the training, the model parameters are estimated with Baum-welch algorithm from a set of characters. Experiments have been conducted, and a good classification score has been obtained. The discrimination between characters models has been improved. Experiments showed that this approach can better capture the dynamic nature of arabic script.

Experimental Car Plate Recognition System By Neural Networks And Image Processing

Authors:
R.Parisi, E.D.Di Claudio, G.Lucarelli, G.Orlandi, *University of Rome "La Sapienza" (ITALY)*

ABSTRACT

In this paper a system for Italian-style car license plate recognition is described, based on the use of general purpose feedforward neural networks and basic image processing algorithms. The neural network replaces the entire pattern matching module encountered in similar systems. The algorithm actually does not make use of large databases, and experiments were conducted by training the neural classifier on a purely artificial training set. Nevertheless, final results are similar to those of more complicated approaches, and could be further improved by on line learning. The use of neural networks, coupled with the fast BRLS training algorithm, can substantially enhance the global recognition speed.

Parameter Estimation Of Conics: Application To Handwritten Digits

Authors:

M. Amara, P. Courtellemont, D. de Brucq, *Université de Rouen (FRANCE)*

ABSTRACT

We expose a method for modeling handwriting thanks to conic sections described by Cartesian equations under an implicit form. The parameter estimation is processed by an extended Kalman filter, taking as minimization criterion, the squared orthogonal distance between a point and the conic. The state equation is here constant, and the observation is a system of two equations: the first one characterizes the minimization of the criterion, and the second one is a normalization constraint of the parameters. The method provides a robust and invariant estimation of parameters, and an unique solution allowing the classification of modeled patterns. We apply this method to the coding of handwritten digits. A geometrical criterion allows to locate model changes. For a large interval of the used thresholds, we observe a great stability of the estimated parameters and of the instants of model changes. The method is evaluated in terms of accuracy, but equally by the data reduction rate, compared to other modeling techniques.

Multisensor Object Identification On Airports Using Autoassociative Neural Networks

Authors:

K. Haese, *German Aerospace Center(DLR) (GERMANY)*

ABSTRACT

This paper presents a neural identification method of traffic participants at airports. The traffic objects are observed by several dissimilar sensors. Their information is fused to possible object feature sets. These feature sets are processed by a neural network based identifier, so that traffic participants are identified on the basis of very few observations. The neural network identifier is composed of several RBF networks including networks with autoassociative network architecture.

Neural Network-Based Event Detection For Surveillance Applications

Authors:
M. Trompf, U. Knoblich, B. Boudisca, *Alcatel Alsthom Corporate Research Center (GERMANY)*

ABSTRACT

In our current work we investigate neural network-based event detection for surveillance tasks. After signal segmentation and feature extraction, we take a first segment-based decision between the three classes silence, activity, and alarm. Higher-level decisions are taken subsequently from a temporal combination of multiple segments. This technique allows for the detection of predefined complex intrusion scenarioes. Based on the different types of sensors of a surveillance system, data fusion techniques are used for joint processing of multisensor information to reach optimal results in terms of detection and false alarm rates. In addition to the signal classificator, we desbribe the architecture of our neural network-based event detector for a fence security system and give evaluation results from field tests.

Noise Elimination In Approximation And Time Series Prediction With Hinging Hyperplanes

Authors:
S. R. Baldomir, D. Docampo, *Universidade de Vigo (SPAIN)*

ABSTRACT

This paper presents a method to approximate multi-dimensional functions using the Sweeping Hinge Algorithm (SHA) in combination with the Truncated Hinging Hyperplanes (THH) as approximation units. The paper focuses on the real learning problems where some noise is present in the available examples. We show how the method provides good approximations, due to the simplicity of the selected units and to the fact that the convergence properties of the SHA algorithm are not influenced by the noise present in the data. Fair simulations contribute to support the good behavior of this purely constructive method against the noise.

Multiscale Versus Multiresolution Analysis For Multisensor Image Fusion

Authors:
Y. Chibani, A. Houacine, *Institute of Electronics - USTHB (ALGERIA)*

ABSTRACT

In this paper, we propose a multisensor image fusion method based on the multiscale decomposition. In this representation, the redundant wavelet transform is performed on input images to emphasise the dominant details present at each scale. A fusion rule is applied between wavelet coefficients in order to produce a fused image. Two different fusion rules are used in the wavelet domain. A quantitative evaluation shows the superiority of our approach over the existing multiresolution method.

Randomized Regression In Differentiators

Authors:
S. Siren, P. Kuosmanen, *Tampere University of Technology (FINLAND)*

ABSTRACT

Differentiation of a signal is required in many applications in the field of signal processing. Linear differentiators fail to give good results for signals corrupted by both Gaussian and impulsive type of noise. For such cases nonlinear methods can be used in order to obtain better results. In this paper we propose a method, which we call randomized regression differentiator, based on random sampling and giving very good results in the presence of Gaussian and impulsive types of noise. We also present a modification of this general method which is designed for piecewise linear signals and utilizes random samplings of one window position in the next one. The output of this differentiator has sharp transitions when the slope value changes and the constant slope areas are smooth having only few small deviations around the correct slope value.

A Study On Discrete Wavelet Transform Implementation For A High Level Synthesis Tool

Authors:

J.-M. Tourreilles, C. Nouet, E. Martin, *LESTER - University of South Brittany (FRANCE)*

ABSTRACT

Actually, the error from the fixed-point implementation is not taken in account in the high level synthesis tools. The processing signal algorithms are implemented without quality evaluation. This paper presents a fixed-point implementation methodology applied to the Discrete Wavelet Transform.

A 500 Mhz 2d-DWT VLSI Processor

Authors:

A. Brizio, G. Masera, G. Piccinini, M. Ruo Roch, M. Zamboni, *Politecnico di Torino (ITALY)*

ABSTRACT

The Discrete Wavelet Transform (DWT) is a well-known mathematical tool, which has been proved to offer very good results for Digital Signal Processing applications. The DWT requires a massive computation, especially when real-time constraints are to be met, as it happens in a wide spectrum of applications; therefore dedicated processors are needed to achieve the necessary performance. In this paper a very high-speed VLSI implementation of a two-dimensional DWT processor suitable for image processing is presented. The architecture is based on a bit-serial arithmetic; linear systolic arrays are employed for row filtering and parallel arrays for column filtering. Each basic building block of the arrays has been designed using a True Single Phase Clocking logic, then a manual place and route of the derived cells has been carried out. The processor works at 500 MHz and occupies less than 50 mm^2 in a 0.7 ìm CMOS technology.

Improving "Performance Vs. Silicon Size" Tradeoffs Using Coprocessors A Case Study: G.721 On OAK And Pine DSP Cores

Authors:

A. Pegatoquet, *University of Nice - VLSI Technology (FRANCE)*
M. Auguin, *University of Nice (FRANCE)*
E. Gresset, *VLSI Technology (FRANCE)*

ABSTRACT

The increase in performance and internal memory of current DSP cores allows most applications to have software only implementations. However, recent applications in telecommunications (e.g. MPEG-4) require more and more computational power. A solution to this problem is to implement a subset of the algorithm in hardware. In this paper we present a software only and a mixed hardware/software solution for the G.721 [1] Adaptative Differential Pulse Code Modulation (AD-PCM) on VLSI Technology VVF3000 and VVF3500 DSP cores which stand for DSP Group PineDSPCore TM and OakDSPCore TM [2]. We will show the benefits of a coprocessor based solution and describe our methodology to extract hardware blocks from a software implementation.

Single Chip DSP Array Processor: 100 Million + Transistors With Multithreading Approach

Authors:
R. Sernec, *BIA Ltd. (SLOVENIA)*
M. Zajc, J. F. Tasiè, *Fakulteta za elektrotehniko (SLOVENIA)*

ABSTRACT

We propose an efficient programmable parallel architecture for DSP and matrix algebra applications that can exploit parallelism at algorithm (topology) level via systolic/SIMD array processing and at instruction level via multiple-issue control processor capable of multithreading. Our premise is: *”One array – one chip“* and integration of systolic/SIMD processing on the same processor array with required data storage. Multithreading on systolic/SIMD arrays is analysed on examples, which show that substantial speedups are possible (100% - 800%) when up to four threads are interleaved on cycle-by-cycle basis. We are targeting processor element (PE) granularities in word range and employ support for floating-point operations. Furthermore the array integrates data memory at two levels of hierarchy: local per PE (SIMD) and global for the whole processing array (systolic). Complexity of such a system is explored in detail and is shown that 32 PE array can be implemented on a 120 million transistor chip.

Asynchronous Timing Model For High-Level Synthesis Of DSP Applications

Authors:

O. Dedou, D. Chillet, O. Sentieys, *LASTI - ENSSAT - Universite de Rennes I (FRANCE)*

ABSTRACT

In an asynchronous system, initiation and completion of operations are events that can occur at any instant and the operations have delays which are data dependent. Thus if an asynchronous timing model is considered, we can provide scheduling, resource-allocation strategy. Since one of the principal feature of the asynchronous systems is to exhibit average computation time, it will be interesting to use it as a timing model. In this paper we present a statistical approach to derive the average computation time of asynchronous components, first step toward High Level Synthesis. This method allows to reduce the critical path which in the case of a real-time application will be an excellent issue to reduce the number of operators.

DSP Architecture
For Real Time Digital Video Converter

Authors:
L. Franchina, *La Sapienza University (ITALY)*

ABSTRACT

In this paper we propose the architecture of a digital video standard converter based on the use of DSP processors (AD 21060 SHARC) completely programmable and able to perform real time elaboration. The purpose is to verify experimentally the possibility to implement very complex moto compensation algorithm (for example in real time digital television standard converter), on a multiprocessing DSP system. In this way is possible to make very versatile digital elaborators (because programmable) and at low cost. This system find out applications in:

- Digital Video Broadcasting (DVB) [1-2, 4-6]
- Artificial intelligence
- Robotics and military applications in motion detection system.

A PRML Equalizer For Hard Disk Drives With Low Sensitivity To Sampling Phase Variation

Authors:
A. Gerosa, G.A. Mian, *University of Padova (ITALY)*

ABSTRACT

This work presents a new architecture for the realization of Hard Disk Drive Read Channels, using EPR-IV equalization. The pre-equalization is realized as a sampled-data system in order to have precise pulse shaping and simply tunable bandwidth. The equalization is then completed by a fractionally spaced equalizer, which ensures low sensitivity to sampling phase variation. The system complexity is reduced with respect to typical solutions, while the performances are within the usual specs in every operating condition, as confirmed by careful simulations. The architecture was designed to be integrated in a standard CMOS IC.

A Recursive Algorithm For The Generation Of Space-Filling Curves

Authors:
G. Breinholt, C. Schierz, *Swiss Federal Institute of Technology Zürich (ETHZ) (SWITZERLAND)*

ABSTRACT

Space-filling curves have intrigued both artists and mathematicians for a long time. They bridge the gap between aesthetic forms and mathemtical geometry. To enable construction by computer, an efficient recursive algorithm for the generation of space-filling curves is given. The algorithm is elegant, short and considerably easier to implement than previous recursive and non-recursive algorithms, and can be efficiently coded in all programming languages that have integer operations. The algorithmic technique is shown applied to the generation of the Hilbert and a form of the meandering Peano curve. This coding technique could be successfully applied to the generation of other regular space-filling curves.

Automated Parallel-Pipeline Structure Of FFT Hardware Design For Real-Time Multidimensional Signal Processing

Authors:
A. Petrovsky, *University of Technology Bialystok (POLAND)*
M. Kachinsky, *Belarusian State University of Informatics and Radioelectronics(BELARUS)*

ABSTRACT

Offered algorithm allows to build the structure of the parallel pipeline FFT processors (PPFFT-processor) computing the vector DFT in real time with the minimum structural complexity at given parameters: speed of input data receipt; structure of a computing element (arithmetic device) and time of the butterfly operation execution. The considered approach to structural synthesis of the PPFFT-processors for R-dimensional signal processing allows to receive the structure of the processor under given restrictions of a specific problem and is the basis for resolving the questions of automated design of PPFFT-processors at a structural level.

Parallel Implementation Of A Face Location Algorithm Based On The Hough Transformation

Authors:
F.Yang, E.Drege, M.Paindavoine, H.Abdi, *Université de Bourgogne, (FRANCE)*

ABSTRACT

In order to localize the face in an image, our approach consists to approximate the face oval shape with an ellipse and to compute coordinates of the center of the ellipse. For this purpose, we explore a new version of the Hough transformation : the Fuzzy Generalized Hough transformation. To reduce the computation time, we present also a parallel implementation of the algorithm on 2 Digital Signal Processors and we show that an acceleration of factor 1.6 has been obtained.

GSLC Architecture For Sequence Detectors Using Spatial Diversity

Authors:
M. A. Lagunas, A. I. Perez-Neira, J. Vidal, *Campus Nord UPC (SPAIN)*

ABSTRACT

The role of advanced front-ends including spatial diversity, has been considered as an independent part of peak-distortion equalizers, Wiener and Viterbi equalizers. This involves that the optimum processing to remove point or distributed sources, together with inner and outer intersymbol interference is analyzed independently at the beamformer and at the equalizer stages. Recently, based on extensions of the works performed with forward equalizers and optimal combining in communications systems with spatial diversity, several solutions to the joint design of sequence detectors and spatial combiners have been reported. All these solutions have in common the principle that the optimum design holds the constraint of matching the spatial response of the combiner to the DIR (Desired Impulse Response) of the sequence detector. This work enhances the matched DIR concept with the Generalized Sidelobe Canceller architecture; proving that, for stationary Intersymbol Interference (ISI) for the desired user, the GLSC represents a suitable spatial processing. The GSLC allows continuous updating of the combiner either in order to reject late arrivals and co-channel interferers, without requiring the presence of training sequence or to maximize the effective SNR.

A Pipelined Architecture For DLMS Algorithm Considering Both Hardware Complexity And Output Latency

Authors:

T. Kimijima, K. Nishikawa, H. Kiya, *Tokyo Metropolitan University (JAPAN)*

ABSTRACT

In this paper we propose a new pipelined architecture for the DLMS algorithm which can be implemented with less than half an amount of calculation compared to the conventional architectures. Although the proposed architecture enables us to reduce the required calculation, it can achieve good convergence characteristics, a short latency and high throughput characteristics simultaneously.

A G-transform Based Systolic Array For Least Squares Problems

Authors:
E. Zervas, *TEI Athens, (GREECE)*
A. Alonistrioti, *N.C.S.R. Demokritos (GREECE)*
N. Passas, *University of Athens, (GREECE)*

ABSTRACT

A new type of systolic array for the solution of Least Squares problems, based on the G transform, is presented. The G-transformation matrices are permuted Hessenberg matrices and their use for the solution of LS problems based on a systolic array implementation offers some advantages as square-root free implementation and reduced number of multiplications.

VLSI Implementation And Complexity Comparison Of Residue Generators Modulo 3

Authors:

S. J. Piestrak, *Universite de Rennes 1 (FRANCE)- Wroclaw University of Technology (POLAND)*
F. Pedron, O. Sentieys, *Universite de Rennes 1 (FRANCE)*

ABSTRACT

A generator modulo 3 (mod 3) is a circuit that generates a residue mod 3 from a binary vector. It is an essential circuit used to construct the encoding and checking circuitry for arithmetic error detecting codes, such as residue codes mod 3 and the 3N code, as well as some residue number system hardware. In this paper, we compare speed and area of varius VLSI implementations of 16-input generators modulo 3. It is shown that the generator built of full-adders consumes the least area. On the other hand, the generator built as a tree of special 4-input modules is twice as fast, although at the cost of increasing the area by a factor of 1.7.

Adaptive Filters Implementation Performances Under Power Dissipation Constraint

Authors:
S. Gailhard, N. Julien, E. Martin, *LESTER-IUP (FRANCE)*

ABSTRACT

Power consumption is an essential criteria in embedded systems. Therefore, it is important to decrease it at every stage of the design flow. It is well known that the choice of a signal processing algorithm has a great impact on the power dissipation. For this purpose, a HLS (High Level Synthesis) CAD (Computer Aided Design) tool (gaut_w) has been developed. It allows to estimate the power dissipation of a dedicated VLSI circuit from the algorithmic description of the application. It also reduces the power dissipation during the architectural synthesis in order to target low power time constrained VLSI circuits. This tool has been applied to different adaptive filters as the NLMS, the BLMS and the GAL filters for a radio-communication application. The power dissipation on a low-power TMS320C50 DSP (Digital Signal Processing) has been also estimated.

Behavioral Synthesis Of Digital Filters Using Attribute Grammars

Authors:

G. Economakos, G. Papakonstantinou, P. Tsanakas, *National Technical University of Athens (GREECE)*

ABSTRACT

Recently, a formal framework to perform behavioral synthesis using attribute grammars has been presented, its main advantages being modularity and declarative notation in the development of design automation environments. From a practical point of view, most modern behavioral synthesizers are best suited for the development of special purpose hardware to implement digital signal processing algorithms. In this paper, the attribute grammar formalism and the corresponding framework are extended to handle the development of special purpose hardware for both FIR and IIR filters. The proposed methodology is capable to construct hardware implementations in various technologies (FPGA, CPLD, ASIC) from behavioral filter specifications (difference equations or discrete convolution), utilizing the VHDL standard hardware description language. Overall, a formal specification for fast and efficient filter implementation is proposed.

The Impact Of Data Characteristics On Hardware Selection For Low-Power DSP

Authors:
G. Keane, J. R. Spanier, R. Woods, *The Queen's University of Belfast (UK)*

ABSTRACT

Adders and multipliers are key operations in DSP systems. The power consumption of adders is well understood but there is little knowledge of the choice of multipliers available. This paper considers the power consumption of a number of multiplier structures such as array and Wallace Tree multipliers and examines how the power varies with data wordlengths and different data streams (e.g. image and speech). In all cases results were obtained from EPIC PowerMill™ simulations of synthesised circuit layouts, a process which is accepted to be within 5% of the actual silicon. Analysis of the results highlights the effects of routing and interconnect optimization for low power operation and gives clear indications on choice of multiplier structure and design flow for the rapid design of DSP systems.

Partial Differential Equations In Image Analysis: Continuous Modeling, Discrete Processing

Authors:
P. Maragos, *Georgia Institute of Technology (U.S.A.) - Institute for Language & Speech Processing (GREECE)*

ABSTRACT

This paper presents an overview of selected topics from an emerging new image analysis methodology that starts from continuous models provided by partial differential equations (PDEs) and proceeds with discrete processing of the image data via the numerical implementation of these PDEs on some discrete grid. We briefly discuss basic ideas, examples, algorithms, and applications for PDEs modeling nonlinear multiscale analysis, geometric evolution of curves and signals, nonlinear image/signal restoration via shock filtering, and the eikonal PDE of optics. Wherever possible, we compare the PDE approach with the corresponding all-discrete method. The PDE approach is very promising for solving (or improving previous all-discrete solutions of) many problems in image processing and computer vision because it provides new and more intuitive mathematical models, has connections with physics, gives better approximations to the Euclidean geometry of the problem, and is supported by efficient discrete numerical algorithms based on difference approximations.

Nonlinear Signal Processing For Adaptive Equalisation And Multi-User Detection

Authors:
B. Mulgrew, *The University of Edinburgh (UK)*

ABSTRACT

This paper examines the application of nonlinear signal processing techniques to the development of adaptive equalisers for frequency domain multiple access (FDMA) and multi-user detectors for code division multiple access (CDMA). Current issues are discussed and key problems identified.

Channel Equalization For Coded Signals In Hostile Environments

Authors:
K. Georgoulakis, S. Theodoridis, *University of Athens (GREECE)*

ABSTRACT

In this paper the detection of Trellis Coded Modulated signals corrupted by Intersymbol Interference, Co-Channel Interference and nonlinear impairments is treated as a classification task by means of a Clustering Based Sequence Equalizer-Decoder. The receiver performs jointly decoding and equalization of trellis encoded signals. No specific model is required for the channel or for the interference and the noise, and no code knowledge is needed at the receiver. Complexity reduction of the equalizer is obtained through two suboptimal techniques, a) Cluster's grouping and b) the M-Algorithm. The robust performance of the proposed scheme is illustrated by simulations.

A Combined LMS-SOM Algorithm For Time-Varying Non-Linear Channel Equalization

Authors:

S. Bouchired, M. Ibnkahla, W. Paquier, *National Polytechnics Institute of Toulouse (FRANCE)*

ABSTRACT

The paper proposes a self-organizing map (SOM) approach to equalize time-varying non-linear channels. The approach is applied to a satellite mobile communication channel which is composed of time invariant linear filters, a non-linear memoryless amplifier and a time-varying multi-path propagation channel. The SOM is combined to a Transversal Linear Equalizer (TLE) which is trained by the LMS algorithm. The paper studies the performance and convergence behavior of the SOM. The paper also illustrates the capability of the SOM to track the channel changes in particular for complicated modulation schemes.

Non-Linear Equalizers That Estimate Error Rates During Reception

Authors:
J. Cid-Sueiro, *Universidad de Valladolid (SPAIN)*
A. R. Figueiras-Vidal, *Universidad Carlos III (SPAIN)*

ABSTRACT

Neural Networks can be used to estimate the a posteriori probabilities of the transmitted symbols in digital communication systems. In this paper we apply this property to the on-line estimation of the bit error rate (BER) in the receiver, without using any reference signal. We discuss two different approaches to BER estimation: (1) computing the a posteriori symbol probabilities from estimates of the conditional distributions of the received data, and (2) estimating probabilities by gradient minimization of a special type of cost functions. We show that Importance Sampling (IS) techniques can be combined with the first approach to reduce drastically the variance of the probability estimates. Finally, we analyze the effect of channel variations during transmission.

On The Nonlinearity Of Linear Prediction

Authors:
G. Kubin, *Vienna University of Technology (AUSTRIA)*

ABSTRACT

This paper analyzes adaptive linear prediction and the effects of the underlying optimality criterion on the prediction error. It is well known that the signal-dependent optimization process converts the linear filter into a nonlinear signal processing device and that this will in uence the statistics of the filter output in a way not expected from linear filter theory. For minimum-phase L_p-optimal linear predictors, we can show that the prediction error is maximally close to an i.i.d. process whose probability density function is given by Aexp(-L|X|^p). This result is applied to linear predic- tive analysis-by-synthesis coding of speech and to predictive decision-feedback equalization of channels with nongaussian noise. Implications for testing time series for linearity or gaussianity are discussed, too.

A Likelihood Framework For Nonlinear Signal Processing With Finite Normal Mixtures

Authors:
T. Adali, B. Wang, X. Liu, J. Xuan, *University of Maryland (USA)*

ABSTRACT

We introduce a likelihood framework for nonlinear signal processing using partial likelihood and use the result to derive the information geometric em algorithm for distribution learning through information-theoretic projections. We demonstrate the superior convergence of the em algorithm as compared to least relative entropy (LRE) algorithm by simulations. The performance of finite normal mixtures (FNM) based equalizers with different number of mixtures and different dimension observation vectors is also discussed.

Analysis Of A Nonlinear Equalizer Based On High-Order Statistics

Authors:

J.B.D. Filho, G. Favier, *CNRS/UNSA (FRANCE)*

J. M. T. Romano, *University of Campinas (BRAZIL)*

ABSTRACT

In this paper, we analyse the equalizer structure when the transmitted signal is iid. Conditions for recovering an iid signal are established, which points out limitations of the classical linear equalizer and suggests the application of nonlinear structures. A simple nonlinear equalizer, previously proposed in [1], is then compared to the classical CMA equalizer.

Bit Error Rate Optimization Of Ds-Cdma Receivers

Authors:

I. N. Psaromiligkos, S. N. Batalama, *State University of New York at Buffalo (USA)*

ABSTRACT

In search of acceptable cost versus performance trade-off points for DS-CDMA receivers linear tap-weight filters are considered. Based on stochastic approximation concepts a recursive algorithm is developed for the adaptive optimization of linear filters in the minimum Bit Error Rate (BER) sense. The recursive form is decision driven and distribution free. For AWGN channels, theoretical analysis of the BER surface of linear filter receivers identifies the subset of the linear filter space where the optimal receiver lies and offers a formal proof of guaranteed global optimization with probability one for the 2-user case. To the extent that the output of a linear DS-CDMA filter can be approximated by a Gaussian random variable, a minimum-mean-square-error optimized linear filter approximates the minimum BER solution. Numerical and simulation results indicate that for realistic AWGN DS-CDMA systems with reasonably low signature cross-correlations the linear minimum BER filter and the MMSE filter exhibit approximately the same performance. The linear minimum BER receiver is superior, however, when either the signature cross-correlation is high or the channel noise is non-Gaussian.

Constrained Pulse Shape Synthesis For Digital Communications

Authors:
P. L. Combettes, *City University of New York (USA)*
P. Bondon, *CNRS - Laboratoire des Signaux et Systemes (FRANCE)*

ABSTRACT

The synthesis of optimal pulse shapes for digital data transmission over communication channels typically involves conflicting specifications. We consider the problem of finding a pulse satisfying some mandatory "hard" constraints while violating as little as possible the remaining "soft" constraints. The problem is formalized as that of minimizing a weighted sum of squared distances to the soft constraint sets over the intersection of the sets associated with the hard constraints. This constrained problem is analyzed and a numerical algorithm is proposed. Simulation results are presented.

Sequential Local Transform Algorithms For Gray-Level Distance Transforms

Authors:

P. J. Toivanen, H. Elmongui, *Lappeenranta University of Technology (FINLAND)*

ABSTRACT

In this paper, new algorithms are presented for the calculation of gray-level distance transforms, in which the distance values of pixels are proportional to the gray-value differences of minimal paths [5], not the gray values themselves, which is the case in [1] and [3]. The presented algorithms are sequential local transform algorithms. The performance of the algorithms are evaluated by testing how quickly they converge to the error-free distance image. It is shown that the presented algorithms are faster than the previously presented algorithms [5] for gray-level distance transforms. It turns out that the 4-neighbor, 4-raster algorithm is the fastest, converging to a distance image with no erroneous pixels in only two iteration rounds. Furthermore, because of the raster scanning approach they are easily applied other image grids than the rectangular one, and are easy to implement.

Hierarchical Skeleton Extraction Based On A Deformable Particle System

Authors:
F. Angella, P. Baylou, *ENSERB and PRC-GDR ISIS, CNRS (FRANCE)*
O. Lavialle, *ENITA (FRANCE), ENSERB and PRC-GDR ISIS, CNRS (FRANCE)*

ABSTRACT

The use of a particle system as a skeleton extractor is introduced here. This system behaves as an active contour model embedding variable topology properties. As the particles propagate inside tree-shaped objects, we build their skeleton which is useful to determine the neighborhoods used for the computation of the regularization and interaction forces. This connected skeleton is directly returned and avoids a two-step approach based on a morphological operation followed by a tree analysis. In addition, the method gives information on the hierarchy of structures. Using this method, it is possible to generate a cartography of structures such as veins or channels.

Shape Representation For Object Correspondence Based On Sub-Graph Matching And Fourier Descriptors

Authors:
F. Marques, G. Guitierrez, *Universitat Politecnica de Catalunya (SPAIN)*

ABSTRACT

This paper studies a new shape representation for object correspondence. It relies on the division of complex objects into elementary regions and their description in terms of a graph. Each node in the graph is associated to a region and contains its shape information. This shape information is represented using Fourier descriptors. Links in the graph are weighted. Link values are computed using the shape information common to the associated regions. Based on this representation, an object matching procedure is proposed. It relies on sub-graph matching and follows a bottom-up approach. Finally, to decouple the assessment of the shape representation from the problem of image segmentation, the technique is applied to cartoon contents.

Rapid Location Of Convex Objects In Digital Images

Authors:
E. R. Davies, *University of London (UK)*

ABSTRACT

This paper studies sampling strategies for the rapid location of objects in digital images, and shows how point sampling can be used to minimise computational effort. The process can be extremely efficient, especially when the image space is sparsely populated and large convex objects are being detected. In the case of ellipses, exact location is considerably aided by the new 'triple bisection' algorithm. The approach has been applied successfully to the location of well separated nearly elliptical cereal grains which are to be scrutinised for damage and varietal purity.

Boundary Tracking In 3D Binary Images To Produce Rhombic Faces For A Dodecahedral Model

Authors:
E. Garduno, G. T. Herman, H. Katz, *University of Pennsylvania (USA)*

ABSTRACT

An algorithm is presented for tracking boundaries in three-dimensional (3D) binary images based on rhombic dodecahedral voxels. The algorithm produces a list of all the rhombic voxel faces in such a boundary.

Efficient Curvature-Based Shape Representation For Similarity Retrieval

Authors:
F. Mokhtarian, S. Abbasi, J. Kittler, *University of Surrey England (UK)*

ABSTRACT

The Curvature Scale Space (CSS) image is a multi-scale organisation of the in ection points of a closed planar curve as it is smoothed. It consists of several arch shape contours, each related to a concavity or a convexity of the curve. In our recent work, we have used the maxima of these contours to represent the boundary of objects in shape similarity retrieval. In our new approach, each segment of a shape is represented by the relevant maximum of the CSS image as well as the average curvature on the segment at certain level of scale. In this paper we explain how this new representation together with its matching algorithm improve the performance of our shape similarity retrieval system. To evaluate the proposed method, we created a small classified image database. We then measured the performance of the system on this database. The quantified results of this test provided supporting evidence for the performance superiority of the proposed method.

Model Acquisition And Matching In Tagged Object Recognition (TOR)

Authors:
L. M. Soh, J. Matas, J. Kittler, *University of Surrey, (UK)*

ABSTRACT

General object recognition is difficult. We had proposed a novel solution for object recognition in an unconstrained environment using a tag (Matas[2]). This simplifies the problem by placing tags of special pattern on the objects that allows us to determine the pose easily. A robust calibration chart detector was developed for the first stage of the solution (Soh[1]). This paper investigates the next part of the solution, i.e. the model acquisition and matching using the Chamfer matching algorithm. The algorithm is reasonably simple to implement and very efficient in terms of computation. We experiment with this technique extensively to prove the reliability of the approach. Using this approach, the objective of Tagged Object Recognition (TOR) can be realised and we should be able to perform object recognition wherever a tag is located. The technology will facilitate applications in landmark and object recognition, mobile robot navigation and scene modelling.

Form Identifiction And Skew Detection From Projections

Authors:
N. Liolios, N. Fakotakis, G. Kokkinakis, *University of Patras, (GREECE)*

ABSTRACT

In this paper we describe a system we have built to solve the preprinted forms identification and field extraction problem for Optical Character Recognition (OCR) applications. The strength of this system is that unlike other approaches it solves the problem in the most general and unrestricted sense. It works equally well for any type of preprinted form because it does not rely on any special features like patterns of line crossings or other symbols found only in a particular type of form. We have used the power spectrum as a shift invariant feature vector of the form's horizontal projection from which we identify the type of form and detect rotation. The horizontal and vertical projections themselves are also used to detect the shift of the form. Unlike the expected loss in response time to the benefit of generality, the proposed system is fast, highly accurate, even at reduced resolutions and with minimal user intervention it can be trained to recognize new types of forms.

The Use Of Steerable Filters For Hand-Written Character Recognition

Authors:
E. Tufan, *University of Istanbul (TURKEY)*
V. Tavsanoglu, *South Bank University (UK)*

ABSTRACT

We develop a system for image recognition using steerable filters and neural networks where the main role of steerable filters is in the pre-processing of the image. This process enables the characterisation of the image by the local dominant orientation rather than global image. The Kohonen's self organising feature map is used for the recognition task.

Stereo Vision Via Connected-Set Operators

Authors:
R. Harvey, K. Moravec, J. A. Bangham, *University of East Anglia (UK)*

ABSTRACT

The use of connected operator morphology, in particular operators that decompose by area, are examined for the problem of determining dense depth maps from stereo images. It is shown that the use of connected operators can augment and improve conventional processing algorithms. This paper describes how the new algorithm works and presents a comparison of its performance.

Optimum Oversampling In The Rectangular Gabor Scheme

Authors:
M. J. Bastiaans, *Technische Universiteit Eindhoven (NETHERLANDS)*

ABSTRACT

The windowed Fourier transform of a time signal is considered, as well as a way to reconstruct the signal from a sufficiently densely sampled version of its windowed Fourier transform using a Gabor representation; following Gabor, sampling occurs on a two-dimensional time-frequency lattice with equidistant time intervals and equidistant frequency intervals. For sufficiently dense sampling, the synthesis window (which appears in Gabor's reconstruction formula) may be constructed such that it resembles a rather arbitrarily given function; this function may or may not be proportional to the analysis window (which is used in the windowed Fourier transform). It is shown that the resemblance can already be reached for a rather small degree of oversampling, if the sampling distances in the time and frequency directions are properly chosen. A procedure is presented with which the optimum ratio of the sampling intervals can be determined.

Half-Quadratic Regularization Of Time-Frequency AR Analysis For Recovery Of Abrupt Spectral Discontinuities & Their Detection By A Recursive Siegel Metric Based On Information Geometry

Authors:

F. Barbaresco, *THOMSON-CSF AIRSYS (FRANCE)*

ABSTRACT

We have proposed a Thikonov approach of Burg algorithm regularization based on a local quadratic potential function which yields a smoothness constraint with a long AR model. We propose to extend regularization approach in time-frequency domain with a temporal smoothness constraint. First, we use an half-quadratic regularization that avoids classical oversmoothing effect of quadratic potential function and preserves abrupt spectral changes and discontinuities. Secondly, to improve time-frequency dissimilarity detection, we propose a new recursive Siegel metric for AR models based on information differential geometry introduced by Rao through Fisher information matrix and Shannon entropy functional.

Representation Of Pseudoperiodic Signals By Means Of Pitch-Synchronous Frequency Warped Wavelet Transform

Authors:
S. Cavaliere, G. Evangelista, *University "Federico II," Napoli (ITALY)*

ABSTRACT

In this paper we discuss a novel representation of pseudoperiodic signals based on a frequency warped pitch-synchronous transform. The unitary warping operation is performed by means of the discrete Laguerre transform and controlled by the Laguerre parameter. This parameter can be adapted to the characteristics of the signal. In particular, for a large class of signals, by means of frequency warping one can regularize the spacing of the partials, so that the resulting signal is pseudo-harmonic. By applying the Pitch-Synchronous Wavelet Transform to the regularized signal one can achieve an interesting separation of noise and transients from the resonant components. These concepts are integrated in a single unitary transform. The Pitch-Synchronous Frequency Warped Wavelet Transform has applications in sound analysis, coding and synthesis.

Algorithm For The Instantaneous Frequency Estimation Using Time-Frequency Distributions With Adaptive Window Width

Authors:
L. Stankovic, *Ruhr University Bochum (GERMANY)*
V. Katkovnik, *University of South Africa (SOUTH AFRIKA)*

ABSTRACT

A method for the minimization of mean square error of the instantaneous frequency estimation using time-frequency distributions, in the case of a discrete optimization parameter, is presented. It does not require knowledge of the estimation bias. The method is illustrated on the adaptive window length determination in the Wigner distribution.

Tracking Of Spectral Lines
In An ARCAP Time-Frequency Representation

Authors:
M. Davy, C. Doncarli, *IRCyN UMR 6597 (FRANCE)*
B. Leprettre, N. Martin, *LIS BP 46 (FRANCE)*

ABSTRACT

ARCAP time-frequency representations of narrow-band signals are made of instantaneous characteristics (frequencies and amplitudes), without any time links. In order to extract the frequency modulations (or sprectral trajectories), we propose to re-create them on the basis of ballistic integrator models. The analytic expression of the corresponding asymptotic Kalman filter gains allows a very simple implementation of association procedures including trajectory birth or death. The points being associated, a Fraser filtering leads to the smoothed spectral trajectories.

Detection And Tracking Of Multi-Periodic Signals

Authors:
I J Clarke, G Spence, *DERA (UK)*

ABSTRACT

Periodic signal analysis is an important tool in signal processing, there are many phenomena that exhibit a periodic nature. A number of analysis techniques aimed at estimating the periodicities from sensor data already exist but most use stationary harmonically related Fourier components as the basis. The performance can be seriously degraded when there are multiple signals present and/or a period is time-varying. In this paper a novel time-domain tracking method is pro-posed. This is based on a modified Incremental Multi-Parameter (IMP) algorithm [1] that is able to detect and track several periodic components in a single time series. The method exploits pseudo-integration, a novel method aimed at reducing tracking lag. Two forms of the algorithm are discussed: a) block mode, using an iterative approach on a batch of sampled data and b) recursive mode for updating parameters in a real-time practical situation.

Sound Signature Analysis Using Time-Frequency Signal Processing: Application To Active Stall Avoidance In Axial Compressors

Authors:
T. Le, T. Dombek, M. Glesner, *Darmstadt University of Technology (GERMANY)*

ABSTRACT

We examine the promising approach of using Time-Frequency Analysis based on Cohen's Class to characterize the unstable operation (stall) of an axial compressor. Stall precursors are time-localized transients which indicate a coming stall inception. The analysis uses the acoustic vibration signal to reveal the non-stationary behavior when approaching the stall region. The results in this paper show that it is possible to visualize stall precursors by relying on the microphone signal which represents a low cost setup compared to the use of dynamic pressure probe array. The design goal of an active stall avoidance system in the future requires us to consider already at the algorithm design an effective computing scheme. We adopt the approach of representing the discrete Time-Frequency Distribution (TFD) as a sum of spectograms which is advantageous for real-time implementation issues of the highly complex algorithm. Approximation is possible by taking only significant terms of the decomposition into account. The result of this work serves as a preprocessing step for the later design of a detection/classification system.

Extended Wavelet Transforms in Acoustic Diagnosis

Authors:
G. Wirth, D. A. Mlynski, *Universitat Karlsruhe (GERMANY)*

ABSTRACT

We propose an efficient strategy for qualitative acoustic diagnosis of electric motors based on fast wavelet transforms and fuzzy set theory. First an extended wavelet transform is applied to the acoustic data recorded by a microphone. Thus a decomposition of the frequencies in the data sets is obtained and only a few typical features (wavelet coefficients) remain for the identification process. Then a fuzzy c-means algorithm separates data into various clusters thus giving a qualitative interpretation of the present fault due to the calculated membership values.

Quantization Effects In Implementation Of Distributions From The Cohen Class

Authors:

V. Ivanovic, L. Stankovic, Z. Uskokovic, *University of Montenegro (YUGOSLAVIA)*

ABSTRACT

The paper presents an analysis of the finite register length in uence on the accuracy of results obtained by the time-frequency distributions (TFDs). In order to measure quality of the obtained results, the variance of the proposed model is found, signal-to-quantization noise ratio (SNR) is defined and appropriate expressions are derived. Floating- and fixed-point arithmetic are considered. It is shown that commonly used reduced interference distributions (RID) exhibit similar performance with respect to the SNR. We have also derived the expressions establishing relationship between the number of bits and required quality of representation (defined by the SNR), which may be used for register length design in hardware implementation of TFDs.

Cross-Terms Free Forms Of Some Quadratic And Higher Order Time-Frequency Representations

Authors:
L. Stankovic, S. Stankovic, *University of Montenegro (YUGOSLAVIA)*

ABSTRACT

The S-method based time-frequency analysis is presented. In the case of multi-component signals, it can produce a sum of the pseudo Wigner distributions of each signal component separately. The only condition is that the spectrogram is cross-terms free. The realization is based on the short-time Fourier transform, for which well studied software methods and hardware systems exist, what makes this method attractive for applications and implementations in time-frequency analysis, including higher-order distributions.

A Nonlinear Dynamical Model For Compression And Detection Of ECG Data

Authors:
T. Schimming, H. Dedieu, *Circuits and Systems Group (SWITZERLAND)*
M. Ogorzalek, *University of Mining and Metallurgy (POLAND)*

ABSTRACT

We propose a low-dimensional nonlinear model explaining the ECG dynamics, suitable for data compression and possibly feature detection. Tests on real clinically measured ECG signals confirmed very good performance of the model in terms of modeling errors and compression ratio.

Analysis Of
Brain Electroencephalograms With The
Instantaneous Maximum Entropy Method

Authors:
Y. Takizawa, M. Ishiguro, *Institute of Statistical Mathematics*
S. Uchida, *Institute of Psychiatry, Tokyo (JAPAN)*
A. Fukasawa, *Chiba University*

ABSTRACT

Analysis of electroencephalogram (EEG) is presented for sleep stages featured by sleep onset, light NREM (Non Rapid Eye Movement), deep NREM, and REM. Appearance and continuation of featuristic waves are not steady in EEG. The Instantaneous Maximum Entropy Method (IMEM) is proposed to analyze nonstationary waveform signals of EEG. The characteristics of these waves responding to epoch of sleep are analyzed. The results of analysis are given as follows; (a) time dependent frequency of continuous oscillations of alpha rhythm was observed precisely, (b) sleep spindles were detected clearly within NREM and these parameters of time, frequency, and peak energy were specified, (c) delta waves with very low frequencies and sleep spindles were observed simultaneously, and (d) the relationship of sleep spindles and delta waves was first detected with negative correlation along time-axis. The analysis by the IMEM was found effective comparing conventional analysis method of FFT, bandpass filter bank.

Fast Optimal Beam Determination For Conformation Radiotherapy Treatment Planning

Authors:

Y. Yuan, *Dicomit Imaging Systems Corp (CANADA)*

W. Sandham, T. Durrani, *University of Strathclyde (SCOTLAND)*

C. Deehan, *Western Infirmary (SCOTLAND)*

ABSTRACT

Conformation radiotherapy, which involves intensity modulation of photon treatment beams, offers considerable advantages compared to conventional radiotherapy, since it has the potential for accurately matching the prescribed and delivered dose distributions, hence enabling the effective treatment of complex tumour scenarios. Associated (inverse) treatment planning methods address a constrained linear optimisation problem involving the determination of intensity modulation functions from the prescribed target dose, optimisation criteria and imposed constraints. Many of the reported inverse planning techniques require a considerable number of iterations for algorithm convergence, making them unattractive for clinical use. This paper reports the significant improvement in convergence time possible using a dynamic relaxation technique applied to a Bayesian optimisation process. Performance of the algorithm is illustrated using a complex concave tumour scenario.

Single Channel Analysis Of Sleep EEG : An Adaptive Database Method

Authors:
C. Berthomier, J. Prado, *Dept. Signal, E.N.S.T. (FRANCE)*
O. Benoit, *Laboratoire d'Exporations Fonctionnelles (FRANCE)*

ABSTRACT

An automatic procedure for the spectral analysis of an all-night sleep electroencephalogram (EEG) is presented. This method relies on a fixed database initializing a procedure which adapts parameters so that they match to the signal to be analyzed. Parameters coming from database are normalized power spectra in predefined frequency bands. The novelty of our approach is to use exible sleep stage patterns rather than fixed ones, these patterns being iteratively updated (thanks to a short/long term analysis) in order to cope with the EEG variability. The main part of the procedure, performed on-line, is followed by avery short off-line processing yielding a real time implementation. The procedure adaptation ability is shown on detection of Rapid Eyes Movement (REM) events, the latter being well known for their inter as well as for their intra individual extreme variability.

Blood Glucose Prediction For Diabetes Therapy Using A Recurrent Artificial Neural Network

Authors:

W. Sandham, D. Hamilton, C. MacGregor, *University of Strathclyde (SCOTLAND)*

D. Nikoletou, *University of Strathclyde (SCOTLAND)*

K. Paterson, A. Japp, *Diabetes Centre, Royal Infirmary (SCOTLAND)*

ABSTRACT

Expert short-term management of diabetes through good glycaemic control, is necessary to delay or even prevent serious degenerative complications developing in the long term, due to consistently high blood glucose levels (BGLs). Good glycaemic control may be achieved by predicting a future BGL based on past BGLs and past and anticipated diet, exercise schedule and insulin regime (the latter for insulin dependent diabetics). This predicted BGL may then be used in a computerised management system to achieve short-term normoglycaemia. This paper investigates the use of a recurrent artificial neural network for predicting BGL, and presents preliminary results for two insulin dependent diabetic females.

4-D Reconstruction Of The Left Ventricle From A Single Cycle Ultrasound Acquisition

Authors:
C. Bonciu, R. Weber, *Universite d'Orleans (FRANCE)*
L. D. Nguyen, *Centre Hospitalier Regional d'Orleans (FRANCE)*

ABSTRACT

A new acquisition system using a fast rotating 2-D ultra-sound probe is proposed to reconstruct the deformations of the left cardiac ventricle. During only one cardiac cy-cle, the high resolution probe acquires successive conic sections of the left ventricle. Then, assuming that the contours have been accurately detected, the set of sparse spatio-temporal data, which correspond to the successive intersection points of the moving ultrasound beam with the endocardiac wall, is spatially and temporally inter- polated to estimate the continuous variation of the volume. This reconstruction is achieved through an iterative algorithm based on the harmonic 4-D model of the volume considered as a periodic multidimensional signal [1]. Results are clinically promising.

Ultrasonic Array Imaging Using CDMA Techniques

Authors:

Y. S. Avrithis, A. N. Delopoulos, G. C. Papageorgiou, *National Technical University of Athens (GREECE)*

ABSTRACT

A new method for designing ultrasonic imaging systems is presented in this paper. The method is based on the use of transducer arrays whose elements transmit wideband signals generated by pseudo-random codes, similarly to code division multiple access (CDMA) systems in communications. The use of code sequences instead of pulses, which are typically used in conventional phased arrays, combined with transmit and receive beamforming for steering different codes at each direction, permits parallel acquisition of a large number of measurements corresponding to different directions. Significantly higher image acquisition rate as well as lateral and contrast resolution are thus obtained, while axial resolution remains close to that of phased arrays operating in pulse-echo mode. Time and frequency division techniques are also studied and a unified theoretical model is derived, which is validated by experimental results.

A Comparison Of Several Interpolation Methods In 3d X-Ray Cone Beam Reconstruction

Authors:
J. G. Donaire, I. Garcia, *University of Almeria (SPAIN)*

ABSTRACT

This work is concerned with the use of iterative methods for image reconstruction from projections in 3D cone beam transmission tomography. It is aimed at the anal-ysis of several 2D interpolation methods used to obtain the projection matrix and a set of parameters that play an important role in the reconstruction process, such as the relaxation parameter ë and the interpolation footprint radius, and the use of a Global Optimization algorithm for the best choice of these free parameters.

Weigthing Hyperparameters For 3D Bayesian Estimation In Eddy Current Tomography

Authors:

O. Venard, *L.E.Si.R. - ENS Cachan (FRANCE), L.S.M. - ESIEE (FRANCE)*
D. Premel, *L.E.Si.R. - ENS Cachan (FRANCE)*

ABSTRACT

The skin effect is a major limitation for Eddy Current Tomography ECT. It limits its ability to image aws inside electricaly conductive material. In this paper we tackle the inverse problem in a Bayesian framework which imply the computation of the hyper-parameters defining the weight of theapriori in the estimate. To improve its performance regarding buried aws, we suggest that hyperparameters of the inverse problem can't be the same for each layer of discretization and we propose a method to bias them accordingly to the skin effect.

Enhancement Of Mammographic Images For Detection Of Microcalcifications

Authors:

D. Seršiæ, S. Lonèariæ, *University of Zagreb (CROATIA)*

ABSTRACT

A novel approach to image enhancement of digital mammography images is introduced, for more accurate detection of microcalcification clusters. In original mammographic images obtained by X-ray radiography, most of the information is hidden to the human observer. The method is based on redundant discrete wavelet transform due to its good properties: shift invariance and numeric robustness. The procedure consists of three steps: low-frequency tissue density component removal, noise filtering, and microcalcification enhancement. The experimental results have shown good properties of the proposed method.

Voice Source Parameters For Speaker Verification

Authors:
A. Neocleous, P. A. Naylor, *Imperial College (UK)*

ABSTRACT

In this paper we report on a study of the variability of voice source parameters in the context of speaker characterisation, and we propose a speaker verification system which incorporates these parameters. The motivation for this approach is that, whilst we have conscious control over the action of our vocal tract articulators such as the tongue and jaw, we have only limited voluntary muscle control over the vocal cords. The conjecture is, therefore, that impostors are less likely to be able to mimic vocal cord effects than vocal tract effects. The hybrid speaker verification system that is proposed incorporates two sub-systems to improve the overall performance: (i) a cepstral-based HMM with cohort normalisation and (ii) voice source parameters derived from Multi-cycle Closed-phase Glottal Inverse Filtering (MCGIF). Preliminary experimental results show that the hybrid system performs better than either of the sub-systems in terms of the equal error rate (EER). Specifically, the hybrid system improved the performance of the cepstral-based HMM system by 78% on average, resulting in a mean EER of 0.42% for the specific tests conducted.

A Nonlinear Algorithm For Epoch Marking In Speech Signals Using Poincare Maps

Authors:
I. Mann, S. McLaughlin, *University of Edinburgh (UK)*

ABSTRACT

A novel nonlinear epoch marking algorithm is proposed for use with voiced speech signals. Epoch detection is useful for speech coding, synthesis and recognition purposes, as it provides both the moment of glottal closure and the instantaneous pitch. Our technique functions entirely in state space, by operating on a three dimensional reconstruction of the speech signal which is formed by embedding. By using the fact that one revolution of this reconstructed attractor is equal to one pitch period, we are able to find points which are pitch synchronous by the use of a Poincare section. Evidently the epoch pulses are pitch synchronous and therefore can be marked. Results using real speech signals are presented to illustrate the performance of the technique.

A Geometric Algorithm For Voice Activity Detection In Nonstationary Gaussian Noise

Authors:
H. Ozer, S. G. Tanyer, *Baskent University (TURKEY)*

ABSTRACT

A new algorithm for voice activity detection in additive nonstationary noise is presented. The algorithm utilizes the differences of the probability distribution properties of noise and speech signal. The Magnitude Density (mdf) and the Magnitude Distribution Functions (MDF) are defined. The noise level is monitored for automatic threshold estimation. The estimate is shown to be accurate also when analysis windows do not fully contain non-speech signals and in the presence of nonstationary noise. The algorithm has been applied different type of noises (traffic, water, restaurant, ect.). The voice activity detection algorithm is shown to operate reliably in SNRs down to 0 dB and noise variance up to 10 dB/sec.

Comparison Of Some Time-Frequency Analysis Methods For Classification Of Plosives

Authors:

E. £ukasik, S. Grocholewski, *Poznañ University of Technology (POLAND)*

ABSTRACT

The paper deals with the context independent recognition of unvoiced plosives (/p/, /t/, /k/). In several solutions the best feature vectors are being sought in the burst segments of plosives. It has been proved that the difference between stops fade out very quickly after the burst onset. The short time of the burst duration implies the need of the higher time resolution in time-frequency analysis. The paper presents the results of the application of selected methods of high resolution time-frequency distributions for the recognition of stops. Apart from the traditional Short Time Fourier Transform based spectrogram, Gabor Spectrogram and cone shaped distribution have been used to calculate input parameters (cepstral coefficients) to the neural network used for classification.

Speech Parameters Vector Based On Arithmetic Fourier Transform

Authors:

E I. Bovbel, I. E. Kheidorov, *Belarusian State University (BELARUS)*

ABSTRACT

This paper is devoted to the developing of speech character vector based on the arithmetic Fourier transform for speech recognition tasks. The calculation complexity and accuracy of three spectrum methods - fast Fourier transform (FFT), linear prediction (LP) method and arithmetic Fourier transform (AFT) were estimated. It was found that AFT is very useful to calculate speech mel-parameters and provides a simple parallel VLSI realization. If to apply the cos transformation for the AFT mel-spectral parameters we achieve robust parameters for speech signal variations. As a result of the researches the speech parameter vector based on AFT coefficients was designed.

CELP With Priority To Critical Segments

Authors:
L. Martins da Silva, A. Alcaim, *CETUC - PUC/Rio (BRAZIL)*

ABSTRACT

At low bit rates (around and below 4 kbit/s), the use of a rigid coding configuration in the CELP (Code Excited Linear Prediction) speech coding structure leads to a bad reconstruction of the voiced onsets. In this paper we describe a 4 kbit/s CELP algorithm that gives priority to these critical speech segments. Our scheme employs different combinations of codebooks and bit allocations for different speech classes. Mean opinion scores (MOS) reveal that at 4 kbit/s the speech quality obtained with our coding algorithm is significantly better than that achieved with a CELP codec that does not use different coding strategies for different classes of speech.

On The Combination Of Redundant And Zero-Redundant Channel Error Detection In Celp Speech-Coding

Authors:

N. Goertz, *University of Kiel (GERMANY)*

ABSTRACT

In this paper a new algorithm is described for selective detection and handling of speech codec parameters which were corrupted by bit-errors on the transmission channel. A combination of classical forward error detection schemes using additional (redundant) bits and parameter-correlation based techniques without redundant bits (zero-redundant error detections) is used for this purpose. The algorithm is optimized by informal listening tests rather than by maximization of mathematically tractable measures (e.g. SNR) which usually do not re ect subjective speech quality well. No additional delay and almost no additional memory and complexity is required for the new algorithm. The speech quality resulting at the decoder output is strongly improved compared to standard bad-frame handling techniques if coded speech is transmitted over disturbed channels, e.g. the GSM-full-rate channel which is used for performance evaluation.

A New Intraframe LSP Interpolation Technique For Low Bit Rate Speech Coding

Authors:
J. S. Mao, S. C. Chan, K. L. Ho, *The University of Hong Kong (HONG KONG)*

ABSTRACT

This paper presents a linear LSP interpolation between neighboring frames. Usually there are two to four formants in speech spectrum envelope, and the LSF(Line Spectral Frequency) parameters have an order property. In this paper, we divide the LSF parameters into three sub-vectors, and predict them by the vectors of previous and next frames. An efficient and reliable LSP vector distance measure is proposed for this interpolation algorithm. The interpolated LSP vectors are utilized in our mixed excitation LPC vocoder, which is operated at 1.5 kbps. Informal listening tests indicate that the synthesized speech sounds natural and intelligible.

Backward Adaptive Warped Lattice For Wideband Stereo Coding

Authors:

Aki Harma, U. K. Laine, M. Karjalainen, *Helsinki University of Technology (FINLAND)*

ABSTRACT

In this paper an extremely low delay perceptual audio codec is presented. The codec is based on warped linear prediction which inherently utilizes auditory frequency resolution and frequency masking characteristics of hearing. In the current version of the codec the coding delay is the minimum. This is achieved using backward adaptive lattice methods where waveform modeling is completely based on already transmitted data. Coding technique is applied separately to the two channels but the quantization processes are unified to gain more bit rate reduction.

An Algorithm For Wideband Audio Coding Based On LMS/RLS Switched Predictors

Authors:
E. Mumolo, M. Scagnol A.Carini, *Universita' di Trieste (ITALY)*

ABSTRACT

This paper describes an algorithm for coding high quality audio signals using a switched ADPCM approach. Several theoretical and practical issues are considered as outlined in the paper.

Adaptive System Identification Using the Normalized Least Mean Fourth Algorithm

Authors:
A. Zerguine, *Dept. of Physics KFUPM (SAUDI ARABIA)*
M. Bettayeb *Dept. of Electrical Eng. KFUPM (SAUDI ARABIA)*

ABSTRACT

In this work we propose a novel scheme for adaptive system identification. This scheme is based on a normalized version of the least mean fourth (LMF) algorithm. In contrast to the LMF algorithm, this new normalized version of the LMF algorithm is found to be independent of the input sequence autocorrelation matrix. It is also found that it converges faster than the normalized least mean square (NLMS) algorithm for the lowest steady-state error reached by the NLMS algorithm. Simulation results confirm the superior performance of the new algorithm.

Novel Adaptive Algorithm Based On Least Mean p-Power Error Criterion For Fourier Analysis In Additive Noise

Authors:
Y. Xiao, K. Shida, *Saga University (JAPAN)*

ABSTRACT

This paper presents a novel adaptive algorithm for the estimation of discrete Fourier coefficients (DFC) of sinusoidal and/or quasi-periodic signals in additive noise. The algorithm is derived using a least mean p-power error criterion. It reduces to the conventional LMS algorithm when p takes on 2. It is revealed by both analytical results and extensive simulations that the new algorithm for p=3, 4 generates much improved DFC estimates in moderate and high SNR environments compared to the LMS algorithm, while both have similar degrees of complexity. Assuming the Gaussian property of the estimation error, the proposed algorithm including the LMS algorithm is analyzed in detail. Elegant dynamic equations and closed form noise misadjustment expressions are derived and clarified.

The Summational Projection Algorithm For The Adaptive Volterra Filter

Authors:
Y. Kajikawa, Y. Nomura, *Kansai University (JAPAN)*

ABSTRACT

In this paper, we propose a summational projection algorithm with block length control. This algorithm has the convergence properties of high speed and high accuracy under high noise for adaptive Volterra filters. And this algorithm has a computational complexity of $O(p \cdot N^2)$. The proposed algorithm realizes these convergence properties by controlling the block length in the updating algorithm. In addition, the algorithm can track the variation of the impulse response of an unknown system and the power variation of an additive noise. We show the effectiveness of the proposed algorithm by computer simulations in this paper.

Convergence Analysis And Fast Algorithms Of Volterra Adaptive Filters

Authors:

J. Chao, A. Inomata, S. Uno, *Chuo University, (JAPAN)*

ABSTRACT

In this paper, tight upper and lower bounds are obtained for all of the eigenvalues of covariance matrices in quadratic Volterra adaptive filters with Gaussian input signal. It is shown that the error surface of quadratic Volterra ADF is always extremely steep in one particular direction but relatively flat in the rest directions. Based on these analysis, fast converging adaptive algorithms for both white and colored input signals are presented.

Evolving Complex Adaptive IIR Structures

Authors:

S. Sundaralingam, K. Sharman, *The University of Glasgow (UK).*

ABSTRACT

A new Evolutionary Algorithm (EA) has been formulated for evolving adaptive complex infinite impulse response (IIR) structures. It is a variation of the standard Genetic Algorithm (GA) in which a new operator called "immigrant" is introduced in addition to the standard genetic operators (crossover and mutation). This operator aims to avoid premature convergence and to increase the search points while maintaining a small population size. Filter stability is ensured by removing any unstable systems from the population as it evolves. The algorithm is applied to the design of a channel equalisation filter and it shows a significantly better mean square error performance (MSE) when compared with an equivalent filter optimised through traditional GAs.

Fractionally-Spaced Equalization For Time-Varying Channels

Authors:
M.L. Alberi, I. Fijalkow, *Equipe de Traitement des Images et du Signal, ETIS / ENSEA (FRANCE)*

ABSTRACT

The good convergence tracking properties of spatio-temporal equalizers are pointed out and analyzed when there is effective diversity. The analysis is illustrated in the case of a frequency offset between the baud and sampling clocks that induces important time-variations.

Adaptive Blind Separation Of Convolved Sources Based On Minimization Of The Generalized Instantaneous Energy

Authors:

I. Kopriva, *University of Zagreb (CROATIA)*

ABSTRACT

Generalization of the energy concept is proposed resulting in a class of novel on-line algorithms for blind separation of convolved signals. Signal separation is achieved when signal energy of the appropriated order is minimal. The resulting learning rules have the similar form as those recently discussed to be optimal for blind separation of instantaneously mixed signals. Algorithms are tested on the separation of two real-world signals. It is believed that for the first time the blind signal separation (BSS) theory is applied to the light sources localization problem. With proposed algorithms better separation quality is obtained than when using adaptive decorrelation, recently proposed separation algorithm based on entropy maximization and neural network separator based on nonlinear odd activation functions.

Adaptive Blind Equalisation For Asynchronous DS-CDMA Systems Based On RLS

Authors:
G. Leus, M. Moonen, *Katholieke Universiteit Leuven - ESAT (BELGIUM)*

ABSTRACT

This paper proposes a new adaptive blind equalization scheme for asynchronous direct-sequence code-division multiple-access (DS-CDMA) systems based on recursive least squares (RLS). Linear equalizer based blind direct symbol estimation is considered, which is related to subspace based blind direct symbol estimation. Compared with the subspace based algorithm, the proposed algorithm has increased dimensions but does not make use of the computationally demanding singular value decomposition (SVD). Furthermore, simulations show that the performance is more robust against multi-user interference (MUI).

Improved Neural Network Equalization By The Use Of Maximum Covariance Weight Initialization

Authors:

A. Kantsila, M. Lehtokangas, J. Saarinen, *Tampere University of Technology (FINLAND)*

ABSTRACT

In this paper we focus on adaptive equalization of binary telecommunication signals in a baseband digital communication system. We have studied the use of multilayer perceptron (MLP) networks for equalizing binary data bursts in a channel that introduces both intersymbol interference and noise to the transmitted signal. Conventionally the weights of the MLP network are initialized randomly. Here we have studied the use of maximum covariance (MC) initialization scheme in the weight initialization. By applying MC initialization we have been able to speed up the convergence and decrease the total computational load of the system. This is very important in telecommunications, where it is often not possible to use systems that require a lot of computation.

Adaptive Nonlinear Filtering
With The Support Vector Method

Authors:

D. Mattera, F. Palmieri, *Universita degli Studi di Napoli Federico II (ITALY)*
S. Haykin, *McMaster University (CANADA)*

ABSTRACT

The recently introduced Support Vector Method (SVM) is one of the most powerful methods for training a Radial Basis Function (RBF) filter in a batch mode. This paper proposes a modification of this method for on-line adaptation of the filter parameters on a block-by-block basis. The proposed method requires a limited number of computations and compares well with other adaptive RBF filters.

Adaptive Weighted Vector Median Filter Using A Gradient Algorithm

Authors:
L. Lucat, P. Siohan, *France Telecom CNET/DMR/DDH (FRANCE)*

ABSTRACT

In this paper, a gradient based approach to adapt the parameters of the Weighted Vector Median Filter is presented. The validity of the method is inspected through a convergence test of the filter parameters and with results of noisy image filtering.

Smoothing Of Noisy AR Signals Using An Adaptive Kalman Filter

Authors:

G. Doblinger, *Vienna University of Technology (AUSTRIA)*

ABSTRACT

In this paper, we describe a new and computationally efficient adaptive system for the enhancement of autoregressive (AR) signals which are disturbed by additive white or colored noise. The system is comprised of an adaptive Kalman filter operating as a fixed lag smoother and a subsystem for AR parameter estimation. A superior performance is achieved by implementing a feedback loop between the Kalman filter output and the parameter estimation. Accordingly, the AR parameters are obtained from the enhanced signal and the influence of the disturbing noise on the parameter estimation is damped down. Another advantage of the adaptive Kalman filter is its tracking capability for short-time stationary signals.

Accurate LDA-Spectra
By Resampling And ARMA-Modeling

Authors:
S. de Waele, P.M.T. Broersen, *Delft University of Technology, (NETHERLANDS)*

ABSTRACT

With Laser-Doppler Anemometry (LDA) the velocity of gases and liquids is measured without disturbing the flow. The velocity signal is sampled at irregular intervals; a regularly resampled signal is extracted using Nearest Neighbor Resampling or Linear Interpolation. From the resampled data the spectrum is estimated using AR-MA time series modeling. AR-MA modeling yields a more accurate description of the spectral structure than the best Windowed Periodogram. The accuracy of the spectral description is established with an objective measure: the model error at time scale T.

SINTRACK Analysis.
Application To Detection And Estimation Of Flutter For Flexible Structures

Authors:
M. Jeanneau, C. Pendaries, *SUPAERO - ONERA-CERT (FRANCE)*
P. Mouyon, *ONERA-CERT (FRANCE)*

ABSTRACT

The low computation cost, short delay and accuracy of SINTRACK, makes it a very interesting real-time signal processing tool for detection and estimation of damped sinusoids in noise. A noise analysis, as well as a parameters' optimal adjustment analysis are provided. SINTRACK was successfully applied for detection and estimation of a HALE wing bending and torsion oscillations, validating a theoretical flutter prediction law.

Nonlinear Filtering Of Mr Images Using Geometrically And Statistically Controlled Diffusion

Authors:
I. Bajla, V. Witkovsky, *Slovak Academy of Sciences (SLOVAKIA)*
M. Hanajik, *Slovak Technical University, (SLOVAKIA)*

ABSTRACT

In this paper a novel approach tothe filtering of multivalued Magnetic Resonance (MR) images is proposed. The proposed method is essentially a nonlinear diffusion with a statistically and geometrically controlled conductance. The user is required to define samples of individual tissue classes in the input image, and their statistics are exploited during the image filtering. The method can be used in medical diagnostics for the enhancement and segmentation of medical images.

A Novel Algorithm For Digital Image Processing Applications

Authors:
S. Boussakta, *University of Teesside (UK)*

ABSTRACT

This paper introduces an efficient algorithm for the calculation of 2-D convolution and correlation for image processing applications. The method combines the new 2-D Mersenne with the 2-D Fermat numbers transforms using the 2-D mixed radix conversion. The moduli of the transforms are selected to be close to one another, thus, the choice of the dynamic range and the constraint between the transform sizes and the world length becomes more flexible. The residue transforms are independent and can be calculated in parallel for high speed and high throughput. Hence the technique is suitable for image processing applications such as error free image filtering and enhancement.

Stable Nonlinear Filters With Spatial Prediction

Authors:

J. Abbas, M. Domañski, *Poznañ University of Technology (POLAND)*

ABSTRACT

The paper deals with nonlinear two-dimensional filters for image restoration. The filter employs classic predictor structure with nonlinear predictor and prediction error processed by a static (memoryless) nonlinear element or by a nonlinear two-dimensional filter. The filters discussed are both recursive and nonrecursive. Even in the case of recursive structures stability is guaranteed. The filters are suitable for impulsive noise removal from color images. Preservation of textures and even one-pixel thin lines are advantages of the filters proposed. The experimental data prove that these filters outperforms classic nonlinear median-based filters like vector median, recursive median and weighted median.

From Continuum Model
To A Detailed Discrete Theory Of Median Shifts

Authors:
E. R. Davies, *University of London (UK)*

ABSTRACT

This paper presents a new discrete theory of median shifts. It relates the theory to the older continuum model, predicts angular variations, and also gives an accurate figure for the active area of any discrete neighbourhood, thereby making the continuum model more accurate. The work shows that at low curvature values a quadratic law applies, this being followed by the previously known linear variation. It also explains why the observed linear behaviour is significantly larger than indicated by the continuum model. Overall, there is now very good agreement between the new discrete theory, the continuum model and the observed results.

Fuzzy Colour Filtering Using Relative Entropy

Authors:

A. Fotinos, N. Laskaris, S. Fotopoulos, *University of Patras (GREECE)*

ABSTRACT

Relative entropy (**E_{rel}**) has long been used as a distance measure between two sets of sampled data. In this work **E_{rel}** is used for filtering of colour images. Local statistical characteristics of the image are estimated by means of Parzen estimators. Two indices for the noise are derived. These indices are associated with membership functions. Filtering of the image is achieved by using a small number of fuzzy rules.

Decomposition And Order Statistics In Filtering

Authors:

D. Coltuc, P. Bolon, *University of Savoie (FRANCE)*

ABSTRACT

The paper investigates a three stage filtering scheme, namely: 1) signal decomposition, 2) filtering and 3) signal reconstruction. If marginal rank order filtering is used in step 2, the derived filtering scheme generalizes the classical order statistics one. Based on this idea, a new family of nonlinear filters, called decomposition filters, is proposed and investigated. The most interesting feature of the new filters is their dependency on signal decomposition. Three decomposition procedures that exhibit certain minimum and symmetry properties are investigated. They are the canonical decomposition of functions, the Jordan decomposition of bounded variation functions and the parity decomposition, respectively. The properties of the derived filters are discussed.

A True Order Recursive Algorithm For Two-Dimensional Least Squares Error Linear Prediction And Filtering

Authors:
G.-O. Glentis, *TEI of Heraklion (GREECE)*

ABSTRACT

In this paper a novel algorithm is presented for the efficient Two-Dimensional (2-D), Least Squares (LS) FIR filtering and system identification. Causal filter masks of general boundaries are allowed. Efficient order updating recursions are developed by exploiting the spatial shift invariance property of the 2-D data set. Single step order updating recursions are developed. During each iteration, the filter coefficients set is augmented by a single new element. The single step order updating formulas allow for the development of an efficient, true order recursive algorithm for the 2-D LS causal linear prediction and filtering.

One-Dimensional Scale-Space Preserving Filters

Authors:
R. Harvey, A. Bosson, J. A. Bangham, *University of East Anglia (UK)*

ABSTRACT

We show how graph-morphology processors may be specialized to one dimension and how in this case they amount to parsers of extrema. We compare these scale-space proces-sors on the basis of their performance in additive and replacement noise. Of the filters studied we find that *M* and *N*-filters behave similarly to the recursive median filter and hence inherit their robustness whereas multiscale openings and closings produce much less stable representations.

Transfer Function Models
For Continuous And Discrete Multidimensional Systems

Authors:

R. Rabenstein, *Universitat Erlangen-Nurnberg (GERMANY)*

ABSTRACT

Continuous multidimensional systems, which are described by partial differential equations are usually discretized by standard methods from numerical mathematics. Here, a general approach for the transfer function description of multidimensional systems is presented. It allows a correct representation of initial and boundary values also for problems with spatially varying coefficients, general boundary conditions, three spatial dimensions and general differentiation operators. In spite of this generality, the resulting discrete systems can be realized with standard signal processing elements and are free of implicit loops.

The Periodic Step Gradient Descent Algorithm - General Analysis And Application To The Super Resolution Reconstruction Problem

Authors:
T. Sagi, A. Feuer, *Technion - Israel Institute of Technology (ISRAEL)*
M. Elad, *Hewlett Packard Laboratories - Israel (ISRAEL)*

ABSTRACT

Solving image reconstruction problems, especially complex problems like Super Resolution reconstruction, is very demanding computationally. Iterative algorithms are the practical tool frequently used for this purpose. This paper reviews the *Periodic Step Gradient Descent* (PSGD) algorithm, suggested as a sub-optimal algorithm for solving reconstruction problems (with emphasis on Super Resolution reconstruction problems). The PSGD differs from well-known iterative algorithms in the way the data of the problem at hand is processed. Whereas iterative algorithms process the entire given data in order to update the result, the PSGD updates the result progressively. This paper provides an analysis of the PSGD. We show that the PSGD has an efficient implementation, easy to achieve convergence conditions and fast convergence speed when applied to a Super Resolution reconstruction problem. The performance of the PSGD when applied to a Super Resolution reconstruction problem, is demonstrated by simulations and compared to the performance of other well-known algorithms.

Reconstruction Of Locally Homogeneous Images

Authors:

M. Nikolova, *Universite Rene Descartes (FRANCE)*

ABSTRACT

The reconstruction of images involving large homogeneous zones from noisy data, given at the output of an observation system, is a common problem arising in various applications. A popular approach for its resolution is regularized estimation: the sought image is defined as the minimizer of an energy function combining a data-fidelity term and a regularization prior term. The latter term results from applying a set of potential functions (PFs) to the differences between neighbouring pixels and it can be seen as a Markovian energy. We formalize and perform a mathematical study of the possibility to obtain images comprising either strongly homogeneous regions or weakly homogeneous zones, using regularized estimation. Our results reveal that the recovery of zones of either type in an estimated image depends uniquely on the smoothness at zero of the PFs involved in the prior term. These theoretical results are illustrated on the deblurring of an image.

Statistical Restoration Of Images Using A Hybrid Bayesian Approach

Authors:

D. Hudson, M. Razaz , *University of East Anglia (UK.)*

ABSTRACT

The desire to view smaller and smaller attributes within biological specimens means that confocal microscopes are often used at the limit of their resolution. For quantitative analysis of smaller sized attributes, and as a necessary pre-processing stage for automatic recognition and classification of objects it is essential that confocal images are restored. A fast new hybrid statistical restoration algorithm is presented which makes use of deterministic methodology to speed up optimisation of the posterior probability. Additionally, a prior probability model based on the bayesian classifier is proposed. Restorations of real confocal image data using the above technique and prior are presented and discussed. Quantative analysis of the improvement gained through our hybrid approach is also presented.

A New Approach For Restoration Of NMR Signals

Authors:
M.Razaz, R.A.Lee, *University of East Anglia (UK)*
P.S.Belton, K.M.Wright, *Institute of Food Research (UK)*

ABSTRACT

NMR is a widely used technique for analysing the structure of matter, however the signals produced can be of very poor quality even with the averaging of hundreds or thousands of individual scans. It has therefore become common practice to use a restoration algorithm to improve the signal and remove degradation effects such as line broadening or blurring caused by the NMR instrument itself. The dominant restoration method currently used is maximum entropy. However this is very slow even with signals of 16000 data points and with newer NMR instruments the sampling rate is increasing dramatically enabling signals of over 100,000 data points or more to be collected. Clearly maximum entropy processing of these signals will take hours. Here we present an alternative non-linear restoration method, projection onto convex sets (POCS), which is capable of producing restored signals of equal or better quality to maximum entropy and in much less time.

Enhancing Handwritten Character Images Thanks To A Re-Sampling Process Based On Convex Hull Extraction

Authors:

B. Gosselin, *Faculté Polytechnique de Mons (BELGIUM)*

ABSTRACT

In this paper, we propose a new method that allows, by finding the convex hull of a character image, to set out in one pass only, the control parameters of a particular character distortion process. This character distortion method can then be applied to normalize the character image, i.e. to reduce the within-class scatter of images of handwritten characters, which could lead to a significant improvement of recognition performance. Many tests have been performed on unconstrained handwritten uppercase letters extracted from the NIST3 database. Finally, the combination of two classifiers, one using the proposed normalization method, and the other one not, has allow reducing the overall error rate from 5.24% to 3.88%.

A Sequential Projections Based Postprocessor For Block-Based Image Coding

Authors:
J.-H. Chang, *Samsung Electronics Co. (KOREA)*
S. H. Lee, J.-K. Kim, C. W. Lee, *Seoul National University (KOREA)*

ABSTRACT

In this paper, we propose a novel post-processing technique to reduce blocking artifacts for block-based image coding schemes. Our approach focuses mainly on the reconstruction of the surface continuity, including the continuity of edges and textures in the image objects. To do this, we introduce a set of continuous functions suitable for characterizing edges and, by a linear combination of these functions, estimate the original artifact-free image. We propose the Projections onto the OverComplete Basis (POCB) algorithm to find linear coefficients satisfying a constraint which controls the difference between the blocky and estimated images. The proposed and conventional techniques are tested on various images compressed by the JPEG standard and vector quantization. The simulation results show that the proposed technique yields better results both objectively and subjectively.

Enhancement Of Sketch Contours On Paintings Infrared Photographies: A Comparison

Authors:
A. de Albuquerque Araujo, B. V. Coelho, *Universidade Federal de Minas Gerais (BRAZIL)*
R. M. Hadad, *Fundacao Centro Tecnologico de Minas Gerais (BRAZIL)*

ABSTRACT

This work reports and illustrates the application of optimal filtering to enhance sketch contours on infrared photograhies of wood paintings. In [14] the authors tested three approaches of the symmetric exponential filter of an infinitely large window size proposed by Shen and Castan [13]: the DRF, GEF and SDEF filters. The DRF filter presented the best results and was chosen to have a fine tuning of its parameters. In this paper the DRF filter results obtained in [14] are now compared to methods proposed by Marr [9] and Canny [12].

An Integrated System For Object Tracking And Progressive Coding Based On Statistical Morphology

Authors:
C. S. Regazzoni, A. Teschioni, *University of Genoa (ITALY)*
G. Foresti, *University of Udine (ITALY)*

ABSTRACT

This work presents an integrated system for progressive and predictive image sequences coding and human body movement tracking. The object shape representation, based on the morphological skeleton, is used for both coding and tracking and it allows achieving a high compression ratio. Kalman Filters are used to track the shape of the human body. Practical applications of the focused proposed strategy concern with the surveillance of public areas (supermarkets, stations, stadiums, etc.).

An Inverse Problem: Histogram Equalization

Authors:
D. Coltuc, P. Bolon, *University of Savoie, (FRANCE)*

ABSTRACT

The well-known histogram equalization algorithm is not reversible, namely, given an equalized image and its initial histogram, the original image cannot be recovered. The paper proposes a solution to histogram inversion problem. An ordering on images, closely related to the human perception of brightness, is defined. The proposed ordering refines the normal ordering on graylevels up to a strict ordering. Based on the assumption that the ordering is conserved by histogram equalization algorithm, the inverse problem is further solved. The experimental results show a very good recovery of the original.

Rapid Design Of Discrete Transform Cores

Authors:
J. K.Hunter, J.V.McCanny, *The Queen's University of Belfast (NORTHERN IRELAND)*

ABSTRACT

A new modular approach for the rapid design of application specific transform cores is presented. This approach allows the creation of hardware description language generators for the automated design and VLSI synthesis of specific integrated circuits for a range of multimedia applications. Resulting transform cores are portable across many silicon foundries and their hierarchical nature means that they can also be easily migrated to FPGA and PLD implementations. Exemplar core designs have been produced for the Discrete Cosine Transform (DCT) and are shown to compare very favorably in terms of area, performance and power consumption with DCT circuits developed using more conventional methods. However, design times are reduced considerably. Most significantly, no constraints are placed on the transform size and area/speed trade-offs can be rapidly analyzed. Discrete Sine Transform cores have also been developed.

Low-Power Implementation Of Discrete Wavelet Transform

Authors:
K. Masselos, P. Merakos, T. Stouraitis, C. E. Goutis
University of Patras (GREECE)

ABSTRACT

This paper presents a novel methodology for low power implementation of one and multidimensional discrete wavelet transform. The basic computation performed by forward and inverse wavelet transform is the computation of inner products between vectors of data (either input data or wavelet coefficients of previous stages) and filter coefficients. The proposed methodology aims at reducing the switching activity of the inner product computations required by the wavelet transform, by reordering the sequence of evaluation of the partial products. The total hamming distance of the sequence of coefficients (sum of hamming distances between successive coefficients) is used as the cost function driving the reordering. Minimization of this cost function leads to switching activity reduction at the inputs of the computational units. The proposed methodology is applicable to both custom hardware and instruction set architectures. Experimental results show that the proposed methodology leads to significant switching activity and thus power consumption savings.

Efficient Implementation Of The Row-Column 8X8 IDCT On VLIW Architectures

Authors:

R. Sakellariou, *University of Manchester (UK)*
C. Eisenbeis, *INRIA (FRANCE)*
P. Knijnenburg, *Leiden University (THE NETHERLANDS)*

ABSTRACT

This paper experiments with a methodology for mapping the 8 X 8 row-column Inverse Discrete Cosine Transform on general-purpose Very Long Instruction Word architectures. By exploiting the parallelism inherent in the algorithm, the results obtained indicate that such processors, using sufficiently advanced compilers, can provide satisfactory performance at low cost without need to resort to special-purpose hardware or time-consuming hand-tuning of codes.

Relaxed Look-Ahead Technique For Pipelined Implementation Of Adaptive Multiple-Antenna Cdma Mobile Receivers

Authors:
R. Baghaie, S. Werner, T. Laakso, *Helsinki University of Technology (FINLAND)*

ABSTRACT

In this paper, Relaxed Look-Ahead technique for pipelined implementation of an adaptive Direct-Sequence Code Division Multiple Access receiver is proposed when multiple antennas are utilized for mobile communications. Adaptive multiple-antenna receivers can provide insensitivity to the interfering powers. They also provide room for more users or require smaller number of antennas than the matched filter solution. A number of approximation techniques are utilized to pipeline the adaptive algorithm used for the proposed multiple-antenna receiver. The resulting pipelined receiver achieves a higher throughput or requires lower power as compared to the receiver using the serial algorithm. With the aid of simulations, for different levels of pipelining and different number of antennas, the signal-to-interference ratio and the bit error rate versus the relative interfering power are illustrated.

A Scheme For The VLSI Implementation Of FIR Digital Filters With Reduced Latency

Authors:
C. Gr. Caraiscos, *Technological Educational Institution (TEI) of Piraeus, (GREECE)*
K. Z. Pekmestzi, *National Technical University of Athens, (GREECE)*

ABSTRACT

A·modular bit-parallel implementation scheme for an FIR digital filter which is systolic in the bit-level and exhibits a latency of three clock cycles, is presented. In the word-level, the scheme is that of the merged two-way pipeline systolic array in which each processing element computes two product terms and adds them to a partial sum. In this way high throughput and immediate response are obtained. Merging is also used in the bit-level for the realization of the processing elements in order to obtain low latency. Moreover, the cells of the processing elements are internally pipelined which gives high operation rate, limited by the propagation delay of a gated full-adder and a latch.

Efficient Bit-Level Design Of An On-Board Digital TV Demultiplexer

Authors:
S. Calvo, J.Sala, A. Pages, G. Vazquez, *Universitat Politecnica de Catalunya (SPAIN)*

ABSTRACT

A bit-level description of the signal processing stage of an on-board integrated VLSI multi-carrier demodulator is presented in this paper, along with a description of the optimization procedure that has been developed for the signal processing functions. The demultiplexer is capable of handling a varying number of carriers in a 36 MHz bandwidth on the satellite up-link. Its architecture has been optimized at bit-level in a way dependent on the known input signal statistics and carrier distributions allowed by the frequency plan.

A Distributed Adaptive Block Matching Algorithm : Dis-Abma

Authors:
F. Vermaut, Y. Deville, B. Macq, X. Marichal, *Dept. of Computing Sciences and Engineering / Dept. of Electrical Engineering (BELGIUM)*

ABSTRACT

Variable block size algorithms for motion estimation, like the Adaptive Block Matching Algorithm (ABMA) [10], have been proposed to better match to "objects in motion" compared to the classical BMA [4, 2]. However, the variable block size grid derivation and the related motion estimation relies on a regularization process which implies heavy iterative and inter-dependent computation. Though the parallelization of the BMA is straightforward, the ABMA needs a deeper analysis before its implementation in a distributed environment : this is the goal of this paper. We first designed a modelization of the motion detection of ABMA. This model can lead to several di_erent distributed versions. A specific distributed model, with one master and several slaves, is then described. An implementation of this model has been realized (in C++, using the PVM platform). Experimentations show a linear speedup with respect to the number of processors.

Prediction Of Decoding Time For MPEG-4 Video

Authors:
M. Mattavelli, S. Brunetton, *Swiss Federal Institute of Technology, (SWITZERLAND)*

ABSTRACT

This paper presents a new technique capable of predicting the processing time of video decoding tasks. Such technique is based on the collection and transmission in the compressed video bit-stream, of a suitable statistic information relative to the decoding process. Simulation results show that very accurate decoding time predictions can be obtained without the need of the actual decoding. These predictions are useful for the fundamental problem of guaranteeing, at the same time, realtime performance and an efficient use of the processing power in processor based video decoders. This new approach gives the possibility of implementing optimal resource allocation strategies and new interaction schemes between the decoder and the real-time OS. Moreover, it is the key for providing a high quality of services when the required decoding resources exceed the available ones. The described method, which is of general application on any processor platform, has been included in the new video MPEG-4 ISO standard.

Real Time Image Rotation Using B-Spline Interpolation On FPGA's Board

Authors:

C.Berthaud, E.Bourennane, M.Paindavoine, C.Milan, *Université de Bourgogne (FRANCE)*

ABSTRACT

There are a lot of methods to interpolate a discrete signal. Among them, there are the SINC method, the Tchebycheff's polynomial, the Lagrange's polynomial that give a good interpolation quality but require a lot of calculation. It's also difficult to implement such algorithm onto FPGAs. So we choose an interpolation method using B-spline functions. This method is very adapted to linear filtering and is easy to implement. In this article, we remind a few definitions about B-spline functions. Then, we apply B-spline interpolation to the image rotation problem in order to implement this algorithm onto FPGA. We describe the way we made this algorithm and we present a calculation of roundoff noise which allows us to determine the coding of the data.

Fully Digital, Highly-Modular Architecture of a Hamming-like Neural Network

Authors:

A. Pavlidis, *University of Athens - NCSR "Demokritos", Inst. of Nuclear Physics (GREECE)*
A.Arapoyianni, *University of Athens (GREECE)*
D.Loukas, *NCSR "Demokritos", Inst. of Nuclear Physics (GREECE)*

ABSTRACT

A new, fully-digital, highly-modular architecture of a Hamming-like neural network is proposed. This architecture aims at a simple and efficient hardware implementation in FPGA or semi-custom VLSI technology that could be the basic cell of image processing and pattern recognition systems. The main advantages of the proposed architecture, compared with other analog or hybrid implementations, is its high modularity, the low cost of implementation, and its robustness to ambient noise, temperature and silicon process variation.

A Fast, Two-Stage, Translational And Warping Motion Compensation Scheme

Authors:
D.B.Bradshaw, N.G.Kingsbury, *University of Cambridge (UK)*

ABSTRACT

This paper describes a motion compensation scheme that combines a hierarchical translational stage (aimed at compensating for large, primarily translational motion) with a fast gradient based warping stage [3] (aimed at compensating for smaller more complicated types of motion). The resulting scheme has low computational complexity and accurately compensates for a wide variety of sequences.

A Fast Full Search Block Matching Algorithm Using Subblocks

Autthors:
M. Brunig, W. Niehsen, *Rheinisch-Westfalische Technische Hochschule (RWTH) (GERMANY)*

ABSTRACT

A fast algorithm for motion estimation is presented. The SAD is used as the matching criterion. It is shown that a lower bound for the error measure exists for every search position. This lower bound can be increased by splitting the examined blocks into subblocks. The bound is exploited to determine which search positions can be skipped. Although the number of search positions and therefore the computational requirements are drastically reduced, the search result is identical to the exhaustive search.

A New Technique For Motion Estimation And Compensation Of The Wavelet Detail Images

Authors:

G. Van der Auwera, A. Munteanu, J. Cornelis, *Vrije Universiteit Brussel (BELGIUM)*
G. Lafruit, *IMEC - Leuven (BELGIUM)*

ABSTRACT

This work proposes a new block based motion estimation and compensation technique applied on the detail images of the wavelet pyramidal decomposition. The algorithm implements two block matching criteria, namely the absolute difference (*AD*) and the absolute sum (*AS*). To assess the coding performance of this method, we have implemented a software simulation of a wavelet video encoder based on our square partitioning coder. Coding results indicate a considerable image quality gain, expressed by PSNR values, of up to 3.4 dB compared to intra wavelet coding for the same bit rate. The quality gain, obtained by using two matching criteria (*AS* and *AD*) instead of one (*AD*), varies between 0.3 and 0.5 dB.

New Closed-Form Solutions To Image Flow Equations For Ego-Motion Applications Considering Variations Of The Focal Distance

Authors:

J. M. Menendez, N. Garcia, L. Salgado, E. Rendon, *Universidad Politecnica de Madrid (SPAIN)*

ABSTRACT

Ego-motion requires the determination of the position and trajectory of the moving vehicle just by analysing the sequence of images acquired along its movement. Here, we present an analytical procedure for the estimation of the three-dimensional kinematic parameters of the vehicle, and the depth map that describes the three-dimensional structure of the world that surrounds the acquisition camera (only monocular gray-level images are considered). Since we do not only consider the possibility of translational and rotational movement in the acquisition camera, but we also allow for variations in the focal distance of the lens (zoom), This proposal clearly extends the one proposed by Waxman et al. [10]. Our proposed solution also provides the way to obtain an estimation of the variation of the focal distance that the camera has applied. The procedure is simple, and relies on the assumption of second order surface patches on the visualised objects.

Motion Segmentation And Tracking Using A Seeded Region Growing Method

Authors:
I. Grinias, G. Tziritas, *Institute of Computer Science - FORTH (GREECE), University of Crete (GREECE)*

ABSTRACT

We describe a user-guided system for the segmentation of an image sequence. A first segmentation which involves two images of the sequence is presented, as well as the tracking of its result during a number of frames. The segmentation algorithm is a Region Growing algorithm. The main segmentation feature is the motion of the objects presented in the image, which is combined with the information obtained by their intensity or colour {if neccessary. Two "post-processing" techniques are proposed to stabilize the object boundaries over time. A sense of layering is embedded in the whole process.

Motion Estimation Based On Affine Moment Invariants

Authors:

G. Tzanetakis, M. Traka, G. Tziritas *Institute of Computer Science - FORTH (GREECE), University of Crete (GREECE)*

ABSTRACT

A method is proposed for parametric motion estimation of an image region. It is assumed that the region considered undergoes an afine transformation, which means that the motion is composed of a translation and a pure afine function of pixel coordinates. The solution of the object correspondence problem is assumed to be known. The estimation of the six motion parameters is based on the moments of the corresponding image regions. Moments up to order three are needed. For the motion computation each region is transformed to a standard position which is defined using afine invariants. The result of motion estimation is checked in the construction of a mosaic image.

A Pseudo 3d Motion Estimator For Moving Object Estimation In Noisy Video Sequences

Authors:
C. L. Topping, J. A. Chambers, *Imperial College (UK)*

ABSTRACT

This paper presents a three-dimensional motion estimator for use in cases where we have noisy video sequences containing one moving object on a stationary background. The motion estimator is to be used as part of an image enhancing pre-processing step. A Parallel Extended Kalman Filter (PEKF) developed by J.B. Burl is at the heart of this motion estimator, together with additional data mapping techniques for its expan-sion to what is referred to as a Pseudo 3D motion estimator. This motion estimator is shown with simulations to have much potential for object motion estimation at low image SNR levels, (< 5dB).

An Object-Based Hierarchical Motion Compensation Technique Using The Greedy Method

Authors:
J. S. Lee, S. U. Lee, *Seoul National University (KOREA)*
R. C. Kim, *Hansung University (KOREA)*

ABSTRACT

In this paper, we propose a rate-constrained object-based hierarchical motion compensation technique, by extending the hierarchical grid interpolation (HGI)[4]. First, we propose an object-based HGI (OHGI) technique, which is an extended version of HGI, to employ for the object-based video coding. In order to achieve the rate-distortion trade-off, inherent in the quadtree segmentation, we employ the greedy algorithm, modified for the OHGI technique, to exploit the rate-distortion trade-off effectively. The performance of the proposed technique is evaluated through the simulation, and it is shown that the proposed technique provides better performance than the conventional object-based motion compensation techniques.

Video Object Manipulation Based On Mosaic Representation For Post-Production Applications

Authors:
H. Nicolas, *IRISA/INRIA (FRANCE)*

ABSTRACT

This paper proposes new methods for the creation of mosaic representations of the scene which enable object-based manipulations of the original video data for post-production applications. For such applications, there is no real-time constraint (off-line image processing) and the whole sequence is fully available to process the original data. When a video sequence is acquired by a moving camera, it contains generally both moving objects and a static background. By computing an object-based segmentation of the scene for each image, the background can be separated from the rest of the scene in order to create the mosaic image. This global information can therefore be used for video editing (for example suppression of an object), or to virtually change the camera point of view, i.e., the camera location and its focal. This kind of automatic tools is of great interest for the post-production world in which, with the existing techniques, long and tedious manual efforts are necessary to achieve such manipulations. In this context, this paper presents new techniques which allow an efficient creation of a mosaic image, the suppression of a moving object from an original image sequence and the artificial modification of the camera point of view and its focal.

Tracking Markers For Human Motion Analysis

Authors:
P. J. Figueroa, N. J. Leitey, R. L. Barros, R. Brenzikofer, *State University of Campinas (BRAZIL)*

ABSTRACT

In this work, we consider the problem of tracking markers in sequences of images for the human movement analysis. The images are captured from a common CCD camera in a non-supervised environment. As we will see, the image processing chain consists mainly of morphological segmentation, prediction, and matching operations.

Acoustic Echo And Noise Control - A Long Lasting Challenge

Authors:
P. Dreiseitel, E. Hansler, H. Puder, *Darmstadt University of Technology (GERMANY)*

ABSTRACT

Hand-free operation of telephones, incorporating echo cancellation and noise reduction, has been discussed for over a decade. This paper presents an overview of the wide range of algorithms which are applicable to echo cancellers and noise reduction. Practical problems associated with implementation and overall system control are also discussed.

Speech Quality Evaluation Of Hands-Free Telephones During Double Talk: New Evaluation Methodologies

Authors:
H. W. Gierlich, F. Kettler, *HEAD acoustics GmbH (GERMANY)*
E. Diedrich, *Deutsche Telekom (GERMANY)*

ABSTRACT

During the past years auditory test procedures -conversational tests, listening tests and double talk tests- have been developed in order to quantify the speech quality of hands-free telephones subjectively. This auditory test results are the basis for instrumental evaluation of hands-free telephones. They allow the creation of test procedures as well as the determination of performance parameters and limits. The following paper describes objective test parameter specifically for the double talk situation which is the most critical one for hands-free telephones and requires the most advanced analysis technique. Using the example of one hands-free telephone, the test methodology is described.

Combined Systems For Noise Reduction And Echo Cancellation

Authors:
C. Beaugeant, P. Scalart, *CNET DIH/DIPS (FRANCE)*

ABSTRACT

The performance of three different combined systems for noise reduction and acoustic echo cancellation is compared in this paper. This comparison is made in the specific context of hands-free radiotelephony in cars, where both noise and acoustic echo highly degrade speech quality. This article shows that the best system is based on a cascaded combination of a conventional echo canceller with a filter reducing noise and residual acoustic echo. Nevertheless, recent novel approaches based on the use of a single Wiener filter are to be efficient enough in the specific context of hands-free sound pick-up in cars.

Combined Residual Echo And Noise Reduction: A Novel Psychoacoustically Motivated Algorithm

Authors:
S. Gustafsson, P. Jax, *Institute of Communication Systems and Data Processing (GERMANY)*

ABSTRACT

In this paper we focus on the problem of acoustic echo cancellation and noise reduction for hands-free telephony devices. A standard echo canceller is combined with a frequency domain post-filter, which applies a novel psychoacoustically motivated weighting rule. The algorithm makes use of the masking threshold of the human auditory system to achieve a perceived reduction of noise and residual echo equal to some pre-defined levels.

In contrast to conventional methods, the proposed one preserves the nature of the original background noise and doesn't introduce any audible artifacts. At the same time it can attain a very high reduction of the residual echo.

Speech Enhancement For Mobile Telephony Based On Non-Uniformly Spaced Frequency Resolution

Authors:

P. Dreiseitel, H. Puder, *Darmstadt University of Technology (GERMANY)*

ABSTRACT

In this paper, we present a speech enhancement method based on spectral subtraction in non-uniformly spaced sub-bands. The main advantage of using different frequency resolutions for the various bands is the perception property of the human ear, which is able to separate low frequencies more precisely than high frequencies. The parameters of the noise reduction algorithm are chosen appropriately to the respective signal to noise ratio, which yields a performance superior to an uniform frequency resolution with the same number of sub-bands. A two-stage cascaded filter-bank is used for the decomposition of the signal.

An Acoustic Echo Canceller With Compensation Of Nonlinearities

Authors:
A. Stenger, R. Rabenstein, *University of Erlangen-Nuremberg (GERMANY)*

ABSTRACT

Traditional adaptation algorithms for acoustic echo cancellers are sufficiently robust to give satisfying performance even in the presence of nonlinear distortions, e.g. those induced by low-cost audio equipment in speech communication products for the consumer market. However, the performance of echo compensation can be improved if special measures are taken to acount for nonlinear behaviour in the electro-acoustic transmission system. To this end, a compensation method for memoryless nonlinearities, as encountered in overdriven amplifiers, is presented. Measurements demonstrate that the echo return loss enhancement can be improved without slowing down convergence. It is thus posssible to compensate the nonlinear effects of cheap audio components with few additional operations in the digital part of the device.

On The Use Of New Adaptive Subband Structures In Acoustic Echo Cancellation

Authors:
M. R. Petraglia, A. Petraglia, *Federal University of Rio de Janeiro (BRASIL)*
R. G. Alves, *Federal University of Rio de Janeiro - INMETRO National Institute of Metrology (BRASIL)*

ABSTRACT

A new family of adaptive structures which employ filter banks or wavelets to decompose the input signal and reduced-order adaptive filters in the subbands is applied to the acoustic echo control problem. Structures with sparse adaptive subfilters and no down-sampling of the subband signals, as well as structures with critical sampling of the subband signals, are investigated. Both types of structures yield exact modeling of FIR systems. Computer simulations are presented to illustrate the convergence behavior of the adaptive subband structures investigated in the paper for acoustic echo cancellation.

Steady-State Solutions Of The Extended LMS Algorithm For Stereophonic Acoustic Echo Cancellation With Leakage Or Signal Conditioning

Authors:
T. Hoya, J. A. Chambers, N. Forsyth, P.A.Naylor, *Imperial College of Science (UK)*

ABSTRACT

Acoustic echo cancellers are widely employed in teleconferencing systems to reduce the undesired echos resulting from coupling between loudspeaker and microphone. The single-channel case has been widely studied. Of particular recent interest is the stereophonic case which is not difficult to solve due to the strongly correlated two-channel inputs. In this paper we compare the steady-state solutions of the Leaky eXtended LMS (XLMS) algorithm with XLMS having inputs conditioned by additional zero-memory non-linearities. Modification of the correlation matrix of the two channel-inputs is analysed. We also describe a new configuration for the zero-memory non-linearities which does not impact upon sound quality whilst maintaining improved algorithm convergence properties. Simulation results to support the analyses of these two different de-correlation methods are also included, which suggest that for deterministic parameter settings the performance of the Leaky XLMS algorithm is superior to the case where a half-wave rectifying non-linearity is used. Moreover, simulation results where time variations in the transmission room are considered in order to represent more realistic situations are given, which, on the other hand, suggest that the performance of the XLMS with a signal conditioning method using an Half-Wave Rectifier (HWR) is superior to that of the Leaky XLMS algorithm.

DSP Implementation And Performance Evaluation Of A Stereo Echo Canceler With Pre-Processing

Authors:
Y. Joncour, A. Sugiyama, A. Hirano, *NEC Corporation (JAPAN)*

ABSTRACT

This paper presents implementation and performance evaluation of a stereo echo canceler with pre-processing. A two-tap time-varying filter located in one of the two channels periodically delays the input signal by one sample. By this pre-processing, the correct echo-path identification is achieved. This stereo echo canceler is implemented by four 32-bit floating-point digital signal processors (ADSP-21062). Experimental results show that the implemented echo canceler can reduce the echo by approximately 25 dB for a white Gaussian signal and by 23 dB on average for a speech signal. The ERLE is not degraded by talker movements in the remote room. The Mean Opinion Score for the implemented echo canceler is 0.55-point and 0.48-point higher than that for the echo canceler based on linear combination for round-trip delays of 100 ms and 600 ms, respectively.

A General Maximum Likelihood Classifier For Modulation Classification

Authors:
C. Le Martret, *Centre d'ÉLectronique de l'ARmement (FRANCE)*
D. Boiteau, *Centre d'Études de Systemes et de Techniques Avancees (FRANCE)*

ABSTRACT

This paper deals with maximum likelihood (ML) classification of digital communication signals. We first propose a new approximation of the average likelihood function. Then we introduce the General Maximum Likelihood Classifier (GMLC) based on this approximation which can be applied to a wide range of classification problem involving finite mean power signals. Derivation of this classifier equations are given in the case of linear modulations with an application to the M PSK / M' PSK problem. We show that the new tests are a generalization of the previous ones using ML approach, and don't need any restriction on the baseband pulse. Moreover, the GMLC provides a theoretical foundation for many empirical classification systems including those of that exploit cyclostationarity property of digital modulated signals.

Approximation Of á-Stable Probability Densities Using Finite Gaussian Mixtures

Authors:

E. E. Kuruoglu, W. J. Fitzgerald, *University of Cambridge (UK)*
C. Molina, *Anglia Polytechnic University (UK)*

ABSTRACT

In this paper, we introduce a new analytical model for the á-stable probability density function (p.d.f.). The new model is based on a corollary of the mixing theorem for symmetric á-stable (SáS) random variables (r.v.) [1] which states that a SáS r.v. can be expressed as the product of a Gaussian r.v. and a positive-stable r.v. We also extend this model to provide an analytical approximation for a subclass of multivariate á-stable p.d.f.s, namely the sub-Gaussian á-stable p.d.f.s. Simulation results indicate the success of our technique. The new analytical representation opens path to the application of maximum likelihood and Bayesian techniques for problems involving á-stable random variables. The paper is concluded with the examples of possible application areas.

Wavelet Thresholding
For A Wide Class Of Noise Distributions

Authors:
D. Leporini, J.-C. Pesqaet, *Laboratoire des Signaux et Systèmes (FRANCE)*

ABSTRACT

Wavelet thresholding techniques are becoming popular in the signal processing community for denoising applications. Near-minimax properties were in particular established for simple threshold estimates over wide classes of regular functions. In this paper, we establish close connections between wavelet thresholding techniques and MAP estimation using exponential power prior distributions for a wide class of noise distributions, including heavy-tailed noises. We subsequently prove that a great variety of estimators are derived from a MAP criterion. A simulation example is presented to substantiate the proposed approach.

Information Criteria Based Edge Detection

Authors:
F. Jouzel, *University of Rouen (FRANCE)*
C. Olivier, *University of Poitiers (FRANCE)*
A. El Matouat, *University of Le Havre (FRANCE)*

ABSTRACT

We propose an algorithm to detect ruptures in a signal by means of information criteria along with its application to edge detection on grey-level images. Information criteria are made of two terms: a likelihood term and an original penalization factor. They enable us to find in an optimal way autoregressive models and their number to model the signal. The model changes represent the edges of the objects in the image.

Estimating The Predictability And The Linearity Of A Process By Kernels

Authors:

A. Poncet, G. S. Moschytz, *Swiss Federal Institute of Technology ETH (SWITZERLAND)*

ABSTRACT

On the basis of discrete-time process data for system identification (or time-series prediction), it would be very desirable to determine a priori how unpredictable and how nonlinear a process is. Showing how this can be done by adopting the framework of statistical estimation theory is the purpose of this paper. Inferring the predictability of a process is important for estimating in advance which prediction performance can be realistically expected from a model. The "degree" of nonlinearity of the underlying process should also be assessed before the design of a suitable model is undertaken. If

the data do not reveal a markedly nonlinear character, the irrelevance of nonlinear models will be noticed in advance, thereby saving time which would otherwise be lost on an unnecessary search.

Multi-User Channel Estimation Exploiting Pulse Shaping Information

Authors:

E. Lindskog, J. Strandell, *Uppsala University (SWEEDEN)*

ABSTRACT

A method for joint estimation of wireless communication channels to multiple users utilizing pulse shaping knowledge is presented. The pulse shaping in the transmitter and the receiver is incorporated into the channel estimation by approximating it with a set of pulse shaping functions. In joint multi-user channel estimation the number of parameters to be estimated grows linearly with the number of users while the number of equations remains constant. Utilizing the pulse shaping information is therefore especially useful since this economizes on the number of parameters to be estimated per user. This is illustrated with a multi-user channel estimation example.

A Subspace Fitting-Like Method For Almost Low Rank Models

Authors:

M. Bengtsson, *Royal Institute of Technology (SWEDEN)*

ABSTRACT

Subspace fitting methods have grown popular for parameter estimation in many different application, for example sensor array signal processing, blind channel iden-tification and identification of linear state space systems. Here we show that similar procedures can be used even for data models where the noise free signal gives a full rank contribution to the covariance matrix. A general weighting is introduced and the optimal weight matrix is given together with the resulting asymptotic covariance of the parameter estimates. The method works well when the number of dominating eigenvalues still is fairly small. As an example, we study estimation of direction and spread angle of a source subject to local scattering, using a uniform linear array of sensors. As the algorithm is computationally expensive, the results are not primarily intended for practical implementations, rather they show the theoretical limit for any estimation procedure that uses a low rank approximation of the covariance matrix.

Map Based Schemes For Detection Of Abrupt Changes For Fading Channels

Authors:
C. Carlemalm, F. Gustafsson, *Linkoping University (SWEDEN)*

ABSTRACT

An algorithm for discrimination and detection of the two phenomena double talk and abrupt changes in the echo path is proposed for fading channels. Being able to discriminate and detect these two phenomena is crucial since the echo canceler must react differently. The suggested detection scheme is based on a sequential detection approach. The communication channel is modeled as a randomly time-varying linear system. An autoregressive model is used to describe the time evolution of the channel taps. The channel parameters are identified using a Kalman filter coupled with a recursive least squares algorithm, and, based on model assumptions, the maximum a posteriori probabilities corresponding to double talk and abrupt echo path changes are calculated. The proposed scheme is verified experimentally by way of computer simulations.

Separate Temporal And Spatial Parametric Channel Estimation

Authors:
J. Strandell, E. Lindskog, *University of Uppsala (SWEDEN)*

ABSTRACT

A temporally parameterized channel estimate can be improved by projection onto a spatially parameterized subset. The projection is performed in a spectrum norm sense, and is investigated by means of simulations. By utilizing the channel estimation method in a multidimensional MLSE detector, significant improvements in the bit-error rates can be obtained, compared to using the initial estimate of the channel.

Parameter Estimation In Partitioned Nonlinear Stochastic Models

Authors:
O. Markusson, H. Hjalmarsson, *Royal Institute of Technology (KTH) (SWEDEN)*

ABSTRACT

The application of maximum likelihood estimation and prediction error methods on dynamical systems require that it is possible to compute the innovations of the systems model which more or less implies that it must be possible to invert the system model. For nonlinear stochastic models this can be very difficult. In this contribution it is shown how this can be done very efficiently for a very rich class of non-linear models by way of exact linearization. The method is illustrated on two non-trivial examples.

Source Localization In Shallow Water
In The Presence Of Sensor Depth Uncertainty

Authors:
A. Jakoby, J. Goldberg, H Messer, *Tel-Aviv University (ISRAEL)*

ABSTRACT

This paper studies passive source localization performance in shallow water using a vertical array whose sensor depths are unknown. The performance degradation with respect to the case of known sensor depths is studied via the Cramer-Rao Bound for a single far field narrow band source. Examination of the bound indicates that there is no inherent singularity in the Fisher Information Matrix due to uncertainty in the sensor depths (as opposed to the case of localization in free space). Numerical examples show that the performance degradation in source localization due to the need to estimate the sensor depths in a typical scenario is approximately 3-5dB.

Reconstructing Missing Regions In Colour Images Using Multichannel Median Models

Authors:

S. Armstrong, A. C. Kokaram, P. J.W. Rayner, *Cambridge University (UK)*

ABSTRACT

This paper presents a method for reconstructing missing regions in colour images. A multichannel median model is proposed as the underlying image model and a statist-ical framework is employed to generate sampled realisations of the missing data. The nature of the model leads to a posterior expression for the missing data that does not involve an easy to manipulate multivariate probability distribution. Therefore, the problem of sampling is solved using a numerical approach. Results are included which show that this approach leads to excellent reconstructions.

Unsharp Masking-Based Approach For Color Image Processing

Authors:

F. A. Cheikh, L Khriji, M. Gabbouj, *Tampere University of Technology (FINLAND)*

ABSTRACT

In this paper, we present an unsharp masking-based approach for noise smoothing and edge enhancing. And apply it to color image processing. The proposed structure is similar to the conventional unsharp masking structure, however, a nonlinear function is added to control the behavior of the operator. The proposed scheme enhances the true details, limits the overshoot near sharp edges and attenuates the noise in at areas. Moreover the use of the control function eliminates the need for the subjective coefficient ë used in the conventional unsharp masking technique. Simulations show that the processed image presents sharp edges which makes it more pleasant to the hu-man eye. Moreover, the amount of noise in the image is clearly reduced.

The Role Of Gamma Correction In Colour Image Processing

Authors:

W. Kubinger, M. Vincze, M. Ayromlou, *Vienna University of Technoloy (AUSTRIA)*

ABSTRACT

In this paper we analyse the in uence of gamma correction of digital colour cameras on computer vision algorithms. These nonlinear signals R′, G′ and B′ yield a non-linear shift in colour space models and can cause algorithms to fail. Experiments of colour segmentation and tracking confirm the claims. We suggest to use linear light signals R, G, and B for calculating most colour space models instead of the nonlinear signals R′, G′and B′. To obtain linear light signals either the camera has to be configured linear using the gamma switch (if available) or by removing the nonlinearity analytically using the inverse operation (gamma re-correction).

Quality Limits In Color Printing With More Than Three Primary Colors

Authors:
W. Praefcke, *Aachen University of Technology (GERMANY)*

ABSTRACT

A widely used type of a mathematical color printing model, the Neugebauer model, is investigated. In a three color printer the produced colors depend on the spectral characteristics of the three primary inks as well as on the driving parameters, which are the CMY values. Using the so-called Neugebauer equations it is possible to obtain the re ectance spectrum and finally the color, depending on the incident illumination. At first a strategy is presented to calculate optimal spectral re ectances for the three primaries together with the optimal driving parameters CMY. Afterwards the three color Neugebauer printing model is extended to an arbitrary number of primary inks and evaluate the achievable performance gain for 4 and 5 inks.

Image Halftoning Using Optimized Dot Diffusion

Authors:

M. Mese, P.P. Vaidyanathan, *California Institute of Technology (USA)*

ABSTRACT

The dot diffusion method for digital halftoning has the advantage of parallelism unlike the error diffusion method. However, image quality offered by error diffusion is still regarded as superior to other known methods. In this paper we show how the dot diffusion method can be improved by optimization of the so-called class matrix. By taking the human visual characteristics into account we show that such opti-mization consistently results in images comparable to error diffusion, without sacrificing the parallelism.

A Technique For Image Quality Assessment Based On A Human Visual System Model

Authors:
W. Osberger, N. Bergmann, *Queensland University of Technology, (AUSTRALIA)*
A. Maeder, *University of Ballarat (AUSTRALIA)*

ABSTRACT

This paper describes an objective quality measure for compressed pictures based on a model of the human visual system (HVS). The structure of natural images and of the coding errors introduced by compression schemes is taken into consideration when determining appropriate HVS parameters. Higher level perceptual factors are also taken into account through the use of Importance Maps. This allows a weighting of the visible errors, since errors occurring in regions of high visual importance have a greater effect on our overall perception of picture quality than those in lower importance regions. The results show the improved predictions achieved by this technique, in particular for images coded with spatially varying quality.

Subjective Measure Of Edge Degradation For Vector Quantized Color Images

Authors:

C. Charrier, H. Cherifi, *Universite Jean Monnet (FRANCE)*

ABSTRACT

Edges are of fundamental importance in the analysis of images, and of course in the field of image quality. To incorporate the edge information as coded by the Human Visual System (HVS), we have developped a classification strategy to take into account edges in the codebook. Psychophysical experiments have been performed to adjust the optimal amount of edge information.

Towards A Visual Quality Metric For Digital Video

Authors:
A. B. Watson, *NASA Ames Research Center (USA)*

ABSTRACT

The advent of widespread distribution of digital video creates a need for automated methods for evaluating the visual quality of digital video. In previous work, we have developed visual quality metrics for evaluating, controlling, and optimizing the quality of compressed still images 1, 2, 3, 4 . These metrics incorporate simplified models of human visual sensitivity to spatial and chromatic signals. Here I describe a new video quality metric that is an extension of these still image metrics into the time domain. Like the still image metrics, it is based on the Discrete Cosine Transform. An effort has been made to minimize the amount of memory and computation required by the metric, in order that might be applied in the widest range of applications. To calibrate the basic sensitivity of this metric to spatial and temporal signals we have made measurements of visual thresholds for temporally varying samples of DCT quantization noise.

Psychovisual Measurement And Distortion Metrics For Image Sequences

Authors:
E. M. Yeh, A. C. Kokaram, N. G. Kingsbury, *MIT (USA) - University of Cambridge (UK)*

ABSTRACT

A number of perceptual distortion measures have been developed for both still images and image sequences. Many of these measures, however, derive from visual experiments using stimuli quite unlike digital image coding artifacts. We present the results from six visual experiments characterizing human visual sensitivity to edge-like blocking artifacts under a number of spatiotemporal conditions, in an attempt to establish a firm experimental basis for a tuned perceptual distortion measure for blocking artifacts in image sequences.

Monitoring The Quality Of MPEG Coded Video

Authors:

K. T. Tan and M. Ghanbari, *University of Essex, (UK)*

ABSTRACT

A novel two-stage video distortion meter exploring the psychological aspect of human beings is proposed. The first stage is a distortion weighting stage, where picture artifacts are weighted according to their visibility. The second stage, denoted the Cognitive Emulator, simulates the psychological effect of time-varying picture quality on the judgement process of human observers. Experimental results show that with the cognitive emulator, one can predict the result that would be obtained using otherwise the SSCQE methodology for video sequences.

A Human Vision System Model For Objective Image Fidelity And Target Detectability Measurements

Authors:
J. Lubin, Ph.D., *Sarnoff Corporation (USA)*

ABSTRACT

An algorithm is described that accurately predicts human perceptibility of differences between two image sequences, across a broad range of signal types (e.g., consumer video, mammography), difference types (e.g., DCT quantization noise, presence/absence of target of interest), and tasks (e.g., subjective quality rating, target detection). The algorithm, Sarnoff's Just-Noticeable Difference (JND) Vision Model, is based on known physiology and psychophysics of vision, and is calibrated using simple psychophysical tasks of low contrast sine grating detection and sine grating contrast discrimination. Model outputs are derived from psychophysical units of JNDs, in which one JND is defined as a difference between two signals that is at the threshold of detectability. Model results are presented showing excellent predictive performance across a broad range of conditions, as well as some results in which attentional effects demonstrate the need for additional model development.

Modeling 3d Textured Objects
By Fusion Of Multiple Views

Authors:

T. Jost, C. Schütz, H. Hügli, *University of Neuchâtel (SWITZERLAND)*

ABSTRACT

For modeling 3D objects, the object geometry is often not sufficient. Better appearance can be obtained by texture mapping which efficiently combines good appearance resolution with small geometric complexity. This paper presents a way to create such texture mapped 3D models of free-form objects, by integrating textured views obtained from range images. The three main steps consist of the following: first, texture-mapped views of the object are obtained from a 3D scanner. Then, the different views are registered, considering geometric and color information of overlapping grids. Finally, meshes can be fused by using a new overlap-removing method that keeps most of the existing triangulation and texture mapping unchanged.

Scene Analysis Using Fusion Of Range And Color Data

Authors:
P. Pujas, I.U.T. - *Université Montpellier II (FRANCE)*
M.-J. Aldon, *LIRMM - UMR C55060 - CNRS / Université Montpellier II (FRANCE)*

ABSTRACT

In this paper, we present a segmentation algorithm based on 3D-color (3DC) images acquired with a calibrated heterogeneous sensor composed of a video camera and a laser range finder. The first part of the paper is dedicated to the presentation of the multi-sensor system and the specific calibration process which allows it to register color and range data in a unique frame. The next part describes the region growing algorithm used for the 3DC image segmentation. The color criterion used to select homogeneous pixels is defined in a perceptual representation space. A 3D criterion is used to select voxels which belong to the same planar face in the observed scene. We show how these two criterion may be combined to segment the 3DC image into planar faces which are characterized by homogeneous colors. Results obtained with real dense 3DC images illustrate the performance of this method.

Tracking Camera Calibration In Multi-Camera Sequences Through Automatic Feature Detection And Matching

Authors:
F. Pedersini, A. Sarti, S. Tubaro, *Politecnico di Milano (ITALY)*

ABSTRACT

When using multi-camera acquisition systems, in order to extract accurate 3D information on the viewed scene, the geometrical, optical and electric characteristics of the camera system must be known with good accuracy. If the acquisition of the sequence of multi-views requires a certain time to be completed, mechanical shocks, vibrations or thermal effects on the cameras and their support can cause a drift of the initial camera parameters. In this paper we propose a technique that, during the acquisition session, is able to track the camera parameters and, whenever possible, to correct them according to the occurred modifications. This technique does not need any a-priori knowledge or test objects to be placed in the scene, but exploits luminance features already present in the scene, such as luminance corners and spots.

Range Image Registration For Free-Form Surfaces Using Curvature Information As Invariance

Authors:
M. Rieder, N. Guil, R. Dillmann, *University of Karlsruhe, (GERMANY)*

ABSTRACT

A curvature based technique for fast range image registration is described in this work. With this technique the rotation and the displacement of a surface with respect to another one is calculated. The proposed algorithm uses Gaussian curvature information from each image point to extract a number of feature points. Two bounds are applied to the curvature values. Points with these bound curvature value are classified into two classes of feature points. In order to calculate the rotation angle between the two range images, the feature points are paired. That means the angle and the distance of the point pairs have to be transformed into a general view coordinate system. In the orientation matching procedure, we compare each pairing in the first image with each possible in the second one. This process is independent of the translation calculation which is calculated in a sequencing step. This pipelining principle speeds up the registration process if correspondences are calculated constantly.

Visualization Of Video-Conference Image Sequences Using VRML 2.0

Authors:
I. Kompatsiaris, M. G. Strintzis, *Aristotle University of Thessaloniki (GREECE)*

ABSTRACT

In this paper a procedure for visualisation of video-conference image sequences using Virtual Reality Modeling Language (VRML) 2.0 is described. First, image sequence analysis is performed in order to estimate the shape and motion parameters of the person talking in front of the camera. For this purpose, we propose the K-Means with connectivity constraint algorithm as a general segmentation algorithm combining information of various types such as colour and motion. The algorithm is applied “hierarchically” in the image sequence and it is first used to separate the background from the foreground object and then to further segment the foreground object into the head and shoulders regions. Based on the above information, the 3D shape parameters are estimated for each sequence and a 3D model is automatically adapted. The rigid 3D motion is estimated next for each sub-object. Finally a VRML file is created containing all the above estimated information and can be viewed using any VRML 2.0 compliant browser.

Object Segmentation In 3-D Images Based On Alpha-Trimmed Mean Radial Basis Function Network

Authors:

A. G. Bors , I. Pitas, *University of Thessaloniki (GREECE)*

ABSTRACT

This paper presents a new approach for 3-D object segmentation. Objects from a stack of images are represented as overlapping ellipsoids. Graylevel statistics and shape features are simultaneously employed for object modeling in an unsupervised approach. The extension of the Hough Transform in the 3-D space is used for finding the ellipsoid centers. Each ellipsoid is modeled by a Radial Basis Function (RBF) and the entire structure is represented by means of an RBF network. The proposed algorithm is applied for blood vessel segmentation from tooth pulp in a stack of microscopy images.

Classification Of Archaeological Fragments Using A Description Language

Authors:

R. Sablatnig, C. Menard, W. Kropatsch, *Vienna University of Technology (AUSTRIA)*

ABSTRACT

This paper presents an application of 3d-reconstruction and graph theory in the field of archaeology. The classification and reconstruction of ancient pots and vessels out of fragments (so-called sherds) is an important aspect of archaeological research work. Up to now this is a time consuming, inaccurate, and subjective task which leads to tons of unclassified fragments in archives. Computer aided classification could help to get a better understanding of ancient cultures, since all data of an excavation would be accessible to the public, not only selected parts as it is now. We propose a bottom-up strategy to classify fragments. The profile section (which is a section a the fragment in the direction of the rotational axis) is segmented into its primitives (with certain properties like length) and relations among this primitives (like position and curvature of connecting points). These primitives and the relations form a description language, different profiles have different descriptions.

On The Application Of Light Field Reconstruction For Statistical Object Recognition

Authors:

B. Heigl, J. Denzler, H. Niemann, *Universitat Erlangen (GERMANY)*

ABSTRACT

In this paper we apply light field reconstruction and rendering of object views to the problem of automatic generation of training material for a statistical object recognition system. The advantages of using a light field instead of real images are shown. We evaluate with respect to the error rate of the classifier, whether the reconstructed light field can be applied to the training step. We also show how the recognition rate of the classifier trained by the light field depends on its resolution.

Virtual Environment Generation By CAD-BASED MethodologyFor Underwater Vehicle Navigation

Authors:

L. De Floriani, *Universita di Genova (ITALY)*

V. Murino, G. G. Pieroni, *Universita di Udine (ITALY)*

E. Puppo, *Istituto per la Matematica Applicata - Consiglio Nazionale delle Ricerche (ITALY)*

ABSTRACT

We describe a recognition framework to generate a virtual environment through CAD-based vision techniques from optical data. Descriptions of objects of the environment in terms of aspect graphs, and suitable recognition strategies for them are compiled off-line. A relational graph of image features is obtained on-line by processing optical data, and matching occurs between such a graph, and descriptions of objects in the framed scene. Multiresolution techniques are used in order to adapt recognition strategies to the distance and relevance of objects within the field of view

Combining Intensity And Stereo Data To Improve Satellite Urban Scenes Modeling

Authors:

N. Paparoditis, *ESGT (FRANCE)*
M. Cord, *ETIS / URA CNRS 2235 (FRANCE)*

ABSTRACT

A global scheme combining intensity and stereo data to improve urban landscapes modeling from aerial or satellite stereopairs is presented. Both intensity and stereo data are used for building detection, recognition and scene reconstruction. Grey-level intensity data features are used to improve the quality of depth maps stereo processing

and depth maps features are used to improve monocular analysis and especially perceptual grouping for building shapes reconstruction.

A New FPGA Architecture For Image Processing : CYCLOP

Authors:
P. Guermeur, *ENSTA (FRANCE)*

ABSTRACT

This paper presents CYCLOP, a multi-FPGA board architecture designed to allow real-time implementation of low-level image processing algorithms. A set of such identical boards is to be integrated into a workstation. It offers the users different algorithm implementation possibilities, depending on the trade-off speed area desired: - for applications requiring high speed computation rates, the user will have to consider the possibility to partition his algorithm into several boards (pipeline processing);
- another method, illustrated in this paper, will consist in iterating the different steps of the algorithm on a single board. In the latter case, we implement a two-dimensional filtering environment. It is built on a one-dimensional second order recursive filtering unit, whose coefficients can be loaded via the VME bus. We illustrate the filtering performances by an immediate application to the Deriche edge detection algorithm.

Multi FPGA Processor For Gray-Scale Morphology

Authors:
M. Akil, S. Zahirazami, *Grupe ESIEE*, *(FRANCE)*

ABSTRACT

Local operations in image processing are often used, namely during the preprocessing step. On one hand, their implementation is expensive, on the other hand, they become efficient when they use a wide area neighborhood.

In this paper we propose a general method for synthesis of linear and non linear filters (mathematical morphology operators). The main interest of this method is that it can be applied to various types of filters: separable or not, factorizable or not and for any size of the filter kernel (convolution) or the structuring element (mathematical morphology). From this method, one can get an architecture in a straightforward way. We propose a reconfigurable architecture, based upon reprogrammable circuits (FPGA: Field Programmable Gate Array). This multi FPGA architecture allows a real time implementation of any r x c size kernel (processing at video rate).

Radon Transform
For Internal Wave Detection And Orientation

Authors:
J. A. Rodenas, N. Mandelert, R. Garello, *Ecole Nationale Superieure des Telecommunications de Bretagne (FRANCE)*

ABSTRACT

This paper studies the feasibility of using the Radon transform to detect and orientate internal wave features from SAR images since it seems to be well suited for extracting quasi-linear features even at low signal-to-noise levels. The approach is based on the localized Radon transform where the intensity integration is performed over short line segments rather than across the entire image. The results of this testing demonstrate the algorithm's robustness in the presence of speckle noise, as well as its ability to detect, localize, and orientate internal wave features that are significantly shorter than the image dimensions or that display some curvature.

Application Of The Momentary Fourier Transform To SPECAN SAR Processing

Authors:
S. Albrecht, I. Cumming, *The University of British Columbia (CANADA)*

ABSTRACT

The momentary Fourier transform computes the DFT of a discrete-time sequence for every new sample in an efficient recursive form. We describe the arithmetic requirements of the MFT, and give examples where it is more efficient than the FFT. We describe how DFTs are used in the SPECAN method of Synthetic Aperture Radar (SAR) processing, which includes overlapped inverse DFTs. We apply the MFT to SPECAN processing and show what advantages and disadvantages it has compared to the FFT.

Synthetic Aperture Radar Interferometry Using Ground Slope Vector To Phase Unwrapping

Authors:
X. Dupuis, P. Mathieu, M. Barlaud, *Université de Nice-Sophia Antipolis (FRANCE)*

ABSTRACT

This paper deals with 2-dimentional synthetic aperture radar interferometry. Because of high sensitivity to noise, we propose a two-step phase unwrapping approach. The purpose of preprocessing is to organize the phase by an adaptive filter to improve the estimation of the ground slope vector. The main process smoothes the phase, using the ground slope components, and unwraps the signal. We present numerical and experimental results for synthetic data.

Automatic Pitch Marking For Speech Transformations Via TD-PSOLA

Authors:
Y. Laprie, V. Colotte, *LORIA (FRANCE)*

ABSTRACT

This paper describes an automatic pitch marking method which can be used in the context of modifying speech signals with TD-PSOLA (Time Domain Pitch Synchronous Overlap-Add). Unlike other approaches where glottal closure instants are searched directly in the speech signal, our approach exploits results of a pitch extraction algorithm. The principle is to optimize the propagation of pitch marks from one pitch period to the following by means of dynamic programming. The first step consists of extracting extrema on regularly spaced segments, the size of which is the smallest pitch period in the signal under investigation. Then, an optimal subset of extrema, which represent pitch marks, is found by a dynamic programming algorithm which derives from a smoothing algorithm proposed by Ney. Pitch marks obtained through this post-synchronization algorithm allow very good speech signal transformations. Furthermore, this algorithm has the advantage that it is independent from the pitch determination step which allows it to be easily combined with any algorithm for pitch extraction.

The Method Of Pitch Frequency Detection On The Base Of Tuning To Its Harmonics

Authors:

V. Sercov, A. Petrovsky, *University of Technology Bialystok, (POLAND)*

ABSTRACT

The major part of speech compression algorithms from the first simple formant vocoders to widely used at presence LP- and MBE-vocoders in spite of different principals of coding use by the analysis-synthesis procedures information about the frequency of pitch [1-3]. There is a number of methods for its detection [4-6]: on the base of auto-correlation processing, cepstrum analysis, after linear prediction remaining signal. The methods provide good results when analysing parts of speech with pronounced vocalisation by insignificant noise level. When there is a strong noise component in the input signal the number of errors and accuracy of pitch frequency detection is greatly increased [7]. To eliminate the errors they use generally either method of linear smoothing of pitch contour that additionally decrease the accuracy.

The paper presents the method and algorithm of pitch frequency detection which doesn't provide gross errors and has the high accuracy.

Acoustic Measure Of Noise Energy In Vocal Folds Operated Patients

Authors:
C. Manfredi, M. D'Aniello, P. Bruscaglioni, *University of Florence (ITALY)*

ABSTRACT

Voiced sounds are the result of a periodic excitation of the vocal tract due to the vocal folds vibration. They are characterised by the fundamental frequency of the phonation, named pitch, which is the rate of vibration of the folds. In pathologic voices, pitch variations are indicative of the patient status, hence robust pitch estimation methods are required in order to track such variations and to quantify the degree of voice deterioration. This paper compares some pitch estimation methods, with modifications suitable for the present application. A measure of the signal "noise" level is adopted which allows tracking its variations with time. The methods are tested on simulated signals and then applied to real signals coming both from healthy and pathologic voices.

Analysis Of Pitch-Synchronous Modulation Effects By Using Analytic Filters

Authors:

U. K. Laine, *Helsinki University of Technology (FINLAND)*

ABSTRACT

Gabor type of analytic filters are used together with Teager-Kaiser energy operator based DESA-1 and two other algorithms to study pitch-synchronous modulation effects in speech. The methods are tested with synthetic and natural speech. Common problems and limitations of these methods are discussed.

Perceptual Evidence For F0 Variability Constraints Of Phrase-Initial Accents In Greek

Authors:

C. Malliopoulos, *National Technical University of Athens (GREECE)*
G. Carayannis, *Institute for Language and Speech Processing (GREECE)*

ABSTRACT

The pitch or *F0* contour is a carrier of multiple information, from segmental to intonational (i.e. grammatical and syntactic to semantic to pragmatic). In this paper it is investigated the relation of certain configurations of *F0* to the structural organization of an utterance. First some analytical observations are made. Then, in order to verify our analytical observations we conducted an experiment to test their perceptual relevance. Specifically we examine *F0* accent configurations on phrase initial positions. The two experiments conducted in order to determine whether the leading *F0* peak in an intonation phrase is free to vary unconditionally or its variability is constrained by tonal configurations of adjacent phrases. The results showed that the perceived prominence of *F0* over phrase-initial accents is linguistically controlled by sentence structure.

The Impact Of Relative Average Duration Of Vowels In Greek On The Perception Of A Constant Synthetic Speech Rate

Authors:

S.-E. F.Fotinea, *National Technical University of Athens (GRRECE)*
M. A.Vlahakis, G. V.Carayannis, *Institute for Language and Speech Processing (GRRECE)*

ABSTRACT

The relative average duration of vowels were found to have a significant impact on the perception of a given speech rate (tempo) of a Text to Speech Synthesis System (TTS) developed for the Greek language. Even though the duration ratios vary slightly according to segmental context and word prominence, their modification beyond a certain level of tolerance, may result in the perception of a mismatch for the tempo of the modified word in comparison with the tempo selected for the rest of the phrase.

Text-To-Speech Synthesis In Slovenian Language

Authors:
T. Šef, A. Dobnikar, M. Gams, *Jozef Stefan Institute (SLOVENIA)*

ABSTRACT

This paper presents a text-to-speech (TTS) system, capable of synthesis of continuous Slovenian speech. The system is based on the concatenation of basic speech units, diphones, using TD-PSOLA technique improved with a variable length linear interpolation process. Input text is processed by a series of modules which are described in detail. A special attention is given to modeling the F0 contour, mainly based on the so-called superpositional approach. This system is experimentally used in an employment agent EMA that provides employment information through the Internet.

Comparison
Of Two Different Text-To-Speech Alignment Systems: Speech Synthesis Based Vs. Hybric HMM/ANN

Authors:

O. Deroo, F. Malfrere, T. Dutoit, *Faculté Polytechnique de Mons (BELGIUM)*

ABSTRACT

In this paper we compare two different methods for phonetically labeling a speech database. The first approach is based on the alignment of the speech signal on a high quality synthetic speech pattern, and the second one uses a hybrid HMM/ANN system. Both systems have been evaluated on French read utterances from a speaker never seen in the training stage of the HMM/ANN system and manually segmented. This study outlines the advantages and drawbacks of both methods. The high quality speech synthetic system has the great advantage that no training stage is needed, while the classical HMM/ANN system easily allows multiple phonetic transcriptions. We deduce a method for the automatic constitution of phonetically labeled speech databases based on using the synthetic speech segmentation tool to bootstrap the training process of our hybrid HMM/ANN system. The importance of such segmentation tools will be a key point for the development of improved speech synthesis and recognition systems.

A Fuzzy Approach To Text-To-Speech Synthesis

Authors:
E. Mumolo, W. Costanzo, *Elettronica ed Informatica Universita' di Trieste via Valerio (ITALY)*

ABSTRACT

In this paper we describe a fuzzy approach for the synthesis of a speech waveform from a phonetic description in Italian language. The system is based upon a set of fuzzy rules which linguistically describe the transitions between phonemes. The fuzzy system has several interesting properties, such as an easy management of the set of rules and the possibility to continuously improve the system by adding more knowledge to it using linguistic descriptions. Some experimental results are included, and future possible developments are outlined.

A Method to Choose An Appropriate Concatenating Position For Automatically Generated Synthesis Units

Authors:
H. J. Park, *Bumil Information & Communication Co. Ltd (KOREA)*
S. H. Kim, M. S. Han, *Electronics and Telecommunications Research Institute (KOREA)*
J. H. Chung, *Inha University (KOREA)*

ABSTRACT

To make synthesized speech sounds more natural, one would prefer using a larger size of speech data base. However, when we adopt manual segmentation method in making a large size data base, the procedure would be very time-consuming and the constructed data base would be in consistent. Consequently, automatic segmentation using speech recognition system has been used for making a large size of speech synthesis data base. However, when the automatic segmentation were applied to make synthesis speech, significant concatenation distortion happens due to phoneme boundary error. The purpose of our study is to choose an appropriate concatenating position of automatically generated synthesis units which may have errors on their boundaries. We have performed MOS (Mean Opinion Score) tests and analyzed the shape of spectrograms to evaluate our proposed algorithm. Our test results have shown that the quality of concatenated speeches with proposed algorithm is superior to that of concatenated ones generated by automatic segmentation only.

A High-Performance Vowel Spotting System Based On A Multi-Stage Architecture

Authors:

J. Sirigos, N. Fakotakis and G. Kokkinakis, *University of Patras (GREECE)*

ABSTRACT

In this paper we present a novel multi-level vowel detection system with improved accuracy. Multi layer perceptrons (MLP), Discrete Hidden Markov Models (DSHMM) and heuristic rules are combined in three different levels to reduce the probability of false acceptance and rejection of vowel sounds. The TIMIT database was used to train and test this system. The rules are variable and are automatically customized by statistics extracted from the database, which concern the duration, the energy and the distance between vowels. The proposed method can easily be extended to languages other than English as long as a proper database exists for training the system. Its accuracy was measured to 99.22% using all the test data sets of the TIMIT database. Thus, the proposed vowel detection process can be reliably used for speech processing applications (speaker or speech recognition) where accurate vowel spotting algorithms are necessary.

On A Time-Varying Complex Speech Analysis

Authors:

K. Funaki, Y. Miyanaga, K. Tochinai, *Hokkaido University (JAPAN)*

ABSTRACT

LPC methods are successfully employed in a broad range of speech processing. However, the LPC methods can not estimate time-varying parameters since observed signals are assumed as stationary within the analysis interval. On the other hand, several complex LPC methods for analytic signals have already been proposed. These methods take some advantages over conventional LPC, i.e., more accurate spectral estimation, smaller computation, and so on. This paper presents a new non-recursive complex signal analysis for analytic speech signals by introducing a time-varying AR model in which the parameters are represented by complex basis expansion. The complex AR coefficients can be efficiently estimated by solving linear equation by means of an extended version of LDU decomposition. The experimental result with Japanese natural speech demonstrates that the proposed method offers accurate estimate of time-varying speech spectrum.

Autocorrelation Analysis Of Speech Signals Using The Number Theoretic Transforms Over The Direct Sums Of Finite Fields

Authors:

N. S. Rubanov, E. I. Bovbel, *Belarussian State University (BELARUS)*

ABSTRACT

In this paper we consider the application of the number theoretic transforms (NTTs) over the direct sums of finite fields R_m, m=0,1,2,... to autocorrelation analysis of a speech signal. We show that these NTTs are high-efficient and allow more flexible segmentation of the speech signal frame in comparison with the classical discrete Fourier transform (DFT). As a result, using the NTTs over R_m, m=0,1,2 reduces considerably the computational complexity of a speech signal frame autocorrelation function.

Speech Analysis And Synthesis Using Instantaneous Amplitutes

Authors:

G. Li, *Nanyang Technological University (SINGAPORE)*
L. Qiu, *MOTOROLA ELECTRONICS Pte. Ltd.*

ABSTRACT

In this paper, we propose an instantaneous amplitude (IA) based model for speech signal representation. Unlike the traditional representation, we represent each component with two parametrized instantaneous amplitudes and one constant `center' frequency. This can avoid the difficulty in dealing with the time-varying phases and allows us to carry out an optimization procedure easily such that the synthetic signal can be made as close to the original one as possible. Experiments show that the synthetic speech with the developed technique is of excellent quality and almost perceptually indistinguishable from the original speech.

Spectral Estimation Of Voiced Speech With Regressive Linear Prediction

Authors:
S. Varho, P. Alku, *University of Turku (FINLAND)*

ABSTRACT

A new predictive algorithm for the spectral estimation of voiced speech, Regressive Linear Prediction (RLP), is presented in this paper. Unlike conventional linear prediction (LP), RLP first combines the values that preceed the sample x(n) into groups of three consecutive samples (i.e., x(n-i), x(n-i-1), x(n-i-2)). Each of these p groups is then used to determine a regression line the value of which at time n are used as a data sample in the prediction. The prediction is optimised by minimising the square of the error between the original sample and the predicted value according to the autocorrelation criterion. With the same number p of unknowns in the normal equations, RLP yields an all-pole filter of order p+2, whilst the all-pole filter given by LP is of order p. The additional poles of the RLP filter improve the spectral modelling of voiced speech. This is shown in the present study by comparing the performance of conventional LP and the RLP method.

The Performance Of A Novel Pre-Processor For The Learning Vector Quantization Classifier

Authors:

H. Kelleher, E. Chilton, *University of Surrey (UK)*

ABSTRACT

In this paper a novel pre-processing technique, known as the pseudo-cepstrum, is described. The technique is applied to a speaker independent, continuous speech recogniser, using Learning Vector Quantization (LVQ) for the classification stage. The results of a set of performance comparisons between the pseudo-cepstrum, the conventional log-cepstrum and linear prediction will be presented. For each of these three pre-processors the effect of using a mixture of instantaneous, and first and second time differences are investigated. It will be shown that the pseudo-cepstrum, using a mixture of instantaneous and first time differences gives superior results. When this is translated into the lip shape classification, which is the intended output of this recognition system, the results are further improved.

Low Delay Phone Recognition

Authors:
A. R. Fonollosa, E. Batlle, J. B. Mario, *Universitat Politecnica de Catalunya (SPAIN)*

ABSTRACT

In the frame of the ACTS VIDAS project (VIDeo ASsisted with audio coding and representation) the authors have compared different schemes for low-delay phone recognition. In the VIDAS project, the phonetic recognizer was proposed for lip-synchronization in videophone applications, but the results are also of interest for very low rate speech coding (Phonetic Vocoding) and for driving synthetic mouth gestures in real time.

Design Of A Command Interface
With A Dynamic Grammar Speech Recognition Engine

Authors:

S. Kruger, S. Chennoukh, J. Flanagan, L. Rothkrantz, *Delft University of Technology (NETHERLANDS)*

ABSTRACT

This paper presents the design of an open grammar speech recognition engine. The engine is built to operate in real-time and kept simple. The objective of this paper is to show how task dependent knowledge can improve speech recognition accuracy by cancelling out parts of the command set depending on the state of the interface. The system is a phone-based recognizer and therefore the vocabulary and grammar can easily be extended without retraining the system. The hypothesis of this paper is that speech recognition accuracy will improve if higher level knowledge about the state of a task is used by the recognition engine.

ADPCM With Non-Linear Prediction

Authors:
M. Faundez-Zanuy, *Escola Universitària Politècnica de Mataró (SPAIN)*
O. Oliva-Suarez, *Signal Theory & Communications Department (UPC) (SPAIN)*

ABSTRACT

Many speech coders are based on linear prediction coding (LPC), nevertheless with LPC is not possible to model the nonlinearities present in the speech signal. Because of this there is a growing interest for nonlinear techniques. In this paper we discuss ADPCM schemes with a nonlinear predictor based on neural nets, which yields an increase of 1-2.5dB in the SEGSNR over classical methods. This paper will discuss the block-adaptive and sample-adaptive predictions.

Non-Linear Processing In Cochlear Spaced Sub-Bands Using Artificial Neural Networks For Multi-Microphone Adaptive Speech Enhancement

Authors:
A. Hussain, D. R. Campbell, *University of Paisley (UK)*

ABSTRACT

A general class of single hidden-layered, linear-in-the-parameters feedforward Artificial Neural Networks is proposed for processing band-limited signals in a multi-microphone sub-band adaptive speech enhancement scheme. The sub-band spacing within the adaptive speech enhancement system is set according to a published cochlear function. Comparative results achieved in simulation experiments demonstrate that the proposed sub-band scheme is capable of significantly outperforming conventional linear processing based wide-band and sub-band noise cancellation methods, in the presence of non-linear interference.

Optimization Of Filter Banks Based On Properties Of The Input Signal

Authors:
P. P. Vaidyanathan, *California Institute of Technology (USA)*

ABSTRACT

Filter banks and wavelets have found applications in signal compression, noise removal, and in many other signal processing contexts. In this tutorial we review a number of recent results on the optimization of filter banks based on the knowledge of the input. The main emphasis will be the minimization of error due to subband quantization, and its connection to principal component reconstruction. Both uniform and nonuniform filter banks are considered.

A Block Adaptive DFE In The Frequency Domain Based On Tentative Decisions

Authors:
K. Berberidis, P. Karaivazoglou, *University of Patras (GREECE)*

ABSTRACT

In this paper a new block adaptive DFE implemented in the frequency domain is derived. The new algorithm is suitable for rejecting ISI due to multipath echoes, especially to wireless transmission systems which involve channels with long impulse response. The novel idea is to use tentative decisions properly derived by minimizing a frequency domain criterion. The algorithm has a steady-state performance which is practically identical to that of the symbol-by-symbol DFE. At the same time it offers a faster convergence rate and substantial savings in complexity. Additionally, the level of the steady-state MSE achieved by the algorithm is insensitive to the length of the block. Thus the new algorithm allows a trade off between complexity and processing delay without affecting performance.

Implementation Of A Block Adaptive Filter Working In The Frequency Domain Combined With A Robust Adaptation Control

Authors:
B. Nitsch, *Darmstadt University of Technology, (GERMANY)*

ABSTRACT

Adaptive filters for hands-free telephone systems need an adaptation control which locks or slows down the adaptation process during double talk periods. In this contribution a block adaptive filter algorithm working in the frequency domain which combines a significant reduced numerical complexity with the same tacking abilities as the NLMS algorithm is presented. Moreover we control the stepfactor of the adaptive filter with an adaptation control based on a correlation criterion and several signal power estimators. The adaptation control is robust against double talk.

A Fast Prewhitened Affine Projection Algorithm

Authors:
K. Maouche, D. T.M. Slock, *Institut Eurecom (FRANCE)*

ABSTRACT

We derive a new adaptive filtering algorithm called the Instrumental Variable Affine Projection (IVAP) algorithm and give its fast version (IVFAP algorithm). The IVAP algorithm departs from the AP algorithm and uses an IV. The IV process is generated in a way such that the new algorithm combines between the AP and the Fast Newton Transversal Filter (FNTF) algorithms. Simulations show that the IVAP algorithm is more robust to noise than the AP algorithm. With the IV, the sample covariance matrix loses its Hermitian property and its displacement structure is different from the one of the AP algorithm. Consequently, the derivation of a fast version is done by deriving the IV Sliding Window Covariance Fast Transversal Filter (IV SWC FTF) algorithm. Using this and other ingredients, we derive the IVFAP algorithm whose computational complexity is nearly the same as the one of the FAP algorithm.

A Fast Multichannel Adaptive Filtering Algorithm

Authors:
M. Bhouri, M. Mboup, M. Bonnet, *Universite Rene Descartes-Paris 5 (FRANCE)*

ABSTRACT

In this paper we present a new QR decomposition (QRD) based fast multichannel adaptive filtering algorithm. This algorithm is based on a block reduction technique which leads to a substantial reduction of complexity compared to other QRD-based algorithms. It has a complexity of O (N) where N is the sum of the channels orders which may be different. It, also, has good numerical properties and is amenable to systolic architecture. Simulation shows a better robustness than the QRD-RLS in the context of highly intercorrelated channels inputs.

Simplified FLS Algorithm
For Linear Phase Adaptive Filtering

Authors:
L. S. Resende, *Universidade Federal de Santa Catarina, (BRAZIL)*
J. M. T. Romano, *Universidade Estadual de Campinas, (BRAZIL)*
M. G. Bellanger, *Conservatoire National des Arts et Métiers, (BRAZIL)*

ABSTRACT

The purpose of the present paper is to show that the linear phase property can be obtained with only a minor change in the regular FLS algorithm, with no additional multiplication, using the standard adaptation gain g(n). The proof is based on the linearly constrained filter called Generalized Sidelobe Canceller. Simulation results illustrate the efficacy of the simplified algorithm.

A Delayless Subband Adaptive Filter Using The Projection Method

Authors:
H. Sakai, S. Miyagi, *Kyoto University (JAPAN)*

ABSTRACT

Two topics associated with a delayless subband adaptive filter using the Hadamard transform are treated. One is to improve convergence speed and the other is to consider another aspect of subband adaptive filtering. Instead of LMS type algorithms which are often used in many applications, a projection method is employed as an adaptive algorithm to accelerate the convergence speed. The two filters corresponding to the secondary path transfer function in an ANC system are included in the conversion rule from subband filters to a fullband filter. From this fact, a new structure which is similar to the conventional fullband filtered-X algorithm is derived. The increase of the convergence speed and the stability in steady state with these methods are demonstrated by computer simulations.

Performance Limitations Of Subband Adaptive Filters

Authors:
S. Weiss, R.W. Stewart, *University of Strathclyde (UK)*
A. Stenger, R. Rabenstein, *Universitat Erlangen-Nurnberg (GERMANY)*

ABSTRACT

In this paper, we evaluate the performance limitations of subband adaptive filters in terms of achievable final error terms. The limiting factors are the aliasing level in the subbands, which poses a distortion and thus presents a lower bound for the minimum mean squared error in each subband, and the distortion function of the overall filter bank, which in a system identification setup restricts the accuracy of the equivalent fullband model. Using a generalized DFT modulated filter bank for the subband decomposition, both errors can be stated in terms of the underlying prototype filter. If a source model for coloured input signals is available, it is also possible to calculate the power spectral densities in both subbands and reconstructed fullband. The predicted limits of error quantities compare favourably with simulations presented.

Step-Size Optimization Of The BNDR-LMS Algorithm

Authors:
J.A. Apolinario Jr., *COPPE/Universidade Federal do Rio de Janeiro (BRAZIL) - Helsinki University of Technology (FINLAND) - Instituto Militar de Engenharia (BRAZIL)*
P.S.R. Diniz, *COPPE/Universidade Federal do Rio de Janeiro (BRAZIL)*
T.I. Laakso, *Helsinki University of Technology (FINLAND)*
M. L. R. de Campos, *Instituto Militar de Engenharia (BRAZIL)*

ABSTRACT

The binormalized data-reusing least mean squares (BNDR-LMS) algorithm has been recently proposed and has been shown to have faster convergence than other LMS-like algorithms in cases where the input signal is strongly correlated. This superior performance in convergence speed is, however, followed by a higher misadjustment if the step-size is close to the value which allows the fastest convergence. An optimal step-size sequence for this algorithm is proposed after considering a number of simplifying assumptions. Moreover, this work brings insight in how to deal with these conflicting requirements of fast convergence and minimum steady-state mean square error (MSE).

LMS Tracking Behavior
Under Periodically Changing Systems

Authors:
M. Rupp, *Bell-Labs - Lucent Technologies (USA)*

ABSTRACT

The tracking behavior of Least Mean Squares (LMS) and Recursive Least Squares (RLS) algorithms have been under considerations for several years. For system identification problems it is usually assumed that the system under consideration is of FIR type with coefficients that are statistically independent random processes with identical distribution (IID), a somewhat artificial assumption that can hardly be found in existing systems. In this article it is assumed that the system is periodically changing over time. This is an important case that occurs in every wireless system where the carrier frequency provided by the transmitter suffers an offset compared to the local oscillator of the receiver. The article presents an analysis of the algorithm tracking behavior as well as algorithms to estimate the frequency offset so that this particular corruption can be removed.

A Controlled Block Implementation Of The NLMS Algorithm

Authors:
K. Fujii, J. Ohga, *Fujitsu Laboratories Ltd. (JAPAN)*

ABSTRACT

The block implementation of the normalized least mean square (NLMS) algorithm has another parameter governing its convergence property. Controlling its block length enables to keep the estimation error at a desired level against the fluctuation of the power ratio of the reference signal to the noise. This block length control technique can be also applied to improving the convergence property. This technique provides almost the same convergence rate as the recursive least square (RLS) algorithm does.

Automatic 3D Model Acquisition And Generation Of New Images From Video Sequences

Authors:
A. Fitzgibbon, A. Zisserman, *University of Oxford, (UK)*

ABSTRACT

We describe a method to completely automatically recover 3D scene structure together with 3D camera positions from a sequence of images acquired by an unknown camera undergoing unknown movement. Unlike "tuned" systems which use calibration objects or markers to recover this information, and are therefore often limited to a particular scale, the approach of this paper is more general and can be applied to a large class of scenes. It is demonstrated here for interior and exterior sequences using both controlled-motion and handheld cameras.

The paper reviews Computer Vision research into structure and motion recovery, providing a tutorial introduction to the geometry of multiple views, estimation and correspondence in video streams. The core method, which simultaneously extracts the 3D scene structure and camera positions, is applied to the automated recovery of VRML 3D textured models from a video sequence.

Synthesis Of Virtual Views Of A Scene
From Two Or Three Real Views

Authors:

Georges M. Quenot, *LIMSI-CNRS (FRANCE)*

ABSTRACT

This paper presents two methods for synthesizing arbitrary views of a scene from two or three real views. Both methods rely on the extraction of a dense and accurate matching field between the images. The matching field is obtained from an optical ow computation technique based on dynamic programming. First method performs simple image interpolation with motion compensation in order to synthesize intermediate views. Transformations of the matching field allows to synthesize other views that are not simply intermediate ones. Second method is based on the reconstruction of a textured 3D surface. Both methods are demonstrated with two- and three-image sets.

3D Photography Using Shadows

Authors:
J.-Y. Bouguet, *California Institute of Technology (USA)*
P. Perona, *California Institute of Technology (USA), Universita di Padova, (ITALY)*

ABSTRACT

A simple and inexpensive approach for extracting the three-dimensional shape of objects is presented. It is based on `weak structured lighting;' and requires little hardware besides the camera: a light source (a desk-lamp or the sun), a stick and a checker-board. The object, illuminated by the light source, is placed on a stage composed of a ground plane and a back plane; the camera faces the object. The user moves the stick in front of the light source, casting a moving shadow on the scene. The 3D shape of the object is extracted from the spatial and temporal location of the observed shadow. Experimental results are presented on three different scenes (indoor with a desk lamp and outdoor with the sun) demonstrating that the error in reconstructing the surface is less than 0:5% of the size of the object.

Towards Flexible 3-D Digitizing Systems

Authors:
P. Hebert, J. Tremblay, F. Blais, H. Chotard, S. Dyck, *National Research Council of Canada (CANADA)*

ABSTRACT

The next generation of 3-D digitizing systems based on a laser range sensor must be more flexible. Systems must be portable, easy to set up, easy to use and could be hand-held. To reach this goal, three important issues are addressed: i) the integration of a hand-held sensor with a positioning device, ii) the portability and system set up, and iii) the user interaction with the digitizing system. The discussion is supported by the implementation of a prototype system.

Acquisition And 3D Registration Of Image Sequences For Structured Light Based Free-Form Surface Reconstruction

Authors:
C. Schoenenberger, P. Graebling, E. Hirsch, *Université Louis Pasteur, (FRANCE)*

ABSTRACT

Accurate measurements on manufactured objects including free-form surfaces are of prime interest for (industrial) computer vision. However, resolution of standard CCD sensors is a strong limitation for objects of large extent. To overcome this shortage, image sequences covering the areas of interest are acquired and registered. Making use of structured light for obtaining 3D information requires registration to rely on the extracted 3D data. Novelty of our highly accurate registration scheme for pairs of consecutive images lies in interpolation of the imaged surfaces before matching 3D points in the automatically determined overlap regions, allowing to significantly reduce the size of the search space for matching without spoiling registration accuracy. Further, the algorithm computes actual displacements between images, enabling to express the measured points in a reference frame. Application to spatial image sequences assess efficiency, robustness and accuracy (15 to 50µm·depending on object nature and size of image sequences).

Evaluation Of The Construction Of Novel Views By A Combination Of Basis Views

Authors:
B. Buxton, Z. Shafi, *University College London (UK)*
J. Gilby, *Sira Technology Centre (UK)*

ABSTRACT

A technique for synthesising novel views of an object or scene from a linear combination of basis images, originally proposed by Ullman and Basri, is briefly reviewed, extended and evaluated in a series of experiments on simple test objects. A symmetric, but overcomplete set of linear equations relating a small number of control points in the novel view to corresponding points in the basis images is used to calculate the geometry of the object as seen in the novel view.The image intensity is then calculated from a rendering model based on the distance of the novel from the basis views. Comparison of synthesised and actual images of the objects shows that the reconstructed image geometry and intensity are both accurate unless perspective effects are large. The use of an overcomplete set of linear equations to calculate the reconstructed image geometry does not lead to stability problems.

Three Dimensional View Registration
By A Frequency Domain Technique

Authors:
G.M. Cortelazzo, G. Doretto, L. Lucchese, S. Totaro, *University of Padova (ITALY)*

ABSTRACT

This work presents the algorithmic variations of a 3-D motion estimation method which make it suitable to 3-D view registration. Important characteristics are that the method is suitable for unsupervised 3-D registration and that texture information can be incorporated with beneficial effects on the overall 3-D registration performance. A peculiarity of the method is that it operates in the frequency domain.

Multi-View Modeling & Synthesis

Authors:
C. Brestel, S. Ullman, *The Weizmann Inst. of Science (ISRAEL)*

ABSTRACT

In this work we introduce a method for generating a complete object representation based on its cover by a set of 2D views. This view-based representation has applications in several areas such as object recognition, computer graphics and video compression. The notion of image-based modeling received considerable attention in recent years. The basic motivation is to avoid the computational-intensive process of acquiring a 3D model followed by rendering. Instead, the approach uses a number of model images taken from different poses of the 3D object or scene as a basis. From these views any pose can be synthesized directly. Prior work used a small number of views to represent the object from a limited range of viewing directions. In the current work, a complete model that allows a presentation of the object from any viewing direction is introduced. A second novel aspect of the current work is a method for obtaining correspondence between views using a special pattern illumination. The correspondence is obtained with complexity of O(Nlog(N)) where N is the number of pixels in the image.

Adaptive Learning Algorithms For Semantic Object Extraction

Authors:

N. Doulamis, A. Doulamis, S. Kollias, *National Technical University of Athens (GREECE)*

ABSTRACT

In this paper a novel approach is proposed for semantic video object segmentation. In particular, it is assumed that several neural network structures have been stored in a system database or memory. Each network have been learned to be appropriate to a specific application. Then, a retrieval mechanism is introduced which selects that network from the memory which better approximates the current environment. Since, however, the retrieved network does not exactly correspond to the current conditions a small adaptation of its weights will be necessary in most cases. For that reason, an efficient training algorithm has been adopted based on both the former and the current network knowledge. The former one corresponds to knowledge existing in the memory while the latter is provided by the training set selection module based on the user assistance and a color segmentation algorithm. Experimental results are presented illustrate the performance of the proposed method to real life applications.

Recurrent Neural Networks For Signal Processing Trained By A New Second Order Algorithm

Authors:
P. Campolucci, M. Simonetti, A. Uncini, *Università di Ancona*, *(ITALY)*

ABSTRACT

A new second order algorithm based on Scaled Conjugate Gradient for training recurrent and locally recurrent neural networks is proposed. The algorithm is able to extract second order information performing two times the corresponding first order method. Therefore the computational complexity is only about two times the corresponding first order method. Simulation results show a faster training with respect to the first order algorithm. This second order algorithm is particularly useful for tracking fast varying systems.

Neural Networks With Hybrid Morphological/Rank/Linear Nodes And Their Application To Handwritten Character Recognition

Authors:

L. F. C. Pessoa, *Georgia Institute of Technology (USA)*

P. Maragos, *Georgia Institute of Technology (USA), Institute for Language and Speech Processing (GREECE)*

ABSTRACT

We propose a general class of multilayer feed-forward neural networks where the combination of inputs in every node is formed by hybrid linear and nonlinear (of the morphological/rank type) operations. For its design we formulate a methodology using ideas from the back-propagation algorithm and robust techniques to circumvent the non-differentiability of rank functions. Experimental results in a problem of handwritten character recognition are described and illustrate some of the properties of this new type of nonlinear systems.

Decision Level Fusion By Clustering Algorithms For Person Authentication

Authors:
V. Chatzis, A. G. Bors, I. Pitas, *Aristotle University of Thessaloniki (GREECE)*

ABSTRACT

In this paper, the use of clustering algorithms for decision level data fusion is proposed. Person authentication results coming from several modalities (e.g. still image, speech), are combined by using the fuzzy k-means (FKM) and the fuzzy vector quantization (FVQ) algorithms, two modification of them that use fuzzy data FKMfd and FVQfd, and a median radial basis function (MRBF) network. The modifications of the FKM and FVQ algorithms are based on a novel fuzzy vector distance definition, and they utilize the quality measure of the results, that is provided by the authentication methods. Simulations show that the proposed algorithms have better performance compared to classical clustering algorithms and other known fusion algorithms.

Supervised Multisensor Tracking Algorithm

Authors:
V. Nimier, *ONERA (FRANCE)*

ABSTRACT

We propose, in this paper, a method and an algorithm for combining symbolic and numerical information for data fusion in tracking application. The objective is to have a tracking algorithm supervised by contextual analysis. This analysis is able to detect the sensors which are reliable and those which are not in operational situations. Then, automatically the algorithm increase, in the fusion process, the importance of measurements of reliable sensor and decrease the importance of those provided by disturbed sensors. A simulation shows the effectiveness of the method and the gain in performance. This gain is essentially based on the fact that the system has a better robustness in disturbed operational situations.

Second Order Hebbian Neural Networks And Blind Source Separation

Authors:
K. I. Diamantaras, *Technological Education Institute (GRRECE)*

ABSTRACT

The adaptive blind source separation problem has been traditionally dealt with the use of nonlinear neural models implementing higher-order statistical methods. In this paper we show that second order Cross-Coupled Hebbian rule used for Asymmetric Principal Component Analysis (APCA) is capable of blindly and adaptively separating uncorrelated sources. Our method enjoys the following advantages over similar higher-order models such as those performing Independent Component Analysis (ICA): (a) the strong independence assumption about the source signals is reduced to the weaker uncorrelation assumption, (b) there is no constraint on the sources pdf's, i.e. we remove the assumption that at most one signal is Gaussian, and (c) the higher order statistical optimization methods are replaced with second order methods with no local minima, and(d) the kurtosis of the sources becomes irrelevant. Simulation experiments shows that the model successfully separates source images with kurtoses of different signs.

Divide And Conquer Algorithms For Constructing Neural Networks Architectures

Authors:
K. Koutroumbas, A. Pouliakis, N. Kalouptsidis, *University of Athens (GREECE)*

ABSTRACT

In this paper two algorithms for the construction of feedforward neural network architectures are discussed. Their close relation with a special class of decision tree classifiers is commented. Finally, the performance of the algorithms is assessed on a complex cytological pattern classification application.

Implementing Size-Optimal Discrete Neural Networks Require Analog Circuitry

Authors:
V. Beiu, *Los Alamos National Laboratory, (USA)*

ABSTRACT

This paper starts by overviewing results dealing with the approximation capabilities of neural networks, as well as bounds on the size of threshold gate circuits. Based on a constructive solution for Kolmogorov's superpositions we will show that implementing Boolean functions can be done using neurons having an identity transfer function. Because in this case the *size* of the network is minimised, it follows that *size*optimal solutions for implementing Boolean functions can be obtained using analog circuitry. Conclusions and several comments on the required precision are ending the paper.

Fast Formation Of Invariant Feature Maps

Authors:
S. McGlinchey, C. Fyfe, *University of Paisley (UK)*

ABSTRACT

We present a neural network that uses competitive learning and a neighbourhood function in a similar way to the self-organising map (SOM) [1]. The network consists of a number of modules that are positioned in an array (normally in one or two dimensions) where each module performs a subspace projection and the learning rate used within each module is weighted by the neighbourhood function. By using a subspace method that always gives convergence of orthonormal basis vectors within the modules, we demonstrate that this method can allow fast formation of filters of basic invariant features. The adaptive subspace self-organising map (ASSOM) [1,2] is in some ways similar and it has been proposed as a modular neural network for invariant filtering, providing invariances in scale, phase, rotation, translation etc. In this paper we show that our method can also provide these invariances with much improved performance in terms of the processing time required during training.

Performance Evaluation Of Adaptive Subspace Detectors, Based On Stochastic Representations

Authors:
S. Kraut, L. Scharf, *University of Colorado (USA)*

ABSTRACT

In this paper we present a technique for evaluating the moments of "adaptive" detectors, where the noise covariance is estimated from training data, rather than assumed to be known a priori. It is based on the method of "stochastic representations" recently presented in [1]. These representations express adaptive detectors as simple functions of the same set of five statistically independent scalar random variables. They may be applied to a whole class of detectors, which includes the adaptive versions of the matched subspace detectors shown to
be UMP-invariant and GLRT in [2] and [3], and to the adaptive GLRT detector of Kelly [4]. Using a stochastic representation, the moments of any member of this class may be evaluated without the need to derive its density or characteristic function. The first two moments give a convenient measure for how the SNR loss improves as the number of training vectors, M, increases.

Low Power Detection

Authors:
M. Nafie, A.Tewfik, *University of Minnesota (USA)*

ABSTRACT

Low power and low complexity algorithms for signal processing applications have gained increasing importance with the deployment of portable communication equipment. Most of the current power reduction techniques rely on reducing the power by VLSI implementation. This approach is expensive and limited by technology. Hence algorithm design optimization is a must for low energy consumption. Here, we propose using finite memory detection algorithms as low complexity- low energy near optimal detection algorithm that trades a small amount of detection performance for a reduction in complexity and power consumption. The negligible loss in detection performance is easily accommodated in wireless video and audio transmission applications. In data applications, this small loss can be further reduced with error correcting codes at the expense of a slight loss in communication bandwidth. We present simple algorithms for deriving the near optimum finite memory detectors in the time invariant and time variant case. The same algorithms can be used in tandem configurations in decenteralized detection.

Mlse And Spatio-Temporal Interference Rejection Combining With Antenna Arrays

Authors:
David Asztely and Bjorn Ottersten, *Royal Institute of Technology (SWEDEN)*

ABSTRACT

By using multiple receiving antennas and modeling co-channel interference (CCI) together with the noise as additive temporally white Gaussian noise with some spatial color, CCI may be suppressed. This paper proposes spatio-temporal interference rejection combining by modeling the CCI and noise as an autoregressive Gaussian process. In this way, the joint spatial-temporal properties of the CCI may be taken into account. A training sequence based estimator is proposed, and simulations show large gains in CCI rejection as compared to only spatial processing for small antenna arrays in interference limited GSM urban scenarios. An example with data collected with a dual polarized antenna in a suburban environment is also presented.

Interference Suppression In Ss Systems: A Comparison Between Ptv P/S-Type And W-Type Filters

Authors:
G. Gelli, *Seconda Universita di Napoli, (ITALY)*

ABSTRACT

The recently proposed LCL-PTV structures for narrow-band interference suppression in DS/SS systems are examined and compared in terms of output SIR. New LCL-PTV prediction/subtraction filters are proposed, which are shown to provide the same SIR performances of the LCL-PTV whitening structures, with the additional advantage of being amenable to blind adaptive implementation. Simulation results are provided to support and expand the theoretical claims.

A Space/Time Pre-equalization Technique For Down-link Signal Transmission In Time-Division-Duplex (TDD) Mobile Multimedia Communications

Authors:

S.Tomisato, K.Fukawa, T. Matsumoto, *NTT Mobile Communications Network Inc. (JAPAN)*

ABSTRACT

This paper proposes a new spatial and temporal (S/T) pre-equalization technique for time-division-duplex (TDD) mobile multimedia communication systems. Base station (BS) employs an adaptive array antenna (AAA) and an adaptive decision feedback equalizer (DFE) for up-link signal reception. The weight vectors for the up-link AAA and DFE are also used for pre-equalization of signal to be transmitted. With the S/T pre-equalization at the transmitter, the needs for equalizers at mobile station (MS) can be eliminated. In order to evaluate performances of the proposed S/T pre-equalization scheme, this paper uses a detailed propagation model which looks into the mobile propagation mechanism more in detail than conventional multipath models. Performances of the proposed S/T pre-equalizer are then demonstrated through computer simulations. The results show that the proposed scheme is effective in reducing effects of both inter-symbol and co-channel interferences.

Fast Phase Sequences Spreading Codes For CDMA Using FFT

Authors:
E. Del Re, R. Fantacci, L. S. Ronga, *Universita di Firenze, (ITALY)*

ABSTRACT

In multi-path environments, Spread Spectrum multiple access techniques offer attractive properties of rejection of undesired signals and an high spectral efficiency. In this paper is presented a design method for spreading codes which employs Perfect Sequences, a class of complex sequences with singular correlation properties. The main advantage from the use of Perfect Sequences is the oppportunity of a fast DSP implementation with FFT blocks, where the analog section is extremely contained. Simulations in a multipath enviroment confirm the useful caracteristics of the proposed receiver for asyncronous trasmissions.

A New Orthogonal Sequence For DS/CDMA Systems

Authors:
S. I. Park, H.G. Kim, I. Song, S. C. Kim, Y. H. Kum, J. Lee, *Korea Advanced Institute of Science and Technology (KOREA)*

ABSTRACT

In this paper, new polyphase sequences and a sequence generation method are suggested. The correlation properties of the sequences are investigated. These sequences have good correlation properties. Since the suggested generation method consists only of integer sums and modular techniques, sequence generation is also easy. The performance of the sequences is investigated for QS-CDMA systems in frequency selective, time nonselective, slow Nakagami fading channel with additive white Gaussian noise.

Modelling Sea Clutter Using Conditional Heteroscedastic Models

Authors:

J. L. Noga, W. J. Fitzgerald, *Cambridge University (UK)*

ABSTRACT

In this paper a class of conditional heteroscedastic models is introduced in the context of sea clutter modelling. In particular, an Auto-regressive (AR) process driven by conditional heteroscedastic (CH) errors (AR-CH model) is proposed as a model for the time evolution dynamics of the modulating component of sea clutter. The CH process parameters of the AR-CH model determine the weight of the tails of the marginal distribution, while the AR component largely determines the correlation structure. Different functional forms of conditional variance models are investigated using real sea clutter data.

Robust Image Restoration Matched With Adaptive Aperture Formation In Radar Imaging Systems With Sparse Antenna Arrays

Authors:
I. Prudyus, S. Voloshynovskiy, T. Holotyak, *State University "Lvivska polytechnika" (UKRAINE)*

ABSTRACT

The paper presents a complex approach to radar imaging system development. The approach consists of two main stages. The first stage is adaptive aperture formation and the second one is robust adaptive image restoration. The use of adaptive aperture formation strategy makes possible to estimate the principle image components in spatial frequency domain and increase the reliability of received data. The robust adaptive image restoration allows to compensate the blurring effect of sparse aperture in the presence of mixed noise (i.e. Gaussian and impulse). The efficiency of the proposed approach is investigated on numerous test examples.

A Real-Time Performance Evaluation Technique For Future DSP-Based "Software" Wireless Radio

Authors:
L. Prosperi, Dr. L. Franchina, *University of Rome "La Sapienza" (ITALY)*
D. Gianfelici, Dr. S. A. Kosmopoulos, *Space Engineering S.P.A. (ITALY)*

ABSTRACT

The desired performance of a wireless system depends on the interaction of the number of users carried by the system, hence, influencing the choice of the multiple access scheme (e.g., FDMA vs. TDMA vs. CDMA). This choice has a major impact on the selection of the power control techniques and on the physical layout of the entire system. It is the intention of this paper to provide a fast converging semi-analytical evaluation of the Bit Error Rate (BER) performance of A/DS-CDMA transmission schemes, incorporating a Wiener filter to countermeasure the effects of the near-far effect present in future wireless communication systems.

An Efficient Order Recursive Algorithm For Volterra System Identification

Authors:
G.-O. Glentis, *TEI of Heraklion (GREECE)*
N. Kalouptsidis, *University of Athens (GREECE)*

ABSTRACT

In this paper nonlinear filtering and identification based on finite support Volterra models is considered. A set of primary signals, defined in terms of the input signal, serve for the efficient mapping of the nonlinear process to an equivalent multichannel format. An efficient order recursive method is presented for the determination of the Volterra model structure. The efficiency of the proposed methods is illustrated by simulations.

Signal Modeling
Using Piecewise Linear Chaotic Generators

Authors:

T. Schimming, M. Goetz, W. Schwarz, *Technische Universitat Dresden (GERMANY)*

ABSTRACT

In this paper the modeling of a signal by achaotic generator with respect to a specified signal statistic will be considered. To accomplish that, the considered class of n-dimensional piecewise linear Markov generators will first be analyzed analytically, yielding an algebraic expression for the statistical quantity in question. Based on this analytical result an optimal set of parameters minimizing the modeling error with respect to the considered statistical quantity will be calculated.

Analysis Of Synchronization Of Chaotic Systems Based On Local Contitional Lyapunov Exponents

Authors:

Z. Galias, *University of Mining and Metallurgy (POLAND)*

ABSTRACT

In this paper we consider the problem of synchronization of coupled chaotic systems. First we show the limitations of existing techniques for studying of the synchronization. Then we introduce the notion of local conditional (transversal) Lyapunov exponents. We show that they can be successfully used in investigations of synchronization properties. We develop a new criterion for synchronization based on local conditional Lyapunov exponents. The discussion is supported by numerical examples.

Inversion Of H-ARMA Models

Authors:

D. Declerq, P. Duvaut, J. Soubielle, *ETIS URA CNRS 2235 (FRANCE)*

ABSTRACT

We present in this contribution the problem of nongaussian H-ARMA models inversion. We show that very classical methods of parameters identification based on the likelihood are unefficients in our case and we have chosen a fractionnal distance minimisation approach to estimate the nonlinearity. The ARMA coefficients are identified with maximum likelihood estimators and a comparison study with the cumulant based method has been conducted on synthetic data.

A Time-Frequency Method For Nonlinear System Classification In Presence Of Noise

Authors:
L. Galleani, L. Lo Presti, *Politecnico di Torino (ITALY)*

ABSTRACT

This paper presents a method for the classification of nonlinear systems through the study of the free oscillations in the time-frequency plane, when the measured data are affected by noise. Nonconservative SDOF (Single Degree Of Freedom) oscillators described by a non-linear second order differential equation are considered. The nonlinearity is due to a nonlinear function of the state variable, which produces free oscillations with a time-variant spectrum. The method used for the classification is a substantial modification of a basic algorithm proposed by the same authors for noise-free data. In presence of noise improved performances are obtained with the new algorithm.

Nonlinear Frame-Like Decompositions

Authors:

B. Pouye, J.-C. Pesquet, *LSS / University Paris Sud (FRANCE)*
A. Benazza-Benyahia, *LS Telecom (TUNISIE)*
I.Pollak, *SSG - LIDS - MIT (USA)*
H. Krim, *ECE Dept./CACC - NCSU (USA)*

ABSTRACT

In this paper, we revisit a number of concepts which have recently proven to be useful in multiscale signal analysis, specifically by replacing the now classical linear scale transition operators by nonlinear ones. Connections between nonlinear perfect reconstruction filter banks and PDE operators used in scale-space theory are established. An application of the proposed nonlinear tools is then given for extracting a signal embedded in noise. We also develop the important case of time invariant nonlinear representations.

Numerical Integration Of Nonlinear PDEs Using MD Passive Circuits

Authors:
F. N. Koumboulis, *University of Thessaly, (GREECE)*
B. G. Mertzios, *Democritus University of Thrace, (GREECE)*

ABSTRACT

Numerical integration of nonlinear (NL) partial differential equations (PDEs) is studied via transformation of the system and then realization of the transfofmed system to a passive generalized Kirchhoff circuit. The Kirchhoff circuit is appropriately discretized in order to be represented by a discrete multidimensional (Ivm) system, using principles of wave digital filters (VWFs).

Digital Compression Of Analytic Signals Using Nth Root

Authors:

M. Bellanger, K. Ouaissa, *CNAM - Electronique (FRANCE)*

ABSTRACT

The computation of Nth root produces a compression of the spectrum for some types of analytic signals. Once the spectrum has been shrinked in the frequency domain, the signal can be decimated in the time domain, leading to compression. The Nth root calculation raises some problems which are discussed, like phase jumps related with zero crossings of the amplitude. Results of simulations with sinusoids and speech signal are reported. More works is needed to make the approach practical.

Generation Of Idempotent Monotone Boolean Functions

Authors:
I. Shmulevich, *University of Nijmegen (NETHERLANDS)*

ABSTRACT

This paper focuses on the class of idempotent monotone Boolean functions. Monotone Boolean functions correspond to an important class of non-linear digital filters called Stack Filters. The idempotence property implies that applying such a filter produces a root signal in one pass. We present an algorithm for testing a given monotone Boolean function for idempotence and provide the set of idempotent monotone Boolean functions of 5 variables.

Nonlinear Decision Feedback Equalization Using RCPL Network

Authors:
L. Demirekler, T. Adalz, X. Liu, *University of Maryland Baltimore County, (USA)*

ABSTRACT

In this paper, we introduce a nonlinear decision feedback equalizer (DFE) structure which decomposes the nonlinear equalization problem into feedback and feedforward parts, thus allows flexibility in the choice of suitable structures for a given problem and improves on the performance of the conventional DFE and the nonlinear equalizer proposed in [1]. We show that the given general structure for this DFE can be represented by a recurrent canonical piecewise linear (RCPL) network [2] and that it satisfies the dynamic properties given in [3].

A New Hybrid CELP-Harmonics Speech Coder At Low Bit Rates

Authors:
L. Hubaut, F. Coulon, *Faculté Polytechnique de Mons, (BELGIUM)*
O. van der Vrecken, *BaBel technologies SA, (BELGIUM)*

ABSTRACT

This paper describes a new speech coding system working at a bit rate of 2400 bps. Like other coders at this bit rate, the BTC-S24 is based on a linear prediction (LP) model that uses a bimodal voicing decision in order to make an efficient coding of voiced and unvoiced frames. Such coders have proved to produce a good intelligibility of the synthesized voice but often at a quality that is inadequate for network communication applications like internet telephony. For this raison, some features have been added to the coding system in order to improve the perceptual quality of the produced voice without introducing high complexity. These features are implemented by the way of an hybrid coding procedure based on a pitch harmonics reconstruction for voiced frames and a CELP like algorithm for unvoiced frames. A voiced proportional ratio is also applied in voiced frames to take into account the proportion of voiced and unvoiced energies in several frequency bands. This coder has been tested by different people and results show that the perceptual quality is greatly improved against the original LPC-10.

Increasing Quality Of CELP Coders By Source-Filter Interrelation Using Self Organising Maps

Authors:
G. Avkarogullar, *Havelsan Inc. (TURKEY)*
T. Ciloglu, *Middle East Technical University (TURKEY)*

ABSTRACT

There are various alternatives for secondary excitation formulation for CELP type speech coders. In this paper we present a secondary excitation codebook generation and search algorithm based on the information derived from the linear prediction filter. Source-filter interrelation is extracted using Kohonen Learning algorithm and filter parameters (LSFs) are clustered according to topographic neighbourhood. For each cluster a secondary excitation shape-gain codebook is generated. Using the class information that current LSFs belong, only the associated codebook is searched. Shape codebooks of size 128 and gain codebooks of size 32 lead to statistically indifferent synthetic voice quality according to listening test when compared to FS1016 CELP coder.

On The Application Of A Psychoaccoustically Motivated Speech-Quality Measure In CELP Speech-Coding

Authors:
M. Hauenstein, N. Goertz, *University of Kiel (GERMANY)*

ABSTRACT

The crucial task in a CELP speech codec consists of finding the optimal excitation vector for the synthesis filter. This is usually done in an 'analysis-by-synthesis' structure by minimizing the mean-square error of the original and the coded/decoded speech frame. It is a common assumption that distance measures other than MSE and

adapted to the human auditory perception should result in better speech quality. Such measures could be based on scientific results provided by psychoacoustics. However, due to the computational load there is no possibility to implement complex psychoacoustical models in real-time speech codecs and, for the time being, we are restricted to the MSE. Nevertheless, it is interesting to study the potential of psychoacoustic distance measures to improve speech codecs if complexity restrictions are neglected. This paper shows how a psychoacoustics-based distance measure can be integrated into a CELP codec, and the unexpected results are presented.

Fast LSP Calculation And Quantization With Application To The CELP FS1016 Speech Coder

Authors:

S. Grassi, M. Ansorge, F. Pellandini, *University of Neuchâtel (SWITZERLAND)*

ABSTRACT

Line Spectrum Pair (LSP) representation is used for spectral quantization in the CELP FS1016 speech coder, where the LSPs are first calculated, and then quantized using 34-bit non-uniform scalar quantization. In the algorithm proposed in this paper, computational complexity is decreased by searching the zero-crossings on the grid formed by the values of the quantization tables. As the actual LSPs are not calculated, two criteria to select the "closest" quantized LSPs are proposed. These criteria take into account the interaction between successive LSPs. The efficiency and reliability of the proposed algorithm are improved using the interlacing property of the LSPs and knowledge of the direction of the sign-change at every zero-crossing. The proposed algorithm is compared with the existing Kabal's algorithm (followed by quantization), showing similar quantization performance. The computational complexity on a fixed-point DSP56001 implementation is reduced by 66 %, using the proposed algorithm.

A Preliminary Study Of An Audio-Visual Speech Coder: Using Video Parameters To Reduce An LPC Vocoder Bit Rate

Authors:
E. Foucher, G.Feng, L. Girin, *Institut de la Communication Parlée INPG/ENSERG/Université Stendhal (FRANCE)*

ABSTRACT

Today there exists numerous speech coding techniques which allow to transmit and stock efficiently the acoustic signals. But speech is both auditory and visual : audio and video information are complementary and the lip shape of the speaker can help the listeners to better understand what is said. This paper aims to show the interest of video information in the speech coding domain in general and in terms of transmission rate in particular.

Warped Linear Predictive Audio Coding In Video Conferencing Application

Authors:
K. Palomäki, A. Härmä, U. K. Laine, *Helsinki University of Technology (FINLAND)*

ABSTRACT

A codec for wideband 12kHz speech and audio in a video conferencing application is proposed in this paper. The codec is based on warped linear predictive coding algorithm which utilizes the auditory Bark frequency resolution. The structure of the codec is described and the main issues on the real-time implementation are discussed. The codec is integrated to a video conferencing product which is a video codec PC-board. The algorithm is implemented in a Texas TMSC31 digital signal processor.

Variable-Rate Speech Coding: Coding Unvoiced Frames With 400 Bps

Authors:

W. Ehnert, *University of Kiel (GERMANY)*

ABSTRACT

The following article describes a new variable-rate speech coder which combines the fortes of the principles of Harmonic Coding (HC) and Linear Prediction (LP). It presents an algorithm for the classification of voiced and unvoiced frames as well as a new method for coding unvoiced plosive and fricative phonemes with only 400 bps and 450 bps, respectively. Using the Improved Multiband Excitation (IMBE) version of 4.15 kbps for voiced frames results in an average rate of approximately 3.3 kbps for English speech.

Performance Of Discrete Fourier Transform With Small Overlap In Transform-Predictive-Coding-Based Coders

Authors:

A. Jbira, *ENST (FRANCE)*

ABSTRACT

In this paper, we evaluate the performance of Discrete Fourier Transform with small overlap in Transform-Predictive-Coding-based coders - e.g. Transform Coder eXcitation (TCX) and Transform Predictive Coding (TPC). Three di_erent time-frequency analysis techniques are compared : the Discrete Fourier Transform, the Modified Discrete Cosine Transform (MDCT) with 50% overlap and the Discrete Fourier Transform with 3% overlap. We show how a DFT with small overlap is more attractive, in a transform-predictive-coding based coder, than a simple DFT or a high frequency resolution MDCT.

Improved Lost Frame Recovery Techniques For ITU-T G.723.1 Speech Coding System

Authors:

G. Ho, S. Yeldener, M. Baraniecki, *COMSAT Laboratories (USA)*

ABSTRACT

Accepted by the ITU in 1996, the G.723.1 dual rate speech coder has since been used for application in low bit-rate videophone systems. Specifically, it has been used to encode voice for multimedia services over packet-switching networks including IP, ATM, Frame Relay, as well as over mobile communications networks. Due to the near toll quality of the G.723.1 speech coder, it is ideal for real-time voice communications over private and local area networks (LANs) where packet loss is minimal. However, over wide area networks (WANs), global area networks (GANs), and mobile communications networks, congestion is often severe and increasing rates of packet loss may generate significant distortions in output speech quality if unremedied. This paper describes a lost frame recovery technique for reconstructing lost packets at the speech decoding system. For packet loss rates up to 15%, simulation results show a substantial improvement in output speech quality over the current strategy provided for the G.723.1 speech coder, at the price of increasing algorithmic delay.

An Adaptive Quantization Using IIR Filter Bank For Speech Compression

Authors:
H. Saito, S. Nakamura, *Tokyo Denki University (JAPAN)*

ABSTRACT

Data compression techniques, such as image compression and speech compression, are useful in communication applications. We propose a simple speech compression algorithm using sub-band division and Sub-Adaptive Quantization. Although speech data are stored in a semiconductor memory device, its capacity and an available network capacity are limited. Therefore it is necessary to compress the data as much as possible. However there are two conditions to be satisfied: One is that the reconstructed datum is understood correctly. The other is that we can identify who is the sender. Signals with a rate of 64kbit/s are compressed at a ratio of about 1/13 using the proposed IIR sub-band division and Sub-Adaptive Quantization.

Spatial Coherence Exploitation Which Yields Non-Stationary Noise Reduction In Subband Domain

Authors:
R. Atay, *Faculte des Sciences (MAROKO)*
E. Mandridake, D. Bastard, M. Najim, *Equipe Signal et Image ENSERB and PRC - GDR ISIS (FRANCE)*

ABSTRACT

This paper deals with the enhancement of speech corrupted by real additive noises in a car when two observations are available. As far as we know, no enhancement system was capable of improving both the quality and the intelligibility of the noisy signals. We propose an enhancement method using thresholding, segmentation and filtering in subband domain. The main idea is to expand in subband signals the two observation speech signal and to exploit the spatial coherence of sound from the subband expansion. Noise reduction is conducted using two methods according to the degree of correlation of the subbands of the two observations. The proposed noise reduction approach is applied to realistic situations like speech signals, which are corrupted by non-stationary noises of diverse origins.

Adaptive Kalman Filter For Speech Enhancement From Colored Noise

Authors:
M. Gabrea, *Technical University of Timisoara (ROMANIA)*
E. Mandridake, M. Najim, *Equipe Signal et Image, ENSERB (FRANCE)*

ABSTRACT

In the framework of speech enhancement we propose a new approach for signal recovering in colored noise based on adaptive Kalman filter. The approaches proposed in the past, in this context, operate in two steps: they first estimate the noises variances and the parameters of the signal and noise models and secondly estimate the speech signal. In this paper we propose a new parameters estimation method based on the EM (Expectation-Maximisation) algorithm.

A Robust Begin-End Point Detector For Highly Noisy Conditions

Authors:
R Martinez, A. Alvarez, P. Gómez, V Nieto, V Rodellar, M. Rubio and M. Pérez,
Universidad Politécnica de Madrid, (SPAIN)

ABSTRACT

Most recognition methods, which have shown to be highly efficient under noise-free conditions fail dramatically with S/N ratios around or below 10 dB. One of the consequences of these high noise levels is that most Begin-End Point Detectors fail to separate properly the speech segments of the noise ones. Therefore, the speech recognition mechanisms will not have a clear boundary to start the processing of the signal, and as a consequence, speech segments will be lost, and noisy segments will be used in recognition. The overall reliability of the Speech Recognition System will dramatically experience the consequences of this impairment in its results. What is being proposed in this paper is to use the side information provided by an Adaptive Noise Canceller for the dynamic detection of word boundaries.

On Real Time Implementation Aspects Of A Source Separation Algorithm

Authors:
U. A. Lindgren, *Ericsson Mobile Communications AB (SWEDEN)*

ABSTRACT

The present paper presents a source separation algorithm used for noise suppression. The complexity of the algorithm at hand is treated. A reformulation of the algorithm reduces the number of arithmetic operation by a factor of five. Measurement noise is discussed and remedies are suggested and presented in simulations. An alternative estimation of correlations is suggested which might be more suited for use in a custom circuit.

A Double Talk Detector Based On The Partial Coherence Function

Authors:

R. le Bouquin Jeannes,G. Faucon, *Université de Rennes 1 (FRANCE)*

ABSTRACT

The growth of mobile radio and teleconference communications requires the design of efficient and robust hands-free systems. In this context, optimisation of acoustic echo and noise reduction is needed. This operation often requires double talk detection, either to choose between different structures or to stop the adaptation of the acoustic echo canceller. In this paper, two microphones and one loudspeaker are considered and a double talk detector based on the partial coherence is investigated. Results are presented on simulated and real signals.

Speaker Normalization For Automatic Speech Recognition - An On-Line Approach

Authors:
I. Dologlou, *Katholieke Universiteit Leuven - ESAT - PSI (BELGIUM)*
T. Claes,L. ten Bosch, D. Van Compernolle, H. Van Hamme, *Lernout & Hauspie Speech Products (BELGIUM)*

ABSTRACT

We propose a method to transform the on line speech signal so as to comply with the specifications of an HMM-based automatic speech recognizer. The spectrum of the input signal undergoes a vocal tract length (VTL) normalization based on differences of the average third formant F_3 . The high frequency gap which is generated after scaling is estimated by means of an extrapolation scheme. Mel scale cepstral co-efficients (MFCC) are used along with delta and $delta^2$-cepstra as well as delta and $delta^2$ energy. The method has been tested on the TI digits database which contains adult and kids speech providing substantial gains with respect to non normalized speech.

On The Equivalence Between Predictive Models For Automatic Speech Recognition

Authors:
B. Petek, *University of Ljubljana (SLOVENIA)*

ABSTRACT

This paper addresses the equivalence of input-output mapping functions between the Linked Predictive Neural Networks (LPNN) and the Hidden Control Neural Networks (HCNN). Two novel theoretical results supported with Mathematica experiments are presented. First, it is proved that for every HCNN model there exist an equivalent LPNN model. Second, it is shown that the set of input-output functions of an LPNN model is strictly larger than the set of functions of an equivalent HCNN model. Therefore, when using equal architecture of the canonical building blocks (MLPs) for the LPNN and HCNN models, the LPNN represent a superset of the approximation capabilities of the HCNN models.

Cross-Language Text-Independent Speaker Identification

Authors:
G. Durou, *Faculte Polytechnique de Mons (BELGIUM)*
F. Jauquet, *Royal Military Academy (BELGIUM)*

ABSTRACT

In this paper, we investigate the influence of the language on the text-independent speaker recognition. For this purpose, we have used several automatic text-independent speaker recognition methods (Multivariable Auto-Regression, Vector Quantization and Histogram Classifiers). To measure the effect of the language, we have applied these methods on the POLY-COST 250 multi-language database. Among the different experiments, we have seen that the speaker recognition performance has not been really affected if the language used during the training and the test is different.

A Novel Model For Phoneme Recognition Using Phonetically Derived Features

Authors:

N. Harte, S. Vaseghi, P. McCourt, *The Queen's University of Belfast (N.IRELAND)*

ABSTRACT

This paper presents work on the use of segmental modelling and phonetic features for phoneme based speech recognition. The motivation for the work is to lessen the effects of the IID assumption in HMM based recognition. The use of phonetic features which are derived across the duration of a phonetic segment is discussed. In conjunction with the use of these features, a hybrid phoneme model is introduced. In a classification task on the TIMIT database, these features are capable of outperforming standard HMM. The extension of the work to recognition is presented in detail. The challenges are identified and a novel algorithm presented for recognition based on phonetic features and the hybrid phoneme model. The approach is built around a segmentation hypothesis approach employing pruning at a number of levels.

Diverse Processing In Cochlear Spaced Sub-Bands For Multi-Microphone Adaptive Speech Enhancement In Reverberant Environments

Authors:
A. Hussain, D. R.Campbell, T. J. Moir, *University of Paisley (UK)*

ABSTRACT

A multi-microphone sub-band adaptive speech enhancement scheme using a human cochlear model is presented. The effect of distributing the sub-bands non-linearly as in humans is investigated. A new robust metric is developed in order to automatically select the best form of diverse processing within each sub-band. Comparative results achieved in simulation experiments demonstrate that the proposed scheme employing diverse processing in cochlear spaced sub-bands is capable of significantly outperforming conventional noise cancellation schemes.

A General-Tree-Structured Vector Quantizer For Image Progressive Coding

Authors:
L. Y. Tseng, S. B. Yang, *National Chung Hsing University (TAIWAN)*

ABSTRACT

Recently, several tree-structured vector quantizers had been proposed. But almost all trees used are binary trees and hence the training samples contained in each node are forced to be divided into two clusters artificially. We present a general-tree-structured vector quantizer that is based on a genetic clustering algorithm. This genetic clustering algorithm can divide the training samples contained in each node into more natural clusters. A distortion threshold is used to guarantee the quality of coding. Also, the Huffman coding is used to achieve the optimal bit rate after the general-tree-structured coder was constructed. Progressive coding can be accomplished by given a series of distortion thresholds. An experiment result is given to illustrate the performance of this vector quantizer on image progressive coding. A comparison of the performance of this vector quantizer and the other two tree-structured vector quantizers is also given.

Rate Distortion Optimal Contour Compression Using Cubic B-Splines

Authors:
J. Zaletelj, *University of Ljubljana (SLOVENIA)*
R. Pecci, *University of Firenze (ITALY)*
F. Spaan, A. Hanjalic, R.L. Lagendijk, *Delft University of Technology (NETHERLANDS)*

ABSTRACT

Object contours resulting from segmenting images or video frames can be efficiently encoded using B-spline functions. An unsolved problem is how to divide a given contour into segments such that the resulting compression is optimal in rate-distortion sense. In this paper we describe two techniques for finding a close-to-optimal knot assignment. The first technique prunes an accurate B-splines approximation until the desired rate is achieved. The second technique analyzes the curvature function of the original contour to obtain a suboptimal knot assignment. The resulting algorithms are compared experimentally.

Coding Of Arbitrarily Shaped Video Objects Using B-Spline Wavelets

Authors:

M. Sepponen, *University of Oulu (FINLAND)*

V. Koivunen, *Tampere University of Technology (FINLAND)*

ABSTRACT

This paper addresses the problem of encoding and decoding image patches in a model-based image coding scheme. Any arbitrarily shaped video object can be represented using a triangular mesh. A method for coding triangular image blocks using B-spline wavelet filter banks is developed here. Triangles are considered degenerate rectangular blocks hence the same coding can be used for rectangular blocks as well. Triangles at the boundary of a video object are allowed to have curved boundaries in order to have more accurate shape representation using fewer elements. In the decoding end, patches are warped using affine motion estimates. Examples are given using real image data.

Fast QUASI-DCT Algorithm For Shape-Adaptive DCT Image Coding

Authors:
R. Stasinski, *Hogskole i Narvik (NORWAY)*
J. Konrad, *INRS-Telecommunications (CANADA)*

ABSTRACT

In this paper we develop a new variant of the shape-adaptive discrete cosine transform (SA-DCT) recently proposed by Sikora and Makai and currently considered for MPEG-4 as a texture compression engine. We are concerned with the computational complexity of the SA-DCT; although its complexity is acceptable in the context of 8x8 (boundary) blocks as proposed for MPEG-4, it is very high for a true region-based coding where complete regions (e.g., 100 by 100 pixels) need to be processed. We adapt the original SA-DCT scheme by replacing the usual DCT with a quasi-DCT for which some basis functions are identical and some similar to those of the DCT. We test the new method and compare it numerically in terms of the basis restriction error as well as subjectively on some natural images. We conclude that the new method's energy compaction performance is slightly inferior to that of the SA-DCT, but its computational complexity is highly reduced.

Fast Approximation Of DCT Kernel In JPEG

Authors:
A. Marcek, G. Rozinaj, *Slovak University of Technology (SLOVAKIA)*

ABSTRACT

In this paper a new approach to the approximation of Modified Integer Cosine Transform (MICT) without multiplication will be introduced. An application of this method in JPEG standard for a compression of images is analysed and discussed. An error analysis of the approximation shows very good results. Nevertheless, an accuracy of the algorithm vs. speed can be very easily optimised. Combination of standard JPEG and JPEG based on proposed algorithm has been analysed. An example at the end of this paper shows practical experience with the method.

Statistical Modelling Of Full Frame DCT Coefficients

Authors:
M. Barni, F. Bartolini, A. Piva, F. Rigacci, *Universita di Firenze (ITALY)*

ABSTRACT

Because of the increasing influence it has in image watermarking applications, the estimation of the distribution shape of full frame DCT coefficients is here addressed. Based on previous analyses on block-DCT, the coefficients are first assumed to follow a Generalized Gaussian distribution. The shape parameter ν is then evaluated according to the maximum likelihood criterion applied to a set of 170 natural images. The analysis has been further validated by using the χ^2 test-of-fit criterion. In contrast to the block-DCT case, experimental results prove that full frame DCT coefficients can be modeled without appreciable loss of performance by a Laplacian density function having variance decreasing with frequency.

Quality Scalable Coding Of Selected Region

Authors:

W.-J. Kim, J.-W. Yi, S.-D. Kim, *Korea Advanced Institute of Science and Technology(KOREA)*

ABSTRACT

If a region is semantically more important than others, it is appropriate that a image compression scheme is capable of handling the regional semantic difference because the information loss of the interested region is more severe. We propose the quality scalable coding with its model by introducing the quality scale parameter. It is more extended and generalized image compression philosophy than a conventional coding. As an implementation of the proposed quality scalable coding, a H.263 based scheme is presented. This scheme can control the temporal and spatial quality efficiently, and improve the reconstructed image quality of the interested region..

Lossless Compression Of Images By A Search Order Based Entropy Coding

Authors:
J. Jiang, *University of Glamorgan (UK)*
C.V. Brett, *Loughborough University (UK)*

ABSTRACT

In this paper, we propose a simple search order entropy coding algorithm to compress grey level scale images on lossless basis. Extensive experiments are carried out in comparison with JPEG lossless compression mode. The results reveal that for the majority of image samples tested, the proposed algorithm outperforms JPEG. This work provides a competitive alternative yet simple solution to lossless image compression.

A Context-Based Recursive Non-Linear Interpolation For Near-Lossless Coding Of X-Ray Images

Authors:
B. Aiazzi, S. Baronti, F. Lotti, *“Nello Carrara" IROE - CNR (ITALY)*
L. Alparone, *University of Florence (ITALY)*

ABSTRACT

This paper addresses quality issues in Medical Image compression and proposes an approach toachieve near-lossless storage of digitized X-ray plates. An image is normalized to the standard deviation of its noise, which is estimated in an unsupervised fashion. The resulting bitmap is encoded without any further loss. The compression algorithm proposed is based on a two-stage recursive interpolation exploiting nonseparable median filtering a on a quincunx grid. The advantage is twofold: interpolation is performed from all error-free values and is unlikely to occur across edges, thereby reducing the coding cost of the outcome residuals. In addition, classification based on spatial context is employed to improve entropy coding. The scheme outperforms other established methods when applied to X-ray images.

A Study Of A Lossless Image Compression Algorithm Using Adaptive Prediction And Context-Based Entropy Coding

Authors:
G. Deng, *La Trobe University, (AUSTRALIA)*

ABSTRACT

In this paper a context based lossless image compression algorithm is presented. It consist of an adaptive median-FIR predictor, a conditional context based error feed back process and a new error representation. The prediction error is encoded by a context-based arithmetic encoder. Experimental results show that for a set of 18 images of different kinds, the compression performance of the proposed algorithm is very close to that of CALIC and is better than LOCO and S+P. This paper also presents an algorithmic study of the proposed algorithm. The contribution of each of the building blocks to the compression performance is studied. It has been shown that these building blocks can be incorporated into further development of lossless image compression algorithms.

Rate-Distortion Analysis
Of Nonlinear Quantisers For Video Coders

Authors:

Y.-S. Saw, *Next Generation Telecomm's Div. (KOREA)*

P. M. Grantzand, J. M. Hannahz, *University of Edinburgh (UK)*

ABSTRACT

Quantisation is used in video coders such as MPEG in association with rate control scheme to regulate the data rate of compressed video bit stream entering the transmission buffer. When the transmission data rate is limited the quantiser has a crucial effect on video data rate and video quality. The quantiser step size is generally
determined by a linear non-adaptive method with respect to the buffer occupancy. In this paper, we investigate in the framework of rate-distortion theory two adaptive nonlinear quantiser control functions, sigmoidal and unimodal, which achieve superior video rate control performance while maintaining similar video quality to the linear case. Their performance for video rate uctuation has also been analysed in both analytic and experimental ways.

Hierarchical MPEG 2 Video Transmission On ADSL For A Higher Quality Of Service (QoS)

Authors:
M. Colin, M. Gharbi, M. Gazalet, *Universite de Valenciennes (FRANCE)*
C. Modlin, *Texas Instruments Broadband Access Group (Amati)*

ABSTRACT

This paper considers the problem of improving the QoS of MPEG 2 video transmission over ADSL, which is an emerging technology that permits high bit rate transmission over subscriber lines. For this purpose, we propose to combine the use of the DP (Data Partitioning) MPEG 2 scalable mode and a modified coded DMT based ADSL modem that provides two different Bit Error Rates (BERs) for transmission: important video data such as headers and synchronization modules are better protected than less important data, resulting in a QoS improvement.

Fast Color Transformation By Means Of Hierarchical Table Lookups

Authors:
P.L. Dragotti,A.R.P. Ragozini, *Universita Federico II di Napoli (ITALY)*

ABSTRACT

Virtually all image compression techniques are designed to minimize the popular and easy-to-compute mean square error (MSE) measure of distortion. However, for images represented in the RGB color space the MSE is a poor match to the human visual system. Better results could be obtained by transforming the input image in a perceptually uniform color space such as the C.I.E. L*u*v*, but such a color conversion is very time-consuming and for this reason it is rarely used. In this paper we propose to carry out the color space conversion by means of a high-rate hierarchical vector quantizer which requires only table lookup operations. As a consequence, the computational burned is reduced to a minimum and the transformation can be implemented efficiently both in hardware and software. Experimental results show that the proposed fast color space transformation, although lossy, improves the visual quality of compressed images , as compared to that of images compressed without any color conversion.

Robust Tracker
Of Small, Fast-Moving Low-Contrast Targets

Authors:
D. Davies, P. Palmer, *University of Surrey (UK)*
M.Mirmehdi, *University of Bristol (UK)*

ABSTRACT

We present a multiresolution adaptive wavelet transform to locate small low-contrast targets. Our approach expands upon methods which use adaptive filters to remove noise to produce a near real-time robust tracker using a specially adapted Kalman filter. This generates a small set of hypotheses to test. Incorrect hypotheses are removed using an interest operator founded on the error covariance generated by the Kalman filter. We demonstrate the technique using some experimental results.

Accurate Motion Interpolation
By Using A Region Based Motion Estimator

Authors:
R. Lancini, M. Ripamonti, S. Tubaro, P. Vicari, CEFRIEL, *Politecnico di Milano (ITALY)*

ABSTRACT

In this paper an effective motion compensated image interpolation techniques is presented. The proposed algorithm has been developed for the interpolation of missing frames in image sequence and it is based on two principal elements. The first is a region-based motion estimation and representation technique, which combines, for each known image, an approximate initial motion field estimate and an initial image over-segmentation (obtained using both luminance andchrominance information) to produce a very accurate affine-model regularized motion field. The proposed algorithm uses a robust identification technique for the estimation of the motion parameters associated to each region. The second important element is the use of an interpolation technique, that starting from the available motion information, defines, for each point of the image to be interpolated a suitable reconstruction strategy taking into account for different situation that can appear (for example a moving object can occlude either a stationary background or an another object and so on). The principal aim of the proposed algorithm is the reconstruction of the missing frames in a reasonable way, without introducing significant artifacts and assuring the pleasant representation of object displacement.

A Cooperative Top-Down/Bottom-Up Technique For Motion Field Segmentation

Authors:

R. Leonardi, P. Migliorati, G. Tofanicchio, *University of Brescia (ITALY)*

ABSTRACT

The segmentation of video sequences into regions underlying a coherent motion is one of the most useful processing for video analysis and coding. In this paper, we propose an algorithm that exploits the advantages of both top-down and bottom-up techniques for motion field segmentation. To remove camera motion, a global motion estimation and compensation is first performed. Local motion estimation is then carried out relying on a traslational motion model. Starting from this motion field, a two-stage analysis based on affine models takes place. In the first stage, using a top-down segmentation technique, macro-regions with coherent affine motion are extracted. In the second stage, the segmentation of each macro-region is refined using

a bottom-up approach based on a motion vector clustering. In order to further improve the accuracy of the spatio-temporal segmentation, a Markov Random Field (MRF)-inspired motion-and-intensity based refinement step is performed to adjust objects boundaries.

Motion Estimation And Modeling For Video Sequences

Authors:
C. Cafforio, E. Di Sciascio, C. Guaragnella, *Politecníco di Bari (ITALY)*

ABSTRACT

Object based video coding is gaining considerable attention as it should allow possible object-based interactivity and the elimination of mosquito and block artifacts typical of block coding. Algorithms ensuring a good scene segmentation based only on motion information are still far from reliable. In this work we present an algorithm for jointly estimating motion fields and obtaining a segmentation of the scene in coherent motion areas, using only information obtained by block matching. A quadratic model is used to represent inter block dependencies of motion estimates. Region growing is used to gather blocks endowed with compatible matching functions and produce an initial scene segmentation. Refinements are obtained with a subsequent iterative procedure.

Experimental results show a good correspondence of the motion field segmentation with the objects in the scene and prediction errors close to those of block matching, but with a much more regular motion field.

The Impulse RETINA : A Smart Velocity Sensor

Authors:
M. Emmanuel, C. Olga, F. Alain, *Université du Havre (FRANCE)*
C. Christophe, *Esigetel (FRANCE)*

ABSTRACT

The model Retina developed by the laboratory LACOS (University of le Havre, France) in collaboration with the A.B. Kogan research institute for neurocybernetics (Rostov-state University, Russia) is a retinal neural network permitting a pre-processing of foveal vision. This model is adaptative and its multi-resolution allows it to detect a large scale of velocities. We use the Retina to detect the motion and extract the velocity vector of a time sequence image. From impulse output signals of Retina we extract the pertinent parameters which encode the motion by time frequency analysis. The final aim is to implement this sensor on a mobile robot.

A Fast Block Matching Motion Estimation Algorithm Based On Simplex Minimisation

Authors:
M. E. Al-Mualla, N. Canagarajah, D. R. Bull, *University of Bristol (UK)*

ABSTRACT

A fast block matching motion estimation algorithm is presented. The algorithm is based on a generic unconstrained optimisation technique called *simplex minimisation* (SM). In order to apply this method to the constrained minimisation problem of block matching motion estimation, a suitable initialisation procedure, termination criterion, and constraints on the independent variables of the search, are proposed. The algorithm is demonstrated to outperform other fast block matching motion estimation techniques providing better reconstruction quality, a smoother motion field and reduced computational complexity.

Motion Estimation Based On Triads Of Gabor Filters: DSP Board Implementation

Authors:
A. Spinéi, *INPG. (FRANCE)*
D. Pellerin, J. Hérault, *INPG. - Université Joseph Fourier (FRANCE)*

ABSTRACT

The quality and rapidity of motion estimation in image sequences are fundamental in many applications as for instance in artificial vision and three-dimensional scene reconstruction. Among all the existing techniques, we are particularly interested in energy-based methods which are known to furnish high-quality results but usually require intensive calculation. We present a fast energy-based method which combines in a direct manner the energetic responses of Gabor spatio-temporal filters organized in triads and we describe an implementation of this technique on a general purpose DSP board. Our hardware implementation attains a reasonably fast output rate (few images/second). These results open interesting perspectives for real-time implementations, such as in mobile robotics.

Hierarchical Estimation Of Optical Flow Using Spectral Energy Method

Authors:
T. Koike, N. Hamada, *Keio University (JAPAN)*
K. Kondo, *Himeji Institute of Technology (JAPAN)*

ABSTRACT

The estimation of optical flow is an important problem in the dynamic image analysis. There are various methods of estimating the optical flow. The gradient method and matching method are based on the operation in the space-time domain. While spectral energy method is based on the frequency analysis and has robustness against noise. In this paper, we propose a hierarchical estimation of the optical flow using spectral energy method. In the proposed method, we make a set of hierarchical images for each frame. A global motion which is estimated at the coarsest resolution image sequences is corrected by using finer resolution image sequences. Consequently we can estimate both global and local motion accurately with robustness to noise.

Visual Module Integration For Optical Flow Estimation

Authors:

L. Bediniy, A. Cannatay, E. Salernoy, A. Tonazziniy, *Istituto di Elaborazione della Informazione - CNR (ITALY)*

M. Ferraroz, *Universita di Torino (ITALY)*

ABSTRACT

A technique to integrate gradient-based and feature-based modules to estimate the optical flow from a pair of images is proposed. The integration strategy is based on a Bayesian approach, where the optical flow is evaluated as the minimizer of a suitable posterior energy function, containing all the gradient and feature information on the problem. The capability of the technique to constrain the displacement in the neighbourhoods of motion discontinuities has been tested.

A Vector Based Approximation Of KLT And Its Application To Face Recognition

Authors:
N. Tsapatsoulis, V. Alexopoulos, S. Kollias, *National Technical University of Athens (GREECE)*

ABSTRACT

A face recognition scheme is proposed in this paper. The method utilizes a vector based approximation of KLT (VKLT) which eliminates the large memory demands and the singularity of the covariance matrix matters that are the main drawbacks of the "eigenface method" (a face recognition scheme based on KL transform). The reconstruction error of VKLT approaches the one of KLT preserving also its data dependence, which is important for discriminating face images from non-face images. Moreover, the greater advantage of VKLT over KLT is that of keeping intra-class variations small and consequently increasing the robustness of the face recognition system.

Recognition Of Rotated And Scaled Textures Using 2-D Ar Modeling And The Fourier-Mellin Transform

Authors:

C. Cariou, O. Alata, C. Rosenberger, J.-M. Ogier, K. Chehdi, *LASTI - Groupe Image, (FRANCE)*

ABSTRACT

In this communication, we address the problem of the recognition and classification of rotated and scaled stochastic textures. We propose an extension of previous works, which are mostly based upon the use of the Fourier transform for the derivation of translation invariant statistics. More precisely, we suggest the sequential use of a 2-D high resolution spectral estimate, called Harmonic Mean power spectrum density (HM PSD), which presents a good trade off between reliability and complexity, with the Fourier-Mellin transform from which rotational and scaling invariants can be derived. Experimental results on rotated and scaled textures are presented that show the efficiency of this new technique.

Point Pattern Matching Using A Genetic Algorithm And Voronoi Tessellation

Authors:
M. Tico, C. Rusu, *Tampere University of Technology (FINLAND)*

ABSTRACT

Point pattern matching problem consists in identifying similar point patterns in two point sets which differ one each other in scale, orientation angle or position. A new objective function for the problem of point pattern matching is proposed here. The function scores not only the exact matching situations between patterns, but also the inexact ones. It has only one global maximumin the desired solution, and hence implosion cases which occur for very low scale factors are avoided. An efficient algorithm for the evaluation of the objective function is proposed. The algorithm requires a preprocessing stage to label the Voronoi regions of one of the point sets. Then, a genetic algorithm is used to maximize the proposed objective function.

Knowledge Based Interpretation Of Aerial Images Using Multiple Sensors

Authors:
R. Tonjes, C.–E. Liedtke, *Universitat Hannover (GERMANY)*

ABSTRACT

A knowledge based approach for the interpretation of aerial images is presented that combines cues from multiple sensors (visual, infrared, SAR). The sensor fusion is applied at object level. This allows to use prior knowledge to increase the separability of the classes. The prior knowledge is represented explicitly using semantic nets. Interpretation exploits the semantic net to control the sequence of sensor fusion mixing bottom–up and top–down strategies. The presented approach addresses the problem of uncertain and imprecise sensor data by judging the different cues based on possibility theory. Competing interpretations are stored in a search tree. An A*–algorithm selects the most promising, i.e. best judged, interpretation for further investigation.

Automatic Generation Of A VLSI Parallel Architecture For QRS Detection

Authors:
A. Koulouris, N. Koziris, T. Andronikos, G. Papakonstantinou, P. Tsanakas, *National Technical University of Athens (GREECE)*

ABSTRACT

QRS is the dominant complex in the electrocardiogram (ECG). Its accurate detection is of fundamental importance to reliable ECG interpretation and hence, to all systems analyzing the ECG signal (e.g. Heart-monitoring). Syntactic methods are a very powerful tool for QRS detection, since they can easily describe complex patterns, but their high computational cost prevents their implementation for real time applications. In this paper, we present a VLSI architecture for ECG signal processing, automatically derived using a nested loop parallelization method. This architecture detects the QRS complex by parsing the corresponding to the ECG signal input string, based on an attribute grammar describing it.

A Fuzzy Logic Filter for Coherent Detection in Mobile Communication Receivers

Authors:

A.Pérez-Neira, M.A.Lagunas, A. Jové, A. Artés, *Universitat Politecnica de Catalunya (SPAIN)*

ABSTRACT

This paper proposes and demonstrates the use of a fuzzy logic filter for carrier phase synchronization in digital mobile communication receivers. The use of *a priori* knowledge of the dynamics of the phase in mobile fading channels allows the fuzzy filter to achieve fast phase tracking over Rician and Rayleigh flat-fading channels, as well as achieving quick acquisition. Additionally, a fuzzy CEMAC (Cerebellar Model Arithmetic Computer) architecture is proposed to increase the processing speed. The low computational requirements and the satisfactory performance of the proposed system, comparable or better than the Kalman filter for low SNR and low sampling rate, presents the fuzzy filter as a good alternative to other fading compensation techniques.

Viterbi Algorithm With Embedded Channel Estimation Based On Fuzzy Inference

Authors:
L. Favalli, P. Savazzi, *Universita di Pavia (ITALY)*
A. Mecocci, *Universita di Siena (ITALY)*

ABSTRACT

This paper describes a novel approach to the problem of fading channels estimation. Specifically, the proposed system makes use of fuzzy logic in the computation of the metrics for an MLSE equaliser for GMSK signals in GSM typical environments. The comparison with the traditional Viterbi algorithm is performed using the channel models specified by ETSI and shows that the proposed system performs better under all channel conditions: for the same SNR the improvement is about a half decade in BER or, conversely, the same BER can be obtained with a SNR 4dBs lower than using the adaptive Viterbi algorithm alone.

Co-Channel Interference Suppression Using A Fuzzy Filter

Authors:
S. K.r Patra, B. Mulgrew, *University of Edinburgh (UK)*

ABSTRACT

This paper investigates the problem of channel equalisation in digital cellular radio (DCR) application. DCR systems are affected by cochannel interference (CCI), intersymbol interference (ISI) in presence of additive white Gaussian noise (AWGN). Here we propose a fuzzy equaliser to equaliser communication channels with these anamolies. This equaliser performs close to the optimum Bayesian equaliser with a substantial reduction in computational complexity. The equaliser is trained with supervised and unsupervised scalar clustering techniques in sequence, and consist of a fuzzy equaliser with a preprocessor for CCI compensation. Simulation studies have demonstated the performance of the proposed technique.

Blind Equalization Of Nonlinear Channels Using Hidden Markov Models

Authors:
K. Georgoulakis, S. Theodoridis, *University of Athens (GREECE)*

ABSTRACT

A novel blind channel equalizer is proposed which is suitable both for linear and nonlinear channels. The proposed equalizer consists of the following three steps: a) identification of the clusters formed by the received data samples, via an unsupervised learning clustering technique, b) labeling of the identified clusters, by using a Hidden Markov Modeling (HMM) of the process and c) channel equalization by means of a Cluster Based Sequence Equalizer. The performance of the equalizer is investigated for a variety ofchannels (minimum/non-minimum phase, linear/nonlinear channels).

Symbol-By-Symbol Mobile Radio Channel Equalization Using The K-Nn Classifier

Authors:
P. Savazzi, L. Favalli, E. Costamagna, *Universita di Pavia (ITALY)*
A. Mecocci, *Universita di Siena, (ITALY)*

ABSTRACT

This paper illustrates an implementation of a GSM receiver in which channel equalization and demodulation are realised by means of the Nearest Neighbor (NN) classifier algorithm. The most important advantage in using such techniques is the significant reduction in terms of computational complexity compared with the MLSE equalizer. No explicit channel estimation need be carried out, and the whole process involves a simple symbol-by-symbol decision procedure. The performance of the proposed receiver, evaluated through a channel simulator for mobile radio communications, is compared with the results obtained by means of a 16 states Viterbi algorithm and other sub-optimal receivers. Despite the simplicity of the receiver, performance degradation is kept within the limits imposed by the GSM specifications.

Reduced-Complexity Decision-Feedback Equalizer For Nonlinear Channels

Authors:
F. J. Gonzalez-Serrano, *Universidad de Vigo (SPAIN)*
F. Perez-Cruz, A. Artes-Rodriguez, *Ciudad Universitaria (SPAIN)*

ABSTRACT

This paper deals with the compensation for nonlinear distortions introduced by power-efficient amplifiers on linear modulations by means of equalization. In our approach, we employ a Decision-Feedback Equalizer (DFE) based on the Generalized Cerebellar Model Arithmetic Computer. The new scheme is compared with the conventional Linear DFE, the Volterra and the Multi-Layer Perceptron-based DFE in terms of their convergence rates, Bit-error rates and Signal-to-Noise Ratio degradation.

Source Separation Without Explicit Decorrelation

Authors:

O. Macchi, *LSS, Supelec (FRANCE)*
E. Moreau, *S-GESSY, ISITV (FRANCE)*

ABSTRACT

The contrast approach has become classical for separating independent sources. It involves whitening (with decorrelation) as a pre-processing step. Here we propose a new contrast applicable to correlated signals, as long as they have unit power. The corresponding system involves output Automatic Gain Controls (AGC). An adaptive contrast maximization is proposed. Its achievements are shown to outperform the adaptive implementation of the classical pre-whitened contrast.

Blind Separation Of Polarised Waves

Authors:
J.L. Lacoume, F. Glangeaud, J. Mars, *LIS INPG (FRANCE)*

ABSTRACT

In a great many situations polarised waves are received on vectorial sensors. These waves are composed of several polarised sources associated with different modes of propagation within the investigated medium. The objective of our work is to separate the polarised sources without *a priori* knowledge of the polarisation of each source and to apply this technique to elastic waves observed in a seismic sounding.

BLIND MULTICHANNEL ESTIMATION EXPLOITING THE FINITE SYMBOL ALPHABET

Authors:
J. Ayadi, D. T.M. Slock, *Institut EURECOM (FRANCE)*

ABSTRACT

Unlike the recent works dealing with purely blind channel estimation algorithms that are based on the second-order statistics of the received signal, in this paper we address the exploitation of the finite alphabet of the transmitted symbols to improve the blind estimation performance of the channel. The incorporation of the finite alphabet nature leads the symbols present in the problem to act as a training sequence for the channel estimation. Hence, a blind approach that exploits the symbol alphabet outperforms its purely blind version. We propose to incorporate the prior knowledge of the finite alphabet by combining a purely blind channel estimation criterion with a decision-directed linear MMSE equalization criterion. This combined criterion corresponds to an optimally weighted least-squares approach. Simulation results demonstrate that significant improvement can be obtained by exploiting the finite symbol alphabet.

Robustness Of Least-Squares And Subspace Methods For Blind Channel Identification/Equalization With Respect To Channel Undermodeling

Authors:

A. P. Liavas, P. A. Regalia, J.-P. Delmas, *Institut National des Telecommunications (FRANCE)*

ABSTRACT

The least-squares and the subspace methods are well known approaches for blind channel identification/equalization. When the order of the channel is known, the algorithms are able to identify the channel, under the so-called *length* and *zero* conditions. Furthermore, in the noiseless case, the channel can be perfectly equalized. Less is known about the performance of these algorithms in the cases in which the channel order is underestimated. We partition the true impulse response into the significant part and the tails. We show that the m-th order least-squares or subspace methods estimate an impulse response which is “close" to the m-th order significant part of the true impulse response. The closeness depends on the diversity of the m-th order significant part and the size of the “unmodeled" part.

Identifiability Conditions For Blind And Semi–Blind Multichannel Estimation

Authors:

E. de Carvalho, D. T.M. Slock, *Institut EURECOM, (FRANCE)*

ABSTRACT

We investigate the identifiability conditions for blind and semi-blind FIR multichannel estimation in terms of channel characteristics, data length and input symbol excitation modes. Parameters are identifiable if they are determined uniquely by the probability distribution of the data. Two models are presented: in the deterministic model, both channel coefficients and input symbols are considered as deterministic quantities and in the Gaussian model, the input symbols as Gaussian random variables. The Gaussian model appears more robust than the deterministic one as it requires less demanding identifiability conditions. Furthermore, semi-blind methods appear superior to blind methods as they allow the estimation of any channel with only few known symbols.

Blind And Informed Cyclic Array Processing For Cyclostationary Signals

Authors:

P. Chevalier, A. Maurice, *Thomson-CSF Communications (FRANCE)*

ABSTRACT

Limiting the analysis to the exploitation of the second order statistics of the complex data, the optimal Spatio-Temporal (ST) receivers in stationary contexts are Linear and Time Invariant (TI). However, for (quasi)-cyclostationary observations, it is now well-known that the optimal ST complex receivers become (poly)-periodic (PP) and, under some conditions of non circularity, Widely Linear (WL). Using these results and the fact that PP filtering is equivalent to FREquency SHifted (FRESH) filtering, the purpose of this paper is to present a new ST, PP and WL receiver structure, very useful for applications such as passive listening or source separation, taking into account the potential (quasi)-cyclostationarity and non circularity properties of the observations. This new cyclic receiver may be implemented either blindly or from the a priori knowledge or estimation of the useful signal steering vector. The performance computation of this new cyclic receiver shows off the great interest of the latter in cyclostationary contexts and its great capability of interferences rejection even from a one sensor reception.

A Numerical Algorithm For Implementing The Single-Stage Maximization Criterion For Multichannel Blind Deconvolution

Authors:

S. Ohno, Y. Inouye, *Shimane University (JAPAN)*
K. Tanebe, *Osaka University (JAPAN)*

ABSTRACT

We give an algorithm for the single-stage maximization criterion to attain multi-channel blind deconvolution. This criterion determines the coefficients of equalizers for all channels simultaneously. However, the original maximization criterion has many constrains so that it is difficult to directly implementitas a numerical algorithm. By exploiting pre-whitening and lattice representations of paraunitary systems, we can reduce the original maximization problem into a simple constraint-free one and then present a stochastic gradient algorithm. system. Exploiting the factorization of FIR paraunitary systems [8], we reduce the original maximization problem into a simple one without constrains and then present a stochastic gradient algorithm.

DFT based Optimal Blind Channel Identification

Authors:
C.Becchetti, G.Jacovitti, G.Scarano, *Università di Roma "La Sapienza" (ITALY)*

ABSTRACT

Optimal solutions for channel identification from oversampled Pulse Amplitude Modulated signals are presented. While (large sample) optimal identification derived in the time domain involves solution of matrix weighted linear systems of equations solved in a Least Squares sense, using Discrete Fourier Transforms of the received data, it is shown that optimal solutions are rather obtained through scalar weighted Least Squares. Since the matrix of weights is not a priori known, estimation is generally performed in two steps. Here, we describe a less computational demanding single step estimation procedure. In the small samples case, the single step estimation is slightly less accurate than two step procedures.

Blind Identification Of IIR Model Based On Output Over-Sampling

Authors:
L. Sun, W. Nisizawa, W. Liu, A. Sano, *Keio University (JAPAN)*

ABSTRACT

This paper deals with a blind problem for an IIR model in a time domain. Based on an output over-sampling scheme, the proposed algorithm can estimate parameters of an alternative multi-output model description first, then the parameters of original model can be obtained later. It can be clarified that an IIR model can be identified by using over-sampling scheme.

Blind Multichannel Equalization with Controlled Delay

Authors:
A. Touzni, I. Fijalkow, *ETIS / ENSEA-UCP (FRANCE)*

ABSTRACT

In this contribution, we address a new second order approach for multichannel zero forcing equalization with controlled delay. The method basically exploits the second order whiteness input signal properties and condition of left invertibility of the multichannel. Channel identification is investigated in a second step. In comparison of existing methods, the proposed method has the interesting properties to involves some robustness with respect to channel order estimation, and similar complexity than subspace-like methods.

A Disjoint Set Algorithm For The Watershed Transform

Authors:

A. Meijster, J. B.T.M. Roerdink, *University of Groningen (NETHERLANDS)*

ABSTRACT

In this paper the implementation of a watershed transform based on Tarjan's Union-Find algorithm is described. The algorithm computes the watershed as defined by Meyer in [4]. The algorithm consists of two stages. In the first stage the image to be segmented is transformed into a lower complete image, using a FIFO-queue algorithm. In the second stage, the watershed of the lower complete image is computed. In this stage no FIFO-queues are used. This feature makes parallel implementation of the watershed transform much easier.

Electrostatic Formulation For Adaptative Dilation

Authors:

O. Lavialle, *Equipe Signal/Image ENSERB (FRANCE), ENITA de Bordeaux (FRANCE)*

P. Baylou, *Equipe Signal/Image ENSERB (FRANCE)*

ABSTRACT

We introduce a new concept of caricature in order to exacerbate the main morpho-logical characteristics of objects. Here, the term caricature implies that we are looking for a method leading to a simplification or an exaggeration of the particularities of objects. Our technique is based on the shape of the object considered as an insulator charged with unipolar electricity. The electric forces exerted on each pixel belonging to the contour create the desired transformation. The contextual information is also taken into account through the definition of attractive forces between objects. We consider only the case of the dilatation of objects in a binary image. The theory is illustrated through a problem concerning connection of objects in a binary image and through an example of contour closing.

Textured Regions Extraction Using Matching Pursuit Method

Authors:
T. Kayanuma, N. Hamada, *Faculty of Science and Technology Keio University (JAPAN)*

ABSTRACT

This paper shows the method of extracting textured regions from an image using matching pursuit method. Textured region is made by iteration of the texture component that is called “texton”, and its dominant spectrum is in the medium frequency. Matching pursuit algorithm can model images with the 2D wavelets and describe the local properties of the image. The textured region is extracted as the regions that have same properties as ones in its neighborhood.

Hexagonal Wavelet Decomposition For Texture Characterization

Authors:
A. Mojsilovic, *Bell Laboratories, (USA)*
S. Markovic, M. Popovic, *University of Belgrade, (YUGOSLAVIA)*

ABSTRACT

This paper demonstrates new possibilities in texture characterization, offered by hexagonal spectral decomposition schemes. We have described, theoretically analyzed and compared image decompositions based on separable and hexagonal sampling. For both decompositions feature extraction is performed with the filters derived from the same ID wavelet family, and textures are characterized by a set of channel variances calculated at the output of the corresponding filter bank. Experiments with a large number of textures have shown that in standard working conditions both decompositions perform equally, but that hexagonal transform is optimal for fine structural analysis and performs well in the presence of noise. Finally, hexagonal decomposition was incorporated in a simple region growing technique, showing its use in a texture segmentation.

Optimised Wold-Like Decomposition
Of 2D Random Fields

Authors:
P. Campisi, A. Neri, *Universita di RomaTre (ITALY)*
G. Iacovitti, *Universita di Roma "La Sapienza" (ITALY)*

ABSTRACT

In this paper we address the problem of image texture modeling. In particular we adopt here a 2D Wold decomposition that separates a texture into regular and chaotic parts, allowing for simple texture parameterization, and for explicit extraction of periodic structures. The identification scheme based on the proposed decomposition improves the accuracy of the estimation of the model parameters as well as the visual resemblance of the model with respect to samples, owing to compliance of the chaotic component with the Julesz's conjecture.

A New Algorithm CGA For Image Labeling

Authors:
G. D. Guo, S. de Ma, *Chinese Academy of Sciences (CHINA)*
S. Yu, *INRIA (FRANCE)*

ABSTRACT

Many image analysis and computer vision problems can be formulated as a scene labeling problem in which each site is to be assigned a label from a discrete or continu- ous label set with contextual information. In this paper we present a new labeling algorithm based on the game theory. More precisely, we use Markov random fields to model images, and we design an n-person cooperative game which yields a deterministic optimization algorithm. Experimental results show that the algorithm is efficient and effective, exhibiting very fast convergence, and producing better result than the recently proposed non-cooperative game approach. We also compare this algorithm with other labeling algorithms on real world and synthetic images.

Maximum Entropy Contouring And Clustering For Fractal Attractors With Application To Self–Similarity Coding Of Complex Texture

Authors:
K. Kamejima, *Osaka Institute of Technology, (JAPAN)*

ABSTRACT

Based on stochastic modeling of self-similarity processes. capturing probability for not-yet-identified pattern is represented by multi-scale image. Through detecting level set and local maxima of the multi-scale image, smooth contours and finite feature pattern are estimated for fractal attractors. Estimated feature pattern is clustered within the framework of entropy maximization to design a system of reduced affine mappings with fixed points on boundary. Geometric-structural consistency of designed code is verified through computer simulation.

Region Based Analysis Of Video Sequences With A General Merging Algorithm

Authors:
L. Garrido, P. Salembier, *Universitat Politecnica de Catalunya (SPAIN)*

ABSTRACT

Connected operators [4] and Region Growing [2] algorithms have been created in different context and applications. However, they all are based on the same funda-mental merging process. This paper discusses the basic issues of the merging algorithm and presents different applications ranging from simple frame segmentation to video sequence analysis.

A Framework For Interactive Video Sequence Segmentation Based On Multiple Features

Authors:

R. Castagno, T. Ebrahimi, *Swiss Federal Institute of Technology (SWITZERLAND)*

ABSTRACT

In this paper, a scheme for interactive video segmentation is presented. A key feature of the system is the distinction between two levels of segmentation, namely Region and Object Segmentation. Regions are homogeneous areas of the images, which are extracted automatically by the computer. Semantically meaningful objects are obtained through user interaction by grouping of regions, according to application specifications. This splitting relieves the computer of a fully semantic understanding of a scene, and allows a higher level of exibility. The extraction of regions is based on the multidimensional analysis of several image features by a spatially constrained Fuzzy C-Means algorithm. The local level of reliability of the different features is taken into ac-count in order to adaptively weight the contribution of each feature to the segmentation process. Results about the extraction of regions as well as about the tracking of spatio-temporal objects are presented.

Curvature Estimation Of Oriented Patterns

Authors:
M. Donias, P. Baylou, *Equipe Signal et Image ENSERB and PRC-GDR ISIS (FRANCE)*
N. Keskes, *Departement Image CSTJF - ELF Aquitaine (FRANCE)*

ABSTRACT

In digital pictures curvature which characterizes the local geometry of features cannot be calculated exactly and can only be estimated. This paper presents two methods to estimate curvature of oriented patterns at different scales which can be used even if the structures are very close. The first approach assumes that a structure is locally defined by an implicit isointensity contour. Curvature is obtained by a direct computation stemming from the differential geometry. The second approach is based on an explicit representation of a structure defined by a set of points initially extracted. We assume this set represents an osculating circle and fit it with a circular arc. Both methods are applied to curvature estimation of seismic images for geological interpretation.

ARHON: A Multimedia Database Design For Image Documents

Authors:
K. V. Chandrinos, J. Immerkaer, P. E. Trahanias, *Institute of Computer Science, FORTH (GREECE)*

ABSTRACT

In this paper we present the design and initial implementation of a multimedia system for historical documents. The system, currently under development at FORTH, will be used by the Vikelea Municipal Library of archives at Heraklion, Crete. The material consists of historical documents that need to be digitized, indexed and continuously annotated by scholars so that the value of the archive will increase by its use. We present the design architecture and initial implementation of a client-server model for the creation, maintenance and use of such a digital archive. Also, we present a technique for marking-up different media to provide rapid entering of semantic information The user can interactively mark-up parts of the representation on the medium of choice (e.g. areas on a document image) and associate local information with text, images, sound-clips, video or subparts of them without altering the original. The output is XML files that can be parsed by viewers to navigate through the semantic information or used by search agents to refine the accuracy of retrieval in a query. For demonstration purposes we present ImageTagger, a prototype for document image tagging. Developing in Java allows for net-centric administration and use of a number of heterogenous databases through standard Internet browsers.

A Multimedia System For The Surveillance Of Unattended Railway Stations

Authors:
E. Stringa, C. Sacchi, C. S. Regazzoni, *University of Genoa (ITALY)*

ABSTRACT

In this paper a multimedia system for surveying remote environments is presented. The goal of the proposed system is to alert the surveillance operator when an abandoned object is detected in waiting rooms of unattended railway stations. The system is based on a monochromatic TV-camera acquiring video data about the surveyed environment; this data are processed at local level to reduce redundancy and only necessary data and images are sent to the remote control centre. The paper is focused on describing two main aspects: 1) image sequence processing 2) channel coding transmission strategies for security issues.

Down–Sampling Of Compressed Images In The DCT Domain

Authors:
A. N. Skodras, *University of Patras, (GREECE)*
C.A. Christopoulos, *Ericsson Telecom AB, (SWEDEN)*

ABSTRACT

An efficient down-sampling algorithm of DCT (discrete cosine transform) compressed images is presented in this communication. The algorithm operates directly on the compressed data, thus avoiding the need for decompressing, down-sampling in the spatial domain and re-compressing the images. As a result, the quality of the reconstructed images is higher and the computational complexity is lower than similar algorithms appeared in the open literature. Its structure is highly regular, resulting in efficient software and hardware implementations. The algorithm can be used in various applications, such as image and video browsing, video compositing and transcoding, and HDTV to SDTV conversion.

Broadband Multimedia Systems
Architecture And Integration

Authors:
I. Defee, *Tampere University of Technology (FINLAND)*

ABSTRACT

Interactive multimedia systems with guaranteed service quality require complex integration of servers, broadband networks and terminals. A critical issue is development of such systems based on standards in order to achieve interoperability comparable to the Internet. The task for devising standardized interactive networked multimedia systems has been undertaken by the Digital Audio Visual Council (DAVIC). In this paper we describe development of a DAVIC compliant networked multimedia system. This system is based on the implementation and integration of hardware and software based on standards developed by DAVIC, MPEG-2 DSM-CC, MHEG-5, CORBA and ATM.

A Color Segmentation and Classification Scheme for Facial Image and Video Retrieval

Authors:
N.Herodotou, K.N.Plataniotis, A.N.Venetsanopoulos, *University of Toronto (CANADA)*

ABSTRACT

In this paper, a technique is presented to locate and track the facial areas in image and video databases. The extracted facial regions are used to obtain a number of features that are suitable for content-based storage and retrieval. The proposed face localization method consists of essentially two components: i) a color processing unit, and ii) a shape and color analysis module. A number of features such as hair color, skin-tone, and face location and size are subsequently determined from the extracted facial areas. The hair and skin colors provide useful descriptions related to the human characteristics while the face location and size can reveal information about the activity within the scene and the type of image (i.e. portrait shot, complete body).

Algorithms For Compressed Video Processing In Multimedia Applications

Authors:
F. Dardi, G. L. Sicuranza, *University of Trieste (ITALY)*

ABSTRACT

Recently, there have been considerable efforts for detecting scene changes and for efficient matching and clustering of video shots using compressed data. We propose here some algorithms that reduce false detections in cases of significant object motions, subtitling, picture in picture or special effects.

Detecting And Classifying Video Shot Boundaries In MPEG Compressed Sequences

Authors:
Irena Koprinska, Sergio Carrato, *University of Trieste (ITALY)*

ABSTRACT

We present a new method for shot boundaries detection and classification that operates directly on the MPEG compressed video. It is based only on the information about the macroblock coding mode in P and B frames. In order to maintain good accuracy while limiting complexity, the system follows a two-pass scheme and has a hybrid rule-based/neural structure. A rough scan over the P frames locates the potential boundaries of the transitions and the solution is then refined by a precise scan over the B frames of the respective neighborhoods. The "simpler" boundaries are recognized by the rule-based module, while the decisions for the "complex" cases are refined by the neural part. The experimental results demonstrate high speed and accuracy in detecting cuts, fades and dissolves.

Image Region Extraction For Content-Based Image Retrieval

Authors:
D. Androutsos, K.N. Plataniotis, A.N.Venetsanopoulos, *University of Toronto (CANADA)*

ABSTRACT

We present a technique for coarsely extracting the regions of natural color images which contain directional detail, e.g., edges, texture, etc., and smooth color regions, which we then use for image database indexing. As a measure of color activity, we use a perceptually modified distance measure based on the sum-of-angles criterion. We then apply histogram thresholding techniques to separate the image into smooth color regions and busy regions where edge, texture and color activity exists. Color segmentation is performed on the smooth areas using HSV histogram techniques, to obtain image regions with one dominant average color. Database indices are then created from the busy regions using the directional detail histogram technique along with the color vectors representing the segmented smooth areas. Image retrieval is then performed using the histogram intersection method for the detailed areas and color vector distance measures are utilized for retrieval of the smooth color regions.

Wavelets, Filterbanks, And The Karhunen-Loeve Transform

Authors:
M. Unser, *Swiss Federal Institute of Technology (SWITZERLAND)*

ABSTRACT

Most orthogonal signal decompositions, including block transforms, wavelet transforms, wavelet packets, and perfect reconstruction filterbanks in general, can be represented by a paraunitary system matrix. Here, we consider the general problem of finding the optimal P x P paraunitary transform that minimizes the approximation error when a signal is reconstructed from a reduced number of components $Q<P$. This constitutes a direct extension of the Karhunen-Loeve transform which provides the optimal solution for block transforms (unitary system matrix). We discuss some of the general properties of this type of solution. We review different approaches for finding optimal and sub-optimal decompositions for stationary processes. In particular, we show that the solution can be determined analytically in the unconstrained case. If one includes order or length constraints, then the optimization problem turns out to be much more difficult.

Total Variation Based Interpolation

Authors:
F. Guichard, *Inrets-Dart (FRANCE)*
F. Malgouyres, *Univ. Paris-Dauphine (FRANCE)*

ABSTRACT

We propose an reversible interpolation method for signals or images, in the sense that the original image can be deduced from its interpolation by a sub-sampling. This imposes some constraints on the Fourier coefficients of the interpolated image. Now, there still exist many possible interpolations that satisfy these constraints. The zero-padding method is one of them, but gives very oscillatory images. We propose to choose among all possibilities, the one which is the most "regular". We justify the total variation of the image as a good candidate as measure of regularity. This yields to minimise a regularisation functional defined in space domain, with a constraint defined in the frequency domain.

Transform Oriented Technology For Quantitative Analysis Of Fetal Movements In Ultrasound Image Sequences

Authors:
L. Yaroslavsky, B.-Z. Shaick, *Tel Aviv University (ISRAEL)*

ABSTRACT

An image processing technology for featal organ tracking in ultrasound image sequences is presented. The technology includes a variety of algorithms for image calibration, noise suppression, target locations, signal interpolation and image geometrical transformation and is based on an extensive use of nonlinear signal processing in the domain of orthogonal transforms such as DFT and DCT.

Discrete-Time Linear-Phase Nearly Orthogonal Wavelet Banks

Authors:
T. Saramaki, K. Egiazarian, *Tampere University of Technology (FINLAND)*

ABSTRACT

A family of discrete-time linear-phase nearly orthogonal wavelet banks is introduced. These wavelet banks are intermediate cases between orthogonal wavelet banks having nonlinear-phase impulse responses and biorthogonal banks having linear-phase impulse responses. For these banks, the wavelet and scaling functions are made very regular and the wavelet function has several vanishing moments. Strightly speaking, the proposed filter banks are not real wavelet banks in the sense that there are negligible aliased terms and a very small reconstruction error for the unaliased component. Several examples are included showing the usefulness of the proposed banks in signal processing applications.

Filtering In Consecutive Fractional Fourier Transform Domains - A Method To Synthesize General Linear Systems For More Efficient Digital And Optical Implementations

Authors:
M. F. Erden, *Tampere Univ. of Technology (FINLAND)*
H. M. Ozaktas, M. A. Kutay, *Bilkent University (TURKEY)*

ABSTRACT

Either exact realizations or useful approximations of general linear systems may be implemented in the form of repeated filtering operations in consecutive fractional Fourier domains. These implementations are much cheaper than direct implementations of general linear systems. Thus, we may significantly decrease the optical and digital implementation costs of general linear systems with little or no decrease in performance by synthesizing them with the proposed repeated filtering method.

Fast Fractional Fourier Transform

Authors:
E. V. Labunets, V. G. Labunets, *Ural State Technical University (RUSSIA)*

ABSTRACT

The fractional Fourier transform (FRFT) is a one-parametric generalization of the classical Fourier transform. Since it's introduction in 1980th, the FRFT has been found a lot of applications and used widely nowadays in signal processing. Space and spatial frequency domains are the special cases of the fractional Fourier domains. They correspond to the 0th and 1st fractional Fourier domains, respectively. In this paper, we briefly introduce the multi-parametrical FRFT and its fast algorithm.

Solution Of The Super-Resolution Problem Through Extrapolation Of The Orthogonal Spectra Using Multi-Valued Neural Technique

Authors:
I. Aizenberg, *K.U.Leuven (BELGIUM)*
N. Aizenberg, *State University of Uzhgorod (UKRAINE)*
J. Astola, K. Egiazarian *Tampere University of Technology (FINLAND)*

ABSTRACT

Two methods of the orthogonal spectra extrapolation problem are considered in this paper. Both solutions are based on the application of multi-valued neural element as a filter and as an extrapolator. The first method consist of approximation of the spectra in the higher frequency part using an iterative approximation and multi-valued non-linear filtering. The second method is reduced to the prediction of the spectral coefficients corresponding to the higher frequency part using possibility of MVN to predict time-series. Application of the proposed methods to solution of the super-resolution problem are presented.

New Fast Trigonometric Transforms

Authors:
K. Egiazarian, J. Astola, *Tampere University of Technology (FINLAND)*
V. G. Labunets, E. V. Labunets, *Ural State Technical University (RUSSIA)*

ABSTRACT

New fast trigonometric transforms formed as combinations of discrete parametric Pontryagin transforms and discrete conjugate parametric Pontryagin transforms are introduced in this paper. They include as special cases such known orthogonal trigonometric transforms as the Fourier, Hartley, Wang, cosine and sine, Walsh, Chrestenson transforms, and many others. Different methods of efficient computation of the introduced transforms are described.

Fast Algebraic Convolution For Prime Power Lengths

Authors:
R. Creutzburg, T. Minkwitz, *University of Applied Sciences Brandenburg, (GERMANY)*

ABSTRACT

In this paper a new fast convolution algorithm is introduced. The concept of this new algorithm is fully algebraic. Therefore, no roundoff errors occur. The signal length N is a prime power $N = p^n$, $p \geq 2$. Hence the paper generalizes the results of a previous paper [17] for power of 2 lengths. The cyclic convolution of signals of length N is interpreted as a multiplication of polynomials of degree N - 1 modulo the polynomial $X^N - 1$. The calculation in our new method is done in a finite field extension of the field Q of rational numbers. To minimize the number of operations, an extension field Q[ù] is chosen. with ù as a symbolic root of unity and

$$\deg [Q[ù] : Q] = (p-1/p \lceil \sqrt{N} \rceil^{*})$$

Here. $\lceil \ \rceil^{*}$ and $\lfloor \ \rfloor^{*}$ denote the well-known CEILING and FLOOR functions, respectively, but rather a map-ping to the next larger power of p. The method achieves an arithmetic cost, of O(N log N . log log N) operations over Q, which is the same as that of the well- known algorithm by SCHONHAGE-STRASSEN. However, our algorithm avoids the much more complicated FERMAT arithmetic over integers and works for primes $p \neq 2$ as well, although it performs best with small p, preferably p=2 or 3.

On The Issue Of Rank Estimation In Subspace Tracking: The NA-CSVD Solution

Authors:
P. A. Pango, B. Champagne, *Universite du Quebec (CANADA)*

ABSTRACT

The issue of rank estimation in subspace tracking algorithms is adressed. In a recent paper, we proposed a subspace tracking algorithm, the NA-CSVD. We now extend the performance of NA-CSVD to rank tracking by coupling it with a recently proposed rank tracking technique. The paper includes an overview of typical rank+subspace tracking algorithms which, along with our proposed algorithm, are tested in various simulation scenarios. The new algorithm tracks e_ciently the rank and the signal subspace.

The Min-Norm Beamformer :
A New Estimate Of The Propagation Speed Of Waves
In A Car Exhaust

Authors:
G. Piñero, L. Vergara, *Dpto. Comunicaciones (UPV) (SPAIN)*

ABSTRACT

In this paper we present a new estimate of the propagation speed of acoustic waves propagating along a pipe based on array signal processing, the so-called Min-Norm Beamformer (MNB). In a previous contribution, the authors had already presented the formulation of this new estimate for narrowband signals, but the scenery of acoustic waves uses to be broadband. Therefore, we establish now the complete formulation of the MNB estimate for broadband waves for the two unique possible situations: a coherent or an incoherent processing of the data Sample Covariance Matrix (CSM), including the most efficient matrix transformations given in the literature. Finally, we present several results for three experiments carried out in a real environment : a car exhaust with three different engines at work.

Direction-Of-Arrival Estimation Of Cyclostationary Coherent Signals In Array Processing

Authors:

J. Xin, H. Tsuji, Y. Hase, *Communications Research Laboratory, MPT (JAPAN)*

A. Sano, Keio University (JAPAN)

ABSTRACT

Recently many cyclosationarity-based methods have been proposed to improve signal detection capability, however, in the multipath propagation environment due to various reflections, these cyclic techniques perform poorly. For overcoming the coherent source problem, a preprocessing scheme such as the maximum likelihood (ML) method or the spatial smoothing (SS) approach can be used. Although a cyclic least-squares method was proposed, which involves the multidimensional search and results in the more intensive computational burden. In this paper, a new approach is proposed for estimating the directions-of-arrival (DOA) of the coherent signals impinging on a uniform linear array (ULA) by utilizing the spatial smoothing (SS) technique. For improving the robustness of the DOA estimation, the choice problems of the lag parameter and subarray dimension are considered, and we give an algorithm with multiple lag parameters and the optimal subarray size to exploit the cyclic statistical information sufficiently and to handle the coherence effectively. The performance of the proposed method is demonstrated and compared with the conventional methods through numerical examples.

A Downlink Adaptive Transmitting Antenna Method For T/F/SDMA FDD Systems Avoiding DOA Estimation

Authors:
T. Aste, P. Forster, L. Fety, *Conservatoire National des Arts et Metiers, (FRANCE)*
S. Mayrargue, *CNET, (FRANCE)*

ABSTRACT

In this paper, the problem of space only downlink processing for FDD (Frequency Division Duplex) radio-communication systems is considered. It is shown that using criterions based on average C/I, one can avoid DOA estimation and deduce these criterions at downiink frequency from their equivalent at uplink frequency, under some assumptions about array topology. Moreover, here, we introduce a more generic approach than the former one, denoted as pattern synthesis.

DOA Outlier Mitigation
For Generalised Spatial Smoothing

Authors:
Y. I. Abramovich, N. K. Spencer, *Cooperative Research Centre for Sensor Signal and Information Processing (CSSIP) (AUSTRALIA)*

ABSTRACT

This paper considers the problem of DOA (direction-of-arrival) estimation for a small number of fully correlated sources. The standard spatial smoothing technique may be applied to this single-snapshot model, but only for a uniformly-spaced linear antenna array (ULA). In our ICASSP-98 paper, we introduced a special class of nonuniform array geometry with embedded "partial arrays" and a corresponding generalised spatial smoothing (GSS) algorithm. The initialisation stage of GSS (which is followed by a local maximum-likelihood refinement) involves spatial averaging over all suitable noncontiguous sub-arrays with identical inter-sensor separations. These partial arrays are themselves nonuniform in geometry, and have a small number of sensors. It is well known that MUSIC may fail to resolve poorly-separated sources when the SNR and number of spatial averagings are insufficient, due to abnormal DOA estimates ("outliers"). An additional outlier mechanism for partial-array MUSIC occurs because each partial array has some associated "manifold ambiguity". Thus for spatial smoothing, the set of (initial) DOA estimates often contains outliers. This paper introduces an algorithm which aims to identify each outlier and to correct it, if possible.

Fast High Resolution Methods

Authors:
S. Bourennane, M. Frikel, A. Bendjama, *S.D.E.M. URA CNRS (FRANCE)*

ABSTRACT

We consider the problem of direction of arrival (DOA) estimation in the persence of correlated sources. To perform the DOA estimation, we combine two algorithms. The first algorithm exploits the noise projection matrix without eigende composition and without prior knowledge of the number of the radiating sources. Then the high- resolution spectral estimation algorithm is used. We summarize the algorithms herein and present simulations demonstrating the effectiveness of the proposed method.

Geometrical Determination Of Ambiguities In Bearing Estimation For Sparse Linear Arrays

Authors:
A. Flieller, P. Larzabal, *L.E.Si.R. E.N.S. (FRANCE)*
H. Clergeot, *L.T.S.M.M. Faculté de technologie de Guyane (FRANCE)*

ABSTRACT

The aim of this paper is to study the presence of manifold ambiguities in linear arrays. We propose a general framework for the analysis and so we obtain a generalisation of results given in recent publications for any rank ambiguities. We present a geometrical construction able to determine all the ambiguous directions which can appear for a given linear array. This is a geometrical approach closely connected to the analytical method proposed by Proukakis and Manikas. The method allows determination of any rank ambiguities and for each ambiguous direction set the rank of ambiguity is determined. The search is exhaustive. Application of the method requires no assumption for the linear array and is easy to implement. We apply the method to the search of ambiguities for sparse linear arrays, in particular minimum redundant and non redundant arrays. We show how an ambiguous generator set can be associated to each intersensor distance (lags) if the intersensor distances are all multiples of the half wavelength.

Sufficient Conditions For The Unique Localization Of Distributed Sources

Authors:

S. Valaee, Tarbiat Modares University, *Sharif University of Technology (IRAN)*

B. Champagne, *Universite du Quebec (CANADA)*

ABSTRACT

The array output for a distributed source can be approximated by the superposition of the array response to a large number of closely spaced point sources. In the limit, a distributed source corresponds to an infinite number of point sources. In this approximation, the number of free parameters increases with the number of point sources. In this paper, we show that if the point sources (approximation of a distributed source) are related through some parametric constraints, then for any observation at the array output, almost surely, there is a unique solution for the localization problem, provided that the dimensionality of the parameter space satisfies a certain bound. We show this for both coherently and incoherently distributed sources.

Spatial And Temporal Processing Of Cyclostationary Signals In Array Antennas Based On Linear Prediction Model

Authors:

A. Kanazawa, H. Tsuj, J. Xin, *Communications Research Laboratory, M.P.T. of Japan (JAPAN)*
Blagovest Shishkov, *Technical University of Sofia (BULGARIA)*

ABSTRACT

For the directions of arrival (DOA) estimation, the subspace algorithms have received much attention because of their high-resolution. However, these approaches ignore the temporal properties of the desired signals. By exploiting the special temporal property of the modulated signals in communications, some cyclic direction-finding (DF) algorithms were proposed for improving the signal detection capability. In this paper, a new approach is proposed for the estimating DOA of cyclostationary signal that utilizes a linear prediction (LP) technique. For resolving the problem of the choice of the optimal lag parameter, we present a new alternative approach that exploits the cyclic statistical effectively in a forward-backward way, enabling robust high-resolution performance to be achieved. Furthermore, the presented scheme is simpler and more convenient to implement than those using the conventional cyclic algorithms, where the computation of the cyclic array covariance matrix is burdensome. The effectiveness of the presented algorithm is demonstrated and compared with the conventional algorithms through numerical examples.

Improving Signal Subspace Estimation
And Source Number Detection
In The Context Of Spatially Correlated Noises

Authors:
P. Fabry, Ch. Serviere, J. L. Lacoume, *LIS-ENSIEG, (FRANCE)*

ABSTRACT

This paper addresses the issue of Orthogonal Techniques for Blind Source Separation of periodic signals when the mixtures are corrupted with spatiaily correlated noises. The noise covariance matrix is assumed to be unknown. This problem is of major interest with experimental signals. We first remind that Principal Components Analysis (PCA) cannot, provide a correct estimate of the signal subspace in this situation. We then decide to compute the spectral matrices using delayed blocks to eliminate the noise influence. We show that two of these delayed spectral matrices are enough to get the unnoisy spectral matrix. We also introduce a new source number detector which exploits the eigenvectors of a delayed matrix to estimate the signal subspace dimension. Simulation results show that the signal subspace estimation is improved and the source number detector is more efficient in this situation than the usual AIC and MDL criteria.

Blind Separation Of Cyclostationary Signals

Authors:
A. Dapena, D. Iglesia, L. Castedo, *Universidad de La Coruna (SPAIN)*

ABSTRACT

In this paper we propose a new approach to blind source separation that exploits the cyclostationary nature of the sources typically found in communication applications. A new adaptive algorithm is proposed that simultaneously utilizes Higher Order Statistics (HOS) and cyclic moments. The approach enables exploitation of periodicities embedded in the sources such as the carrier frequency or the symbol rate. It is also presented an analysis to obtain the conditions under which the algorithm converges to the desired solutions.

Blind Separation From å-Contaminated Mixtures

Authors:
V. Koivunen, *Tampere University of Technology (FINLAND)*
P. Pajunen J. Karhunen E. Oja, *Helsinki University of Technology (FINLAND)*

ABSTRACT

This paper deals with the problem of Blind Source Separation (BSS). BSS algorithms typically require that observed data are prewhitened. The data are here assumed to be contaminated by highly deviating samples. Hence, covariance matrix used for whitening and determining the number of signals is estimated unreliably. We propose a method where data are first whitened in a robust manner. Sources are then separated using an iterative least squares algorithm. The proposed method is compared to a method based on sample estimates and the influence of outliers is analysed.

Efficient Stochastic Maximum A Posteriori Estimation For Harmonic Signals

Authors:
C. Andrieu, A. Doucet, *University of Cambridge (UK)*

ABSTRACT

In this paper, we address the problems of ML (Maximum Likelihood) parameter estimation and model order selection for harmonic signals using classical criteria. Solving these problems requires the maximization of complex multimodal functions. These optimization problems are shown as being equivalent to the estimation of joint and marginal MAP (maximum a posteriori) estimates under given Bayesian models. Efficient stochastic algorithms based on non-homogeneous Markov chain Monte Carlo methods are presented to solve these problems and their convergence is established. Computer simulations demonstrate the efficiency of these algorithms.

A Fuzzy Reasoning Based ARMA Order Selection Method

Authors:

M. Haseyama, H. Kitajima, *Hokkaido University (JAPAN)*

ABSTRACT

A fuzzy reasoning based approach for ARMA order selection is discussed in this paper. The proposed method attempts to select the optimal ARMA order of a time-varying ARMA model. This method improves model validity-criterion based order selection, such as the AIC (Akaike's Information Criterion) and the MDL (Minimum Description Length), etc with applying both of a fuzzy reasoning method and a fuzzy c-means clustering method. These fuzzy methods are incorporated in the proposed method as the follows: (1) Suppose the ARMA order of the reference time-varying model changes, the suitable ARMA order is selected by utilizing a recursive fuzzy reasoning method. (2) By using the fuzzy c-means clustering method, we detect the time at which the ARMA order of the reference model changes, and the clustering values are used for adaptively setting the forgetting factor in the recursive fuzzy reasoning method. The experimental results show that the proposed method effectively selects the ARMA orders of a time-varying ARMA model.

Spectral Methods For Stationary Harmonizable Alpha-Stable Processes

Authors:
G.A. Tsihrintzis, *Northeastern University (USA)*
P. Tsakalides, C.L. Nikias, *University of Southern California (USA)*

ABSTRACT

We address the problem of estimation of the fractional-power spectrum of certain classes of symmetric, alpha-stable (SáS) processes. We start with a summary of the key definitions and results from the theory of stationary, harmonizable SáS processes and proceed to discuss the performance of fractional-power periodograms. Next, we present a high resolution fractional-power spectrum estimation algorithm that we term "the minimum dispersion distortionless response". The algorithm is a generalization of the classical Maximum Likelihood Method of Capon. Preliminary tests of the algorithms are run on simulated data.

Robust Bayesian Spectral Analysis Via MCMC Sampling

Authors:
A. Doucet, C. Andrieu, *University of Cambridge (UK)*

ABSTRACT

In this paper, the harmonic retrieval problems in white Gaussian noise, non-Gaussian impulsive noise and in presence of threshold observations are addressed using a Bayesian approach. Bayesian models are proposed that allow us to define posterior distributions on the parameter space. All Bayesian inference is then based on these distributions. Unfortunately, a direct evaluation of these latters and of their features requires evaluation of some complicated high-dimensional integrals. Efficient stochastic algorithms based on Markov chain Monte Carlo methods are presented to perform Bayesian computation. In simulation, these algorithms are able to estimate the unknown parameters in highly degraded conditions.

Linear Prediction Modeling For Signal Selective DOA Estimation Based On Higher-Order Statistics

Authors:
H. Tsuji, J. Xin, Y. Hase, *M.P.T. of Japan (JAPAN)*
B. Shishkov, *Technical University of Sofia (BULGARIA)*

ABSTRACT

The direction-finding approach for impinging signals is one of the most important issues in array processing. By exploiting the cyclic statistics and higher-order temporal properties of communication signals, cyclic higher-order statistics (CHOS) direction-finding approaches have been proposed for narrow-band non-Gaussian signals. However, conventional cumulant-based algorithms become very complicated and are computationally intensive when a cumulant higher than the forth-order is used. In this paper, by utilizing a linear prediction (LP) model of the sensor outputs, a new cyclic higher-order method is given to detect the signals of interest (SOI). The proposed method can not only reduce the computational load and completely exploit the CHOS temporal information, but can also correctly estimate the DOA of desired signals by suppressing undesired signals. We also show the effectiveness of the proposed method through simulation results.

Wavelet–Packet Basis Selection For Abrupt Changes Detection In Multicomponent Signals

Authors:

E. Hitti, M.–F. Lucas, *Universite de Nantes, (FRANCE)*

ABSTRACT

We propose a "best basis" selection method, to detect abrupt changes in noisy. multicomponent signals. The basis, selected from a wavelet-packet library with an energetic criterion, separates the different frequency components of the signal while keeping the best possible time localization. From the obtained basis, we reconstruct monocomponent signals in each band. The detector is based on the adequation of this set of signals with an abrupt change model. A performance analysis is realized from synthetic signals, in terms of selected basis and detection results.

Rank Test
For Detection Of The Number Of Cisoids In Noise

Authors:
J. Sorelius, *Uppsala University (SWEDEN)*

ABSTRACT

In this paper, a rank test for detecting the number of cisoids in noise is presented. The method is based on Gaussian Lower triangular - Diagonal - Upper triangular (LDU) decomposition of a Hankel data matrix, and the method is especially useful for short data records.

High Resolution Nearly-ML Estimation
Of Sinusoids In Noise
Using A Fast Frequency Domain Approach

Authors:
Dr M. D. Macleod, *Cambridge University (UK)*

ABSTRACT

Estimating the frequencies, amplitudes and phases of sinusoids in noise is a problem which arises in many applications. The aim of the methods in this paper is to achieve computational efficiency and near-ML performance (i.e. low bias, variance and threshold SNR), in problems such as vibration or audio analysis where the number of tones may be large (e.g. >20). An approach has recently been published for resolved tones. This paper extends that frequency domain approach to the high-resolution problem.

A Period Detector For Pseudo-Periodic Signals

Authors:
B. F. Rice, *Lockheed Martin Technical Operations (USA)*

ABSTRACT

The spreading codes of direct sequence spread spectrum signals are often periodic. From a segment of signal containing multiple periods, this periodicity can be detected and the period estimated. The approach taken is to bandlimit the signal (to introduce variation into the envelope), compute the envelope of the filtered signal (to remove the information), and compute the autocorrelation of the envelope. The autocorrelation function contains a peak at an offset equal to the period. The peak-to-noise ratio can be used to obtain an estimate of the input signal-to-noise ratio. Narrowband interference can corrupt the process, so it can be important to precede application of the technique by a pre-whitening filter. The technique applies even for "featureless" signals, in which measures have been taken to suppress rate lines in the outputs of quadratic or higher-order chip rate detectors.

Relationship Between The Wigner-Distribution And The Teager Energy

Authors:

R. Hamila, F. A. Cheikh, J. Vesma, J. Astola, M. Gabbouj, *Tampere University of Technology (FINLAND)*

ABSTRACT

The link between the Teager energy (TE) operator and the Wigner distribution (WD) is established in this paper. This link results in a simple way to calculate the second conditional moment in frequency of a Wigner distribution. Furthermore, this link helps explain the negative 'energy' via the Teager operator, and the negative conditional second moment via the Wigner distribution. We present the similarity between the WD and the time-variant periodogram. Also, we calculate the TE of a time-variant finite-time complex spectrum, and the TE of a time-variant periodogram.

LPC And CCF Vocal Tract Models In Speech Synthesis

Authors:
R. Vích, *Academy of Sciences (CZECH REPUBLIC)*
Z. Smékal, *Technical University of Brno (CZECH REPUBLIC)*

ABSTRACT
In the paper a new cepstral continued fraction zero-pole model for speech synthesis is presented. It is based on the cosine expansion of the logarithmic short time spectrum of the windowed speech signal and the synthesis is realized by adaptive approximative inverse cepstral transform using continued fraction digital filter structures. These structures were implemented on a digital signal processor. The spectrum modelling properties for this approach and those obtained by linear predictive coding are compared.

Real Time Detector For Cyclostationary RFI In Radio Astronomy

Authors:
R. Weber, *Universite d'Orleans (FRANCE)*
C. Faye, *ETIS-ENSEA (FRANCE)*

ABSTRACT

The negative impact of radio frequency interferences on the quality of radio astronomical observations is a matter of increasing concern for the radio astronomy community. The integrity of the data can be preserved by detecting the interference in order to blank the receiver in real time. The proposed detector uses cyclostationary properties of the interference. It converts the known hidden periodicity into a periodic signal which can be easily detected. In this paper, the method is explained and simulations applied to typical interferences are presented. Finally, a dedicated hardware implementation using programmable logic array is proposed.

Ladder Scheme For
Perfect Reconstruction Modulated Filter Banks

Authors:
M. Gharbi, M. Colin, M. Gazalet, F.X. Coudoux, *ENSIMEV University of Valenciennes (FRANCE)*

ABSTRACT

A product form of the polyphase filter matrix with adjustable overall delay which allows perfect reconstruction, is given for orthogonal and biorthogonal modulated filter banks. The lifting scheme has been used, yielding to simpler implementation and allowing for in-place computations, i.e. the transform can be calculated without allocating auxilary memory. The modulated filter bank system is decomposed into small subsystems in a ladder module configuration, which is profitable for many applications and easies the hardware implementation.

Multistage Implementation Of Optimal Reconstruction In Noisy Filter Banks

Authors:

O. Bradeanu, *Military Technical Academy (ROMANIA)*
U. Appel, *Bundeswehr University (GERMANY)*

ABSTRACT

It is well known that conventional design techniques of filter banks lose their ability to guarantee the perfect reconstruction condition when additive noise disturbs the subband sequences. The alternative synthesis scheme compensates the effects of the subband noise by estimating the transmitted samples minimizing the reconstruction mean squared error, MSE. This process is based on the state-space model of the standard block interpolation process. According to relation between the decimation/interpolation factor and the model order, the appropriate computational efficient implementation schemes were developed.

Optimal Subband Analysis Filters Compensating For Quantization And Additive Noise

Authors:

A. Doulamis, N. Doulamis, A. Delopoulos, *National Technical University of Athens (GREECE)*

ABSTRACT

In this paper, we present an analysis filter design technique which optimally defines the proper decimator so that the quantization noise is compensated. The analysis is based on a distortion criterion minimization using the Lagrange multipliers. The optimal decimation filters are derived through a Ricatti solution which involves both the quantization and the interpolation filters. Experimental results are presented indicating the good performance of the proposed technique versus conventional subband filter banks in the presence of quantization noise.

Multidimensional Multirate Filter Without Checkerboard Effects

Authors:
Y. Harada, S. Muraamatsu, H. Kiya, *Tokyo Metropolitan University (JAPAN)*

ABSTRACT

The checkerboard effect is caused by the periodic time-variant property of inter-polators which compose a multirate system. Although the conditions for some one-dimensional(1D) multirate systems to avoid the checkerboard effect have been shown, the checkerboard effect in multidimensional(MD) multirate systems has not been considered. In this paper, the conditions for MD multirate filter and filter bank to avoid the checkerboard effect are considered. Besides, the properties of the MD filters with no checkerboard effect are given. Simulation example shows that the checker-board effect can be avoided by using the proposed condition.

Inter-Relationships Between Different Structures For Periodic Systems

Authors:
D. McLernon, *The University of Leeds (UK).*

ABSTRACT

A self-contained, clear and consistent notation is developed for five different mathematical models of linear periodically time-varying (LPTV) filters. Then the various inter-relationships between the different structures are derived.

A Design Approach To A Hierarchical Structure Transversal Filter Using GA

Authors:
S. Nakamura, T. Hoshi, S. Watanuki, M. Yoneyama, *Tokyo Denki University (JAPAN)*

ABTRACT

The purpose of this paper is to discribe a direct design approach using a GA(Genetic Algorithm) to a HSTF(Hiearchical Structure Transversal Filter). A direct design approach of the HSTF is not easy because of highly nonlinear problem. So far, we took a two phase approach. However, this approach cannot give an accurate specification of the filter. We applied a GA into a direct design of the HSTF and could get some preferable results. A design example by GA will be given compared with that of the two phase design approach.

Design Of Primitive Operator Digital Filters Using Genetic Algorithms

Authors:
D. W. Redmill, D. R. Bull, *University of Bristol (UK)*

ABSTRACT

This paper considers the design of low complexity digital filters. Complexity is reduced by constraining the filters to have integer coefficients, which can be efficiently implemented using primitive operator directed graphs (PODG). Genetic Algorithms (GAs) are used in conjunction with a heuristic graph design algorithm, to provide a joint optimization of filter performance and complexity. The proposed techniques used to design 1D filters, 2D filters and perfect reconstruction filter banks.

Design Of Equiripple Minimum Phase FIR-Filters

Authors:

A. Groth, H. G. Gockler, *Ruhr-Universitat Bochum (GERMANY)*

ABSTRACT

The numerical robustness and applicability of a recent approach to the design of minimum phase FIR filters by cascading a passband and a stopband filter [Go 80] is improved by the following measures:

i) Spectral factorization (root finding) is avoided by exploiting the Hilbert relations between attenuation and phase

ii) Unbounded gain in don't care regions is restricted by prescription of appropriate desired values.

As a result, highly selective minimum phase FIR filters requiring a considerably reduced number of multiplications are obtained.

Designing Low-Pass Digital Filters With Fewer Parameters

Authors:
G. Li, *Nanyang Technological University (SINGAPORE)*

ABSTRACT

The finite impulse response (FIR) filters are often used in digital filter design. It has been noted that for a given design specification a FIR filter of very high order is required when the desired frequency response is of very narrow bandwidth. In this paper, we propose a new model for linear phase low-pass narrowband digital filter design. The corresponding filter can be implemented using the classical transversal structure but with the shift operator **z**^**{-1}** replaced by an alternative operator **\rho**^**{-1}**. Several design examples are presented. It is shown that for a given specification, the proposed structure requires much less design parameters.

Cluster Filter

Authors:
J. Yli-Hietanen, K. Koppinen, K. Halonen, *Tampere University of Technology (FINLAND)*

ABSTRACT

In this paper, a new form of clustering method is presented where a priori knowledge of the reliability of different samples is used. This knowledge can be inserted into the cluster-finding based computation of the estimator output in the form of sample weights. This kind of method is needed in time-delay based angle of arrival estimation with nonuniform linear sensor arrays.

Least Squares Design Of IIR Filters With Arbitrary Magnitude And Phase Responses And Specified Stability Margin

Authors:

M. C. Lang, *Vienna University of Technology (AUSTRIA)*

ABSTRACT

This paper presents an algorithm for the frequency domain design of infinite impulse response (IIR) digital filters with prescribed magnitude and phase responses according to a least squared error criterion. We use the implications of Rouché's Theorem to modify standard algorithms for nonlinear least squares optimization in such a way that a constraint on the maximum pole radius can be incorporated. Hence, the filters obtained by the proposed method can be guaranteed to be stable and, moreover, exhibit an arbitrary stability margin. This helps to avoid undesired behavior of the magnitude response in the transition bands, decreases the sensitivity to coefficient quantization, and helps to reduce the amplitude of small-scale limit cycles. A design example shows the superiority of the proposed algorithm over several other design methods.

Quadratic FIR-Filter Design
As A Generalized Eigenproblem

Authors:
J. O. Coleman, *Naval Research Laboratory (USA)*

ABTRACT

This work suggests optimizing quadratic performance measures on (possibly multi-rate) DSP systems in order to design one or more embedded FIR filters, which might be real or complex and with or without linear-phase, N-th band, or other structure. The outlined approach replaces the eigenproblem of the eigenfilter design method with a generalized eigenproblem. A general capability results for minimizing an arbitrary nonnegative-definite (purely) quadratic system-performance measure subject to a single arbitrary equality constraint on a second such quadratic. Using this one constraint to fix the impulse-response energy in nonembedded single-filter design results in the ordinary eigenfilter method. A simple example and a geometric inter-pretation show that alternate equality constraints can profoundly affect the solution. Finally a multirate matched-filtering-system example design impossible with the ordinary eigenfilter method hints at the flexibility of the approach.

On The Optimal Structure Of 2-D Digital Filters With L2-Sensitivity Minimization

Authors:
G. Li, *Nanyang Technological University (SINGAPORE)*

ABSTRACT

An expression is derived for the error variance of transfer function of a Two Dimensional system due to FWL errors. The optimal realization problem is then formulated by minimizing this variance with respect to all possible realizations of the system. This problem is shown to be equivalent to the minimization of a pure L2 norm based sensitivity measure and can be solved using any standard minimization algorithm with guaranteed convergence.

Design Of FIR Filters For Images Based On Properties Of Human Vision

Authors:
N. Aikawa, *Tokyo Engineering University (JAPAN)*
O. Sato, *Tokyo Metropolitan University (JAPAN)*

ABTRACT

The design of linear phase filters for image processing based on properties of human visual perception has been shown to realize the minimization of criterion functions in both the spatial domain and the frequency domain. Based on properties of human vision, we propose the design method of FIR filters which have an adequately reduced ringing and keep edge information of images.

Lattice Structure Of Transparent Layered Optical Systems

Authors:
Y. Monden, *Shimane University (JAPAN)*
H. Hirayama, *Hiroshima University (JAPAN)*

ABSTRACT

We will propose a system theoretical approach to lattice characterization of a layered transparent optical system at oblique incidence and classiry and specify completely its optical states as well as their spectra based on a set of ``axiomatic'' properties of its system matrices.

Complexity Constraints
A Way Toward Simpler Structuring Systems

Authors:

P. Koivisto, P. Kuosmanen, *Tampere University of Technology (FINLAND)*

ABSTRACT

A method of incorporating implementation aspects in the algorithm-level design of nonlinear filters is proposed. As a case study, the trade-off between the visual properties and the complexity of soft morphological filters is studied using training-based optimization methods. Specifically, it is shown that the use of the complexity constraints can provide the filter designer valuable information on to what extent it is reasonable to increase the complexity of the filter structure.

An Effective Algorithm
For Filtering And Observer Design

Authors:
S. Ibrir, *Ecole Superieure d' Electricite (FRANCE)*

ABSTRACT

We propose a new numerical algorithm for calculating the continuous filtered signal and its reliable higher derivatives from noisy discrete data. This algorithm is conceived to be implemented as a filter or as an observer for dynamical systems where the measurements are supposed to be available at equally-spaced instants and whose states could be expressed in terms of finite numbers of their higher derivatives. The presented algorithm uses the Generalized cross-validation criterion to compute the optimal value of the smoothing parameter which translates the compromize between the smoothing and the closeness to measurable data. The algorithm has the advantage of being used in a recursive manner, like an adaptive filter, so as to improve the quality of filtering. Numerical simulations are presented to show its efficiency.

Audio-Visual Signal Processing For Mobile Communications

Authors:
P. Haavisto, *Nokia Research Center, (FINLAND)*

ABSTRACT

The speech channels and the available data transmission rates of digital mobile systems, together with mobile handset processing capability, are developing fast enabling extensive exploitation of advances in audio-visual processing to provide new or improved services for end-users. The new Adaptive Multi-Rate codec for GSM will adapt to varying channel conditions to increase quality and capacity while complementing the high basic quality of the Enhanced Full Rate codec. Increased data rates will enable transmission of live video (e.g., video telephony) in GSM during 1999 and the new MobiVideo coder can double the bandwidth efficiency of video coding compared to existing standards. Speech recognition is becoming a standard feature in mobile handsets and the emerging Distributed Speech Recognition standard will enable high quality network based speech recognition over mobile channels.

Nth-Band Filter Design

Authors:
T. Saramaki and M. Renfors, *Tampere University of Technology (FINLAND)*

ABSTRACT

Digital Nth-band linear-phase nonrecursive and Nth-band recursive filters are special digital filter classes playing an important role in various applications. Both these filter classes are named according to their frequency-domain characteristics. This paper reviews the properties of these filters as well as their usefulness in several digital signal processing applications. Also their optimization for various applications is considered.

Fractional Rate Decimator And Interpolator Design

Authors:
T. A. Ramstad, *Norwegian University of Science and Technology (NORWAY)*

ABSTRACT

In this article fractional rate sampling frequency change is considered. A general theory based on combined analog and digital interpolators is presented, a generalized Farrow structure is introduced, and comparisons between cases with different design parameters are given. It is shown that the introduction of the digital interpolator provides simplifications in terms of computational complexity at the cost of increased coefficient storage.

Subband Coding Of Speech And Audio

Authors:

C. D. Creusere, *Naval Air Warfare Center*

ABSTRACT

We discuss here the use of multirate filter banks for perceptually-weighted speech and audio compression. While our primary focus is on audio compression, we also review two recently proposed wideband speech coders that use filter banks to eliminate perceptually redundant information. Our goal is to examine and compare the time-frequency tradeoffs inherent various coding algorithms as they try to exploit the mask-ing of the human auditory system.

Multiresolution Coding Of Image And Video Signals

Authors:
B.Girod, F.Hartung, U.Horn, *University of Erlangen-Nuremberg, (GERMANY)*

ABSTRACT

Multiresolution image and video coding schemes offer both excellent coding efficiency and the ability to support scalability. This paper gives an introduction to the principles of multiresolution coding of image and video signals. Besides critically sampled subband pyramids, oversampled pyramid decompositions are discussed. Oversampled subband pyramids are often better suited for scalable coding schemes. Solutions to the bit allocation problem for subband coders with and without quantization noise feedback are presented. Finally, we briefly review spatio-temporal pyramids as the most promising approach to scalable video coding.

Acoustic Echo Control In Subbands - An Application Of Multirate Systems

Authors:
G. Schmidt, *Darmstadt University of Technology (GERMANY)*

ABSTRACT

This paper discusses the application of multirate signal processing techniques for the design of a system to cancel acoustic echoes in subbands. After a description of the computational complexity and the filterbank design, the need of delaying the microphone signal is explained. Aliasing caused by nonideal filterbanks and especially the noncausal subband impulse responses limit the quality of the system. If certain restrictions (e.g. an upper threshold of delay or a finit amount of memory) have to be fulfilled, it is neccesary to "balance" the entire system. Hence, all components should be in the same "quality-range". Finally we point out the advantages gained by applying multirate processing.

Transmultiplexers - Some New Applications

Authors:
R. D. Koilpillai, K. C. Zangit, *Advanced Development and Research Group Ericsson Inc.,, (USA)*
T. Q. Nguyen, *Boston University, (USA)*

ABSTRACT

Traditionally, transmultiplexers (TMUX) were deployed in telephone networks (speech transmission) for the TDM ~ FDM translation, and for on-board processing in satellites. It is known that the theory of multirate filter banks can be applied to TMUX to obtain novel implementations and to gain new insights into their design. Recently, two communications applications, namely, the software radio, and multi-carrier modulation have gained considerable importance. In this paper, we briefly review the multirate DSP framework for TMUX and discuss these two applications using this framework.

Discrete Wavelet Packet Based Multitone Modulation For Transmitting Complex Symbols

Authors:
T.K. Adhikary, V.U. Reddy, *Indian Institute of Science (INDIA)*

ABSTRACT

Multicarrier modulation (MCM) methods have been attracting considerable attention lately. In this paper, we address MCM techniques based on wavelet packets for QAM (quadrature amplitude modulation) and PM (phase modulation) symbol constellations. We first discuss the design of complex wavelet packets, which serve as orthonormal carriers for MCM, and then describe a DWMT (discrete wavelet multitone) scheme for transmitting QAM and PM symbols. We consider both blind and non-blind schemes for retrieving the symbols at the receiver. Simulations have been carried out to study the performance with both the schemes.

Oversampling A/D And D/A Converters

Authors:
G. C. Temes, *Oregon State University (USA)*

ABSTRACT

This tutorial paper explains the principles and areas of application of oversampling data converters. A comparison between Nyquist-rate and oversampling converters is given, and some commonly used architectures discussed for both D/A and A/D oversampling converters.

Permutation Spreading In Wavelet OFDM Systems

Authors:
F. Dovis, M. Mondin, *Politecnico di Torino (ITALY)*
F. Daneshgaran, *Calif. State Univ. (USA)*

ABSTRACT

In this paper the use of permutations as a mean of spreading the spectrum of the frequency channels in a general multicarrier modulation scheme is presented. Our first objective is to demonstrate how signature waveforms suitable for multiple access communications can be generated by applying permutation transformations to orthogonal channels generated with filterbanks or DFT techniques. The second objective is to motivate the use of such techniques as a mean of achieving immunity to timing errors, which can cause severe Adjacent Channel Interference (ACI) in multicarrier systems, due to large spectral overlap among adjacent channels. An efficient implementation of the interleaver implementing the considered permutations is also described.

Performance Analysis Of Wavelet-Based Generalized Sidelobe Cancellers

Authors:
Y. Chu, W.-H. Fang, *National Taiwan University of Science and Technology (TAIWAN)*
S.-H. Chang, *National Taiwan Ocean University (TAIWAN)*

ABSTRACT

In this paper, we carry out a performance analysis of the recently addressed wavelet-based generalized side-lobe canceller (GSC) which employs a set of regular M-band wavelet filters in the design of the blocking matrix involved. We begin our analysis by developing a closed form expression for the output signal-to-interference-plus-noise-ratio (SINR) of the GSC with a linear constraint under one jammer environment. Then, basing on this expression, we examine the relationship between the choices of the parameters involved in the wavelet-based GSC and the corresponding output SINR. Some simulations are provided to justify these results.

A Decorrelating Multiuser Receiver For DS/CDMA Systems Using TCM

Authors:

K. S. Kim, H. G. Kim, I. Song, S. R. Park, S. H. Yoon, H.M. Kim, *Korea Advanced Institute of Science and Technology, (KOREA)*

ABSTRACT

In this paper, we propose and analyze a multiuser receiver using a decorrelating filter and Viterbi decoders for trellis coded DS/CDMA systems with biorthgonal signal constellation in asynchronous channels. The biorthogonality is implemented by user signature waveforms and the decorrelating filter. The performance of the proposed system is investigated: it is shown that the proposed system can provide us with some coding gain and near-far resistance.

Bearings-Only Target Motion Analysis By Estimation Of Densities

Authors:
M. Spigai, J.-F. Grandin, *THOMSON-CSF-RCM (FRANCE)*

ABSTRACT

The aim of this paper is to compare, in the domain of the Bearings-only Target Motion Analysis (BTMA) with two observers, three approaches in terms of tracking performances: the Interacting Multiple Models (IMM) based on the Extended Kalman Filtering, the Hidden Markov Models (HMM) and the approximated densities filtering based on the maximum of entropy.

PLL Frequency Synthesizer System Utilizing Multi-Programmable Dividers

Authors:
Y. Sumi, Tottori SANYO Electric Co. (*JAPAN)*
K.Syoubu, S. Obote, Y. Fukui, Y. Itoh, Tottori University (*JAPAN*)

ABSTRACT

In this paper, we propose a new PLL frequency synthesizer utilizing multi-programmable dividers which can attain a higher speed lock-up time. Proposed PLL can increase the loop gain without the increase of reference frequency. Effectiveness of PLL with multi-programmable dividers and multi-phase detectors will be shown by the theoretical considerations and experimental results.

DSP-FLL For FSK Demodulation

Authors:
S. Obote, K. Syoubu, Y. Itoh, Y. Fukui, *Tottori University (JAPAN)*
Y. Sumi, *Tottori SANYO Electric Co. (JAPAN)*

ABSTRACT

In this paper, we propose a DSP type Frequency Locked Loop (DSP-FLL) using a Frequency Difference Detector (FDD). Since the FDD is employed, the DSP-FLL is controlled by the frequency. Therefore, the transfer function becomes first order and ringing does not occur. Furthermore, it can be understood from the detection property of the FDD that the cycle slip does not occur and DSP-FLL can pull in the frequency up to half of the sampling frequency.

Some Improvements Of A Rotation Invariant Autoregressive Method. Application To The Neural Classification Of Noisy Sonar Images

Authors:
H. Thomas, C. Collet, K. Yao, *Ecole Navale Groupe de Traitement du Signal (FRANCE)*
G. Bure, *Universite de Brest (FRANCE)*

ABSTRACT

This paper presents some improvements of a rotation invariant method based on AutoRegressive (AR) 2D Models to classify textures. The basic model and our improved version are applied to natural sidescan sonar images (with multiplicative noise) in order to extract a reduced set of relevant rotation invariant features which are then used to feed a MultiLayer Perceptron (MLP) for identification task. The basic method provides three AR parameters, estimated over a 3*3 pixel neighbourhood. We propose an extension of this method to a 5*5 pixel neighbourhood in order to take spatial interactions into account more efficiently. Three new features are estimated. Some analyses are conducted over these features to evaluate their interest. Classification results on four types of sidescan sonar images illustrate the efficiency of the proposed approach.

A New Approach To ECF Design Is Proposed Based On IRLS Algorithm

Authors:
A. Petrovi, A. Zejak, I. Simi, B. Zrni, *IMTEL- Institute of Microwave Techniques and Electronics (YUGOSLAVIA)*

ABSTRACT

A new approach to ECF design is proposed based on IRLS algorithm utilization. The obtained results show better performance in the case of a nonfeasible solution scenario

On A New Stochastic Version Of The EM Algorithm

Authors:
C. Campbell, S. Godsill, *Cambridge University (UK)*

ABSTRACT

We present an algorithm in which the Maximisation step of the EM algorithm is replaced by a Sampling step. We describe an application of the algorithm to noise reduction for an audio signal. The results of various simulations on synthetic data are presented and compared to the results obtained using the EM algorithm and the Gibbs Sampler. A major limitation of the EM algorithm is that it can converge on local stationary points. The results we present show how our algorithm successfully overcomes this limitation.

Variable Selection By A Reversible Jump MCMC Approach

Authors:

P. M. Djuric, *State University of New York at Stony Brook (USA)*

ABSTRACT

In this paper we address the problem of selecting the best subset of predictors in linear models from a given set of predictors. In computing the posterior probabilities of the various models, we propose to use the method of reversible jump Markov chain Monte Carlo sampling which cyclicly sweeps through the set of possible predictors and includes or removes them from the model one at a time. Special emphasis is given to a scheme that does not require sampling of the model coefficients and is based on predictive densities. Numerical results are provided that show the performance of the proposed approach.

A Comparison Of Sample Based Filters And The Extended Kalman Filter For The Bearings-only Tracking Problem

Authors:
N. Gordon, *Pattern and Information Processing, DERA, (ENGLAND)*
M. Pitt, *Imperial College of Science (ENGLAND)*

ABSTRACT

In this paper we are concerned with the development and evaluation of effective filtering methods for the bearings-only tracking problem. For this problem we consider a stationary observer who only obtains measurements of the bearing of a moving object subject to noise. We assume a linear Gaussian evolution in the states describing the motion of the object. We develop new sample based methods for filtering for this extremely non-linear model. We compare the performance of these methods to existing sample based methods and to the extended Kalman filter. Simulated scenarios are considered for evaluating the relative efficiency of the methods considered. Finally, an actual scenario arising from recordings made on a civilian ship is considered.

Some Examples Of Inverse Problems In Geophysics

Authors:
M. Lavielle, *Universite Paris V (FRANCE)*
D. Marquez, *Universite de Marne-la-Vallee (FRANCE)*

ABSTRACT

Most of the problems arising in geophysics consist in recovering some characteristics of the earth (reflectivity, resistivity, ...), from some measurements usually made at the top of the earth (seismic data, electromagnetic data, ...). The observations can be seen as the output data of a noisy (linear or non linear) system, that can be known, or unknown. When the system is unknown, it can be estimated, together with the noise variance, by using a stochastic version of EM algorithm. When the system is identified, some adaptations of MCMC techniques allow one to estimate the a posteriori distribution of the input sequence, according to the prior distribution. This prior can take into account the regularity of the input sequence, as well as the presence of abrupt changes in this sequence.

Bayesian Decomposition Trees
With Application To Signal Denoising

Authors:
D. Leporini, *CNRS/UPS and GDR ISIS, ESE, Gif-sur-Yvette Cedex (FRANCE)*

ABSTRACT

Tree-structured dictionaries of orthonormal bases (wavelet packet/Malvar's wavelets) provide a natural framework to answer the problem of finding a "best representation" of both deterministic and stochastic signals. In this paper, we reformulate the "best basis" search as a model selection problem and present a Bayesian approach where the decomposition operators themselves are considered as model parameters. Denoising applications are subsequently presented to substantiate the proposed methodology.

MCMC Methods For Restoration Of Nonlinearly Distorted Autoregressive Signals

Authors:
P. T.Troughton, S. J. Godsill, *University of Cambridge (ENGLAND)*

ABSTRACT

We approach the problem of restoring distorted autoregressive (AR) signals by using a cascade model, in which the observed signal is modelled as the output of a nonlinear AR process (NAR) excited by the linear AR signal we are attempting to recover. The Volterra expansion of the NAR model has a very large number of possible terms even when truncated at fairly small maximum orders and lags. We address the problem of subset selection and uncertainty in the nonlinear stage and model length uncertainty in the linear stage through a hierarchical Bayesian approach, using reversible jump Markov chain Monte Carlo (MCMC) and Gibbs sampling. We demonstrate the method using synthetic AR data, and extend the approach to process a long distorted audio time series, for which the source model cannot be considered to be stationary.

Multidimensional Optimisation Of Harmonic Signals

Authors:
P. J. Walmsley, S. J. Godsill, P. J. W. Rayner, *University of Cambridge (UK)*

ABSTRACT

Harmonic models are a common class of sinusoidal models which are of great interest in speech and musical analysis. In this paper we present a method for estimating the parameters of an unknown number of musical notes, each with an unknown number of harmonics. We pose the estimation task in a Bayesian framework which allows for the specification of (possibly subjective) a priori knowledge of the model parameters. We use indicator variables to represent implicitly the model order and employ a Metropolis-Hastings algorithm to produce approximate maximum a posteriori parameter estimates. A novel choice of transition kernels is presented to explore the parameter space, exploiting the structure of the posterior distribution.

Undermodelled Equalization: Extrema Of The Godard/Shalvi-Weinstein Criterion

Authors:

P. A. Regalia, *Institut National des Telecommunications (FRANCE)*
M. Mboup, *Universite Rene Descartes (FRANCE)*

ABSTRACT

The more promising results to date in blind equalization are restricted to so-called sufficient-order settings, in which all configurations in the combined (channel-equalizer cascade) impulse response space are attainable for some setting of the equalizer coefficients. Here we address issues related to equalizer undermodelling, in which only a proper subset within the combined impulse response space is attainable by adjusting the equalizer coefficients. We derive an analytic characterization of stationary points for the Godard and Shalvi-Weinstein criteria in undermodeled cases, an establish relations to the convergence of super-exponential algorithms in undermodeled cases.

Blind Single-Channel Interference Rejection Using Godard's Criterion

Authors:
D. Nussbaum, *TDF-C2R (FRANCE)*
O. Macchi, *LSS/CNRS/ESE (FRANCE)*

ABSTRACT

It is well known that the performance of the Constant Modulus Algorithm (CMA) for interference cancellation is limited by a so-called notch compromise. This paper presents a new recursive structure based on Godard's Criterion for blind interference suppression which overcomes this drawback. This interference-rejection structure is based on the linear prediction of the interference. The effectiveness of the new structure is studied in the presence of Co-Channel Interference (CCI) and additive Gaussian noise. It is shown that this structure can cancel predictable CCI.

Robust Second-Order Blind Equalization Of Polyphase Channels

Authors:

C. Papadias, *Lucent Technologies (Bell Labs Research) (USA)*

D. Gesbert, A. Paulraj, *Stanford University (USA)*

ABSTRACT

This work deals with the problem of linear polyphase blind equalization (BE), i.e. we are interested inequalizing the output of a single-input-multiple-output (SIMO) channel, without observing its input. A recent result by Liu and Dong [1] showed that if the sub-channel polynomials are co-prime, the equalizer output whiteness suffices for the equalization of a white input. Based on this observation, we propose a simple decorrelation criterion for second-order based blind equalization. Due to its second-order nature, this criterion is insensitive to the distance of the input from Gaussianity, hence it performs BE even for Gaussian or non-Gaussian inputs. Moreover, unlike other second-order techniques, our approach bypasses channel estimation and computes directly the equalizer. By doing so, it avoids the problem of ill-conditioning due to channel order mismatch which is crucial to other techniques. Combined to its good convergence properties, these characteristics make the proposed technique an attractive option for robust polyphase BE, as evidenced by both our analysis and computer simulation results.

A Constant Modulus Approach
To Multiple Access Interference Rejection

Authors:

J. Miguez, L. Castedo, *Universidade da Coruna (SPAIN)*

ABSTRACT

This paper addresses the problem of blind Multiple Access Interference (MAI) suppression in Direct Sequence (DS) Code Division Multiple Access (CDMA) systems. The Constant Modulus (CM) criterion is used to devise a blind adaptive DS CDMA receiver that achieves the same performance as the Minimum Mean Square Error (MMSE) receiver for high values of the Signal to Noise Ratio (SNR). The main limitation of the CM receiver is that an interferent user may be extracted instead of the desired one. Two approaches are investigated that practically overcome this problem when an estimate of the desired user code is available. The first one is an adequate choice of the adaptive receiver initial conditions and the second one is the incorporation of linear constraint.

Constrained Normalized Adaptive Filters For CDMA Mobile Communications

Authors:

J.A. Apolinario Jr., *Instituto Militar de Engenharia (BRAZIL) - Helsinki University of Technology (FINLAND) - COPPE/Universidade Federal do Rio de Janeiro (BRAZIL)*

S. Werner, T.I. Laakso, *Helsinki University of Technology (FINLAND)*

P.S.R. Diniz, *COPPE/Universidade Federal do Rio de Janeiro (BRAZIL)*

ABSTRACT

This work presents an extension of the classical LMS-based Frost algorithm to include both the Normalized LMS (NLMS) and the Binormalized Data-Reusing LMS (BNDR-LMS) algorithms. Two simple versions of these algorithms are derived for the Frost structure. These new algorithms were applied to a DS-CDMA mobile receiver. The results showed a considerable speed up of the convergence rate compared to the LMS Frost.

Direct Exploitation Of Non-Gaussianity As A Discriminant

Authors:
I. J. Clarke, *DERA (UK)*

ABSTRACT

The main purpose of this paper is to provide some additional insight into the several methods of cochannel blind signal separation that are based on the established concept of Independent Component Analysis (ICA). We compare published versions with a robust algorithm that has been devised and developed by the author. Most ICA algorithms are based on maximising the magnitudes of auto-cumulants and/or minimising various cross-cumulants of orthonormal principal components. In an alternative approach, the objective of independence between multiple signals is obtained by applying unitary rotations estimated from the rotational symmetry observed in joint probability distribution functions (jpdf). We show that the pairwise rotation-sensitive statistic, as used in the latter method, involves bivariate higher order statistical (HOS) terms common to other methods of ICA (but with differing relative weights). With this insight, we observe that, for optimum separation in non-Gaussian noise, the relative weighting applied to individual samples can also be modified. Another difficulty is that of measuring the performance achieved by different blind algorithms. This arises because the weighting of cumulants selected in a test of independence of the output waveforms can unfairly favour the algorithm that uses a similar weighting as it's objective function.

Performance Of Blind Discrete Source Separation

Authors:
O.Grellier, *I3S- CNRS (FRANCE)*
P.Comon, *I3S CNRS - Eurecom Institute (FRANCE)*

ABSTRACT

The aim of this paper is two-fold. In a first part, we investigate the ultimate performances of source separation in the case of BPSK and 4-PSK sources. These ultimate bounds are computed by looking first for the most favourable mixing matrices, for various SNRs, number of sources and number of sensors, and then by calculating the associated error probabilities. In a second part, we compare the behaviour of two 4-QAM source separation algorithms. One is based on the Constant Modulus (CM) property of these sources, and the other is based on the constance of their fourth power. The goal here is to see what kind of improvement brings the knowledge of the source distribution.

A Residual Bound For The Mixing Matrix In ICA

Authors:

L. De Lathauwer, J. Vandewalle, *K.U.Leuven (BELGIUM)*

ABSTRACT

In this paper, we derive a prewhitening-induced lower-bound on the Frobenius-norm of the difference between the original mixing matrix and its estimate in Independent Component Analysis. The derivation makes use of a lemma, stating that the sum of singular values of a matrix product cannot be larger than the sum of the products of the singular values of the distinct matrices.

Generalization Of A Maximum-Likelihood Approach To Blind Source Separation

Authors:

V. Zarzoso, A. K. Nandi, *University of Strathclyde (UK)*

ABSTRACT

In the two-source two-sensor blind source separation scenario, only an orthogonal transformation remains to be disclosed once the observations have been whitened. In order to estimate this matrix, a maximum-likelihood (ML) approach has been suggested in the literature, which is only valid for sources with the same symmetric distribution and kurtosis values lying in certain positive range. In the present contribution, the expression for this ML estimator is reviewed and generalized to include almost any source distribution.

De-Noising Of Experimental Signals From Pyroelectric Sensors By A Source Separation Method

Authors:
R. Huez, D. Nuzillard, A. Billat, *Moulin de la Housse (FRANCE)*

ABSTRACT

The aim of this work is to de-noise experimental signals from pyroelectric sensors. Its originality lies in the use of a double pyroelectric sensor especially designed to receive an instantaneous linear mixing of signals. The centre of the sensor receive mainly the signal, its periphery picks up mainly noise. The signal is processed with a separation source method enabling in this way to exploit the possibilities of this new device. The setting up of such a device has never been done previously neither in the view of experimentation nor in the view of information processing.

Hierarchical Mesh-Based Motion Estimation Using A Differential Approach And Application To Video Coding

Authors:

P. Lechat, H Sanson, *France Telecom (FRANCE)*
M, Ropert, *Atlantide (FRANCE)*

ABSTRACT

In this paper, we present a motion estimation scheme based on a non-uniform hierarchical triangular mesh and the nodal motion vectors optimization by means of a multi-resolution differential method. The mesh pyramid is obtained by locally splitting triangular elements, where the prediction error is high. The multiresolution source image pyramid is coupled to the mesh pyramid in such a way that the coarsest mesh corresponds to the lowest image resolution, leading to efficient and fast motion estimation and motion field refinement. To illustrate the performances of the method, results concerning the motion mesh and the prediction error are presented and compared to classical approaches with one single mesh level.

Compact Representation Of Motion Parameters For Object-Based Video Coding

Authors:

R. A Packwood, M. K Steliaros, G. R Martin, *University of Warwick (UK)*

ABSTRACT

We describe motion coding algorithms for use with the arbitrary shaped objects of emerging video coding standards. These algorithms exploit the temporal redundancy of the object shape description under conditions of concentrated diverse motion. In Variable Sized Block Matching (VSBM), the ambiguity of small area matching is reduced by coalescing blocks to yield accurate motion information over larger blocks. The technique is extended to exploit irregularly shaped areas of uniform motion within a small object. This Modified VSBM (MVSBM) can be enhanced by coding areas of identical motion within the spatial structure of the object. This spatial structure itself has temporal redundancy that can be reduced by inter-frame differential coding. For the MPEG-4 test sequences “Stefan” and “Children” the total amount of data required to specify motion shows improvements of 20% and 17% respectively.

Object Oriented Coding Using 3D Motion Estimation

Authors:

G. Calvagno, V. Orsatti, R. Rinaldo, L. Sbaiz, *Dipartimento di Elettronica e Informatica (ITALY)*

ABSTRACT

In this work, we apply 3D motion estimation to the problem of motion compensation for video coding. We model the video sequence as the perspective projection of a collection of rigid bodies which undergo a rototranslational motion. Motion compensation of the sequence frames can be performed once the shape of the objects and the motion parameters are determined. The motion equations of a rigid body can be formulated as a non linear dynamic system whose state is represented by the motion parameters and by the scaled depths of the object feature points. An extended Kalman filter is then used to estimate both the motion and the object shape parameters simultaneously. We found that the inclusion of the shape parameters in the estimation procedure is essential for reliable motion estimation. Our experiments show that the proposed approach gives the following advantages: the filter gives more reliable esti-mates in the presence of measurement noise in comparison with other motion estimators that separately compute motion and structure; the filter can effectively track abrupt motion changes; the structure imposed by the model implies that the reconstructed motion is very natural as opposed to more common block-based schemes; the parametrization of the model allows for a very efficient coding of motion information.

Adaptive Orthogonal Filter Bank Trees For Encoding Of Motion-Compensated Frame Differences

Authors:

W. Niehsen M. Wien, *Rheinisch-Westfalische Technische Hochschule (RWTH) (GERMANY)*

ABSTRACT

An algorithm for adaptive orthogonal filter bank trees for encoding of motion-compensated frame differences is presented. The proposed method is based on the hy-brid video coding scheme. For frame prediction, overlapped block motion compensation is applied. The prediction error frames are encoded using an adaptive wavelet packet algorithm based on a two-channel orthogonal FIR filter bank. Haar filters and minimum-phase Daubechies filters are employed. The signal boundaries are decomposed by means of optimized boundary filters. The proposed method is compared to the DCT-based hybrid video coding scheme.

Architectures For Vector-Tracing Based Motion Estimation For Mpeg2 Type Coding For TV And HDTV

Authors:
M. Gumm, F. Mombers, S. Dogimont, I. Remi, D. Mlynek, *Swiss Federal Institute of Technology (SWITZERLAND)*

ABSTRACT

A class of motion estimation VLSI architectures is presented which has been developed for the use in studio quality MPEG2 encoders. A new, fast motion estimation algorithm is applied which exploits both, temporal and spatial redundancies in motion vector fields and delivers near full search quality on large search windows. The proposed architectures are MIMD based, scalable both on chip and system level, and provide high flexibility according to a programmable RISC/co-processor approach. A chip tailored to TV resolution requirements is under design. The same architecture principle can be used to build HDTV capable motion estimation devices

Vector Tracing Techniques For Motion Estimation Algorithms In Video Coding

Authors:
M. Mattavelli, G. Zoia, *Swiss Federal Institute of Technology (SWITZERLAND)*

ABSTRACT

Despite the several efforts aiming at reducing the complexity, block motion estimation remains the most computationally demanding stage of video compression algorithms. This is particularly evident when sequences contain large displacements. Very large search windows are needed to achieve high quality coding when such critical conditions occur. This paper presents a block motion estimation technique based on the combination of motion trajectories tracing and of a modified genetic search heuristic. The proposed method is able to provide motion estimates in very large search windows with optimal coding results. The complexity reduction factor ranges up to more than two orders of magnitudes. The technique can be applied to any macroblock-based video compression standard and to any group of picture (GOP) structure.

A Background Memory Update Scheme For H.263 Video Codec

Authors:
K. Zhang, J. Kittler, *University of Surrey (UK)*

ABSTRACT

In the majority of video conference applications, the camera position and its focus are fixed and the scene in the picture is stationary. When the subject moves, some occluded background reappears in the scene. Such uncovered background part usually has to be spatially encoded. In this paper, we propose a background memory update scheme for H.263 video codec which can use the previously transmitted background scene as the reference so that a higher coding efficiency can be achieved. The proposed method only uses available information in the macroblock level to update the background memory. Thus the extra computational load is small and the synchronisation of the background memory is easy to achieve. The experimental results show that the proposed scheme gives improved performance both in terms of objective and subjective criteria.

Connected Operators For Sprite Creation And Layered Representation Of Image Sequences

Authors:
P. Salembier, O. Pujol, L. Garrido, *Universitat Politecnica de Catalunya (SPAIN)*

ABSTRACT

This paper proposes and discusses the use of motion-oriented connected operators for sprite creation. Motion-oriented connected operators are tools allowing the simplification of frames by removing objects that do not follow a given motion. They combine features of filtering and segmentation tools. They are, however, less computationally expensive than most motion-oriented segmentation algorithms. In this paper, we show how they can be used to effciently remove outliers with respect to the dominant motion and to create layered representation of sequences.

Low - Bitrate Video Coding With Third Order Geometric Transformations

Authors:

C. H. Slump, M. A.J.A. van Veen, F. J. de Bruijn, *University of Twente (NETHERLANDS)*

ABSTRACT

This paper describes low-bitrate video compression based upon the characterization of the new frame as a set of geometric transformations of objects of the previous frame. Objects with motion are detected and the motion is estimated. The estimated motion (motion field) is used to obtain the parameters for the geometric transformations. The pertinent geometric transformations are rotation, translation, zooming and isotropic and anisotropic distortion. The motivation for choosing this set of third-order transformations is that we have at our disposal special ASICS for real-time video processing. We only want to transform moving objects and therefore the boundaries of the moving objects must be known. The boundaries of the objects are represented by closed contours.

Generalized Quadratic Minimization And A Signal Processing Application

Authors:
A. Gorokhov, *Laboratoire des Signaux & Systemes(FRANCE)*
P. Stoica, *Uppsala University (SWEDEN)*

ABSTRACT

This paper deals with the problem of quadratic minimization subject to linear equality constraints. Contrary to the standard formulation, we assume the most general case of a possibly singular quadratic form. As we explain, the existing formal solution to this problem has several drawbacks. Our new approach is free from most of these drawbacks. It has a simple physical interpretation and is relatively easy to implement. Practical importance of this result lies in its numerous applications : filter design, spectral analysis, direction finding and blind deconvolution of multiple FIR channels. Here we focus on the blind deconvolution application.

Frisch Filtering Of Noisy Signals

Authors:
P. Guidorzi, *University of Bologna, (ITALY)*

ABSTRACT

The Frisch scheme, that considers additive independent noises on the measures of the input and output of a process, has recently led to the development of specific identification procedures. The obtained models, that have already been used to implement smoothing procedures congruent with the scheme, are here used to develop a filtering algorithm.

An Algebraic Approach
To The Subset Selection Problem

Authors:

A. Tewfik, M. Nafie, *University of Minnesota (USA)*

ABSTRACT

The need for decomposing a signal into its optimal representation arises in many applications. In such applications, one can usually represent the signal as a combination of an over-complete dictionary elements. The non-uniqueness of signal representation, in such dictionaries, provides us with the opportunity to adapt the signal representation to the signal. The adaptation is based on sparsity, resolution and stability of the signal representation. In this paper, we propose an algebraic approach for identifying the sparsest representation of a given signal in terms of a given over-complete dictionary. Unlike other current techniques, our approach is guaranteed to find the solution, given that certain conditions apply. We explain these conditions.

Person Identification Via The EEG Using Computational Geometry Algorithms

Authors:
M. Poulos, *University of Piraeus (GREECE)*
M. Rangoussi, *National Technical University of Athens (GREECE)*
E. Kafetzopoulos

ABSTRACT

A direct connection between the electroencephalogram (EEG) and the genetic information of an individual has been suspected and investigated by neuro-physiologists and psychiatrists since 1960. However, most of this early as well as more recent research focuses on the classification of pathological EEG cases, aiming to construct diagnosis tests from the EEG. The present work aims to establish an one-to-one correspondence between the genetic information of a (healthy) individual and certain features of his/her EEG, and - as a further goal - to develop a test for person identification based on EEG-extracted features. At the present stage the proposed method uses spectral information extracted from the EEG via the FFT and employs computational geometry algorithms to classify an unknown EEG as belonging to one of a finite number of individuals. Correct classification scores at the level of 95\%, in a limited scale experiment conducted on real data, show evidence that the EEG indeed carries genetic information and that the proposed method can be used to construct person identification tests based on EEG features.

A Vector Quantization Schema For Non-Stationary Signal Distributions Based On Ml Estimation Of Mixture Densities

Authors:
N. A. Vlassis, K. Blekas, G. Papakonstantinou, A. Stafylopatis

ABSTRACT

We show that by selecting an appropriate distortion measure for the encoding-decoding vector quantization schema of signals following an unknown probability density p(x), the process of minimizing the average distortion error over the training set is equivalent to the Maximum Likelihood (ML) estimation of the parameters of a Gaussian mixture model that approximates p(x). Non-stationary signal distributions can be handled by appropriately altering the parameters of the mixture kernels.

Fourth-Order Cumulant-Based Algorithms For Non-Minimum Phase MA System Identification

Authors:
D. P. Ruiz, A. Gallego, M. C. Carrion, *University of Granada (SPAIN)*

ABSTRACT

The purpose of this communication is the formulation of three new linear algorithms for blind non-minimum phase MA system identification. These methods have been derived starting from new equations involving system coefficients and q-slices of the k-th order cumulant sequence of the ouput MA system. In particular, these new algorithms use only fourth-order cumulants, and thus, are specially useful when the driving system noise has symmetric probability density function and a possible additive Gaussian noisy process contaminates the system ouput.

Stochastic System Identification For ATM Network Traffic Models: A Time Domain Approach

Authors:
K. De Cock, B. De Moor, *K.U.Leuven, ESAT-SISTA, (BELGIUM)*

ABSTRACT

In our paper we discuss a new time domain approach to the traffic identification problem for ATM networks. The Markov modulated Poisson process is identified in two steps.

By applying a nonnegative least squares algorithm we obtain in a very fast way a description of the first order statistics of the data. This first order characterisation includes also an estimate of the model order. Consequently, we are able to identify a Markov modulated Poisson process without a priori knowledge of the model order.

The identification of the second order statistics is based on unconstrained optimisation algorithms.

Blind Identification Of Sparse Multipath Channels Using Cyclostationary Statistics

Authors:
K. Abed-Meraim, Y. Hua, *The University of Melbourne (AUSTRALIA)*

ABSTRACT

Blind identification of a wireless communication channel is an important issue in communication system design. Most existing blind system identification techniques process the unknown information of the system from its output only. However, in many practical situation partial knowledge of the system transfer function is available.

By relying on this known information, the performance of channel identification and equalization can be significantly enhanced. In this paper, we introduce a new system identification technique that exploits both the a priori knowledge of the pulse shape filter and the multipath channel propagation model. The approach consists first in processing the cyclo-spectrum of the system output that is shown to be superimposed exponential function of the channel propagation delays and attenuations. Then, the frequency parameters, i.e., channel propagation parameters, are later estimated using the Matrix Pencils (MP) frequency estimation method.

On The Linearly Constrained Blind Multichannel Equalization

Authors:

S. Zazo, J. M. Páez-Borrallo, *ETS Ingenieros de Telecomunicación - Universidad Politécnica de Madrid (SPAIN)*

ABSTRACT

Working at baud rate regime (one sample per symbol), it is well known the equivalence between the linear prediction problem and a proper linearly constrained power minimization criterium. However, the success of this criterium as a blind equalization technique is limited to the minimum phase channel condition. On the other hand, if it is assumed a multichannel model (several samples per symbol or several sensors), it has been shown, under certain hypotheses, the minimum phase character of the multivariate transfer function; this property is in fact which allows one of the main approaches of multichannel blind equalization as a multivariate linear prediction problem. Our main goal is to introduce a family of adaptive algorithms dealing with the formulation of the blind equalization task as a linearly constrained cost function in order to generalize the baud rate case results: a detailed analysis of the mentioned cost functions is included and also supported by several computer simulations.

Evolutionary Multimodel Partitioning Filters For Multivariable Systems

Authors:
G. N. Beligiannis, D. A. Fotakis, S. D. Likothanasis, *University of Patras - Computer Technology Institute (GREECE)*
K. G. Berketis, *University of the Aegean (GREECE)*

ABSTRACT

It is known that for the adaptive filtering problem, the Multi Model Adaptive Filter (MMAF) based to the Partitioning Theorem is the best solution. It is also known that Genetic Algorithms (GAs) are one of the best methods for searching and optimization. In this work a new method, concerning multivariable systems, which combines the effectiveness of MMAF and GAs' robustness has been developed. Specifically, the a-posteriori probability that a specific model, of the bank of the conditional models, is the true model can be used as fitness function for the GA. Although the parameters' coding is more complicated, simulation results show that the proposed algorithm succeeds better estimation of the unknown parameters compared to the conventional MMAF, even in the case where it is not included in the filters bank. Finally, a variety of defined crossover and mutation operators is investigated in order to accelerate algorithm's convergence.

Adaptive AR Model Identification Based On The FAEST Filters

Authors:
S. D. Likothanassis, *University of Patras - Computer Technology Institute (C.T.I.) (GREECE)*
E. N. Demiris, *University of Patras (GREECE)*
D. G. Karelis, *T.E.I. of Patras (GREECE)*

ABSTRACT

A new adaptive approach for simultaneously selecting the order and identifying the parameters of an AutoRegressive model (AR) is presented. The proposed algorithm is based on the reformulation of the problem in the standard state space form and the subsequent implementation of a bank of fast a posteriori error sequential technique (FAEST) filters, each fitting a different order model. The problem is reduced then to selecting the true model, using the Multi-Model Partitioning (MMP) theory. Simulations illustrate that the proposed method is selecting the correct model order and identifies the model parameters, even in the case that the true model order does not belong to the bank of FAEST filters. The use of FAEST filters, reduce the computational effort, especially in the case of large order AR models. Finally, the algorithm is parallel by nature and thus suitable for VLSI implementation.

Markovian Approach To Random Fields Nonlinear Detection And Estimation Problems

Authors:
A.B. Shmelev, *Academician Mints Radiotechnical Institute*, *(RUSSIA)*

ABSTRACT

Applications of the Markovian approach to random fields nonlinear estimation and detection problems are described. The problem of radiowave phase estimation in presence of Gaussian noise is considered. It includes non-stationary estimation, stationary filtering and smoothing of phase fluctuations caused in particular by the wave distortions in turbulent medium. Some general relations between optimal detection and estimation problems for random fields observed in Gaussian noise background are obtained.

Blind And Semi-Blind Maximum Likelihood Techniques For Multiuser Multichannel Identification

Authors:

E. de Carvalho, L. Deneire, D.T.M. Slock, *EURECOM Institute (FRANCE)*

ABSTRACT

We investigate blind and semi-blind maximum likelihood techniques for multiuser multichannel identification. Two blind Deterministic ML methods based on cyclic prediction filters are presented [1]. The Iterative Quadratic ML (IQML) algorithm is used in [1] to solve it: this strategy does not perform well at low SNR and gives biased estimates due to the presence of noise. We propose a modification of IQML that we call DIQML to "denoise" it and explore a second strategy called Pseudo-Quadratic ML (PQML). As proposed in [2], PQML works well only at very high SNR. The solution we present here makes it work well at rather low SNR conditions and outperform DIQML. Like DIQML, PQML is proved to be consistent, asymptotically insensitive to the initialisation and globally convergent. Furthermore, it has the same performance as DML. A semi-blind extension combining these algorithms with training sequence based approaches is also studied. Simulations will illustrate the performance of the different algorithms which are found to be close to the Cramer-Rao bounds.

A Robust Method For Reflections Analysis In Color Image Sequences

Authors:
A. Teschioni, C. S. Regazzoni, *University of Genoa (ITALY)*

ABSTRACT

A robust algorithm for the detection of reflections in colour image sequences is here presented. The algorithm directly works on the RGB space and it performs an analysis at two different resolution levels, i.e. the pixel level and the region level. The proposed algorithm is used to improve the localisation and tracking capabilities of a video-based surveillance system. Results showing the goodness of the proposed approach are presented.

Locating Text In Color Document Images

Authors:
E. Ortacag, B. Sankur, *Bogazici University (TURKEY)*
K. Sayood, *University of Nebraska at Lincoln (USA)*

ABSTRACT

A novel text extraction algorithm from cluttered color document images is developed and tested. The algorithm consists of a color segmentation stage followed by rule-based filtering of non-text regions. Extraction of text segments algorithm uses the measurement of geometrical properties as well as characterness properties and a set of heuristic rules. The algorithm includes a fusion cycle of three different segmentation maps, and a restitution cycle to restore any deleted characters and/or their diacritical marks. The proposed method, proven successful in extraction of texts from many color document images, has applications in color image indexing and retrieval.

Characterization Of Circular Objects Based On The Hough Transform

Authors:
B. Correia, *INETI- DOP (PORTUGAL)*
F. Carvalho Rodrigues, *INETI- DOP, Universidade Independente (PORTUGAL)*

ABSTRACT

This paper presents a new image analysis approach based on the Hough Transform, for the characterisation of natural objects of approximately circular shape in real situations. To accomplish this purpose, an innovative interpretation and filtering methodology of the three dimensional Hough Transform (HT) accumulation space for circles was conceived and implemented, which proved to be crucial when dealing with shape and texture irregularities of natural objects. Some results that demonstrate the robustness of the developed approach under real testing conditions are also presented, namely concerning its application into a vision system that automates the volume measurement of piles of timber logs, allowing the evaluation of their diameter distributions.

How To Detect Dominant Points On 3-D Curves

Authors:
K. Sugimoto, *RWCP Multi-Modal Functions SANYO Laboratory, (JAPAN)*
F. Tomita, *Electrotechnical Laboratory, (JAPAN)*

ABSTRACT

A method for detecting dominant points on a 3-D digital curve is presented. The essential point is how accurately local properties such as curvatures and tangent vectors at every point are computed. This procedure first computes temporary properties by line or circle approximation using k-neighbors, then determines the optimum approximation by comparing the error value and correcting properties, and finally only for the point which has a large error value, adapts a threshold value K which defines the neighboring points for approximation. This method can detect not only corners but inflection points where two curves which have different convexities smoothly join and transition points where a straight line and a curve connect smoothly.

Curvature Variation Of Projected Cross-Sections From Straight Uniform Generalized Cylinders

Authors:

W. Puech, *University of Toulon (FRANCE)*

J.-M. Chassery, *Grenoble University (FRANCE)*

ABSTRACT

In this paper, we present a new approach for reconstructing Straight Uniform Generalized Cylinder (SUGC) with a scene mapped on its surface. In monocular vision, by using a priori knowledge about the support surface of pictures and projected cross-sections we reconstruct the surface in order to backproject the image on the surface. To reconstruct the surface we have to analyze the curvature of several projected cross-sections. An overview on this domain is found in the papers [Eggert 93, Huang 95, Chen 96]. The curvature variation analyses of surfaces [Kasa 96] and projected curves [Faugeras 95, Bruce 96] show the complexity of this problem.

3-D Measurements Using Conoscopy And Application To Ophthalmology

Authors:
C. Moser, G. Barbastathis, D. Psaltis, *California Institute of Technology (USA)*

ABSTRACT

In this paper we present a novel method to measure 3-D quasi planar or quasi spherical reflective surfaces with submicron depth accuracy. Two implementations are presented: a scanning and a non-scanning system. The non-scanning device allows fast measurements and can be applied for eye-shape measurements. The paper is organized as follows: in the introductory section, we first demonstrate the principle of the conoscopic effect leading to the formation of the interferogram. The second and third sections explain respectively, the scanning and non-scanning methods based on the conoscopic effect. We present the experimental results from a simple measurement and show how they conform with theory.

Projection-Based Registration Of Radiological Images

Authors:
C. Eroglu, A. E. Cetin, *Bilkent University (TURKEY)*

ABSTRACT

This paper introduces a projection-based method for registration of radiological images. The proposed method estimates the scale, translation and rotation parameters which define the global affine motion of an image with respect to a reference image. Using these estimated parameters, radiological images can be properly re-aligned to help diagnosis. The proposed method tries to match the horizontal and vertical projections of the image to be registered with the corresponding projections of the reference image in a hierarchical manner. After coarse estimates of the parameters are obtained using low resolution images, higher resolution images are utilized for the refinement of the parameter estimates. The experiments carried out on Computed Tomography (CT) and Magnetic Resonance (MR) images show that the performance of the algorithm in estimating the affine motion parameters is encouraging.

Lip Features For Speech And Speaker Recognition

Authors:
R. Auckenthaler, *Technical University Graz (AUSTRIA)*
J. Brand, J. S. Mason, *University of Wales Swansea (UK)*

ABSTRACT

This paper implicitly differentiates between the quality of visual representation necessary for speech and speaker recognition and assesses the performance of visual lip features with respect to well established audio features. Blue lip highlighted data is used to show how variations in lip measurements can influence speech and speaker recognition. From these experiments and other researchers results [1] it is postulated that the fine detail of the lips is critical for speaker recognition, but conversely, the same amount of detail does not noticeably improve visual speech recognition. Visual error rates of 26.3% and 70% are achieved for cross-digit speaker and cross-speaker speech recognition respectively.

Precise Eye Detection from Image Sequences

Authors:
W. Huang, J.-K. Wu, Q. Sun, C. P. Lam, *Kent Ridge Digital Labs (SINGAPORE)*

ABSTRACT

Automatic face location in complex scenes is extremely challenging in human face recognition system. The accuracy of the location is also a critical factor to the recognition performance. This paper presents a scheme to accurate eye detection from face sequences. After detecting the preattentive features by using the Gaussian steerable filter, face models are investigated to locate the whole face and the facial features such as eyes, nose and mouth. From the positions of eyes detected roughly, an approach to the precise detection of eyes is proposed, which is implemented with orientation calibration and phase registration. Because face is almost rigid object in image sequences, the direct registration can then work quite well.

A New Stereo Matching Algorithm Based On Probabilistic Diffusion

Authors:
S. H. Lee, C. W. Lee, *Seoul National University (KOREA)*
J. Il Park, *MIC3 - ATR (JAPAN)*

ABSTRACT

In this paper, the general formula of disparity estimation based on Bayesian Maximum A Posteriori (MAP) algorithm is derived and implemented with simplified probabilistic models. The probabilistic models are independence and similarity among the neighboring disparities in the configuration. The formula can be implemented into the some different forms corresponding to the probabilistic models in the disparity neighborhood or configuration. And, we propose new probabilistic models in order to simplify the joint probability distribution of disparities in the configuration. According to the experimental results, the proposed algorithm outperformed the other ones, such as sum of squared difference(SSD) based algorithm and Scharstein's method, and the proposed probabilistic models are reasonable and approximate the pure joint probability distribution very well with decreasing the computations to 0.01% of the generalized formula.

An Efficient Balanced Hierarchical Data Structure For Multiversion Accesses To Spatio-Temporal Data

Authors:
H. Dekihara, Y. Nakamura, *Hiroshima City University (JAPAN)*

ABSTRACT

In the management of spatio-temporal data, a data structure must manage multiple versions of a data structure efficiently, and provide quick and flexible search methods not only for temporal or spatial queries, but also for the combined queries of spatial and temporal intervals. The persistent MD-tree, called the PMD-tree is developed by extending a hierarchical data structure to support accesses to multiple versions. The PMD-tree has the novel properties that the tree representing any time aspect of a data structure is always balanced, and that the storage utilization rate is more than 66.6\%. The algorithms of the PMD-tree, space and time analyses, and search performances compared to the MD-tree are described in the paper.

Vision Navigation Of An Autonomous Vehicle By Fuzzy Reasoning

Authors:
W. Li, *Tsinghua University (CHINA)*
F. M. Wahl, *Technical University of Braunchweig (GERMANY)*

ABSTRACT

This paper presents a method for vision navigation of an autonomous vehicle based on fuzzy reasoning. Autonomous vehicles operating in real world require fast image processing and robustness with respect to noisy sensor readings and with respect to environment changes. The key subject of this paper is the integration of some special domain knowledge into a fuzzy rule base in order to enhance image segmentation performance. The proposed method is applied to navigate the THMR-III autonomous vehicle in out-door environments.

Measuring Tree-Ring Parameters Using The Generalised Fisher Ratio

Authors:
S. Zheng, C. G. Molina, *Anglia Polytechnic University (UK*)

ABSTRACT

A new technique is proposed to locate, count and measure the modes of a well-separated multimodal density distribution. It is based on a generalisation of the Fisher ratio for Gaussian mixture models. Under the assumption that the optimal mixture model that matches the right number of modes in a well-separated multimodal distribution is the one that maximises the Fisher ratio, we can find the optimal mixture model using a merging technique. We apply this new criterion to count, locate and measure tree-rings in dendrochronology and compare it to the most widely used method, the Hybrid Edge Detection algorithm.

ABRLS Algorithm For Time Variant Channel Equalisation

Authors:

T. Shimamura, *Saitama University, (JAPAN)*

C. F. N. Cowan, *The Queen's University of Belfast, (UK)*

ABSTRACT

This paper proposes a non-linear adaptive algorithm, the ABRLS algorithm, as an adaptation procedure for time variant channel equalisers. In the ABRLS algorithm, a coefficient matrix is updated based on the amplitude level of the received sequence. To enhance the tracking capability of the ABRLS algorithm, a parallel adaptation scheme is deployed which involves the structures of decision feedback equaliser(DFE). Computer simulations demonstrate that the novel ABRLS DFE provides a significant improvement related to the conventional RLS DFE on rapidly time variant communication channels.

The Use Of Evolutionary Optimisation In Channel Equalisation

Authors:
F. Sweeney, P. Power, C.F.N. Cowan, *The Queens University of Belfast (UK)*

ABSTRACT

This paper outlines the use of an Evolutionary Algorithm (EA) to perform the Equalisation of a non minimum phase channel. Conventional techniques utilising first and second order approximations of the error surface, have been demonstrated to be ineffective inachieving an optimal solution in continuous simulations, and have proved incapable of dealing with the more difficult non minumum phase problems. Using an EA, this paper will show how a consistent, near optimal, solution can be achieved.

An Algorithm For Channel Equalization With Adaptive Tap Position Control

Authors:

M. T. Arvind, P.G. Poonacha, *Silicon Automation Systems Pvt. Ltd. (INDIA)*

ABSTRACT

A stochastic adaptive algorithm is proposed for simultaneous channel equalization and reference tap placement. It is shown that the algorithm is gradient following in an expected sense. It is further shown that the long-term behaviour of the algorithm can be approximated by an associated differential equation. The stability of the equlibrium points of the differential equation are analyzed to yield pointers to the behaviour of the algorithm. The speed of convergence of the algorithm is investigated; it is shown that the algorithm is parallelizable and its speed performance can be considerably improved.

A Constant Modulus Array For Real Signals

Authors:
W. Pora, S. Lambotharan, J.A. Chambers, A.G. Constantinides, *Imperial College (UK)*

ABSTRACT

A modified Constant Modulus Algorithm (CMA) is proposed for real signals impinging upon an array. The algorithm solves the mix-up problem of CMA which occurs when real signals propagate through complex channels. Moreover, it decreases computational complexity and extends the maximumnumber of real sources which can be resolved by a given array. Simulations are presented to support the analysis.

Blind Iterative Separation Of CDMA Signals By Nonlinear Independent Component Analysis

Authors:

A.Hottinen, *Nokia Research Center (FINLAND)*

ABSTRACT

In this paper we study the use of projection pursuit[12] and independent component analysis[2] in blind demodulation of CDMA signals in the presence of high multiple access interference. The blind single-user receiver is assumed to be unaware of the signals transmitted by co-channel users. Conventionally, linear MMSE receivers have been applied to solve the problem[10]. However, it is well known that linear receivers are suboptimum for the given problem[16]. In this paper we propose a blind nonlinear multiuser receiver, where signal (code and bit) estimates are refined iteratively for the interfering users. This leads to rapid system identification and detection with performance superior to linear MMSE receivers.

A Bayesian Method For GPS Signals Delay Estimation

Authors:
J. Soubielle, I. Fijalkow, P. Duvaut , J.Y. Delabbaye, A. Bibaut, *ETIS / URA-CNRS (FRANCE)*

ABSTRACT

In GPS applications, positioning techniques are based on the characteristics of the pseudo-random code autocorrelation function. They don't take into account the eventuality of multipath propagation. One reflected signal may greatly disturb the measures of the usual Early-Late method. A maximum a posteriori (MAP) estimator is proposed in this study, introducing a multipath model and prior laws on both amplitude and delay parameters. This method deals with deconvolution of known functions with simple forms. Performance depends on data observation length but they greatly improve results.

Radio Frequency Interference Rejection In Radio Astronomy Receivers

Authors:
P. Fridman, *Special astrophysical observatory, (RUSSIA)*

ABSTRACT

Methods of radio frequency interference suppression in radio astronomy are considered. Estimations of signal-to-noise ratio for temporal and frequency methods of RFI rejection are made. Implementation of these methods in real-time digital signal processing could be an effective mean for supporting radio astronomy observations in worsening radio ecology environment. Radio telescope RATAN-600 experience shows the advantages of such a processing.

Spatial-Temporal Processing With Restriction Of Degrees Of Freedom For CDMA-Multiuser Detection

Authors:

O. Munoz, J.A. Fernandez-Rubio, *Dept. of Signal Theory and Communications, UPC, (SPAIN)*

ABSTRACT

Multipath distortion and multiple access interference are major limitations for the performance of wireless CDMA systems. In order to overcome both problems, a base-station antenna array RAKE receiver is proposed in this paper. Each one of the main propagation paths is assigned to one of the branches of the RAKE receiver. After removing the spreading sequence from the selected paths, the de-spread signals are used to estimate the spatial signature of the paths and the received power at each one of them. No training signal nor any a priori spatial information is required for the estimation. After estimating the path spatial signatures, an specific weight vector can be computed for each user. In addition to the multipath combination, this weight vector is able to cancel the interference from other users, achieving substantial robustness against fading and near-far effect at the same time.

An Efficient Non Linear Receiver For High Density Optical Recording

Authors:
L. Agarossi, *Philips Research, (ITALY)*
S. Bellini, *Politecnico di Milano (ITALY)*
F. Bregoli, P. Migliorati, *Univ. of Brescia (ITALY)*

ABSTRACT

This paper presents an innovative Non Linear Receiver (NLR) for the high density optical channel. This receiver is based on the combination of Maximum Likeli-hood Sequence Estimation (MLSE) and nonlinear Inter-Symbol Interference (ISI) cancellation. For the nonlinear channel description a suitable model based on the Volterra series has been adopted. Simulation results show that the proposed NLR performs better than traditional equalizers introduced for nonlinear channels, such as Nonlinear Adaptive Volterra Equalizer (NAVE) and Nonlinear Decision Feedback Equalizer (NDFE), and it offers significant advantages with respect to traditional MLSE.

Noise Reduction Of Image Sequences As Preprocessing For Mpeg2 Encoding

Authors:
P.M.B. van Roosmalen, J. Biemond, *Delft University of Technology (NETHERLANDS)*
A.C. Kokaram, *Trinity College (IRELAND)*

ABSTRACT

This paper investigates the influence of three classes of noise reduction filters as preprocessors to MPEG2 coding of noisy image sequences. From each class we select a representative noise filter that performs well and we investigate the effects of these on the coding efficiency. We also investigate the use of an adjusted MPEG2 compression scheme for simultaneous noise reduction and compression. The quality of the filtered sequences are evaluated objectively and subjectively after MPEG2 coding at bitrates varying from 1.0 to 6.5 Mbit/s.

Variational Principles Applied To Image Filtering

Authors:
L. J. Tardon-Garcia, J. Portillo-Garcia, *ETSI Telecomunicacion-UPM (SPAIN)*

ABSTRACT

In this paper we present a novel image filtering and reconstruction methodology mathematically supported by variational principles and partial diferential equations. We will establish the uni-dimensional formulation of the technique which gives sufficient insight about its use and is far easier to state and implement than the bi-dimensional one and has obvious computational advantages over the latter approach. Some considerations about the bi-dimensional formulation and its relation to the uni-dimensional one will be given.

Removal Of Mixed Noise By Adaptive Linear Combination Of Weighted Order Statistics (LWOS) Filters Based On Local Statistics

Authors:

A. Taguchi, T. Inoue, *Musashi Institute of Technology (JAPAN)*

ABSTRACT

It is well known that median filters are very useful for image restoration. The success of median filters is based on two properties: edge preservation and efficient impulsive noise attenuation. However, the median filter is not a perfect filtering operation. It may remove important image details and not sufficiently attenuate non-impulsive noise. The main reason is that the median filter uses only rank-order information of the input data within the filter window, and discards its original temporal-order information. In order to utilize both rank- and temporal-order information of input data, several classes of rank order based filters have been developed in recent years, such as L*l* filters[1], weighted median filters[2], FIR-WOS(weighted order statistics) hybrid(FWH) filters[3]. J. Song and Y. H. Lee have also proposed a filter whose structure provide a unified framework for representing both linear FIR and median-type nonlinear filters called linear combination of weighted order statistics (LWOS) filters[4]. The LWOS filter is combination of L-filters and WOS filters. For a typical image, each part differs considerably from other parts such that a description by a non-stationary process might be appropriate. Therefore, a time or space invariant filter will not necessarily yield satisfactory performance because the filter may have varying levels of performance over different portions of a signal. As a consequence, the noise filtering should be done by non-stationary or data-dependent methods. In this paper, we propose adaptive LWOS filters based on local statistics called data-dependent LWOS (DD-LWOS) filters in order to restore the signals corrupted by mixed noise (i.e., Gaussian noise and impulse noise are mixed). The DD-LWOS filter is a basically combination of data-dependent a-trimmed mean (DD-ATM) filter[5] and data-dependent weighted median (DD-WM) filters[6]. Thus, the proposed filter is superior to the DD-WM filter and the DD-ATM filter in Gaussian noise attenuation property and signal preservation property, respectively. Several design examples are presented showing the good performance of the proposed filter.

A New Operator For Image Processing Based On Lp-Filter Approximation

Authors:
M. Tabiza, Ph. Bolon, *LAMII/CESALP, Universit de Savoie (FRANCE)*

ABSTRACT

In this paper, we define and analyse some properties of a class of order filters. These filters can be regarded as adaptive L-Filters which can be tuned by setting only one parameter instead of N, where N is the filter size. Deterministic and statistical properties are discussed. Experimental results obtained on both synthetic and real images show that the noise reduction effect of the new flter is similar to that of optimal L-Filters and optimal Lp-Filters whereas the edge preservation is improved.

Signal And Image Denoising In Transform Domain And Wavelet Shrinkage: A Comparative Study

Authors:

R. Oktem, K. Egiazarian, *Tampere University of Technology (FINLAND)*
L. Yarovslavsky, *Tel Aviv University (ISRAEL)*

ABSTRACT

In this work, nonlinear local transform domain filtering is reviewed, and its relation with wavelet denoising is discussed. A postprocessing stage is applied to a number of transform domain denoised signals to obtain a better estimate of the original signal. Simulations are made over different Gaussian noise corrupted one-dimensional signals and images, in DCT and wavelet transform domains. Their performances with respect to threshold, transform basis and window size are compared.

Generation Of The Signature With The Structured Information Of The Image

Authors:
H. Kinoshita, M. Satoh, *Kanagawa University, (JAPAN)*

ABSTRACT

In this paper, we propose a new digital signature system for images. This system generates the signature information based on the structured information of the image. It increases the tolerance resistance against processing and coding work. This characteristics is deployed as an image retrieval method which allows rough sketch matching.

Self-Similarity Based Image Watermarking

Authors:
P. Bas, J.-M. Chassery, *UMR CNRS TIMC (FRANCE)*
F. Davoine, *UMR CNRS Heudiasyc (FRANCE)*

ABSTRACT

In this paper we present an original way to mark images. Firstly the different techniques in watermarking are presented. In a second part our approach is outlined. It consists in searching an IFS (Iterated function systems) in the image to hide new similarities in it. In a third part two different algorithms are presented: the first uses the luminance domain, the second uses the DCT (Discrete Cosine Transform) domain. Finally results and perspectives of our scheme are outlined.

Removing Spatial Spread Spectrum Watermarks By Non-Linear Filtering

Authors:

G. C. Langelaar, R. L. Lagendijk, J. Biemond, *Delft University of Technology (NETHERLANDS)*

ABSTRACT

Many watermarking methods are based on adding pseudo-random noise in the spatial domain. In general, the robustness of the watermark is determined by measuring the resistance to JPEG-compression, adding Gaussian noise and applying linear filters. Using these processing techniques the quality of the image must be affected significantly before the watermark is removed. In this paper a method is proposed to estimate a pseudo-random spread spectrum watermark only from the watermarked image. If this estimated watermark is subtracted from the watermarked image, the watermark is removed without distorting the image significantly.

Robust Image Watermarking In The Subband Or Discrete Cosine Transform Domain

Authors:

D. Tzovaras, N. Karagiannis, M. G. Strintzis, *Aristotle University of Thessaloniki (GREECE)*

ABSTRACT

In this paper a method is presented for copyright protection in digital images. Copyright protection is achieved by embedding an invisible signal, known as digital signature or watermark, in the digital image. The method proposed in this paper casts the signature in the frequency domain by slightly modifying the values of randomly selected DC coefficients of the Discrete Cosine Transform (DCT) of the image. The same method is applied also on the Subband or Wavelet Transform coefficients. An adaptive method is proposed also based on perceptual criteria that guarantees the invisibility of the watermark and avoids the deterioration of the image. Signature detection is done via hypothesis testing, without to use any information from the original image. The watermarks embedded by the proposed method are very resistant to JPEG and other frequently used compression. Experimental results using real image data verify the effectiveness of the method.

Non-Invertible Statistical Wavelet Watermarking

Authors:
G. Nicchiotti, E. Ottaviani, *Elsag Bailey R&D (ITALY)*

ABSTRACT

An important distinction among the watermarking schemes is between the techniques using the original image during the mark detection and techniques which do not use it. Most of the techniques presented in literature belongs to the first category. We will show that this fact has important consequences over the possibility to invalidate the claims of ownership, as it intrinsecally determines the invertibility of the scheme hence allowing the creation of fake originals. Then we will propose a novel non-invertible wavelet watermarking approach. Our watermark philosophy follows the guidelines described in [1] enriching the wavelet scheme we presented in [2] with the non-invertibility property.

User Interaction In Content-Based Video Coding And Indexing

Authors:
P. Correia, F. Pereira, *Instituto Superior Técnico - Instituto de Telecomunicações (POTUGAL)*

ABSTRACT

The current level of request for applications involving content-based video coding and video information retrieval is increasing the relevance of systems able to somehow 'understand' the content of video sequences. This type of systems will enable the provision of services where enhanced user interaction with the visual content is supported. Such an object-based video coder is being specified by ISO under the MPEG-4 project [1], while the standardization of video description capabilities to support content-based video indexing and retrieval is being considered by MPEG-7 [2]. In this paper, we discuss the issue of video content identification and characterization, and the importance of providing the means for the user to interact with the analysis process so that the achieved results can have a meaningful and powerful semantic value.

An Mpeg2 Compliant Psnr Controller For A Constant High Quality Studio Enviorment

Authors:
E. Frumento, R. Lancini, *CEFRIEL, Politecnico di Milano (ITALY)*

ABSTRACT

This paper proposes a new approach for the quality control of an Mpeg2 source. Actually the VBR coder is able to assure an average constant quality (namely less variable PSNR with respect to the CBR one) but it does not in a contribution application case, where the quality constraint plays an important role and a small variance of quality could be unacceptable. The reproduced quality is mostly relayed on the quantization parameters and we have studied how to control them to assure the same constant PSNR on each frame for a VBR-Mpeg2 profile 4:2:2 source.

Adaptive Tree-Structured Lattice Vector Quantization For Video Coding

Authors:
V. Ricordel, *MS-GESSY/ISITV, Universite de Toulon et du Var (FRANCE)*
M. Gabbouj, *DMI/SPL, Tampere University of Technology (FINLAND)*

ABSTRACT

The purpose of this paper is to introduce a new adaptive vector quantization (AVQ) for the compression of digital image sequences. In our previous studies [13, 14] we proposed a lattice vector quantizer (LVQ) based on the hierarchical packing of embedded truncated lattices. Now we investigate the capability of this LVQ for AVQ, through a scheme taking account of its tree-structure. Precisely, in order to fit the spatiotemporal statistics of the image sequences, the codebook is designed using a training procedure and in two parts, with: a stump from several types of training sequences; some branches added according to the sequence to be coded. Experimental results are given with the adaptive LVQ taking place in a subband coder.

Linear Combination Of Face Views For Low Bit Rate Face Video Compression

Authors:

I. Koufakis, B. Buxton, *University College London (UK)*

ABSTRACT

We present a technique for very low bit rate encoding of faces for videoconferencing applications over the Internet or mobile communication networks. The proposed scheme represents each new face view as a linear combination of three basis face views of the same person. The background area of the target face view is also encoded by the same method using the same basis views as used for the face area. Changes in the eyes and mouth regions are encoded using principal components analysis with three training sets of eyes and mouth images of the same person. The scheme results in a very compact representation of the face video data, and good quality images are reconstructed with an estimated bit rate from 1422 to 1620 bits/frame.

Computational Graceful Degradation For Video Decoding

Authors:
M. Mattavelli, S. Brunetton, *Swiss Federal Institute of Technology (SWITZERLAND)*

ABSTRACT

The implementation of software based video decoders is an advantageous solution over traditional dedicated hardware real-time systems. However, guaranteeing real-time performance, needed when processing video/audio bit-streams, by scheduling processing tasks of variable load and that might exceed the available resources, is an extremely difficult and in many cases an impossible challenge. In this paper we introduce the motivations and ideas of the Computational Graceful Degradation (CGD) approach for the implementation of video decoding. The goal is to provide re-sults of possibly lower visual quality, but respecting the real-time constraints, when the required processing needs exceed the available processing resources. This ap-proach is very attractive since it enables a more efficient usage of the hardware processing power without needing to implement resources for worst case complexity decoding. We show how such technique can be applied to the decoding of compressed video sequences. Simulation results for MPEG-4 and H.263 video compression standards showing some of the interesting achievements and potentialities of CGD approach are also reported. The presented approach can be virtually applied to any video compression standard and processor based platform, although the best performances are achieved when complexity prediction information provided as side information by the video standard is available. This option has been included in the new MPEG-4 standard.

Cost-Based Region Growing For Fractal Image Compression

Authors:
H. Hartenstein, D. Saupe, Universtitat Freiburg (GERMANY)

ABSTRACT

For application in fractal coding we investigate image partitionings that are derived by a merge process starting with a uniform partition. At each merging step one would like to opt for the rate-distortion optimal choice. Unfortunately, this is computationally infeasible when efficient coders for the partition information are em-ployed. Therefore, one has to use a model for estimating the coding costs. We discuss merging criteria that depend on variance or collage error and on the Euclidean length of the partition boundaries. Preliminary tests indicate that improved coding costs estimators may be of crucial importance for the success of our approach.

Context-Based Adaptive Arithmetic Coding For Lossy Plus Lossless Image Compression

Authors:
G. Deng, *La Trobe University, (AUSTRALIA)*

ABSTRACT

Lossy plus lossless (LPL) image compression technique provides an efficient way for searching an image data base and for transmitting images such as medical images where lossless compression is necessary. This paper presents a new method for the compression of the prediction error in the LPL technique. The new method is based on context modelling and adaptive arithmetic coding. Experiments show that the new method outperforms a previously published method in that it results in a better quality lossy reconstructed image and a smaller total bit rate.

High Compression Of Chrominance Exploiting Its Correlation With Luminance

Authors:
M. Bartkowiak, M. Domanski, *Politechnika Poznanska (POLAND)*

ABSTRACT

The paper reports research on original techniques for compression of chrominance data in video sequences transmitted at very low bit rates. Two chrominance components are converted into one scalar chrominance signal using vector quantization in chrominance plane. Three methods for scalar chrominance compression based on transform and predictive coding are described. The important issue is that presented algorithms exploit mutual correlation between vector quantized chrominance and the luminance component of video data. In transform coding techniques, the activity of reconstructed luminance estimated for its transform coefficients controls quantization steps of respective transform coefficients of scalar chrominance. In the approach based on predictive coding, detection of active areas in the luminance component allows for efficient coding of chrominance prediction error, where unnecessary transmission of zero{valued errors is avoided. The paper presents experimental results obtained for the proposed methods.

Progressive Medical Image Compression Using A Diagnostic Quality Measure On Regions-Of-Interest

Authors:
A. Signoroni, R. Leonardi, *University of Brescia (ITALY)*

ABSTRACT

Dealing with lossy compression of medical images requires particular attention whether for still images, video or volumetric slice-sets. In this work we propose an approach based on a selective allocation of coding resources that is directly related to the diagnostic task. We introduce the concepts of Region of Diagnostic Interest (RODI) and Diagnostic Quality as key links between the radiological activities and responsibilities and the functioning of a selective coding algorithm. The coding engine is a modified version of Shapiro's EZW algorithm and the coded bit-stream is fully progressive. The RODI selectivity corresponds to the choice of a set of subband weighting masks that depends on a small set of parameters handled and validated by the radiologist in a very natural manner. In conclusion, we present some experimental results that give interesting insights in favor of using lossy compression in a controlled fashion by a competent physician.

Multidimensional Companding For Lattice Vector Quantization Of Circularly Symmetric Densities

Authors:

S. F. Simon, W. Praefcke, *Aachen University of Technology (RWTH) (GERMANY)*

ABSTRACT

In order to realize a flexible vector quantizer with a large codebook but low complexity we consider a system which consists of a nonlinear mapping (compressor), a lattice vector quantizer, and the inverse of the compressor (expander). In general, the nonlinear expansion of the Voronoi regions of the lattice will increase their normalized second moments and hereby cause a loss. Recently, we derived this loss for the practically important class of optimal lattices [1]. This paper focuses on the special case of a source with a spherically symmetric probability density and a compander (compressor/expander) which is only a function of the magnitude of the input vector. Given the density of the source it is shown how the compander can be determined using high-rate quantization theory. Further, on the basis of a simple compander as an example, it is demonstrated that the loss introduced by the companding operation can be kept very small in practical situations.

Stochastic Gradient Algorithms In Active Control

Authors:
A. Gonzalez, *Universidad Politecnica de Valencia (SPAIN)*

ABSTRACT

This paper deals with some aspects concerning to the practical implementation of the stochastic gradient algorithms in active control. The control system under study is assumed to be a multichannel feedforward system and it is also assumed that there is not feedback signals from the secondary sources measured at the detection sensors. Several iterative algorithms were developed in [1] [2] for a frequency domain model of a multichannel active noise control system. Such iterative algorithms related to the p-norm of the error signal vectors were then applied to control pure tones in time domain [3]. When the disturbance signals can be modelled as stationary stochastic processes, a different framework is needed although there exist some analogies with the frequency domain model. This paper reviews the development and implementation techniques of stochastic gradient algorithms for active control under a general point of view, and then focuses on algorithms called minimax type which were studied in the frequency domain in [1] [2] [3].

Stereophonic Acoustic Echo Canceller Using Single Adaptive Filter Per Channel

Authors:
E. S. Kim, D. H. Youn, *Yonsei University (KOREA)*
K. H. Jeong, *LG Electronics Inc. (KOREA)*
W. C. Lee, *Soongsil University (KOREA)*

ABSTRACT

A conventional stereophonic acoustic echo canceller uses two adaptive filters to derive an estimate of one channel echo signal. Due to the strong correlation between the two channel signals, stereophonic echo canceller have various problems. Moreover, the computational complexity is about four times of the single channel system. In this paper, we introduce a pre-processing block deployed on the stereophonic echo canceller to generate the pseudo stereophonic signals. By utilzing the pseudo stereophonic signals, the echo signal can be estimated with single adaptive filter. The performance degradation due to the changes of the transmission room environment can be alleviated. Subjective tests show the validity of the pseudo stereophonic signals.

Modified Multidelay Adaptive Filter For Acoustic Echo Cancellation

Authors:
T. Trump, *Ericsson Radio Systems AB (SWEEDEN)*

ABSTRACT

In this paper a partial knowledge of measurement noise is incorporated into the multidelay frequency domain adaptive filtering scheme to improve its performance in colored measurement noise scenarios. The proposed algorithm is obtained by minimizing a BLUE criterion function using the stochastic gradient method and then switching over to the frequency domain to reduce the computational complexity. The performance of the algorithm in the situations of colored measurement noise is demonstrated by means of simulations using stationary as well as speech signals.

Audio Subband Coding With Improved Representation Of Transient Signal Segments

Authors:
J. Kliewer, *University of Kiel (GERMANY)*
A. Mertins, *University of Western Australia (AUSTRALIA)*

ABSTRACT

In this paper, we present a subband audio coding scheme with an attack-sensitive framing of the input audio signal, where the frame boundaries closely match both ends of the transient. Since each transient frame is processed as a symmetrically extended finite-length signal with a support-preservative MDFT filter bank, pre-echos can be almost completely avoided. Furthermore, the bit-allocation, which is calculated on a frame-by-frame basis, can be determined more accurately, because the calculation of the masking thresholds is carried out on signal segments with almost "stationary" energy distribution.

Adaptive Context Based Sequential Prediction For Lossless Audio Compression

Authors:

C. D. Giurcaneanu , I. Tabus, J. Astola, *Tampere University of Technology (FINLAND)*

ABSTRACT

In this paper we propose the use of adaptive-context-based prediction in a sequential mode for lossless audio compression. We show that lossless compression algorithms with sequential context based prediction can achieve better compression results than with forward-frame-based linear prediction. Two distinct algorithms are proposed and evaluated for audio signal sampled at 48 kHz with 16 bits/sample. The context quantization and prediction in both algorithms are similar to those used in an algorithm previously proposed for image compression[7] but new solutions are provided for modelling of errors and collecting the coding statistics. The first algorithm uses histogram bucketing in a small number of contexts in conjunction with an arithmetic coder. The second algorithm uses parametric modelling of errors in a large number of contexts in conjunction with Golomb-Rice encoding.

Active Binaural Sound Localization

Authors:

G. Reid, E. Milios, York University (CANADA)

ABSTRACT

Estimating the direction of arrival of sound in three dimensional space is typically performed by generalized time-delay processing on a set of signals from an array of omnidirectional microphones. This requires specialized multichannel A/D hardware. Our work is motivated by the desire to only use standard two-channel audio A/D, which is commonly available on personal computers. To estimate direction of arrival of sound, we change the pose of the microphones by mounting them on a computer-controlled pan-and-tilt unit. In this paper, we describe two approaches. The first uses two omnidirectional microphones on a fixed baseline, which has two rotational degrees of freedom. The second uses a directional microphone with two rotational degrees of freedom. We show algorithms for estimation of sound direction of arrival using these configurations. Preliminary experimental results demonstrate the feasibility of the approach.

Recognition Of Isolated Musical Patterns Using Discrete Observation Hidden Markov Models

Authors:
A. Pikrakis, S. Theodoridis, *University of Athens (GREECE)*
D. Kamarotos, *IPSA-Aristotle University of Thessaloniki (GREECE)*

ABSTRACT

Recognition of pre-defined musical patterns is very useful to researchers in Musicology and Ethnomusicology. This paper presents a novel efficient method for recognizing isolated musical patterns, using discrete observation Hidden Markov Models. The first stage of our method is to extract a vector of frequencies from the musical pattern to be recognized. This is achieved by means of a combination of a moving window technique with a largest-Fourier-peak selection algorithm. Each extracted peak frequency is subsequently quantized to a symbol of a finite and discrete alphabet. The resulting sequence of quantized frequencies is given as input to a set of Hidden Markov Models (HMM). Each HMM has been trained to model a specific pre-defined musical pattern. The unknown musical pattern is assigned to the model which generates the highest recognition probability. We have applied our method for the recognition of isolated musical patterns in the context of Greek Traditional Music. The resulting recognition rate was higher than 95%. Greek Traditional Music has distinct features, which distinguish it from the western equal-tempered tradition. To our knowledge, this paper presents a first effort for musical pattern recognition in the context of Greek Traditional Music using discrete observation Hidden Markov Models. Previous work was based on Dynamic Time Warping [8] .

Toward The Automatic Synthesis Of Nonlinear Wave Digital Models For Musical Acoustics

Authors:
F. Pedersini, A. Sarti, S. Tubaro, R. Zattoni, *Politecnico di Milano (ITALY)*

ABSTRACT

In this paper we propose and describe the principles behind our approach to sound synthesis through nonlinear wave digital modeling. The method is general enough to include a wide variety of nonlinearities that cannot be modeled through classical WDF principles. We also present an automatic synthesis method that, starting from a semantic description of the physical model, generates, validates and initializes an appropriate simulation source code with time-varying parameters.

Number Theoretical Means Of Resolving A Mixture Of Several Harmonic Sounds

Authors:

A. Klapuri, *Tampere University of Technology (FINLAND)*

ABSTRACT

In this paper, a number theoretical method is developed for the purpose of analyzing the spectre of a mixture of harmonic sounds. The method is based on the properties of prime numbers and on non-linear filtering. It is shown that a number theoretical approach is of vital importance in order to detect and observe harmonic sounds in musical polyphonies. The method is verified by applying it to the automatic transcription of piano music.

Identification Of Linear Time-Variant Systems By Spectral Correlation Measurements

Authors:

L. Izzo, A. Napolitano, *Universita di Napoli Federico II (ITALY)*

ABSTRACT

A nonparametric algorithm for the identification of linear time-variant systems is proposed. The class of systems considered maps almost-periodic inputs into almost-periodic outputs and includes as a special case the linear almost-periodically time-variant systems and, hence, the linear time-invariant systems. The proposed identification algorithm is based on spectral cross-correlation measurements between the input and output signals. It provides, under mild conditions, at least in principle, an arbitrarily accurate estimate of the system transmission function, provided that a sufficiently long collect time is considered.

A Polynomial-Algebraic Method For Non-Stationary Tarma Signal Analysis

Authors:
R. Ben Mrad, *University of Toronto (CANADA)*
S.D. Fassois, *University of Patras (GREECE)*
J.A. Levitt, *The Ford Motor Company (USA)*

ABSTRACT

Time dependent AutoregRessive Moving Average (TARMA) models, with parameters belonging to a subspace spanned by time-domain functions, offer appropriate representation for a fairly wide class of non-stationary signals. Yet, their estimation is difficult, due to problems related to local extrema and the need for accurate initial guess parameter values. In this paper a Polynomial-Algebraic (P-A) TARMA estimation method is introduced. The P-A method achieves low computational complexity while eliminating the need for initial guess parameter values and avoiding local extrema problems. Its performance is demonstrated via Monte Carlo simulations.

Performance Analysis Of Some Methods For Identifying Continuous-Time Autoregressive Processes

Authors:
T. Söderström, M. Mossberg, *Uppsala University, (SWEDEN)*

ABSTRACT

Identification of continuous-time AR processes by least squares and instrumental variables methods using discrete-time data in a `direct approach' is considered. The derivatives are substituted by discrete-time differences, for example by replacing differentiation by a delta operator. In this fashion the model is casted into a (discrete-time) linear regression. In earlier work we gave sufficient conditions for the estimates to be close to their true values for large data sets and small sampling intervals. The purpose of this paper is to further analyse the statistical properties of the parameter estimates. We give expressions for the dominating bias term of the estimates, for a general linear estimator applied to the continuous-time autoregressive process. Further, we consider the asymptotic distribution of the estimates. It turns out to be Gaussian, and we characterise its covariance matrix, which has a simple form.

The Bicepstral Distance Between Random Signals: A New Tool For Comparison Of Arma Models Identification Methods Based On Higher-Order Statistics

Authors:
J.-L. Vuattoux, E. Le Carpentier, *Institut de Recherche en Cybernétique de Nantes - UMR CNRS 6597 (FRANCE)*

ABSTRACT

This paper deals with distance measures for signal processing and pattern recognition. It proposes a new distance between stationary random signals, called the bicepstral one, which can be easily converted in a distance between ARMA models. This distance is based on higher order statistics, and therefore is not phase blind. Thus, it provides a good tool for comparison of ARMA model identification methods based on higher order statistics.

Discrete Time Equivalents Of Linear Systems Driven By White Noise

Authors:

N.C. Martins, *Polytechnic Institute of Setubal - Technical University of Lisbon (PORTUGAL)*
A. C. Rosa, *Technical University of Lisbon(PORTUGAL)*

ABSTRACT

Discrete-time equivalencies of linear systems driven by white noise, present technical difficulties which prevent the direct application of the usual methods originally developed for deterministic signals. An approach based on the modification of those methods, for use in presence of white noise, is presented. For the modified zero order hold (ZOH) method, an upper bound for the equivalence error variance is devised.

An HOS Based Statistical Test
For Multiplicative Noise Detection

Authors:
M. Coulon, J-Y Tourneret, *ENSEEIHT / GAPSE, (FRANCE)*

ABSTRACT

This paper addresses the problem of detecting the presence of multiplicative noise, when the information process can be modelled by a parametric AR process. A suboptimal detector based on higher-order cumulants (HOC) is studied. This detector consists of filtering the data by the fitted AR filter. HOC of the residual data are shown to be efficient for the detection problem.

Bussgang Test: A Powerful Non--Gaussianity Test

Authors:

G.Giunta, G.Jacovitti, G.Scarano, *Universita di Roma "La Sapienza"*

ABSTRACT

A process is said Bussgang if the cross-correlation function with its version passed through a zero-memory nonlinearity is proportional to the auto-correlation function of the process (invariance property). Gaussian processes are Bussgang processes too. As a consequence, Bussgangness tests may act as non-Gaussianity tests. Performance analysis shows that Bussgangness tests are more powerful than conventional Gaussian tests for correlated samples for a wide range of correlation coefficients and data lengths.

Nonlinear Constrained Optimization Using Lagrangian Approach For Blind Source Separation

Authors:
B. Stoll, E. Moreau, *Universite deToulon et du Var (FRANCE)*

ABSTRACT

The paper deals with the blind source separation problem. We introduce two new adaptive algorithms based on the minimization of constrained contrast functions using a Lagrangian approach. The algorithms “only” require one stage for separation and the approach is general in the sense that it can be used with any contrasts working with normalized vectors. The computer simulation shows good performances in comparison to the EASI algorithm.

Non--Minimum Phase AR Identification using Blind Deconvolution Methods

Authors:
G.Scarano, G.Panci, Universita di Roma “La Sapienza”

ABSTRACT

Identification of the parameters of non--minimum phase AR processes is formulated in the framework of the Super Exponential blind deconvolution scheme described in Shalvi. We show that the vector of the AR parameters lies in the range of a suitable (rank one) matrix formed using second and higher order statistics measured from the received data. The resulting algorithm is similar to the blind deconvolution scheme presented in [Signal Processing, vol.61, no.3, September 1997], which here is shown to be directly obtained from the optimality criterion underlying the Super Exponential blind deconvolution algorithm.

Limitations Of The ARMA W-Slice Method Using A Linear Combination Of Higher-Order Cumulants

Authors:

D. Zazula, *Faculty of EEand Computer Science (S LOVEN IA)*
J.-L. Vuattoux, *Institut de Recherche en Cybernetique de Nantes (FRANCE)*

ABSTRACT

The paper analyses autoregressive moving-average (ARMA) system identification method. This method belongs to higher-order statistical methods of a linear algebra type, showing a unique feature that the method works for any kind of model, i.e. MA, AR, or ARMA, and that the model's order (p; q) need not be known in advance. Our analyses of the ARMA approach proved that there is a class of systems not being identifiable. All these systems having poles s_i; $i = 1, \ldots, p$, and at least one zero of type of $(s_{i1}\ s_{i2}\ \ldots\ s_{ik})^{-1}$; $i_1, i_2, \ldots, i_{k-1} \in (1, \ldots, p)$ cannot be identified by ARMA w-slices using k^{th}-order cumulants, no matter whether with single cumulants, linear combination of cumulants, 1-D slices, or multidimensional slices. The analytical result is backed by simulations. Finally, we propose a procedure of verification of ARMA identifiability and an extension of ARMA w-slice in order to assure the identifiability.

Modulation Classification
Based On A Maximum-Likelihood Receiver
In The Cyclic-Hos Domain

Authors:

P. Marchand, *LIS-ENSIEG - TEAMLOG (FRANCE)*

C. Le Martret, *LIS-ENSIEG - Centre d' ELectronique de l'Armement (FRANCE)*

J.-L. Lacoume, *LIS-ENSIEG (FRANCE)*

ABSTRACT

A multiple hypothesis QAM modulation classification task is addressed in this paper. The classifier is designed in the framework of decision theory. A characteristic feature is extracted from the signal, and is compared to the possible theoretical features in the maximum-likelihood sense. This feature is composed of a combination between fourth-order and squared second-order cyclic temporal cumulants. The combination between cumulants of different orders is intended to bypass the uncertainty about the power of the signal of interest. As an application, we present simulated performance in the context of 4-QAM vs. 16-QAM vs. 64-QAM classification.

Localization Error Analysis
Of A Fast Broadband Matched Field Method
For A Single Hydrophone

Authors:

E. Sangfelt, B. Nilsson, *National Defense Research Establishment (SWEDWEN)*

ABSTRACT

We investigate the localization errors of a method which gives range and depth coordinates for a broadband sound source using a single hydrophone. It is based on the matching of theoretical multipath time-delays to those measured from the received signal. We focus on developing a fast method but still accurate for short ranges, thus allowing for real-time tracking of the source. We therefor investigate an analytical model for the time-delays based on straight line propagation but where the depth-varying sound velocity is accounted for. It is shown that this model provides much smaller localization errors compared to an isovelocity model, i.e. where a constant sound velocity is used.

Cascaded Scattering Functions For Sonar Signal Processing

Authors:
L. G. Weiss, L. H. Sibul , *The Pennsylvania State University (USA)*

ABSTRACT

This paper derives the total scattering function of a channel as a cascaded convolution of propagation and other scattering functions in a general structure with the narrowband (time-frequency) and wideband (time-scale) details presented as special cases. To do this requires the assumption that the probing signal is sufficiently rich so that the total spreading function can be represented by an inverse transform of the received signal. The derivation of this cascade of scattering functions then exploits properties of reproducing kernel Hilbert spaces (RKHS). Since scattering functions behave similarly to ambiguity functions (there is an increase in ambiguity as the signal propagates through the medium), a convolution of scattering functions can be viewed as propagation of ambiguities through a time-varying multipath medium. The incorporation of cascaded scattering functions into a detection processor then yields an improved technique for detecting signals in more complex time-varying environments.

A Fast Near-Field Bearing Estimation Via Spatial Exponential Kernel Distribution

Authors:
S.-H. Chang, W.-W. Lin, *National Taiwan Ocean University (TAIWAN)*
W.-H. Fang, *National Taiwan University of Science and Technology (TAIWAN)*

ABSTRACT

With its wide applications in various contexts of signal processing, the Direction of Arrival (DOA) of signal emerges to be a cascading issue. The majority of research, however emphasizes far-field estimation instead of near-field one. In addition, almost all of the high-resolution DOA algorithms are based on the eigenstructure decomposition which calls for lots of computations. In this paper, we present a new algorithm by using the spatial Exponential Kernel Distribution (EKD) function in conjunction with a source parameter estimation procedure to evaluate the involved instantaneous parameters. Since the EKD function can suppress cross terms effectively, the new algorithm, as justified by simulations, can render precise estimates for near field, multi-component signal sources. Furthermore, it avoids the computationally intensive eigendecomposition and scanning scheme by using the simpler time-frequency distribution function. To further speed up the processing, the kernel can be initially evaluated and stored in a matrix form.

A System For Seismic Data Processing

Authors:
K. Koester, M. Spann, *The University of Birmingham (UK)*

ABSTRACT

An unsupervised method to extract 2D and 3D inner earth structures from seismic reflection measurements is described. The system is supposed to support an expert interpreting seismic data. The problem is solved by extracting characteristic patterns for seismic events first (feature extraction stage) and then grouping similar patterns to geologic structures (segmentation stage). The feature extraction stage uses a Gabor filter bank for locally emergent frequency estimation. A region growing algorithm based on robust statistics is applied to the segmentation problem. The merging decision is obtained from the comparison of the mutual inlier ratio (MIR) of adjacent regions with a novel theoretically derived discrimination function.

A Feature Extractor Of Seismic Data Using Genetic Algorithms

Authors:
A. V. Adamopoulos, *University of Patras (GREECE)*
S. D. Likothanassis, E. F. Georgopoulos, *University of Patras - Computer Technology Institute (CTI) (GREECE)*

ABSTRACT

A novel signal analysis technique is presented and applied for the analysis of seismic data. The main task of the method is the extraction of feature information of a given timeseries. This task is accomplished by the implementation of a specifically designed Genetic Algorithm that is utilised for the evolution of conditional sets. The term conditional set refers to a set of boundary conditions. These boundary conditions must be fullfilled by subsequent samples of the timeseries. The implemented algorithm was applied on seismic data in order to investigate for the existence of conditional sets that appear more frequently than others. It was found that there exist conditional sets with high probability of appearance. Furthermore, it was found that some of the conditional sets are followed by data samples that apper small deviations from their mean value. It is proposed that conditional sets that combine these two properties (high probability of appearance and small deviation of the next data sample) can account for short-term prediction of seismic events.

Classification Of Microstructure Of Human Sleep Using EEG Modelling

Authors:
J. M. Allen Lima, A. C. Rosa, *Instituto Superior Técnico (PORTUGAL)*

ABSTRACT

This paper presents an automatic classifier for a sleep paradigm. This classification, Cyclic Alternating Pattern Sequence (CAPS) is based on the microstructure of sleep. The classifier is based on one Electroencephalogram, EEG, model to estimate its rhythmic activities. It was tested on 4 normal subjects and achieved better agreement with a visual reference than human-human agreement.

Wavelet Based Neural Network Architecture For ECG Signal Compression

Authors:
S. Kadambe, *Atlantic Aerospace Electronics (USA)*
P. Srinivasan, *Stream Machine Co. (USA)*

ABSTRACT

This paper addresses the problem of compressing Electrocardiogram (ECG) signals using the concept of adaptive sampling. The concept of adaptive sampling relates to optimum estimation of wavelet parameters that best represents a given signal. These wavelet parameters are estimated by minimizing the least mean square error between the original and approximated signal. Such an optimization approach is implemented within the frame work of neural networks by using wavelet non-linear functions in its neurons. We apply this technique for the compression of ECG signals. The experimental details of ECG compression are provided. For these experiments, the standard ECG database that was created by the American Heart Association (AHA) is used.

Process-Adapted Biosignal Processing An Approach Of Wavelet Transformation To Intracardial Leads

Authors:

R. Poll, T. Rauwolf, E. Meisel, *Dresden University of Technology (GERMANY*)

ABSTRACT

This paper presents an approach to surpass the common linearization of electrical cardiac signals and to use signal components of high frequencies for clinical examinations. For this, a high-resolution data acquisition equipment was designed. On-line real time segmentation as well as off-line evaluation of endocardial signals were carried out by wavelet packet transformation. In order to ensure the connection to the signal-generating process a modified Luo-Rudy model of the action potential and a model for two-dimensional spreading of excitation were developed. We found characteristic subbands in the range from 3 kHz to 5 kHz in each of our first 25 recordings. The method is suitable to determine direction and speed of the endocardial excitation front. Thus, it is to be useful for endocardial mapping and ablation technique. We currently are carrying out extended clinical studies.

Wavelet - Entropy Applied To Brain Signal Analysis

Authors:
O. A. Rosso, *Universidad de Buenos Aires (ARGENTINA) - Medical University Lubeck (GERMANY)*
S. Blanco, A. Figliola, *Universidad de Buenos Aires (ARGENTINA)*
R. Quian Quiroga, E. Basar, *Medical University Lubeck (GERMANY)*

ABSTRACT

Since traditional electrical brain signal analysis is mostly qualitative, the development of new quantitative methods is crucial for restricting the subjectivity in the study of brain signals. These methods are particularly fruitful when they are strongly correlated with intuitive physical concepts that allows a better understanding of the brain dynamics. Here we present the application of a new method, the Wavelet Entropy, to the analysis of epileptic seizures as well as visual evoked potentials.

Denoising Of Egg Signals Using Wavelet Shrinkage With Time-Frequency Depended Threshold

Authors:
N. Nikolaev, A. Gotchev, *Bulgarian Academy of Sciences (BULGARIA)*

ABSTRACT

A method for ECG denoising based on wavelet shrinkage approach has been in-vestigated. A shrinkage threshold which is high for the non-informative wavelet co-efficients and low for the informative coefficients is proposed. Its determination is based on suggestion, that the wavelet coefficients, representing different ECG zones, are noise influenced in different ways. The noise variance estimation have been carried out using the wavelet coefficients in high-frequency scales the position of which corresponds to the areas outside QRS complexes. The experimental threshold on the noise variance have been obtained. Limitations for the difference between the thresholds of every two adjacent coefficients throughout the scales in order to avoid Gibbs effects have been set up. The results of denoising show that proposed technique allows suppression of the parasite EMG and in the same way preservation of the parameters of the ECG signals.

Hidden Markov Models
Compared To The Wavelet Transform
For P-Wave Segmentation In Egc Signals

Authors:

L. Clavier, J.M. Boucher, E. Polard, *Ecole nationale supérieure des Télécommunications de Bretagne*

ABSTRACT

The aim of this study is to detect P-wave onset and end of electrocardiograms (ECG). This wave is important for detecting people prone to atrial fibrillation, one of the most frequent heart diseases, but the wave is very difficult to segment accurately because of its small amplitude and the very different shapes it can take. Two different methods are tested for the segmentation : the first one is based on Hidden Markov Models. Though results are good, some particular cases are not well segmented. However a second method based on the Continuous Wavelet Transform can solve those problems.

Hierarchical Neural Network System To Breasts Cancer Diagnosis

Authors:

L.F. Mingo, J. Castellanos,V. Gimenez, *Universidad Politecnica de Madrid (SPAIN)*
A. Villarrasa, *Hospital 12 de Octubre (SPAIN)*

ABSTRACT

This paper describes a hierarchical neural network system used as a help tool in the breast cancer diagnosis. The system divides the image using a tree scheme. The preprocessing of the areas pointed by the tree takes into account the environment that the element to analyze have, so as others computations, that are done using an adaptive filter. The neural network is in charge of classifying the areas into two possible classes, a healthy class or an ill class, using the obtained descriptors from the areas. All the data employed in this research have been obtained from the digitalization of the mammographies taken to some hospitalized patients.

An Instantaneous Frequency Dispersion Estimator For Detecting High Intensity Transient Signals In Human Blood Flow

Authors:
M. Baudry, S. Montrésor, *Laboratoire d'Informatique LIUM (FRANCE)*
P. Abraham, *Laboratoire de Physiologie (FRANCE)*
E. Roy, *Laboratoire d'Informatique LIUM - Laboratoire de Physiologie (FRANCE)*

ABSTRACT

This paper addresses the problem of automatic detection of High Intensity Transient Signals (HITS) in Doppler waveform. The aim of this work is to detect the passage of small particles in the human blood flow in order to prevent the formation of an arterial stenosis. Most existing methods are based on the short time Fourier transform of the Doppler ultrasound signal. We present an instantaneous frequency dispersion estimator for performing the automatic detection of HITS. After presenting the detection procedure, we also compare the detection results to that obtained by human experts. Finally, we comment the results and the limitations of this approach.

Detection Of Microcalcifications Using Non-Linear Filtering

Authors:
D. Meersman, P. Scheunders, D. Van Dyck, *University of Antwerp (BELGIUM)*

ABSTRACT

In the paper we describe a microcalcification detection scheme using non-linear filtering. Our scheme uses a combination of two highly non-linear filters: one for image enhancement, and one for the actual detection of the microcalcifications. Results of our method on the Nijmegen mammographic image database are given.

ECG Compression Using PCA With Polynomial Estimation Of Higher Order Coefficients

Authors:
D. Hamilton, W. Sandham, A. Blanco, *University of Strathclyde (UK)*

ABSTRACT

An ECG compression technique which achieves very low bit rates is presented. Principal Component Analysis (PCA) is used prior to identifying non-linear relationships between the orthonormal eigenvectors. Polynomials are used to approximate higher index transformed coefficients from lower ones. Reconstruction is similar to that used in the Karhunen-Loeve transform (KLT) technique but some of the coefficients have been estimated rather than stored explicitly. Exceptionally low bit rates have been achieved with good reconstruction appearance. For very low bit rates, improvements are consistently seen over KLT and further work is concentrating on generating clinically acceptable performance on all examples through relaxing the bit rate requirements slightly to achieve lower reconstruction errors.

Complete Coding Scheme Using Optimal Time Domain ECG Compression Methods

Authors:

R. Nygaard, D. Haugland, *Høgskolen i Stavanger (NORWAY)*

ABSTRACT

Traditionally, compression of digital ElectroCardioGram (ECG) signals has been tackled by heuristical approaches. However, it has recently been demonstrated that exact optimization algorithms perform much better with respect to reconstruction error. Time domain compression algorithms are based on the idea of extracting representative signal samples from the original signal. As opposed to the heuristical approaches, the exact time domain compression algorithms are based upon a sound mathematical foundation. By formulating the sample selection problem as a graph theory problem, optimization theory can be applied in order to yield optimal compression. The signal is reconstructed by interpolation among the extracted signal samples. Different interpolation methods have been implemented, such as linear interpolation and second order polynomial reconstruction. In order to compare the performance of the two algorithms in a fully justified way, the results have to be encoded. In this paper we develop an efficient encoding method based on entropy coding for that purpose. The results prove good performance of the exact optimization methods in comparison with traditional time domain compression methods.

Segmentation With Predictive Error And Recursive Likelihood Ratio Deviation Methods In Material Characterization

Authors:

S. Femmam, N. K. M'Sirdi, *L.R.P.- U.V.S.Q. (FRANCE)*

ABSTRACT

In this paper, abrupt changes detection techniques are used for segmentation of ultrasound signals in order to characterize materials. We describe the main motivations for the investigation of change detection problems. We show that the use of the Recursive Likelihood Ratio Deviation (RLRD) algorithm provides better results than the Linear Prediction Error (LPE) algorithm. The choice of the ultrasound signal length problem is used in the spectral ratio technique for determining the quality factor of a material.

Adaptive Directional Order Filtera
And Mathematical Morphology
For Road Network Extraction On SAR Images

Authors:

J. Chanussot, I. Issa, P. Lambert, *LAMII/CESALP - Universite de Savoie (FRANCE)*

ABSTRACT

In this paper, a method for automatic detection of linear features in SAR images, with application to road network extraction is proposed. A pre-filtering step is firstly performed using an adaptive directional weighted order filter. It aims to smooth the noise, enhance edges and preserve thin anisotropic structures. Then, the detection operator is applied : it is based on a geometrical model of the roads and uses morphological directional operators with appropriate structuring elements. Results are presented on real SAR ERS-1 satellite data.

Fractal Estimation In A Given Frequency Range Application To Smectite Images

Authors:

R. Harba, M. Cintract, *Université d'Orléans (FRANCE)*
M. Zabat, H. Van Damme, *CNRS et Université d'Orléans (FRANCE)*

ABSTRACT

In this communication, an efficient fractal estimation is proposed in a given frequency range. It is based on the maximum likelihood Whittle approximation for the stationary increments of fractional Brownian motion. Its efficiency is first shown on synthetic data generated by the Cholesky method. Then it is applied to smectite deposits on films analyzed by atomic force microscope. It is shown that the fractal parameter measured in the low frequency region allows to separate 3 groups of smectite clay images, and enables to recover chemical properties of the material.

Radio Frequency Interferences Suppression For Ultra Wide Band Radar

Authors:

B. Juhel, G. Vezzosi, *Université de Rennes (FRANCE)*

M. Le Goff, *Centre d'ELectronique de l'ARmement (DGA) GEOS/SDM (FRANCE)*

ABSTRACT

In this paper, a time domain processing of Ultra Wide Band SAR (Synthetic Aperture Radar) data and an algorithm to remove the Radio Frequency Interferences (RFI) measured with this kind of radar are described and tested. Usually a SAR is a narrow band system, but in our case it transmits nanosecond short pulses without carrier which have a spectral content from 100 MHz to 1 GHz. We investigate the possibilities of Ultra Wide Band radar imaging in order to scan the advantages over conventionnal radar. One of the potential advantage is the accurate target discrimination because of the short pulse duration, but one problem that we encounter is the RFI present in the UWB return. An algorithm based on Least Mean Square method is developped to extract and cancel these RFI.

Segmentation Of Urban Areas In Spot Images Using MRF

Authors:
F. Richard, *INRIA - University Rene Descartes (FRANCE)*
F. Falzon, *Alcatel Alsthom Recherche (FRANCE)*
J. Zerubia , G. Giraudon, *INRIA (FRANCE)*

ABSTRACT

Classification of urban areas from a density measure is required in many application fields. For instance, propagation models for cellular radio networks are parameterized by coefficients whose value is linked to the nature of the soil. Geomarketing is also interested in knowing the population concentration (urban, suburban ...). This paper presents the first results about the construction of an algorithm devoted to the classification of urban areas in SPOT panchromatic images. This algorithm has been inspired by [5, 3] and we point out the differences between their method and ours.

A Supervised Lloyd Algorithm
And Segmentation Of Handwritten Japanese Characters

Authors:

F. Morii, *University of Nara Women's University (JAPAN)*

ABSTARCT

A generalization of a supervised Lloyd algorithm to multilevel image thresholding is provided, whose algorithm is an iterative and fast one based on a minimum weighted-squared distortion criterion. After deriving an important relation between weight coefficients in the criterion and convergent thresholds, we consider a procedure on a reasonable selection of the weight coefficients by a supervisor through segmentation experiments. Using these results, the segmentation of the ETL8 handwritten Japanese character data base is studied.

Image Decomposition Capabilities Of The Joint Wavelet And Radon Transform

Authors:
G. Olmo, L. Lo Presti, E. Magli, *Politecnico di Torino (ITALY)*

ABSTRACT

It is known that the detection of segments in digital pictures can be effectively performed by means of the Radon Transform (RT). We describe a post-processing method based on wavelets, which provides information to be used in the recognition task. We show that the RT can also be successfully applied to the detection of rectangles, and is able to provide information on their width. A further generalization is described in the case of the detection of circular and square spots, along with an application to the ship and wake detection in aerial images of sea regions.

Segmentation Of Boron Carbide Microscopic Images Which Present Twins

Authors:
W. N. Matta, A. de Albuquerque Araújo, *Universidade Federal de Minas Gerais (BRAZIL)*
M. Carneiro de Andrade, *Centro de Desenvolvimento da Tecnologia Nuclear (BRAZIL)*
G. Bertrand, *Groupe ESIEE Paris – DI (FRANCE)*

ABSTRACT

Boron carbide microscopic images may present grains crossed by twins (or macles) viewed as straight lines in the inner part of each grain. The twins preclude the segmentation because they are very similar to the grain borders. To eliminate the twins a preprocessing step, which relies on mathematical morphology tools, is applied to the image. The main operation used is a directional opening, using structuring elements with several sizes and directions. After this step the image is ready to be segmented. A modified watershed algorithm, CBMA, is then used to segment the image. This algorithm employs two geometric attributes: an area and a depth criteria to merge the non-significant catchment basins.

Information Criteria For Histogram Thresholding Techniques

Authors:
P. Courtellemont, F. Jouzel, *University of Rouen (FRANCE)*
C. Olivier, *University of Poitiers (FRANCE)*

ABSTRACT

This paper deals with grey-level images histograms. In a first part, we show how possible it is to reduce the number of levels with the minimum of information loss, thanks to information criteria. The same criteria allow to threshold these histograms, giving the optimal number of thresholds.

CT Image Labeling
Using Simulated Annealing Algorithm

Authors:

Z. Majcenic, S. Loncaric, *Faculty of Electrical Engineering and Computing (CROATIA)*

ABSTRACT

Segmentation of computed tomography (CT) head images is required by many image analysis procedures for quantitative measurements of human spontaneous intracerebral brain hemorrhage (ICH). In this work we describe a stochastic method for segmentation of CT head images based on simulated annealing (SA). In the proposed method, the segmentation problem is defined as the pixel labeling problem with labels for this particular application set to: background, skull and ICH, and brain tissue. The proposed method is based on the Maximum A-Posteriori (MAP) estimation of the unknown pixel labels. A Markov random field (MRF) model has been used for the posterior distribution. The MAP estimation of the segmented image has been determined using the simulated annealing algorithm. Experimental results have demonstrated good results and proved the usability of the method.

Image Segmentation By A Multiresolution Approach

Authors:

N. Benamrane, K. Kies, *Université des Sciences et de la Technologie d'Oran (ALGERIA)*
J. Shen, *Bordeaux-3 University (FRANCE)*

ABSTRACT

Segmentation is the primary step to vision systems. It allows the generation of a compact description of image (edges, regions), but is often one of the most critical processing for subsequent treatments and especially for the matching process. This paper deals with extracting regions (segmentation) which can be used as primitives to facilitate the matching process in stereovision. In the literature, numerous segmentation techniques have been proposed, and they can be divided into three categories: classification methods, edge detection methods and region growing methods. Our technique consists in representing the initial image in a pyramidal form and then applying the region growing process at various resolutions. The linked pyramid is a very flexible data structure representing the image at different resolution levels (multi-resolution). The originality of our method resides on the use of region merging technique with the multi-resolution process. Experimental results for real stereo pair images are presented.

Invariant Supervised Texture Recognition Using Multi-Channel Gabor Filters

Authors:
M. Bresch, *University of Duisburg (GERMANY)*

ABSTRACT

A method of rotation and scale invariant texture recognition is proposed, which can also be employed for multi-object analysis and object recognition based on pattern analysis in noisy images. The recognition method uses data base textures, which are compared with a texture to be recognized and relies on extraction and classification of features. The features are extracted using multi-channel polar logarithmic Gabor filtering of the data base textures and the texture to be recognized with the same definite filter bank. The polar logarithmic orientation of the Gabor filters guarantees rotation and scale invariance. The classification of the features is carried out by symmetric phase-only matched filtering. The performance of the method has been tested on Brodatz textures, hone textures, and textile textures with perfect recognition results. Rotation angle and scale factor can be determined with arbitrary precision by the classification scheme.

Supervised Texture Classification - Selection Of Moment Lags

Authors:
V. Antoniades, A. K. Nandi, *University of Strathclyde (UK)*

ABSTRACT

This papers deals with supervised texture classification. The extracted features are the image second and third order moments. The number of possible moment lags for 2-D signals increases rapidly with the order of the moment even for small lag neighbourhoods. The paper focuses on the selection of moment lags that optimise classification performance. Lag selection also serves another purpose: it waives us from the trouble of calculating a large number of moments every time a new sample is to be classified. Lag selection is performed by a full stepwise feature selection method using four different feature evaluation measures. The selected moments are driven to four classifiers and comparative classification results are obtained.

Unsupervised Texture Segmentation Using Discrete Wavelet Frames

Authors:
S. Liapis, N. Alvertos, G. Tziritas, *University of Crete (GREECE)*

ABSTRACT

Image segmentation could be based on texture features. In this work, an unsupervised algorithm for texture segmentation is presented. Texture analysis and characterization are obtained by appropriate frequency decomposition based on the Discrete Wavelet Frames (DWF) analysis. Texture is then characterized by the variance of the wavelet coefficients. The unsupervised algorithm determines the regions to characterize each different texture content in the image. For applying the algorithm, it is necessary to know only the number of the different texture contents of the image.

Scale Invariant Texture Classification With Mathematical Morpology

Authors:
V. L. Ballarin, E. G. Moler, M. Brun, *Universidad Nacional de Mar del Plata (ARGENTINA)*

ABSTRACT.

One of the most interesting fields of research in Digital Image Processing is the segmentation of an image into different objects. The detection of regions with different textures is one of the techniques used to achieve this objective. These techniques differ among each other in the texture parameters used and the way to obtain them. In this work the characteristics associated to each region will be obtained from the application of the Size Criteria of Successive Openings. This technique of the Mathematical Morphology analyses how the image interacts with different structuring elements. Taking parameters from the images, we obtain size distribution functions associated to each region. We demonstrate that we can obtain characteristics invariant under changes of scale using the statistical moments of the associated normalised density functions. These characteristics can be used as patterns for a further classification.

An Identification Method For Texture AR-2D Modelling Based On Auto- And Partial Correlation Measures

Authors:
I. Claude, A. Smolarz, *Universite deTechnologie de Troyes (FRANCE)*

ABSTRACT

This paper deals with the identification of the model order for a bidimensional autoregressive (AR-2D) texture model. It means the automatic choice of the number of neighbours in the prediction set of the model and their spatial position. The method, called mixed correlation method is based on partial and autocorrelation measures and fastly and efficiently allows to find an adapted model for all microtextures. In a textured samples classification procedure, these adapted models improve the percentage of good classification in comparison with a classical approach consisting in taking the same prediction set for all the textures.

Mean Depth-Width Ratio Of Extrema
As Textural Feature
For Automated Cell Proliferation Analysis

Authors:

C. Gabriel, V. Leticia, *Centro de Instrumentos-UNAM (FRANCE)*
M. Jorge, *Ecole Nationale Supérieure des Télécommunications (FRANCE)*

ABSTRACT

As a step towards automation of *Mitotic Index* estimation for cell proliferation studies, we introduce in this work a roughness feature of surface-intensity images: the *mean depth-width ratio of extrema* (MDWRE). This feature allowed identification of variable-shaped metaphases and interphase nuclei in the presence of many artifacts (one metaphase per hundreds of nuclei and thousands of artifacts). The texture of the cytological objects (seen as rough surfaces) was quantified by scanning in one dimension the lines contained in a closed contour. MDWRE resulted suitable for image magnifications as low as (x10), making possible a faster scanning of the slides. The use of this feature gave +14%, +65%, +133% and +133% better performance figures than classical textural features derived from co-occurrence matrices such as Contrast, Energy, Entropy and Angular 2nd Moment respectively, and +51% better than the Relative Extrema Density (RED). The MDWRE per object and the shape of the histogram of the depth-width ratio of gray-level roughs, have shown to be very useful as textural features for the classification of metaphase images.

On The Initial Label Configuration Of MRF

Authors:
G. D. Guo, *Chinese Academy of Sciences (CHINA)*
S. Yu, S. de May, *INRIA Centre Sophia Antipolis (FRANCE)*

ABSTRACT

Many image analysis and computer vision problems can be formulated as a scene labeling problem. Bayesian modeling of images by Markov random fields is a coherent theoretical framework. It has however some drawbacks, one of which is the computational complexity. Because the energy function has many local minima, most deterministic or local optimization algorithms depend on the starting point, i.e., the better the initialization, the bigger the chance of the final result close to the global optimum. Usually, the initialiation uses maximum likelihood estimation(MLE) for each site and it is not good enough in practice. We propose two approaches to obtain better initialization than the traditional MLE, one is based on circular window sampling, another is \spotlight" operator. From the experiments, we can see the two approaches are very effective and efficient for initializations, and the fast ICM optimization based on them can provide satisfactory labeling results.

Curvature Scale Space Based Image Corner Detection

Authors:
F. Mokhtarian, R. Suomela, *University of Surrey (UK)*

ABSTRACT

This paper describes a new method for image corner detection based on the curvature scale space (CSS) representation. The first step is to extract edges from the original image using a Canny detector. The Canny detector sometimes leaves a gap in T-junctions so during edge extraction, the gaps are examined to locate the T-junction corner points. The corner points of an image are defined as points where image edges have their maxima of absolute curvature. The corner points are detected at a high scale of the CSS and the locations are tracked through multiple lower scales to improve localization. The final stage is to compare T-junction corners to CSS corners and remove duplicates. This method is very robust to noise and we believe that it performs better than the existing corner detectors.

Road Boundaries Detection Using Color Saturation

Authors:
P. Charbonnier, Y. Guillard, *Laboratoire Regional des Ponts et Chaussees de Strasbourg (FRANCE)*
P. Nicolle, J. Charrier, *Laboratoire Central des Ponts et Chaussees (FRANCE*)

ABSTRACT

Our aim is to automatically detect the road borders in a road scene image. This is useful to many road scenes analysis applications, in both fields of vehicle guidance and civil engineering. Difficulties arise because pavements are often heterogeneous and because illumination variations often occur in outdoor scenes. Some vehicle navigation projects use colour images for road borders detection [1, 3], but most of the time they consider RGB features, which are sensitive to shadows and pavement defects. Besides, the proposed classification methods are often computationally intensive [7]. In contrast, chromatic saturation is a discriminant one-band feature, so a simple threshold can be used, making the classification quite fast. Moreover, chromatic saturation is quite insensitive to shadows and pavement variability.

Directional Second Order Derivatives : Application To Edge And Corner Detection

Authors:
S. Guillon, M. Donias, *Equipe Signal et Image ENSERB and PRC - GDR ISIS (FRANCE)*

ABSTRACT
In this paper, we propose an adaptive scheme to design directional second order derivatives orthogonally and tangentially to the local edge. The principle lies on the definition of two adaptive filter masks which estimate the two derivatives along the normal (n) and tangential (t) directions. Both filter masks are controlled by an adaptive set of coefficients tuned in accordance with the local grey level distribution. The two new filters are then applied respectively to edge and corner detection : edge detection is achieved by detecting the zero-crossing of the derivative along n, and corner detection is obtained by thresholding the amplitude of the derivative along t. Results of these detections are provided on synthetise and real-world images, and swow the robustness of the new proposed approach.

Linear Feature Detectors
And Their Application To Cereal Inspection

Authors:

E.R. Davies, D.R. Mason, M. Bateman, *University of London (UK)*

J. Chambers, C. Ridgway, *Central Science Laboratory - Sand Hutton (UK)*

ABSTRACT

This paper extends the vectorial strategy for designing line segment detection masks to large windows, thereby permitting line segments to be detected and orientated efficiently with just two masks in a far greater number of applications than the original formulation [1]. Our own application exemplifies the value of these linear feature detectors for the purpose of locating insects in grain, and 7 ¥ 7 masks seem well suited to this task. This work is expected to be useful in many applications where linear features have to be located, and also those where thin lines have to be tracked – as in fingerprint recognition and document interpretation.

A Probabilistic Framework For The Hough Transform And Least Squares Pose Estimation

Authors:

J. S. Marques, *Instituto Superior Tecnico (PORTUGAL)*

ABSTRACT

The Hough transform and least squares pose estimation are usually considered as unrelated methods based on different assumptions. This paper presents a unified perspective of both approaches, in a probabilistic framework. It is shown that both methods compute maximum likelihood estimates of the object pose, based on different probability distributions. The main properties of the algorithms are discussed and their performance is characterized by Monte Carlo tests. This methodology can be extended to other shape representation algorithms.

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