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1998 IEEE International Conference on Acoustics, Speech and Signal Processing
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ICASSP98 Paper Abstract

On the Robust Incorporation of Formant Features into Hidden Markov Models for Automatic Speech Recognition

Authors:
Philip N Garner, DERA, (U.K.)
Wendy J Holmes, DERA, (U.K.)

Volume 1, page 1, paper no. 1546

Abstract:
A formant analyser is interpreted probabilistically via a noisy channel model. This leads to a robust method of incorporating formant features into hidden Markov models for automatic speech recognition. Recognition equations follow trivially, and Baum-Welch style re-estimation equations are derived. Experimental results are presented which provide empirical proof of convergence, and demonstrate the effectiveness of the technique in achieving recognition performance advantages by including formant features rather than only using cepstrum features.
ICASSP98 Paper Abstract

Exploiting Acoustic Feature Correlations by Joint Neural Vector Quantizer Design in a Discrete HMM System

Authors:
Christoph Neukirchen, Gerhard-Mercator-University Duisburg, (Germany)
Daniel Willett, Gerhard-Mercator-University Duisburg, (Germany)
Stefan Eickeler, Gerhard-Mercator-University Duisburg, (Germany)
Stefan Müller, Gerhard-Mercator-University Duisburg, (Germany)

Abstract:
In previous work about hybrid speech recognizers with discrete HMMs we have shown that VQs, that are trained according to an MMI criterion, are well suited for ML estimated Bayes classifiers. This is only valid for single VQ systems. In this paper we extend the theory to speech recognizers with multiple VQs. This leads to a joint training criterion for arbitrary multiple neural VQs that considers the inter VQ correlation during parameter estimation. The idea of a gradient based joint training method is derived. Experimental results indicate that inter VQ correlations can cause some degradation of recognition performance. The joint multiple VQ training decorrelates the quantizer labels and improves system performance. In addition the new training criterion allows for a less careful way of splitting up the feature vector into multiple streams that do not have to be statistically independent. In particular the usage of highly correlated features in conjunction with the novel training criterion in the experiments leads to important gains in recognition performance for the speaker independent Resource Management database and gives the lowest error rate of 5.0% we ever obtained in this framework.
ICASSP98 Paper Abstract

A NN/HMM Hybrid for Continuous Speech Recognition with a Discriminant Nonlinear Feature Extraction

Authors: Gerhard Rigoll, *Gerhard-Mercator University Duisburg, (Germany)*
Daniel Willett, *Gerhard-Mercator University Duisburg, (Germany)*

Volume 1, page 9, paper no. 1739

Abstract:
This paper deals with a hybrid NN/HMM architecture for continuous speech recognition. We present a novel approach to set up a neural linear or nonlinear feature transformation that is used as a preprocessor on top of the HMM system's RBF-network to produce discriminative feature vectors that are well suited for being modeled by mixtures of Gaussian distributions. In order to omit the computational cost of discriminative training of a context-dependent system, we propose to train a discriminant neural feature transformation on a system of low complexity and reuse this transformation in the context-dependent system to output improved feature vectors. The resulting hybrid system is an extension of a state-of-the-art continuous HMM system, and in fact, it is the first hybrid system that really is capable of outperforming these standard systems with respect to the recognition accuracy, without the need for discriminative training of the entire system. In experiments carried out on the Resource Management 1000-word continuous speech recognition task we achieved a relative error reduction of about 10% with a recognition system that, even before, was among the best ever observed on this task.
ICASSP98 Paper Abstract

Incorporating Voice Onset Time to Improve Letter Recognition Accuracies

Authors:
Partha Niyogi, Bell Labs, Lucent Technologies, (U.S.A.)
Padma Ramesh, Bell Labs, Lucent Technologies, (U.S.A.)

Volume 1, page 13, paper no. 2085

Abstract:
We consider the possibility of incorporating distinctive features into a statistically based speech recognizer. We develop a two pass strategy for recognition with a standard HMM based first pass followed by a second pass that performs an alternative analysis to extract class-specific features. For the voiced/voiceless distinction on stops for an alphabet recognition task, we show that a linguistically motivated acoustic feature exists (the VOT), provides superior separability to standard spectral measures, and can be automatically extracted from the signal to reduce error rates by 48.7% over state of the art HMM systems.
ICASSP98 Paper Abstract

On the Use of Normalized LPC Error Towards Better Large Vocabulary Speech Recognition Systems

Authors:
Rathinavelu Chengalvarayan, Lucent Technologies, (U.S.A.)

Abstract:
Linear prediction (LP) analysis is widely used in speech recognition for representing the short time spectral envelope information of speech. The predictive residues are usually ignored in LP analysis based speech recognition system. In this study, the normalized residual error based on LP is introduced and the performance of the recognizer has been further improved by the addition of this new feature along with its first and second order derivative parameters. The convergence property of the training procedure based on the minimum classification error (MCE) approach is investigated, and experimental results on city name recognition task demonstrated a 8% string error rate reduction by using the extended feature set as compared to conventional feature set.
ICASSP98 Paper Abstract
Use of Periodicity and Jitter as Speech Recognition Features

Authors:
David L. Thomson, Lucent Technologies, (U.S.A.)
Rathinavelu Chengalvarayan, Lucent Technologies, (U.S.A.)

Volume 1, page 21, paper no. 2322

Abstract:
We investigate a class of features related to voicing parameters that indicate whether the vocal chords are vibrating. Features describing voicing characteristics of speech signals are integrated with an existing 38-dimensional feature vector consisting of first and second order time derivatives of the frame energy and of the cepstral coefficients with their first and second derivatives. HMM-based connected digit recognition experiments comparing the traditional and extended feature sets show that voicing features and spectral information are complementary and that improved speech recognition performance is obtained by combining the two sources of information.
ICASSP98 Paper Abstract

Accent Type Recognition and Syntactic Boundary Detection of Japanese Using Statistical Modeling of Moraic Transitions of Fundamental Frequency Contours

Authors:
Keikichi Hirose, University of Tokyo, (Japan)
Koji Iwano, University of Tokyo, (Japan)

Volume 1, page 25, paper no. 2444

Abstract:
Experiments on accent type recognition and syntactic boundary detection of Japanese speech were conducted based on the statistical modeling of voice fundamental frequency contours formerly proposed by the authors. In the proposed modeling, fundamental frequency contours are segmented into moraic units to generate moraic contours, which are further represented by discrete codes. After modeling the accent types and syntactic boundaries, their recognition/detection was done for ATR speech corpus. As for the accent type recognition, 4-mora words were used for the training and testing, and recognition rates around 74% were obtained for speaker open experiments. As for the syntactic boundary detection, detectability of accent phrase boundaries was tested for sentence speech. Although the experiments were conducted only for the closed condition due to availability of speech corpus, the result indicated the usefulness of separating the boundary model into two depending on whether the boundary is accompanied by a pause or not.
ICASSP98 Paper Abstract

A Novel Feature-Extraction for Speech Recognition Based on Multiple Acoustic-Feature Planes

Authors:
Tsuneo Nitta, Toshiba Corporation, (Japan)

Abstract:
In this paper, the author tries to incorporate functions of the auditory nerve system into the feature extractor of speech recognition. The functions include four types of well-known responses to sound stimuli: local peaks of spectrum in steady sound, ascending FM sound, descending FM sound, and sharply rising and falling sound. Each function is realized in the form of a 3-levels differential operator and applied to a time-spectrum pattern X(t,f) of the output of BPF with 26-hannels. The resultant acoustic cue of an input speech represented by multiple acoustic-feature planes (MAFP) is compressed by using KLT, then classified. In the experiments performed on a Japanese E-set (12 consonantal parts of /Ci/) extracted from continuous speech, the MAFP significantly improved the error rate from 34.5% and 29.6% obtained by X(t,f) and X(t,f)+dX(t,f) to 17.0% for unknown speakers.
ICASSP98 Paper Abstract

An Error Correction Approach Based on the MAP Algorithm Combined with Hidden Markov Models

Authors:
Tadashi Yonezaki, Matsushita Communication Ind. Co. Ltd., (Japan)
Koji Yoshida, Matsushita Communication Ind. Co. Ltd., (Japan)
Toshio Yagi, Matsushita Communication Ind. Co. Ltd., (Japan)

Volume 1, page 33, paper no. 1876

Abstract:
The error correction approach which based on a hidden Markov model (HMM) is proposed. The occurrence probability of a code sequence, which is delivered by the HMMs, is used as the measure for the maximum a posteriori probability (MAP) algorithm. The MAP algorithm is based on the assumption that the source is a discrete-time finite-state Markov process, and the HMM which models a Markov source suits well for speech data. Therefore this combination would be useful for a speech coding system. The proposing approach is adapted to the code sequence quantizing line spectrum frequency (LSF) parameters. When the code sequence is sent over a binary symmetry channel (BSC), the proposing approach with 16-state HMMs improves in code error rate and degradation of cepstrum distortion at about 27% and 39% respectively for 3% random errors.
ICASSP98 Paper Abstract
Quantization of the Spectral Envelope for Sinusoidal Coders

Authors:
Thomas Eriksson, AT&T Labs - Research, (U.S.A.)
Hong-Goo Kang, AT&T Labs - Research, (U.S.A.)
Yannis G Stylianou, AT&T Labs - Research, (U.S.A.)

Abstract:
In an effort to efficiently code the spectral envelope of speech signals for wideband speech coding based on sinusoidal models, a robust computation of discrete cepstrum coefficients and their quantization is investigated. A parameterization of the spectral envelope has been proposed which is based on discrete cepstral coefficients using regularization techniques. This paper presents an efficient quantization scheme for these coefficients in order to use them in applications like speech coding. We present results which show a 35% reduction in bitrate when compare to simple scalar quantization. To verify the efficiency of the proposed quantization schemes, informal listening tests were performed in the context of a sinusoidal coder.
ICASSP98 Paper Abstract

Robust Speech Mode Based LSF Vector Quantization for Low Bit Rate Coders

Authors:
Srinivas Nandkumar, Hughes Network Systems, ( U.S.A.)
Kumar Swaminathan, Hughes Network Systems, ( U.S.A.)
Udaya Bhaskar, Hughes Network Systems, ( U.S.A.)

Abstract:
Robust vector quantization of LSF parameters at a low bit rate is essential for voice coders operating below 5 Kbps. A novel aspect of the proposed technique is the use of decorrelated residual LSF vectors from speech mode based backward prediction along with a multi-stage VQ design. Rates as low as 12 bits per 20 ms speech frame for the stationary voiced speech mode and 22 bits/frame for unvoiced and non-stationary voiced frames are shown to result in efficient quantization. In our classification scheme, spectrally stationary voiced frames constitute around 30% of active speech frames resulting in a minimum average bit rate of 19 bits/frame. Objective VQ performance is compared with cellular standard coders such as IS-641 and IS-127. The proposed VQ has been integrated into a speech mode based 4.8 Kbps coder resulting in subjective performance close to that of the 7.4 Kbps IS-641 coder.
ICASSP98 Paper Abstract

A New General Distance Measure For Quantization of LSF and Their Transformed Coefficients

Authors:
Hai Le Vu, Technical University of Budapest, (Hungary)
Laszlo Lois, Technical University of Budapest, (Hungary)

Volume 1, page 45, paper no. 1816

Abstract:
In this paper, we have developed a new general distance measure that not only can be used in a vector quantization (VQ) of the line spectrum frequency (LSF) parameters but performs well in the LSF transformed domain. The new distance is based on the spectral sensitivity of LSF and their transformed coefficients. In addition, the fix scaling factor is used to decrease the sensitivity of spectral error at higher frequencies. Experimental results have shown that the proposed distance measure leads to as good as or better performance of VQ compared to other methods in the field of LSF coding. The use of this distance as the weighting function of the LSFs' transformed parameters is also suggested.
ICASSP98 Paper Abstract
Variable Model Order LPC Quantization

Authors:
Pasi Ojala, Nokia Research Center, (Finland)
Ari Lakaniemi, Nokia Research Center, (Finland)

Abstract:
This paper presents a new method to apply variable bit-rate predictive quantization of the variable model order LPC parameters. In addition, the method is employed to interpolate the parameters within the analysis frame. The LPC model order selection algorithm of this work is based on the characteristics of the input signal and on the performance of the LPC model. Hence, the variable bit-rate LPC quantization is source controlled. The number of quantized parameters needs to be identical in successive frames to be able to apply the predictive quantization and to interpolate parameters inside the frame. Therefore, the order of the LPC model of the previous frame needs to be expanded or reduced to be the same as the current frame LPC model. The advantage of variable model order LPC quantization is the lowered average bit-rate compared to fixed rate while the speech quality remains the same.
Optimal Transform for Segmented Parametric Speech Coding

Abstract:
In voice coding applications where there is no constraint on the encoding delay, such as store and forward message systems or voice storage, segment coding techniques can be used to achieve a reduction in data rate without compromising the level of distortion. For low data rate linear predictive coding schemes, increasing the encoding delay allows one to exploit any long term temporal stationarities on an interframe basis, thus reducing the transmission bandwidth or storage needs of the speech signal. Transform coding has previously been applied in low data rate speech coding to exploit both the interframe and the intraframe correlation. This paper investigates the potential for optimising the transform for segmented parametric representation of speech.
ICASSP98 Paper Abstract

Two Novel Lossless Algorithms to Exploit Index Redundancy in VQ Speech Compression

Authors:
Sridha Sridharan, Queensland University of Technology, (Australia)
John W Leis, University of Southern Queensland, (Australia)

Abstract:
We address the problem of speech compression at very low rates, with the short-term spectrum compressed to less than 20 bits per frame. Current techniques apply structured vector quantization (VQ) to the short-term synthesis filter coefficients to achieve rates of the order of 24 to 26 bits per frame. In this paper we show that temporal correlations in the VQ index stream can be introduced by dynamic codebook ordering, and that these correlations can be exploited by lossless coding approaches to reduce the number of bits per frame of the VQ scheme. The use of lossless coding ensures that no additional distortion is introduced, unlike other interframe techniques. We then detail two constructive algorithms which are able to exploit this redundancy. The first method is a delayed-decision approach, which dynamically adapts the VQ codebook to allow for efficient entropy coding of the index stream. The second is based on a vector sub-codebook approach, and does not incur any additional delay. Experimental results are presented for both methods to validate the approach.
ICASSP98 Paper Abstract

Multi Codebook Vector Quantization of LPC Parameters

Authors:
Costas S Xydeas, University of Manchester, (U.K.)
Thomas M Chapman, University of Manchester, (U.K.)

Abstract:
This paper presents a novel and efficient variable bit rate LPC quantization approach. The proposed MCVQ framework allows a Dynamic Programming based minimum quantization distortion partitioning and quantization process to be performed on input LSP vector tracks in time. Variable duration segments of LSP vector tracks are classified into one of a finite number of language related events. Specific codebooks, designed optimally for each event type, are then employed to vector quantize the individual LSP vectors of a given segment.
ICASSP98 Paper Abstract

Personal Speech Coding

Authors:
Wenhui Jia, Illinois Institute of Technology, (U.S.A.)
Wai-Yip Chan, Illinois Institute of Technology, (U.S.A.)

Volume 1, page 65, paper no. 5005

Abstract:
In existing speech coding systems, all quantizer codebooks are designed to suit the statistical and perceptual characteristics of speech signals of a population of speakers. However, an individual's speech signal does not exhibit, even over a long time, the entire range of characteristics of the population. With the advent of the personal communication systems, personal information might become available and be used to improve the rate-distortion performance of speech coders. In this paper we assess the potential gain of personal speech coding by designing codebooks for individual speakers. Spectral quantization, excitation and pitch lag codebooks of existing CELP coders are redesigned. The gains appear to be modest, suggesting that we need to use a different coding framework. Amongst the components, the spectral quantizer seems to be most amenable to personalization.
ICASSP98 Paper Abstract

A Genetic Approach to the Design of General-Tree--Structured Vector Quantizers for Speech Coding

Authors:
Lin Yu Tseng, National Chung Hsing University, (Taiwan)
Shiueng Bien Yang, National Chung Hsing University, (Taiwan)

Abstract:
The full-search vector quantization suffers from spending much time searching the whole codebook sequentially. 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. An experiment on speech coding was conducted. A comparison of the performance of this vector quantizer and the other two tree-structured vector quantizers is also given.
ICASSP98 Paper Abstract

Minimum Cross-Entropy Adaptation of Hidden Markov Models

Authors:
Mohamed Afify, Universite Henri Poincare, Nancy, (France)
Jean-Paul Haton, Universite Henri Poincare, Nancy, (France)

Volume 1, page 73, paper no. 1812

Abstract:
Adaptation techniques that benefit from distribution correlation are important in practical situations having sparse adaptation data. The so called EMAP algorithm provides an optimal, though expensive, solution. In this article we start from EMAP, and propose an approximate optimisation criterion, based on maximising a set of local densities. We then obtain expressions for these local densities based on the principle of minimum cross-entropy (MCE). The solution to the MCE problem is obtained using an analogy with MAP estimation, and avoids the use of complex numerical procedures, thus resulting in a simple adaptation algorithm. The implementation of the proposed method for the adaptation of HMMs with mixture Gaussian densities is discussed, and its efficiency is evaluated on an alphabet recognition task.
ICASSP98 Paper Abstract

Improving Viterbi Bayesian Predictive Classification via Sequential Bayesian Learning in Robust Speech Recognition

Authors:
Hui Jiang, University of Tokyo, (Japan)
Keikichi Hirose, University of Tokyo, (Japan)
Qiang Huo, University of Hong Kong, (Hong Kong)

Abstract:
In this paper, we extend our previously proposed Viterbi Bayesian predictive classification (VBPC) algorithm to accommodate a new class of prior probability density function (pdf) for continuous density hidden Markov model (CDHMM) based robust speech recognition. The initial prior pdf of CDHMM is assumed to be a finite mixture of natural conjugate prior pdf’s of its complete-data density. With the new observation data, the true posterior pdf is approximated by the same type of finite mixture pdf’s which retain the required most significant terms in the true posterior density according to their contribution to the corresponding predictive density. Then the updated mixture pdf is used to improve the VBPC performance. The experimental results on a speaker-independent recognition task of isolated Japanese digits confirm the viability and the usefulness of the proposed technique.
ICASSP98 Paper Abstract

Discriminative Learning of Additive Noise and Channel Distortions for Robust Speech Recognition

Authors:
Jiqing Han, Systems Engineering Research Institute, (Korea)
Munsung Han, Systems Engineering Research Institute, (Korea)
Gyu-Bong Park, Systems Engineering Research Institute, (Korea)
Jeongue Park, Systems Engineering Research Institute, (Korea)
Wen Gao, Harbin Institute of Technology, (China)
Doosung Hwang, Systems Engineering Research Institute, (Korea)

Abstract:
Learning the influence of additive noise and channel distortions from training data is an effective approach for robust speech recognition. Most of the previous methods are based on maximum likelihood estimation criterion. In this paper, we propose a new method of discriminative learning environmental parameters, which is based on Minimum Classification Error (MCE) criterion. By using a simple classifier defined by ourselves and the Generalized Probabilistic Descent (GPD) algorithm, we iteratively learn environmental parameters. After getting the parameters, we estimate the clean speech features from the observed speech features and then use the estimation of the clean speech features to train or test the back-end HMM classifier. The best error rate reduction of 32.1% is obtained, tested on a Korean 18 isolated confusion words task, relative to conventional HMM system.
ICASSP98 Paper Abstract

A Combination of Discriminative and Maximum Likelihood Techniques for Noise Robust Speech Recognition

Authors:
Kari Laurila, Nokia Research Center, (Finland)
Marcel Vasilache, Nokia Research Center, (Finland)
Olli Viikki, Nokia Research Center, (Finland)

Abstract:
In this paper, we study how discriminative and Maximum Likelihood (ML) techniques should be combined in order to maximize the recognition accuracy of a speaker-independent Automatic Speech Recognition (ASR) system that includes speaker adaptation. We compare two training approaches for speaker-independent case and examine how well they perform together with four different speaker adaptation schemes. In a noise robust connected digit recognition task we show that the Minimum Classification Error (MCE) training approach for speaker-independent modeling together with the Bayesian speaker adaptation scheme provide the highest classification accuracy over the whole lifespan of an ASR system. With the MCE training we are capable of reducing the recognition errors by 30% over the ML approach in the speaker-independent case. With the Bayesian speaker adaptation scheme we can further reduce the error rates by 62% using only as few as five adaptation utterances.
ICASSP98 Paper Abstract

Frame-Synchronous Stochastic Matching Based on the Kullback-Leibler Information

Authors:
Lionel Delphin-Poulat, Telecom CNET/DIH/RCP, (France)
Chafic Mokbel, Telecom CNET/DIH/RCP, (France)
Jerome Idier, Laboratoire des Signaux et Systemes Supelec, (France)

Abstract:
An acoustic mismatch between a given utterance and a model degrades the performance of the speech recognition process. We choose to model speech by Hidden Markov Models (HMMs) in the cepstrum domain and the mismatch by a parametric function. In order to reduce the mismatch, one has to estimate the parameters of this function. In this paper, we present a frame synchronous estimation of these parameters. We show that the parameters can be computed recursively. Thanks to such methods, parameters variations can be tracked. We give general equations and study the particular case of an affine transform. Finally, we report recognition experiments carried out over both PSTN and cellular telephone network to show the efficiency of the method in a real context.
ICASSP98 Paper Abstract

Unsupervised Speaker Normalization Using Canonical Correlation Analysis

Authors:
Yasuo Ariki, Ryukoku University, (Japan)
Miharu Sakuragi, Ryukoku University, (Japan)

Volume 1, page 93, paper no. 1431

Abstract:
Conventional speaker-independent HMMs ignore the speaker differences and collect speech data in an observation space. This causes a problem that the output probability distribution of the HMMs becomes vague so that it deteriorates the recognition accuracy. To solve this problem, we construct the speaker subspace for an individual speaker and correlate them by o-space canonical correlation analysis between the standard speaker and input speaker. In order to remove the constraint that input speakers have to speak the same sentences as the standard speaker in the supervised normalization, we propose in this paper an unsupervised speaker normalization method which automatically segments the speech data into phoneme data by Viterbi decoding algorithm and then associates the mean feature vectors of phoneme data by o-space canonical correlation analysis. We show the phoneme recognition rate by this unsupervised method is equivalent with that of the supervised normalization method we already proposed.
ICASSP98 Paper Abstract

Speaker Independent Acoustic Modeling Using Speaker Normalization

Authors:
Jun Ishii, Mitsubishi Electric Corporation, (Japan)
Toshiaki Fukada, ATR Interpreting Telecommunications Research Labs, (Japan)

Volume 1, page 97, paper no. 1899

Abstract:
This paper proposes a novel speaker-independent (SI) modeling for spontaneous speech data from multiple speakers. The SI acoustic model parameters are estimated by individual training for inter-speaker variability and for intra-speaker phonetically related variation in order to obtain a more accurate acoustic model. The linear transformation technique is used for the speaker normalization to extract intra-speaker phonetically related variation and also is used for the re-estimation of inter-speaker variability. The proposed modeling is evaluated for a Japanese spontaneous speech data, using continuous density mixture Gaussian HMMs. Experimental results from the use of proposed acoustic model show that the reductions in word error rate can be achieved over the standard SI model regardless the type of acoustic model used.
ICASSP98 Paper Abstract

Robust Speech Recognition for Multiple Topological Scenarios of the GSM Mobile Phone System

Authors:
Theodoros Salonidis, Technical University of Crete, (Greece)
Vassilios Digalakis, Technical University of Crete, (Greece)

Volume 1, page 101, paper no. 1623

Abstract:
This paper deals with robust speech recognition in the GSM mobile environment. Our focus is on the voice degradation due to the losses in the GSM coding scheme. Thus, we initially propose an experimental framework of network topologies that consists of various coding-decoding systems placed in tandem. After measuring the recognition performance for each of these network scenarios, we try to increase recognition accuracy by using feature compensation and model adaptation algorithms. We first compare the different methods for all the network topologies assuming the topology is known. We then investigate the more realistic case, in which we don’t know the network topology the voice has passed through. The results show that robustness can be achieved even in this case.
ICASSP98 Paper Abstract

Speaker Verification Using Minimum Verification Error Training

Authors:
Aaron E Rosenberg, AT&T Labs, (U.S.A.)
Olivier Siohan, AT&T Labs, (U.S.A.)
S. Parthasarathy, AT&T Labs, (U.S.A.)

Volume 1, page 105, paper no. 1767

Abstract:
We propose a Minimum Verification Error (MVE) training scenario to design and adapt an HMM-based speaker verification system. By using the discriminative training paradigm, we show that customer and background models can be jointly estimated so that the expected number of verification errors (false accept and false reject) on the training corpus are minimized. An experimental evaluation of a fixed password speaker verification task over the telephone network was carried out. The evaluation shows that MVE training/adaptation performs as well as MLE training and MAP adaptation when performance is measured by average individual equal error rate (based on a posteriori threshold assignment). After model adaptation, both approaches lead to an individual equal error-rate close to 0.6%. However, experiments performed with a priori dynamic threshold assignment show that MVE adapted models exhibit false rejection and false acceptance rates 45% lower than the MAP adapted models, and therefore lead to the design of a more robust system for practical applications.
ICASSP98 Paper Abstract

Speaker Identification Using Minimum Classification Error Training

Authors:
Olivier Siohan, AT&T Labs, (U.S.A.)
Aaron E Rosenberg, AT&T Labs, (U.S.A.)
S. Parthasarathy, AT&T Labs, (U.S.A.)

Volume 1, page 109, paper no. 1783

Abstract:
In this paper we use a Minimum Classification Error (MCE) training paradigm to build a speaker identification system. The training is optimized at the string level for a text-dependent speaker identification task. Experiments performed on a small set speaker identification task show that MCE training can reduce closed set identification errors by up to 20-25% over a baseline system trained using Maximum Likelihood Estimation. Further experiments suggest that additional improvement can be obtained by using some additional training data from speakers outside the set of registered speakers, leading to an overall reduction of the closed-set identification errors by about 35%.
Model Adaptation Methods for Speaker Verification

Authors:
William Mistretta, T-NETIX Inc., (U.S.A.)
Kevin R. Farrell, T-NETIX Inc., (U.S.A.)

Abstract:
Model adaptation methods for a text-dependent speaker verification system are evaluated in this paper. The speaker verification system uses a discriminant model and a statistical model to represent each enrolled speaker. These modeling approaches consist of a neural tree network and Gaussian mixture model. Adaptation methods are evaluated for both modeling approaches. We show that the overall system performance with adaptation is comparable to that obtained by training the model with the additional information. However, the adaptation can be performed within a fraction of the time required to retrain a model. Additionally, we have evaluated the adapted and non-adapted models with data recorded six months after the initial enrollment. The adaptation reduced the error rate for the aged data by 40%.
ICASSP98 Paper Abstract

Robust Model for Speaker Verification Against Session-Dependent Utterance Variation

Authors:
Tomoko Matsui, NTT Human Interface Laboratories, (Japan)
Kiyoaki Aikawa, NTT Human Interface Laboratories, (Japan)

Volume 1, page 117, paper no. 1880

Abstract:
This paper investigates a new method for creating speaker models robust against utterance variation in continuous distribution hidden-Markov-model-based speaker verification. In this method, the distribution of the session-independent features for each speaker is estimated by separately modeling the session-to-session utterance variation as two distinct variations: one session-dependent and the other session-independent. In practice, joint normalization of the session-dependent utterance variation and estimation of the parameters of speaker models is performed based on a speaker adaptive training algorithm. The resulting speaker models more accurately represent session-independent speaker characteristics, and the discriminatory capabilities of these models increases. In text-independent speaker verification experiments using data uttered by 20 speakers in 7 sessions over 16 months, we show that the proposed method achieves a 15% reduction in the error rate.
ICASSP98 Paper Abstract

Speaker Verification in Noisy Environments with Combined Spectral Subtraction and Missing Feature Theory

Authors:
Andrzej Drygajlo, EPFL, (Switzerland)
Mounir El-Maliki, EPFL, (Switzerland)

Volume 1, page 121, paper no. 2014

Abstract:
In the framework of Gaussian mixture models (GMMs), we present a new approach towards robust automatic speaker verification (SV) in adverse conditions. This new and simple approach is based on the combination of a speech enhancement using traditional spectral subtraction, and a missing feature compensation to dynamically modify the probability computations performed in GMM recognizers. The identity of spectral features missing due to noise masking is provided by the spectral subtraction algorithm. Previous works have demonstrated that the missing feature modeling method succeeds in speech recognition with some artificially generated interruptions, filtering and noises. In this paper, we show that this method also improves noise compensation techniques used for speaker verification in more realistic conditions.
ICASSP98 Paper Abstract

A Comparison of A Priori Threshold Setting Procedures for Speaker Verification in the Cave Project

Authors:
Jean-Benoit Pierrot, ENST, (France)
Johan Lindberg, KTH, (Sweden)
Johan Koolwaaij, KUN, (The Netherlands)
Hans-Peter Hutter, Ubilab-UBS, (Sweden)
Dominique Genoud,IDIAP, (Switzerland)
Mats Blomberg, KTH, (Sweden)
Frederic Bimbot, ENST, (France)

Volume 1, page 125, paper no. 5229

Abstract:
The issue of a priori threshold setting in speaker verification is a key problem for field applications. In the context of the CAVE project, we compared several methods for estimating speaker-independent and speaker-dependent decision thresholds. Relevant parameters are estimated from development data only, i.e. without resorting to additional client data. The various approaches are tested on the Dutch SESP database.
ICASSP98 Paper Abstract

Text Dependent Speaker Verification Using Binary Classifiers

Authors:
Dominique Genoud, IDIAP, (Switzerland)
Miguel Moreira, IDIAP, (Switzerland)
Eddy Mayoraz, IDIAP, (Switzerland)

Volume 1, page 129, paper no. 1741

Abstract:
This paper describes how a speaker verification task can be advantageously decomposed into a series of binary classification problems, i.e. each problem discriminating between two classes only. Each binary classifier is specific to one speaker, one anti-speaker and one word. Continuous attribute decision trees are used as classifiers. The set of classifiers is then pruned to eliminate the less relevant ones. Diverse pruning methods are experimented, and it is shown that when the speaker verification decision is performed with an a priori threshold, some of them give better results than a reference HMM system.
ICASSP98 Paper Abstract

Speaker Verification Using Verbal Information Verification for Automatic Enrollment

Authors:
Qi Li, Bell Labs, Lucent Technologies, (U.S.A.)
Biing-Hwang Juang, Bell Labs, Lucent Technologies, (U.S.A.)

Abstract:
A conventional speaker verification (SV) system needs an enrollment session to collect the training data. In [1], we introduced a speaker authentication method called Verbal Information Verification (VIV) which verifies a speaker by verbal contents instead of speech characteristics. Such a system does not need an enrollment session. In this paper, VIV is combined with SV. We propose a system which uses VIV to collect training data during the first few accesses automatically, which are often from different acoustic environments. Then, a speaker dependent model is trained and speaker authentication can be performed by SV. This approach not only avoid formal enrollment session which brings convenience to the user, but mitigates the mismatch problem causing by different acoustic environments between training and test sessions. Our experiments show that the proposed system improved the SV performance over 40% compared to the conventional SV system.
ICASSP98 Paper Abstract

An Adaptive Multi-Rate Speech Codec Based on MP-CELP Coding Algorithm for ETSI AMR Standard

Authors:
Hironori Ito, NEC Corporation, (Japan)
Masahiro Serizawa, NEC Corporation, (Japan)
Kazunori Ozawa, NEC Corporation, (Japan)
Toshiyuki Nomura, NEC Corporation, (Japan)

Volume 1, page 137, paper no. 1909

Abstract:
This paper proposes a speech codec based on the Multi-Pulse based CELP (MP-CELP) coding and a convolutional coding algorithm for the ETSI Adaptive Multi-Rate (AMR) standard. The codec operates at several speech coding rates, maintaining a fixed gross rate including speech and channel coding for the Full-Rate (FR) and Half-Rate (HR) channel modes. MP-CELP has great features of easily changing the speech coding rate by controlling the parameters such as the number of pulses and other parameters. Subjective tests show that the proposed AMR codec in the FR channel mode achieves higher performance than that of the Enhanced FR codec, and the proposed codec in the HR channel mode gives a comparable coding quality to that by the Full-Rate codec, by selecting an optimal coding rate for each channel condition. T-tests based on the test results also show that the proposed speech codec meets about 80% of the seventeen requirements, which are selected from the AMR standard study report. Therefore, the proposed codec is promising for the AMR standard.
ICASSP98 Paper Abstract

GSM EFR Based Multi-Rate Codec Family

Authors:
Janne Vainio, Nokia Research Center, (Finland)
Hannu Mikkola, Nokia Research Center, (Finland)
Kari Järvinen, Nokia Research Center, (Finland)
Petri Haavisto, Nokia Research Center, (Finland)

Volume 1, page 141, paper no. 1576

Abstract:
This paper describes a multi-rate codec family developed as a potential candidate for the GSM Adaptive Multi Rate (AMR) codec standard. The codec family consists of the GSM Enhanced Full Rate (EFR) codec [1] and lower bit-rate extensions thereof. The codec family consists of several codecs, i.e., modes that have different bit-rate partitionings between source coding and error protection. All the source codecs use the same ACELP-method (Algebraic Code Excited Linear Predictive Coding) used also in the GSM EFR codec. The codec operates at gross bit-rates of 22.8 kbit/s in the GSM full rate (FR) channel and 11.4 kbit/s in the GSM half rate (HR) channel. In the full rate channel, the codec provides improved error robustness over the GSM Enhanced Full Rate (EFR) codec. It extends wireline quality (equal to or better than G.726-32 ADPCM) to poor channel error conditions with low C/I-ratios of 7 dB or even below. When operated in the half rate channel, the codec provides improved channel capacity while still providing wireline quality at high C/I-ratios above 16-19 dB.
ICASSP98 Paper Abstract
Removal of Sparse-Excitation Artifacts in CELP

Authors:
Roar Hagen, Ericsson Radio Systems AB, (Sweden)
Erik Ekudden, Ericsson Radio Systems AB, (Sweden)
Bjorn Johansson, Ericsson Radio Systems AB, (Sweden)
W. Bastiaan Kleijn, Royal Institute of Technology, (Sweden)

Abstract:
In CELP, the use of codebooks with entries with only a few non-zero samples provides high speech quality and facilitates fast computation. With decreasing bit-rate, the intervals between the pulses increase, and the quality of the reconstructed signal begins to suffer from a particular type of artifact, which is strongest for noise-like segments. In this paper we describe experiments which show that the perceived artifacts are mainly concentrated at frequencies above 3 kHz, and this is consistent with our understanding of auditory theory. Our analysis leads to simple strategies to eliminate the artifacts, even at lower bit rates. We describe both a non-adaptive and an adaptive post-processing method to remove the artifacts. The methods are demonstrated to be efficient when used in the ACELP algorithm. A closed-loop method for ACELP is also described.
ICASSP98 Paper Abstract
Adaptive Encoding of Fixed Codebook in CELP Coders

Authors:
Hong Kook Kim, Samsung Advanced Institute of Technology, (Korea)

Abstract:
In this paper, we propose an adaptive encoding method of fixed codebook in CELP coders and implement an adaptive fixed code excited linear prediction (AF-CELP) speech coder. AF-CELP exploits the fact that the fixed codebook contribution to speech signal is also periodic as the adaptive codebook (or pitch filter) contribution. By modeling the fixed codebook with the pitch lag and the gain from the adaptive codebook, AF-CELP can be implemented at low bit rates as well as low complexity. Listening tests show that a 6.4 kbit/s AF-CELP has a comparable quality to the 8 kbit/s CS-ACELP.
High Quality Multi-Pulse Based CELP Speech Coding at 6.4 Kb/S and its Subjective Evaluation

Authors:
Kazunori Ozawa, NEC Corporation, (Japan)
Masahiro Serizawa, NEC Corporation, (Japan)

Abstract:
This paper proposes an MP-CELP (Multi-Pulse-based CELP) speech coding at 6.4 kb/s with 10 ms frame. In MP-CELP, amplitudes or signs of multi-pulse excitation are simultaneously vector quantized (VQ). A combination search between multiple pulse location candidates and VQ codebook remarkably improves the quantization performance. In order to improve speech quality for background noise conditions, an adaptive pulse location restriction method is developed. The subjective evaluation results show that speech quality for 6.4 kb/s MP-CELP is higher than that for G.726 at 32 kb/s and is equivalent to that for 6.3 kb/s G.723.1 with 30 ms frame in clean speech and tandem conditions. For background noise conditions, the adaptive pulse location restriction significantly improves MOS value by 0.9. The speech quality is equivalent to that for G.723.1, but still does not reach to that of 24 kb/s G.726, except interference talker condition.
ICASSP98 Paper Abstract

A 13.0 kbit/s Wideband Speech Codec Based on SB-ACELP

Authors:
Juergen Schnitzler, Aachen University of Technology, (Germany)

Volume 1, page 157, paper no. 1377

Abstract:
This paper describes a wideband (7 kHz) speech compression scheme operating at a bit rate of 13.0 kbit/s, i.e. 0.8 bit per sample. We apply a split-band (SB) technique, where the 0-6 kHz band is critically subsampled and coded by an ACELP approach. The high frequency signal components (6-7 kHz) are generated by an improved High-Frequency-Resynthesis (HFR) at the decoder such that no additional information has to be transmitted. In informal listening tests, the subjective speech quality was rated to be comparable to the CCITT G.722 wideband codec at 48 kbit/s.
ICASSP98 Paper Abstract

A Wideband CELP Speech Coder at 16 kbit/s Based on Mel-Generalized Cepstral Analysis

Authors:
Kazuhito Koishida, Tokyo Institute of Technology, (Japan)
Gou Hirabayashi, Tokyo Institute of Technology, (Japan)
Keiichi Tokuda, Nagoya Institute of Technology, (Japan)
Takao Kobayashi, Tokyo Institute of Technology, (Japan)

Volume 1, page 161, paper no. 1704

Abstract:
This paper proposes a wideband CELP coder using frequency warping. Instead of linear prediction, the proposed coder adopts the mel-generalized cepstral analysis, and encodes fullband of the speech signal through a warped frequency scale. It is shown that the subjective quality of the proposed coder at 16 kbit/s is better than that of the ITU-T G.722 at 64 kbit/s. Furthermore, the proposed coder gives a much smaller difference in performance for male and female speakers than the conventional CELP coder. These results indicate that the frequency warping makes a large contribution to the improvement of the subjective quality for wideband speech coding.
ICASSP98 Paper Abstract
A Low-delay Wideband Speech Coder at 24 kbps

Authors:
Anil W Ubale, University of California, Santa Barbara, (U.S.A.)
Allen Gersho, University of California, Santa Barbara, (U.S.A.)

Volume 1, page 165, paper no. 2196

Abstract:
A novel low-delay wideband speech coder, called Multi-band CELP (MB-CELP) achieves a delay of about 10 ms, by exploiting time-domain correlations with a two-stage linear prediction scheme. A low-order forward-adaptive LP stage models coarse shape, and a high-order backward-adaptive LP stage models fine structure of the input spectrum. A conditional pitch prediction method improves the performance of the coder for speech without degrading music performance. A multi-band bank of off-line filtered codebooks generates the excitation signal. A 24 kbps version of the coder has nine multi-band codebooks with nonuniform bandwidth. Subjective comparison tests show that this coder outperforms the G.722 coder at the bit-rate of 56 kbps.
ICASSP98 Paper Abstract
Statistics-Based Segment Pattern Lexicon - A New Direction for Chinese Language Modeling

Authors:
Kae-Cherng Yang, National Taiwan University, (Taiwan)
Tai-Hsuan Ho, National Taiwan University, (Taiwan)
Lee-Feng Chien, Institute of Information Science, Academia Sinica, (Taiwan)
Lin-Shan Lee, Institute of Information Science, Academia Sinica, (Taiwan)

Volume 1, page 169, paper no. 2395

Abstract:
This paper presents a new direction for Chinese language modeling based on a different concept of lexicon. Because every Chinese character has its own meaning and there are not "blanks" in Chinese sentences as word boundaries, also because the wording structure in Chinese language is extremely flexible, the "words" in Chinese are actually not well defined, and there does not exist a commonly accepted lexicon. This makes language modeling very sophisticated in Chinese language, and the "out of vocabulary" (OOV) problem specially serious. A new concept for lexicon is thus proposed in this paper. The elements of this lexicon can be words or any other "segment patterns". They should be extracted from the training corpus by statistical approaches with a goal to minimize the overall perplexity. The language models can then be developed based on this new lexicon. Very encouraging experimental results have been obtained.
ICASSP98 Paper Abstract
Building Class-based Language Models with Contextual Statistics

Authors:
Shuanghu Bai, Institute of Systems Science, National University of Singapore, (Singapore)
Haizhou Li, Institute of Systems Science, National University of Singapore, (Singapore)
Zhiwei Lin, Institute of Systems Science, National University of Singapore, (Singapore)
Baosheng Yuan, Institute of Systems Science, National University of Singapore, (Singapore)

Volume 1, page 173, paper no. 1662

Abstract:
In this paper, novel clustering algorithms are proposed by using the contextual statistics of words for class-based language models. The minimum discriminative information (MDI) is used as a distance measure. Three algorithms are implemented to build bigram language models for a vocabulary of 50,000 words over a corpus of 200 million words. The computational cost of algorithms and resulting LM perplexity are studied. The comparisons between the MDI algorithm and maximum mutual information (MMI) algorithm are also given to demonstrate the effectiveness and the efficiency of the new algorithms. It is shown that the MDI approaches make the tree-building clustering possible with large vocabulary.
ICASSP98 Paper Abstract

Comparison of Part-of-Speech and Automatically Derived Category-Based Language Models for Speech Recognition

Authors:
Thomas R. Niesler, Cambridge University, (U.K.)
Edward W.D. Whittaker, Cambridge University, (U.K.)
Philip C. Woodland, Cambridge University, (U.K.)

Volume 1, page 177, paper no. 2003

Abstract:
This paper compares various category-based language models when used in conjunction with a word-based trigram by means of linear interpolation. Categories corresponding to parts-of-speech as well as automatically clustered groupings are considered. The category-based model employs variable-length n-grams and permits each word to belong to multiple categories. Relative word error rate reductions of between 2 and 7% over the baseline are achieved in N-best rescoring experiments on the Wall Street Journal corpus. The largest improvement is obtained with a model using automatically determined categories. Perplexities continue to decrease as the number of different categories is increased, but improvements in the word error rate reach an optimum.
ICASSP98 Paper Abstract
Balancing Acoustic and Linguistic Probabilities

Authors:
Atsunori Ogawa, Nagoya University, (Japan)
Kazuya Takeda, Nagoya University, (Japan)
Fumitada Itakura, Nagoya University, (Japan)

Volume 1, page 181, paper no. 1986

Abstract:
The length of the word sequence is not taken into account under language modeling of n-gram local probability modeling. Due to this property, the optimal values of the language weight and word insertion penalty for balancing acoustic and linguistic probabilities is affected by the length of word sequence. To deal with this problem, a new language model is developed based on Bernoulli trial model taking the length of the word sequence into account. Not only better recognition accuracy but also robust balancing with acoustic probability compared with the normal n-gram model of the proposed method is confirmed through recognition experiments.
ICASSP98 Paper Abstract

Initial Language Models for Spoken Dialogue Systems

Authors:
Andreas Kellner, Philips GmbH Forschungslaboratorien Aachen, (Germany)

Abstract:
The estimation of initial language models for new applications of spoken dialogue systems without large task-specific training corpora is becoming an increasingly important issue. This paper investigates two different approaches in which task-specific knowledge contained in the language understanding grammar is exploited in order to generate n-gram language models for the speech recognizer: The first uses class-based language models for which the word-classes are automatically derived from the grammar. In the second approach, language models are estimated on artificial corpora which have been created from the understanding grammar. The application of fill-up techniques allows the combination of the strengths of both approaches and leads to a language model which shows optimal performance regardless of the amount of training data available. Perplexities and word error rates are reported for two different domains.
ICASSP98 Paper Abstract

Maximum Likelihood and Discriminative Training of Direct Translation Models

Authors:
Kishore A Papineni, IBM, (U.S.A.)
Salim E. Roukos, IBM, (U.S.A.)
R T Ward, IBM, (U.S.A.)

Volume 1, page 189, paper no. 2195

Abstract:
We consider translating natural language sentences into a formal language using direct translation models built automatically from training data. Direct translation models have three components: an arbitrary prior conditional probability distribution, features that capture correlations between automatically determined key phrases or sets of words in both languages, and weights associated with these features. The features and the weights are selected using a training corpus of matched pairs of source and target language sentences to maximize the entropy or a new discrimination measure of the resulting conditional probability model. We report results in Air Travel Information System (ATIS) domain and compare the two methods of training.
A Telephone Number Inquiry System with Dialog Structure

Authors:
Hsien-Chang Wang, National Cheng-Kung University, (Taiwan)
Jhing-Fa Wang, National Cheng-Kung University, (Taiwan)

Volume 1, page 193, paper no. 1158

Abstract:
A Telephone Number Inquiry System (TNIS) answers caller the phone number he/she wants to know. Traditional system requires the caller to know the full name of the party. If the caller forgets the name, the system fails to retrieve correct information for the caller. In this paper, we propose a novel TNIS with dialog structure that can let caller use a more flexible method while inquiring, i.e., the caller may interact with our system to inquire the phone number by providing just the working, researching area, the surname, or the title, etc. Our system takes the telephone speech as input, after generating the word sequence, it performs a maximum likelihood key-feature matching with knowledge base. If necessary information is not derived, interactive dialog manager is activated to resolve the caller’s requirement. The experimental results show that our novel approach can make the system more natural.
ICASSP98 Paper Abstract
Spoken Dialog Systems for Children

Authors:
Alexandros Potamianos, AT&T Labs, (U.S.A.)
Shrikanth Narayanan, AT&T Labs, (U.S.A.)

Volume 1, page 197, paper no. 1824

Abstract:
In this paper, we outline the main issues when designing interactive multimedia systems for children and propose a unified approach—acoustic, linguistic, and dialog modeling—to system development. Acoustic, linguistic and dialog data collected in a wizard of Oz experiment from 160 children ages 8-14 playing an interactive computer game are analyzed and children-specific modeling issues are presented. Age-dependent and modality-dependent dialog flow patterns are identified. Furthermore, extraneous speech patterns, linguistic variability and disfluencies are investigated in spontaneous children’s speech, and important new results are reported. Finally, baseline automatic speech recognition (ASR) results are presented for various tasks using simple acoustic and language models.
ICASSP98 Paper Abstract
Using Markov Decision Process for Learning Dialogue Strategies

Authors:
Esther Levin, AT&T Labs - Research, (U.S.A.)
Roberto Pieraccini, AT&T Labs - Research, (U.S.A.)
Wieland Eckert, AT&T Labs - Research, (U.S.A.)

Abstract:
In this paper we introduce a stochastic model for dialogue systems based on Markov Decision Process. Within this framework we show that the problem of dialogue strategy design can be stated as an optimization problem, and solved by a variety of methods, including the reinforcement learning approach. The advantages of this new paradigm include objective evaluation of dialogue systems, and their automatic design and adaptability. We show some results on learning a dialogue strategy for an Air Travel Information System.
An Implementation of Partial Parser in the Spoken Language Translator

Authors:
Nam-Yong Han, Electronics and Telecommunications Research Institute, (Korea)
Un-Cheon Choi, Electronics and Telecommunications Research Institute, (Korea)
Youngjik Lee, Electronics and Telecommunications Research Institute, (Korea)

Volume 1, page 205, paper no. 1891

Abstract:
We describe characteristics of the partial parser and evaluations of the output of the spoken language translator with concept-based grammars. This translator translates the Korean utterance generated by a speech recognizer which recognizes spontaneous speech into an English/Japanese utterances through a concept analysis approach. The parsing fails to parse input utterance when the parser finished to medium level tokens because the successful parsing results come from only when all concepts except for the highest top level tokens are reduced into the ones of the highest top level tokens. At this time, the partial parser is ran to analyze those medium level tokens without parsing fail. We obtained 55.2% for the recognized data as the translation rate of meaning based on intention before applying partial parser to the spoken translator, and now obtained 79.1% after applying partial parser.
ICASSP98 Paper Abstract

Experiments in Confidence Scoring Using Spanish Call-home Data

Authors:
Jon G Vaver, Department of Defense, (U.S.A.)

Volume 1, page 209, paper no. 1194

Abstract:
We present results relevant to tasks involved in the confidence scoring of output from a continuous speech recognition system, including the search for predictor variables and model selection. We introduce the DET curve characteristic (DCC) score, which we use along with the normalized cross entropy (NCE) score, to perform the model and predictor variable evaluation. We also show results from experiments that suggest how the NCE and DCC scores vary with recognizer performance.
ICASSP98 Paper Abstract

A New Decoder Based on a Generalized Confidence Score

Authors:
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Chin-Hui Lee, Lucent Technologies, (U.S.A.)
Biing-Hwang Juang, Lucent Technologies, (U.S.A.)

Volume 1, page 213, paper no. 1745

Abstract:
We propose a new decoder based on a generalized confidence score. The generalized confidence score is defined as a product of confidence scores obtained from confidence information sources such as likelihood, likelihood ratio, duration, duration ratio, language model probabilities, supra-segmental information etc. All confidence information sources are converted into confidence score by a confidence pre-processor. We show an extended hybrid as an example of the decoder based on the generalized confidence score. The extended hybrid decoder uses multi-level confidence scores such as frame-level, phone-level, and word-level likelihood ratios, while the conventional hybrid decoder uses the frame-level confidence score. Experimental result shows that the extended decoder gives better result than the conventional hybrid decoder, particularly in dealing with out-of-vocabulary words or out-of-task sentences.
ICASSP98 Paper Abstract

Rejection of Out-of-Vocabulary Words Using Phoneme Confidence Likelihood

Authors:
Takatoshi Jitsuhiro, NTT Human Interface Laboratories, (Japan)
Satoshi Takahashi, NTT Human Interface Laboratories, (Japan)
Kiyoaki Aikawa, NTT Human Interface Laboratories, (Japan)

Abstract:
The rejection of unknown words is important in improving the performance of speech recognition. The anti-keyword model method can reject unknown words with high accuracy in small vocabulary and specified task. Unfortunately, it is either inconvenient or impossible to apply if words in the vocabulary change frequently. We propose a new method for task independent rejection of unknown words, where a new phoneme confidence measure is used to verify partial utterances. It is used to verify each phoneme while locating candidates. Furthermore, the whole utterance is verified by a phonetic typewriter. This method can improve the accuracy of verification in each phoneme, and improve the speed of candidate search. Tests show that the proposed method improves the recognition rate by 4% compared to the conventional algorithm at equal error rates. Furthermore, a 3% improvement is obtained by training acoustic models with the MCE algorithm.
ICASSP98 Paper Abstract
Keyword Verification Considering the Correlation of Succeeding Feature Vectors

Authors:
Jochen Junkawitsch, Siemens AG, (Germany)
Harald Hoege, Siemens AG, (Germany)

Volume 1, page 221, paper no. 1931

Abstract:
The assumption of statistically independent feature vectors within the HMM approach is a well known problem. The aim of this study is to explore a simple and feasible method, that takes the correlation of adjacent feature vectors into account. A so called correlated HMM, that estimates the emission probability of a state with respect to correlated feature vectors, is built by combining two separate knowledge sources. On the one side, a traditional HMM provides an emission probability under the condition of a certain state, whereas on the other side a linear predictor delivers an emission probability considering the previous feature vectors. The efficiency of this method is shown with the help of the German SpeechDat(M) database. The application of the correlated HMM within the verification procedure of a keyword spotter provided an improvement of the Figure-of-Merit from 87.1% to 88.6%.
ICASSP98 Paper Abstract
Using Word Probabilities as Confidence Measures

Authors:
Frank Wessel, RWTH Aachen, ( Germany)
Klaus Macherey, RWTH Aachen, ( Germany)
Ralf Schlüter, RWTH Aachen, ( Germany)

Volume 1, page 225, paper no. 1949

Abstract:
Estimates of confidence for the output of a speech recognition system can be used in many practical applications of speech recognition technology. They can be employed for detecting possible errors and can help to avoid undesirable verification turns in automatic inquiry systems. In this paper we propose to estimate the confidence in a hypothesized word as its posterior probability, given all acoustic feature vectors of the speaker utterance. The basic idea of our approach is to estimate the posterior word probabilities as the sum of all word hypothesis probabilities which represent the occurrence of the same word in more or less the same segment of time. The word hypothesis probabilities are approximated by paths in a wordgraph and are computed using a simplified forward-backward algorithm. We present experimental results on the North American Business (NAB’94) and the German Verbmobil recognition task.
ICASSP98 Paper Abstract
Subword-Based Minimum Verification Error (SB-MVE) Training for Task Independent Utterance Verification

Authors:
Rafid A. Sukkar, Lucent Technologies, (U.S.A.)

Abstract:
In this paper we formulate a training framework and present a method for task independent utterance verification. Verification-specific HMMs are defined and discriminatively trained using minimum verification error training. Task independence is accomplished by performing the verification on the subword level and training the verification models using a general phonetically balanced database that is independent of the application tasks. Experimental results show that the proposed method significantly outperforms two other commonly used task independent utterance verification techniques. It is shown that the equal error rate of false alarms and false keyword rejection is reduced by more than 22% compared to the other two methods on a large vocabulary recognition task.
ICASSP98 Paper Abstract

A Fast Vocabulary Independent Algorithm for Spotting Words in Speech

Authors:
Satya Dharanipragada, IBM, (U.S.A.)
Salim E. Roukos, IBM, (U.S.A.)

Volume 1, page 233, paper no. 2379

Abstract:
In applications such as audio-indexing, spoken message retrieval and video-browsing, it is necessary to have the ability to detect spoken words that are outside the vocabulary of the speech recognizer used in these systems, in large amounts of speech at speeds many times faster than real-time. In this paper we present a fast, vocabulary independent, algorithm for spotting words in speech. The algorithm consists of a preprocessing stage and a coarse-to-detailed search strategy for spotting a word/phone sequence in speech. The preprocessing method provides a phone-level representation of the speech that can be searched efficiently. The coarse search, consisting of phone-ngram matching, identifies regions of speech as putative word hits. The detailed acoustic match is then conducted only at the putative hits identified in the coarse match. This gives us the desired accuracy and speed in wordspotting.
ICASSP98 Paper Abstract

Integration of Utterance Verification with Statistical Language Modeling and Spoken Language Understanding

Authors:
Richard C. Rose, AT&T Labs - Research, (U.S.A.)
Huan Yao, AT&T Labs - Research, (U.S.A.)
Giuseppe Riccardi, AT&T Labs - Research, (U.S.A.)
Jeremy H. Wright, AT&T Labs - Research, (U.S.A.)

Volume 1, page 237, paper no. 5102

Abstract:
Methods for utterance verification (UV) and their integration into statistical language modeling and spoken language understanding formalisms for a large vocabulary spoken understanding system are presented. The paper consists of three parts. First, a set of acoustic likelihood ratio based utterance verification techniques are described and applied to the problem of rejecting portions of a hypothesized word string that may have been incorrectly decoded by a large vocabulary continuous speech recognizer. Second, a procedure for integrating the acoustic level confidence measures with the statistical language model is described. Finally, the effect of integrating acoustic level confidence into the spoken language understanding unit (SLU) in a call-type classification is discussed. These techniques were evaluated on utterances collected from a highly unconstrained call routing task performed over the telephone network. They have been evaluated in terms of their ability to classify utterances into a set of 15 semantic actions corresponding to call-types that are accepted by the application.
ICASSP98 Paper Abstract

Robust Continuous Speech Recognition System Based on a Microphone Array

Authors:
Eduardo Lleida, University of Zaragoza, (Spain)
Julian Fernandez, University of Zaragoza, (Spain)
Enrique Masgrau, University of Zaragoza, (Spain)

Volume 1, page 241, paper no. 1407

Abstract:
In this paper, a robust speech recognition system for videoconference applications is presented based on a microphone array. By means of a microphone array, the speech recognition system is able to know the position of the users and increase the signal-to-noise (SNR) ratio between the desired speaker signal and the interferences from the other users. The user positions are estimated by means of the combination of a direction of arrival (DOA) estimation method with a speaker identification system. The beamforming is performed by using the spatial references of the desired speaker and the interference locations. A minimum variance algorithm with spatial constraints working in the frequency domain is used to design the weights of the broad band microphone array. Results of the speech recognition system are reported in a simulated environment with several users asking questions to a geographic data base.
ICASSP98 Paper Abstract

Hands-free Speech Recognition Based on 3-D Viterbi Search Using a Microphone Array

Authors:
Takeshi Yamada, Nara Institute of Science and Technology, (Japan)
Satoshi Nakamura, Nara Institute of Science and Technology, (Japan)
Kiyohiro Shikano, Nara Institute of Science and Technology, (Japan)

Abstract:
A microphone array is the promising solution for realizing hands-free speech recognition in real environments. Accurate talker localization is very important for speech recognition using the microphone array. However, localization of a moving talker is difficult in noisy reverberant environments. The talker localization errors degrade the performance of speech recognition. To solve the problem, this paper proposes a new speech recognition algorithm which considers multiple talker direction hypotheses simultaneously. The proposed algorithm performs Viterbi search in 3-dimensional trellis space composed of talker directions, input frames, and HMM states. As a result, a locus of the talker and a phoneme sequence of the speech are obtained by finding an optimal path with the highest likelihood. To evaluate the performance of the proposed algorithm, speech recognition experiments are carried out on simulated data and real environment data. These results show that the proposed algorithm works well even if the talker moves.
ICASSP98 Paper Abstract

Using a Real Time, Tracking Microphone Array as Input to an HMM Speech Recognizer

Authors:
Tadd B Hughes, Brown University, (U.S.A.)
Hong-Seok Kim, Brown University, (U.S.A.)
Joseph H DiBiase, Brown University, (U.S.A.)
Harvey F Silverman, Brown University, (U.S.A.)

Volume 1, page 249, paper no. 1858

Abstract:
A major problem for speech recognition systems is relieving the talker of the need to use a close-talking, head-mounted or a desk-stand microphone. A likely solution is the use of an array of microphones that can steer itself to the talker and can use a beamforming algorithm to overcome the reduced signal-to-noise ratio due to room acoustics. This paper reports results for a tracking, real-time microphone-array as an input to an HMM-based connected alpha-digits speech recognizer. For a talker in the very near field of the array (within a meter), performance approaches that of a close-talking microphone input device. The effects of both the noise reducing steered array and the use of a Maximum a posteriori (MAP) training step are shown to be significant. Here, the array system and the recognizer are described, experiments are presented, and the implications of combining these two systems discussed.
ICASSP98 Paper Abstract

Compensation of Speaker Directivity in Speech Recognition Using HMM Composition

Authors:
Franck Giron, NTT Human Interface Laboratories, (Japan)
Yasuhiro Minami, NTT Human Interface Laboratories, (Japan)
Masashi Tanaka, NTT Human Interface Laboratories, (Japan)
Ken'ichi Furuya, NTT Human Interface Laboratories, (Japan)

Volume 1, page 253, paper no. 1669

Abstract:
In hands-free speech recognition the speaker should be able to move freely in front of the speech acquisition device. However, the speech signal is then submitted to variations due to the continuous change of position in the acoustic space. This paper focuses on the role of speaker head rotations as compared with static situations in anechoic conditions. The effect of speaker directivity in speech recognition performance degradation is demonstrated and a compensation method based on HMM composition is proposed to increase the performance.
ICASSP98 Paper Abstract

Subword Unit Based Speech Recognition in Car Environments

Authors:
Alexander Fischer, Philips Research Labs, Aachen, (Germany)
Volker Stahl, Philips Research Labs, Aachen, (Germany)

Abstract:
This paper presents results of speaker-independent speech recognition experiments concerning acoustic front-ends, models and their structures in car environments. The database comprises 350 speakers in 6 different cars. We investigate whole-word models, context-independent phoneme models and context-dependent within-word phoneme models. We studied task-dependent (same vocabulary context in training and test) phoneme models and present first results on task-independent (broad context in training, i.e. phonetically rich material) scenarios. The latter allows flexible vocabulary definition for applications with dynamically changing command words or new applications avoiding an expensive data collection. Acoustic preprocessing is carried out with mel-cepstrum combined with spectral subtraction and SNR normalization. The task-dependent word error rates are well below 3% for both whole-word and phoneme models. The task-independent scenarios have to be worked on further.
ICASSP98 Paper Abstract

Solutions for Robust Recognition over the GSM Cellular Network

Authors:
Lamia Karray, FT-CNET/DIH/RCP, (France)
Abdellatif BenJelloun, FT-CNET/DIH/RCP, (France)
Chafic Mokbel, FT-CNET/HIH/RCP, (France)

Abstract:
This paper deals with automatic speech recognition robustness for noisy wireless communications. We propose several solutions to improve speech recognition over the cellular network. Two architectures are derived for the recognizer. They are based on Hidden Markov Models (HMMs) adapted to adverse noise conditions. Then two more specific solutions aiming to alleviate GSM cellular network defects (holes and impulsive noise) are developed. Holes are detected and rejected. Impulsive noises are modeled using mixture density HMMs and a maximum likelihood criterion. These solutions allow a noticeable recognition error reduction. The last one seems to be promising.
ICASSP98 Paper Abstract

Speech Recognition in Non-Stationary Adverse Environments

Authors:
Zhong-Hua Wang, INRS-Telecommunications, (Canada)
Patrick Kenny, INRS-Telecommunications, (Canada)

Abstract:
In this paper, we introduce a new approach, called nonstationary adaptation (NA), to recognize speech under nonstationary adverse environments. Two models are used: one is a speaker-independent hidden Markov model (HMM) for clean speech, the other is an ergodic Markov chain representing the nonstationary adverse environment. Each state in the Markov chain represents one stationary adverse condition and has associated with it an affine transform that is estimated by maximum likelihood linear regression (MLLR). Three kinds of adverse environments are considered: (i) multi-speaker speech recognition where speaker identity changes randomly and this constitutes a nonstationary adverse condition, (ii) the recognition of speech corrupted by machinegun noise, (iii) the cross-talk problem. The algorithm is tested on the Nov92 development database of WSJF0 with a vocabulary size of 20,000. In multi-speaker speech recognition, NA decreases the error rate by 13.6%. For speech corrupted by machinegun noise, a one-state Markov chain decreases the error rate by 18%, and a two-state Markov chain gives another 14% decrease in error rate. In the cross-talk problem, a one-state Markov chain decreases the error rate by 16.8%. Two-state and three-state Markov chains decrease the error rate by 22% and 24.4%, respectively.
ICASSP98 Paper Abstract
Robust Speech Recognition in Car Environments

Authors:
Makoto Shozakai, Nara Institute of Science and Technology, (Japan)
Satoshi Nakamura, Nara Institute of Science and Technology, (Japan)
Kiyohiro Shikano, Nara Institute of Science and Technology, (Japan)

Volume 1, page 269, paper no. 1561

Abstract:
A user-friendly speech interface in a car cabin is highly needed for safety reasons. This paper will describe a robust speech recognition method that can cope with additive noises and multiplicative distortions. A known additive noise, a source signal of which is available, might be canceled by NLMS-VAD (Normalized Least Mean Squares with frame-wise Voice Activity Detection). On the other hand, an unknown additive noise, a source signal of which is not available, is suppressed with CSS (Continuous Spectral Subtraction). Furthermore, various multiplicative distortions are simultaneously compensated with E-CMN (Exact Cepstrum Mean Normalization) which is speaker-dependent/environment-dependent CMN for speech/non-speech. Evaluation results of the proposed method for car cabin environments are finally described.
ICASSP98 Paper Abstract

TD-PSOLA versus Harmonic Plus Noise Model in Di-iphone Based Speech Synthesis

Authors:
Ann K Syrdal, AT&T Labs, (U.S.A.)
Yannis G Stylianou, AT&T Labs, (U.S.A.)
Laurie F Garrison, AT&T Labs, (U.S.A.)
Alistair Conkie, AT&T Labs, (U.S.A.)
Juergen Schroeter, AT&T Labs, (U.S.A.)

Volume 1, page 273, paper no. 2089

Abstract:
In an effort to select a speech representation for our next generation concatenative text-to-speech synthesizer, the use of two candidates is investigated; TD-PSOLA and the Harmonic plus Noise Model, HNM. A formal listening test has been conducted and the two candidates have been rated regarding intelligibility, naturalness and pleasantness. Ability for database compression and computational load is also discussed. The results show that HNM consistently outperforms TD-PSOLA in all the above features except for computational load. HNM allows for high-quality speech synthesis without smoothing problems at the segmental boundaries and without buzziness or other oddities observed with TD-PSOLA.
ICASSP98 Paper Abstract

A Hybrid Approach to Synthesize High Quality Cantonese Speech

Authors:
Min Chu, Chinese University of Hong Kong, (Hong Kong)
P.C. Ching, Chinese University of Hong Kong, (Hong Kong)

Abstract:
Synthesizing high quality speech necessitates an intelligent modification algorithm to adjust the important prosodic features of the pre-stored speech units to meet the desired requirements, such as smoothness, naturalness and pleasantness. TD-PSOLA is a simple but effective method of varying the pitch and time-scaling of speech and it can produce high quality synthetic output. However, when the prosodic pattern requires a drastic modification in the spectral content, TD-PSOLA often generates speech with reverberation. This paper develops a hybrid synthesis method based on TD-PSOLA and shape-invariant sinusoidal technique to alleviate the problem of reverberation. It is particularly useful for the generation of Cantonese speech, since it can cope with the rapidly changing of the pitch profile of Cantonese, which is a mono-syllabic and tonal language. The proposed method has been applied to construct a Cantonese synthesizer which is shown to be capable of producing high quality Cantonese speech without reverberation.
ICASSP98 Paper Abstract

A System for Voice Conversion Based on Probabilistic Classification and a Harmonic Plus Noise Model

Authors:
Yannis G Stylianou, AT&T Labs - Research, (U.S.A.)
Olivier Cappé, ENST, (France)

Volume 1, page 281, paper no. 2072

Abstract:
Voice conversion is defined as modifying the speech signal of one speaker (source speaker) so that it sounds as if it had been pronounced by a different speaker (target speaker). This paper describes a system for efficient voice conversion. A novel mapping function is presented which associates the acoustic space of the source speaker with the acoustic space of the target speaker. The proposed system is based on the use of a Gaussian Mixture Model, GMM, to model the acoustic space of a speaker and a pitch synchronous harmonic plus noise representation of the speech signal for prosodic modifications. The mapping function is a continuous parametric function which takes into account the probabilistic classification provided by the mixture model (GMM). Evaluation by objective tests showed that the proposed system was able to reduce the perceptual distance between the source and target speaker by 70%. Formal listening tests also showed that 97% of the converted speech was judged to be spoken from the target speaker while maintaining high speech quality.
ICASSP98 Paper Abstract
Spectral Voice Conversion for Text-to-Speech Synthesis

Authors:
Alexander Kain, Oregon Graduate Institute, (U.S.A.)
Michael W Macon, Oregon Graduate Institute, (U.S.A.)

Volume 1, page 285, paper no. 2212

Abstract:
A new voice conversion algorithm that modifies a source speaker's speech to sound as if produced by a target speaker is presented. It is applied to a residual-excited LPC text-to-speech diphone synthesizer. Spectral parameters are mapped using a locally linear transformation based on Gaussian mixture models whose parameters are trained by joint density estimation. The LPC residuals are adjusted to match the target speaker's average pitch. To study effects of the amount of training on performance, data sets of varying sizes are created by automatically selecting subsets of all available diphones by a vector quantization method. In an objective evaluation, the proposed method in found to perform more reliably for small training sets than a previous approach. In perceptual tests, it was shown that nearly optimal spectral conversion performance was achieved, even with a small amount of training data. However, speech quality improved with increases in the training set size.
ICASSP98 Paper Abstract

Speaker Transformation using Sentence HMM Based Alignments and Detailed Prosody Modification

Authors:
Levent M Arslan, Entropic Research Lab, (U.S.A.)
David Talkin, Entropic Research Lab, (U.S.A.)

Volume 1, page 289, paper no. 2580

Abstract:
This paper presents several improvements to our voice conversion system which we refer to as Speaker Transformation Algorithm using Segmental Codebooks (STASC). First, a new concept, sentence HMM, is introduced for the alignment of speech waveforms sharing the same text. This alignment technique allows reliable and high resolution mapping between two speech waveforms. In addition, it is observed that energy and speaking rate differences between two speakers are not constant across all phonemes. Therefore, a codebook based duration and energy scaling algorithm is proposed. Finally, a more detailed pitch modification is introduced that takes into account pitch range differences between source and target speakers in addition to mean pitch level differences. The proposed changes improved the quality of transformed speech. Subjective listening tests showed that for a male to male transformation intelligibility is maintained at the same level as natural speech after the speaker transformation.
ICASSP98 Paper Abstract

Automatic Generation of Synthesis Units for Trainable Text-to-Speech Systems

Authors:
Hsiao-Wuen Hon, Microsoft Research, (U.S.A.)
Alex Acero, Microsoft Research, (U.S.A.)
Xuedong Huang, Microsoft Research, (U.S.A.)
Jingsong Liu, Microsoft Research, (U.S.A.)
Mike Plumpe, Microsoft Research, (U.S.A.)

Volume 1, page 293, paper no. 2571

Abstract:
Whistler Text-to-Speech engine was designed so that we can automatically construct the model parameters from training data. This paper will describe in detail the design issues of constructing the synthesis unit inventory automatically from recording databases. The automatic process includes (1) determining the scaleable synthesis unit which can reflect spectral variations of different allophones; (2) segmenting the recording sentences into phonetic segments; (3) select good instances for each synthesis unit to generate best synthesis sentence during run time. These processes are all derived through the use of probabilistic learning methods that are aimed at the same optimization criteria. Through the automatic unit generation, Whistler can automatically produce synthetic speech that sounds very natural and resembles the acoustic characteristics of the original speaker.
ICASSP98 Paper Abstract

Optimization of a Neural Network for Speaker and Task Dependent F0-Generation

Authors:
Ralf Haury, Siemens AG, (Germany)
Martin Holzapfel, Siemens AG, (Germany)

Volume 1, page 297, paper no. 2432

Abstract:
The generation of a pleasant pitch contour is an important issue for the naturalness of each TTS system. Till now the results are far from being satisfactory. In this paper we present a speaker and task specific approach realized by a neural network. Personal and task specific characteristics are maintained and the demand of generalization decreases. So the results in application can significantly be improved. Using an optimized network structure global and well localized patterns can be covered and trained simultaneously within one network. Correlation analysis of the database versus the sensitivity of the trained network validates the importance of distinctive parameters in training. Based on this comparison we give a discussion of the generalization properties of the nn trained speaker and task dependency. Finally a variation of the context range helps to find an optimized tuning of the input parameter set.
ICASSP98 Paper Abstract

Practical High-Quality Speech and Voice Synthesis Using Fixed Frame Rate ABS/OLA Sinusoidal Modeling

Authors:
E. Bryan George, Texas Instruments, (U.S.A.)

Volume 1, page 301, paper no. 2068

Abstract:
This paper describes algorithms developed to apply the Analysis-by-Synthesis/Overlap-Add (ABS/OLA) sinusoidal modeling system to real-time speech and singing voice synthesis. As originally proposed, the ABS/OLA system is limited to unidirectional time-scaling, and relies on variable frame length to accomplish time-scale modification. For speech and voice synthesis applications, unidirectional time scaling makes effective looping to produce sustained vocal sounds difficult, and variable frame length makes real-time polyphonic synthesis problematic. This paper presents a reformulation of the basic ABS/OLA system to deal with these issues, which is termed Fixed-Rate ABS/OLA (ABS/OLA-FR).
ICASSP98 Paper Abstract

An Automatic Method for Learning a Japanese Lexicon for Recognition of Spontaneous Speech

Authors:
Laura Mayfield Tomokiyo, Carnegie Mellon University, (U.S.A.)
Klaus Ries, Universitaet Karlsruhe, (Germany)

Volume 1, page 305, paper no. 2305

Abstract:
When developing a speech recognition system, one must start by deciding what the units to be recognized should be. This is for the most part a straightforward choice in the case of word-based languages such as English, but becomes an issue even in handling languages with a complex compounding system like German; with an agglutinative language like Japanese, which provides no spaces in written text, the choice is not at all obvious. Once an appropriate unit has been determined, the problem of consistently segmenting transcriptions of training data must be addressed. This paper describes a method for learning a lexicon from a training corpus which contains no word-level segmentation, applied to the problem of building a Japanese speech recognition system. We show not only that one can satisfactorily segment transcribed training data automatically, avoiding human error, but also that our system, when trained with the automatically segmented corpus, showed a significant improvement in recognition performance.
ICASSP98 Paper Abstract
Acoustics-Only Based Automatic Phonetic Baseform Generation

Authors:
Bhuvana Ramabhadran, IBM T.J. Watson Research Center, (U.S.A.)
Lalit R. Bahl, IBM T.J. Watson Research Center, (U.S.A.)
Peter V DeSouza, IBM T.J. Watson Research Center, (U.S.A.)
Mukund Padmanabhan, IBM T.J. Watson Research Center, (U.S.A.)

Volume 1, page 309, paper no. 2275

Abstract:
Phonetic baseforms are the basic recognition units in most speech recognition systems. These baseforms are usually determined by linguists once a vocabulary is chosen and not modified thereafter. However, several applications, such as name dialing, require the user be able to add new words to the vocabulary. These new words are often names, or task-specific jargon, that have user-specific pronunciations. This paper describes a novel method for generating phonetic transcriptions (baseforms) of words based on acoustic evidence alone. It does not require either the spelling or any prior acoustic representation of the new word, is vocabulary independent, and does not have any linguistic constraints (pronunciation rules). Our experiments demonstrate the high decoding accuracies obtained when baseforms deduced using this approach are incorporated into our speech recognizer. Also, the error rates on the added words were found to be comparable to or better than when the baseforms were derived by hand.
ICASSP98 Paper Abstract

Pronunciation Modelling Using a Hand-Labelled Corpus for Conversational Speech Recognition

Authors:
William J. Byrne, Johns Hopkins University, (U.S.A.)
Michael Finke, Carnegie Mellon University, (U.S.A.)
Sanjeev P. Khudanpur, Johns Hopkins University, (U.S.A.)
John McDonough, Johns Hopkins University, (U.S.A.)
Harriet Nock, Cambridge University, (U.K.)
Michael D. Riley, AT&T Labs - Research, (U.S.A.)
Murat Saraclar, Johns Hopkins University, (U.S.A.)
Charles Wooters, US Department of Defense, (U.S.A.)
George Zavaliagkos, BBN, (U.S.A.)

Volume 1, page 313, paper no. 2380

Abstract:
Accurately modelling pronunciation variability in conversational speech is an important component of an automatic speech recognition system. We describe some of the projects undertaken in this direction during and after WS97, the Fifth LVCSR Summer Workshop, held at Johns Hopkins University, Baltimore, in July-August, 1997. We first illustrate a use of hand-labelled phonetic transcriptions of a portion of the Switchboard corpus, in conjunction with statistical techniques, to learn alternatives to canonical pronunciations of words. We then describe the use of these alternate pronunciations in an automatic speech recognition system. We demonstrate that the improvement in recognition performance from pronunciation modelling persists as the system is enhanced with better acoustic and language models.
ICASSP98 Paper Abstract
The Use of Accent-Specific Pronunciation Dictionaries in Acoustic Model Training

Authors:
Jason J Humphries, Cambridge University, (U.K.)
Philip C. Woodland, Cambridge University, (U.K.)

Volume 1, page 317, paper no. 1969

Abstract:
Speech recognition systems are increasingly being built to cover an ever wider range of speaker accents. However, electronically available pronunciation dictionaries (PDs) specific to these accents often do not exist and would be time consuming and expensive to build by hand. This paper explores the use of pronunciation modelling for the synthesis of accent-specific PDs directly from acoustic data, and their use in acoustic model training. It is shown that this is particularly effective when the amount of acoustic data from the new accent region is insufficient to build a new recogniser, and it is necessary to retrain an existing system: a further 15% reduction in word error rate can be achieved over and above the 20% reduction resulting from acoustic model retraining alone. This paper also presents an empirical evaluation of an American English PD which has been synthesised from a British English PD.
ICASSP98 Paper Abstract
Specific Language Modelling for New-Word Detection in Continuous-Speech Recognition

Authors:
Rachida El-Meliani, INRS-Telecommunications, (Canada)
Douglas O'Shaughnessy, INRS-Telecommunications, (Canada)

Volume 1, page 321, paper no. 1264

Abstract:
The objective of this work is to allow the INRS continuous-speech recognizer to process accurately new words and incorporate them into the vocabulary. Until now only a few new-word detectors have been reported, all of them defining an acoustic filler model different from the models used to represent vocabulary words. In this paper, we define several designs using, unlike other researchers, strictly-lexical fillers and a unique process to perform speech recognition, new-word detection and new-word phonetic transcription. Moreover, we propose here four different types of language models differing in the way they use the limited information we gathered on new words. The best combinations are found to be different from the ones we obtained for keyword spotting.
ICASSP98 Paper Abstract

Phonetic Recognition for Spoken Document Retrieval

Authors:
Kenney Ng, MIT, (U.S.A.)
Victor W Zue, MIT, (U.S.A.)

Abstract:
This paper describes the development and application of a phonetic recognition system to the task of spoken document retrieval. The recognizer is used to generate phonetic transcriptions of the speech messages which are then processed to produce subword unit representations for indexing and retrieval. Subword units are used as an alternative to words units generated by either keyword spotting or word recognition. We first investigate the use of different acoustic and language models in the speech recognizer in an effort to improve phonetic recognition performance. Then we examine a variety of subword unit indexing terms and measure their ability to perform effective spoken document retrieval. Finally, we look at some simple robust indexing and retrieval methods that take into account the characteristics of the recognition errors in an attempt to improve retrieval performance.
ICASSP98 Paper Abstract

Topic Extraction with Multiple Topic-Words in Broadcast-News Speech

Authors:
Katsutoshi Ohtsuki, NTT, (Japan)
Tatsuo Matsuoka, NTT, (Japan)
Shoichi Matsunaga, NTT, (Japan)
Sadaoki Furui, Tokyo Institute of Technology, (Japan)

Abstract:
This paper reports on topic extraction in Japanese broadcast-news speech. We studied, using continuous speech recognition, the extraction of several topic-words from broadcast-news. A combination of multiple topic-words represents the content of the news. This is a more detailed and more flexible approach than using a single word or a single category. A topic-extraction model shows the degree of relevance between each topic-word and each word in the article. For all words in an article, topic-words which have high total relevance score are extracted. We trained the topic-extraction model with five years of newspapers, using the frequency of topic-words taken from headlines and words in articles. The degree of relevance between topic-words and words in articles is calculated on the basis of statistical measures, i.e., mutual information or the chi-square-value. In topic-extraction experiments for recognized broadcast-news speech, we extracted five topic-words from the 10-best hypotheses using a chi-square-based model and found that 76.6% of them agreed with the topic-words chosen by subjects.
ICASSP98 Paper Abstract

A Hidden Markov Model Approach to Text Segmentation and Event Tracking

Authors:
Jonathan P Yamron, Dragon Systems Inc., (U.S.A.)
Ira Carp, Dragon Systems Inc., (U.S.A.)
Larry Gillick, Dragon Systems Inc., (U.S.A.)
Steve Lowe, Dragon Systems Inc., (U.S.A.)
Paul Van Mulbregt, Dragon Systems Inc., (U.S.A.)

Volume 1, page 333, paper no. 2335

Abstract:
Continuing progress in the automatic transcription of broadcast speech via speech recognition has raised the possibility of applying information retrieval techniques to the resulting (errorful) text. For these techniques to be easily applicable, it is highly desirable that the transcripts be segmented into stories. This paper introduces a general methodology based on HMMs and on classical language modeling techniques for automatically inferring story boundaries and for retrieving stories relating to a specific event. In this preliminary work, we report some highly promising results on accurate text. Future work will apply these techniques to errorful transcripts.
ICASSP98 Paper Abstract

A Two Stage Hybrid Embedded Speech/Audio Coding Structure

Authors:
Sean A Ramprashad, Bell Labs, (U.S.A.)

Abstract:
A two stage hybrid embedded speech/audio coding structure is proposed. The structure uses a speech coder as a core to provide the minimal bitrate and an acceptable performance on speech inputs. The second stage is transform coder using a MDCT and perceptual coding principles. This stage is itself embedded both in complexity and bitrate, and provides various levels of enhancement of the core output, particularly for general audio signals like music. Informal A-B comparison tests show that the performance of the structure at 16 kb/s is between that of the GSM Enhanced Full Rate coder at 12.2 kb/s, and the G.728 LD-CELP coder at 16 kb/s.
A Bitrate and Bandwidth Scalable CELP Coder

Abstract:
This paper proposes a bitrate and bandwidth scalable CELP speech coder. The proposed coder is based on multi-pulse-based CELP coding and consists of a bitrate scalable base-band coder and a bandwidth extension tool. The bitrate scalable base-band CELP coder employs multi-stage excitation coding based on an embedded-coding approach. The multi-pulse excitation codebook at each stage is adaptively produced depending on the selected excitation signal at the previous stage. The bandwidth scalability is realized by bandwidth-conversion from base-band CELP parameters to those for wideband without a widely used subband structure. The bandwidth-conversion improves base-band coding quality and expands bandwidth, simultaneously. The comparison test results show that the bitrate scalable coder is equivalent in speech quality to the fixed-bitrate CELP coder at the same bitrate for the narrowband speech. In the MOS tests, the proposed 16 kbit/s coder with the bandwidth scalability achieves equivalent coding quality to ITU-T G.722 at 56 kbit/s.
ICASSP98 Paper Abstract
Nonlinear Prediction with Neural Nets in ADPCM

Authors:
Marcos Faundez-Zanuy, Escola Universitaria Politecnica de Mataro, (Spain)
Francesc Vallverdu, Signal Theory & Communications Dept., (Spain)
Enric Monte, Signal Theory & Communications Dept., (Spain)

Abstract:
In the last years there has been a growing interest for nonlinear speech models. Several works have been published revealing the better performance of nonlinear techniques, but little attention has been dedicated to the implementation of the nonlinear model into real applications. This work is focused on the study of the behaviour of a nonlinear predictive model based on neural nets, in a speech waveform coder. Our novel scheme obtains an improvement in SEGNSNR between 1 and 2 dB for an adaptive quantization ranging from 2 to 5 bits.
Mach1: Nonuniform Time-Scale Modification of Speech

Authors:
Michele M Covell, Interval Research Corporation, (U.S.A.)
Margaret M Withgott, Electric Planet, (U.S.A.)
Malcolm G. Slaney, Interval Research Corporation, (U.S.A.)

Abstract:
We propose a new approach to nonuniform time compression, called Mach1, designed to mimic the natural timing of fast speech. At identical overall compression rates, listener comprehension for Mach1-compressed speech increased between 5 and 31 percentage points over that for linearly compressed speech, and response times dropped by 15%. For rates between 2.5 and 4.2 times real time, there was no significant comprehension loss with increasing Mach1 compression rates. In A-B preference tests, Mach1-compressed speech was chosen 95% of the time. This paper describes the Mach1 technique and our listener-test results. Audio examples can be found on http://www.interval.com/papers/1997-061/.
ICASSP98 Paper Abstract

Speech Compression Based on Exact Modeling and Structured Total Least Norm Optimization

Authors:
Philippe Lemmerling, Katholieke Universiteit Leuven, (Belgium)
Ioannis Dologlou, Katholieke Universiteit Leuven, (Belgium)
Sabine Van Huffel, Katholieke Universiteit Leuven, (Belgium)

Abstract:
We present a new speech coding algorithm, based on an all-pole model of the vocal tract. Whereas current Auto Regressive (AR) based modeling techniques (e.g. CELP, LPC-10) minimize a prediction error, our approach determines the closest (in L2 norm) signal, which exactly satisfies an all-pole model. Each frame is then encoded by storing the parameters of the complex damped exponentials deduced from the all-pole model and its initial conditions. Decoding is performed by adding the complex damped exponentials based on the transmitted parameters. The new algorithm is demonstrated on a speech signal. The quality is compared with that of a standard coding algorithm at comparable compression ratios, by using the segmental Signal-to-Noise Ratio (SNR).
Gender Adapted Speech Coding

Authors:
David F Marston, Ensigma Ltd, (U.K.)

Abstract:
Speech coders that are optimized to the characteristics of a particular set of speakers will outperform a speech coder which caters for all speakers; providing that the speaker using it is one of that particular set. This paper describes how speech coders that are optimized to either male or female speech can be an improvement over unoptimised coders. These improvements are bit-rate reduction, speech quality and robustness. A reliable gender identifier is described, which would be practical for the most demanding applications, achieving 95% accuracy after 1 second of speech. The improvements in terms of gender specific speech coding are shown in LSF quantisation with bit-saving, and pitch detection with both bit-saving and robustness.
ICASSP98 Paper Abstract

On Nonlinear Utilization of Intervector Dependency in Vector Quantization

Authors:
Mikael Skoglund, Royal Institute of Technology, (Sweden)
Jan Skoglund, Chalmers University of Technology, (Sweden)

Volume 1, page 361, paper no. 2029

Abstract:
This paper presents an approach to vector quantization of sources exhibiting intervector dependency. We present the optimal decoder based on a collection of received indices. We also present the optimal encoder for such decoding. The optimal decoder can be implemented as a table look-up decoder, however the size of the decoder codebook grows very fast with the size of the collection of utilized indices. This leads us to introduce a method for storing an approximation to the set of optimal decoder vectors, based on linear mapping of a block code vector quantization. In this approach a heavily reduced set of parameters is employed to represent the codebook. Furthermore, we illustrate that the proposed scheme has an interpretation as nonlinear predictive quantization. Numerical results indicate high gain over memoryless coding and memory quantization based on linear predictive coding. The results also show that the sub-optimal approach performs close to the optimal.
ICASSP98 Paper Abstract

A Voice Activity Detector Employing Soft Decision Based Noise Spectrum Adaptation

Authors:
Jongseo Sohn, Seoul National University, (Korea)
Wonyong Sung, Seoul National University, (Korea)

Abstract:
In this paper, a voice activity detector (VAD) for variable rate speech coding is decomposed into two parts, a decision rule and a background noise statistic estimator, which are analysed separately by applying a statistical model. A robust decision rule is derived from the generalized likelihood ratio test by assuming that the noise statistics are known a priori. To estimate the time-varying noise statistics, allowing for the occasional presence of the speech signal, a novel noise spectrum adaptation algorithm using the soft decision information of the proposed decision rule is developed. The algorithm is robust, especially for the time-varying noise such as babble noise.
ICASSP98 Paper Abstract
Towards a Synergistic Multistage Speech Coder

Authors:
Manohar N Murthi, University of California, San Diego, (U.S.A.)
Bhaskar D. Rao, University of California, San Diego, (U.S.A.)

Volume 1, page 369, paper no. 2243

Abstract:
In this paper, we propose some new modeling techniques that provide a more synergistic approach to multistage time-domain speech compression. In particular, we propose a new error criterion for determining all-pole filters, and a unique method for jointly coding the pulse information in excitation vectors. The new error criterion for determining all-pole filters is based upon minimizing the sum of the residual signal's absolute values raised to a power less than one. It is shown to be a desirable cost function for yielding residual signals that are more sparse, and consequently better suited for multistage compression than Linear Prediction residuals. Statistical reasons supporting the new criterion are also provided. Furthermore, exploiting the properties of, and the relationship between, the Linear Prediction and Minimum Variance spectra, we propose a novel parameter set for jointly coding the excitation vector's pulse position, sign, and gain information.
ICASSP98 Paper Abstract
Robust Speech Decoding: Can Error Concealment Be Better Than Error Correction?

Authors:
Tim Fingscheidt, Aachen University of Technology, (Germany)
Peter Vary, Aachen University of Technology, (Germany)
Jesus A. Andonegui, Aachen University of Technology, (Germany)

Abstract:
Digital speech transmission systems use source coding to reduce the bit rate and channel coding to correct transmission errors. Furthermore, in periods of a very poor channel quality error concealment of residual bit errors becomes necessary as channel decoding fails. However, if the channel is clear, channel coding would not be required at all and the speech quality could be improved by allowing a higher bit rate for source encoding. Usually a compromise is taken between speech quality in case of clear channel and error robustness in case of poor channel quality. This paper addresses the problem of a joint optimization of error concealment and source/channel coding. Under the premise of a minimum mean square error criterion for signal reconstruction it turns out that error concealment instead of error correction may be the best choice if source coding leaves sufficient residual parameter correlations by less bit rate reduction.
An Energy-Constrained Signal Subspace Method for Speech Enhancement and Recognition in Colored Noise

Authors:
Jun Huang, University of Illinois, Urbana-Champaign, (U.S.A.)
Yunxin Zhao, University of Illinois, Urbana-Champaign, (U.S.A.)

Abstract:
An energy-constrained signal subspace (ECSS) method is proposed for speech enhancement and recognition under an additive colored noise condition. The key idea is to match the short-time energy of the enhanced speech signal to the unbiased estimate of the short-time energy of the clean speech, which is proven very effective for improving the estimation of the noise-like, low-energy segments in speech signal. The colored noise is modeled by an autoregressive (AR) process. A modified covariance method is used to estimate the AR parameters of the colored noise and a prewhitening filter is constructed based on the estimated parameters. The performance of the proposed algorithm was evaluated using the TIMIT digit database and the TIMIT continuous speech database. It was found that the ECSS method can significantly improve the signal-to-noise ratio (SNR) and word recognition accuracy (WRA) for isolated digits and continuous speech under various SNR conditions.
ICASSP98 Paper Abstract
Removal of Noise from Speech Using the Dual EKF Algorithm

Authors:
Eric A Wan, Oregon Graduate Institute, (U.S.A.)
Alex T Nelson, Oregon Graduate Institute, (U.S.A.)

Volume 1, page 381, paper no. 2119

Abstract:
Noise reduction for speech signals has applications ranging from speech enhancement for cellular communications, to front ends for speech recognition systems. A neural network based time-domain method called Dual Extended Kalman Filtering (Dual EKF) is presented for removing nonstationary and colored noise from speech. This paper describes the algorithm and provides a set of experimental results.
ICASSP98 Paper Abstract

Combined Wiener and Coherence Filtering in Wavelet Domain For Microphone Array Speech Enhancement

Authors:
Djamila Mahmoudi, Swiss Federal Institute of Technology, Lausanne, (Switzerland)
Andrzej Drygajlo, Swiss Federal Institute of Technology, Lausanne, (Switzerland)

Abstract:
Wiener filter based postfiltering has shown its usefulness in microphone array speech enhancement systems. In our earlier work, we developed a postfilter in the wavelet domain where better performance has been obtained compared to the algorithms developed in the Fourier domain. Furthermore, considerable computational savings are provided thanks to the multi-resolution and multi-rate analysis. This contribution shows that the coherence function, calculated between the beamforming output signal and the microphone reference output signal using wavelet transform, provides relevant and exploitable information for further noise suppression. Thus, a nonlinear coherence filtering and Wiener filter are combined in the wavelet transform domain to improve the performance of the Wiener filter based postfilter, especially during pauses. Evaluations of the new algorithm confirm that speech quality is indeed improved with significantly reduced distortions. Finally, the results of the objective measures are presented.
ICASSP98 Paper Abstract

Speech Enhancement in a Bayesian Framework

Authors:
Gaafar M.K. Saleh, *Cambridge University, (U.K.)*
Mahesan Niranjan, *Cambridge University, (U.K.)*

Volume 1, page 389, paper no. 5192

Abstract:
We present an approach for the enhancement of speech signals corrupted by additive white noise of Gaussian statistics. The speech enhancement problem is treated as a signal estimation problem within a Bayesian framework. The conventional all-pole speech production model is assumed to govern the behaviour of the clean speech signal. The additive noise level and all-pole model gain are automatically inferred during the speech enhancement process. The strength of the Bayesian approach developed in this paper lies in its ability to perform speech enhancement without the usual requirement of estimating the level of the corrupting noise from “silence” segments of the corrupted signal. The performance of the Bayesian approach is compared to that of the Lim & Oppenheim framework, to which it follows a similar iterative nature. A significant quality improvement is obtained over the Lim & Oppenheim framework.
ICASSP98 Paper Abstract
Speech Enhancement for Bandlimited Speech

Authors:
David A Heide, Naval Research Laboratory, (U.S.A.)
George S Kang, Naval Research Laboratory, (U.S.A.)

Abstract:
Throughout the history of telecommunication, speech has rarely been transmitted with its full analog bandwidth (0 to 8 kHz or more) due to limitations in channel bandwidth. This impaired legacy continues with tactical voice communication. The passband of a voice terminal is typically 0 to 4 kHz. Hence, high-frequency speech components (4 to 8 kHz) are removed prior to transmission. As a result, speech intelligibility suffers, particularly for low-data-rate vocoders. In this paper, we describe our speech-processing technique, which permits some of the upperband speech components to be translated into the passband of the vocoder. According to our test results, speech intelligibility is improved by as much as three to four points even for the recently developed and excellent Department of Defense-standard Mixed Excitation Linear Predictor (MELP) 2.4 kb/s vocoder. Note that speech intelligibility is improved without expanding the transmission bandwidth or compromising interoperability with others.
A Novel Psychoacoustically Motivated Audio Enhancement Algorithm Preserving Background Noise Characteristics

Authors:
Stefan N.A. Gustafsson, IND, RWTH Aachen, (Germany)
Peter J. Jax, IND, RWTH Aachen, (Germany)
Peter Vary, IND, RWTH Aachen, (Germany)

Volume 1, page 397, paper no. 1183

Abstract:
In this paper we propose an algorithm for reduction of noise in audio signals. In contrast to several previous approaches we do not try to achieve a complete removal of the noise, but instead our goal is to preserve a predefined amount of the original noise in the processed signal. This is accomplished by exploiting the masking properties of the human auditory system. The speech and noise distortions are considered separately. The spectral weighting rule, adapted by utilizing only estimates of the masking threshold and the noise power spectral density, has been designed to guarantee complete masking of distortions of the residual noise. Simulation results confirm that no audible artifacts are left in the processed signal, while speech distortions are comparable to those caused by conventional noise reduction techniques. Audio demonstrations are available from http://www.ind.rwth-aachen.de.
ICASSP98 Paper Abstract

Speech Enhancement based on a Voiced-Unvoiced Speech Model

Authors:
Zenton Goh, Nanyang Technological University, (Singapore)
Kah-Chye Tan, Nanyang Technological University, (Singapore)
B.T.G. Tan, National University of Singapore, (Singapore)

Abstract:
In this work, we attempt to refine the methods based on autoregressive (AR) modeling for speech enhancement [1,2]. As a matter of fact, AR modelling, which is a key strategy of the methods reported in [1,2], is known to be good for representing unvoiced speech but not quite appropriate for voiced speech which is quite periodic in nature. Here, we incorporate a speech model which satisfactorily describes voiced and unvoiced speeches and silence (i.e., pauses between speech utterances) into the enhancement framework developed in [1,2], and specifically devise an algorithm for computing the optimal estimate of the clean speech in the minimum-mean-square-error sense. We also present the methods we use for estimating the model parameters and give a description of the complete enhancement procedure. Performance assessment based on spectrogram plots, objective measures and informal subjective listening tests all indicate that our method gives consistently good results.
ICASSP98 Paper Abstract
Enhancement of Reverberant Speech Using LP Residual

Authors:
Bayya Yegnanarayana, Institute of Technology, Madras, (India)
Philkhana Satyanarayana Murthy, Institute of Technology, Madras, (India)
Carlos Avendano, Oregon Graduate Institute of Science, (U.S.A.)
Hynek Hermansky, Oregon Graduate Institute of Science, (U.S.A.)

Abstract:
In this paper we propose a new method of processing speech degraded by reverberation. The method is based on analysis of short (2 ms) segments of data to enhance the regions in the speech signal having high Signal to Reverberant component Ratio (SRR). The short segment analysis shows that SRR is different in different segments of speech. The processing method involves identifying and manipulating the linear prediction residual in three different regions of the speech signal, namely, high SRR region, low SRR region and only reverberation component region. A weighting function is derived to modify the LP residual. The weighted residual samples are used to excite the time-varying LP all-pole filter to obtain perceptually enhanced speech.
Improved Phone Recognition Using Bayesian Triphone Models

Authors:
Ji Ming, The Queens University of Belfast, (Northern Ireland)
F. Jack Smith, The Queens University of Belfast, (Northern Ireland)

Abstract:
A crucial issue in triphone based continuous speech recognition is the large number of models to be estimated against the limited availability of training data. This problem can be relieved by composing a triphone model from less context-dependent models. This paper introduces a new statistical framework, derived from the Bayesian principle, to perform such a composition. The potential power of this new framework is explored, both algorithmically and experimentally, by an implementation with hidden Markov modeling techniques. This implementation is applied to the recognition of the 39-phone set on the TIMIT database. The new model achieves 74.4% and 75.6% accuracy, respectively, on the core and complete test sets.
ICASSP98 Paper Abstract

Multilingual Phone Recognition of Spontaneous Telephone Speech

Authors:
Cristobal Corredor-Ardoy, LIMSI-CNRS, Orsay, (France)
Lori Lamel, LIMSI-CNRS, Orsay, (France)
Martine Adda-Decker, LIMSI-CNRS, Orsay, (France)
Jean-Luc Gauvain, LIMSI-CNRS, Orsay, (France)

Volume 1, page 413, paper no. 5241

Abstract:
In this paper we report on experiments with phone recognition of spontaneous telephone speech. Phone recognizers were trained and assessed on IDEAL, a multilingual corpus containing telephone speech in French, British English, German and Castillan Spanish. We investigated the influence of the training material composition (size and linguistic content) on the recognition performance using context-independent Hidden Markov Models and phonotactic bigram models. We found that when testing on spontaneous speech data, using only spontaneous speech training data gave the highest phone accuracies for the four languages, even though this data comprises only 14% of the available training data. The use of context-dependent HMMs reduced the phone error across the 4 languages, with the average error reduced to 51.9 % from the 57.4% obtained with CI models. We suggest a straightforward way of detecting non speech phenomena. The basic idea is to remove sequences of consonants between two silence labels from the recognized phone strings prior to scoring. This simple technique reduces the relative average phone error rate by 5.4%. The lowest phone error with CD models and filtering was obtained for Spanish (39.1%) with 4 language average being 49.1%.
ICASSP98 Paper Abstract

Language Adaptation of Multilingual Phone Models for Vocabulary Independent Speech Recognition Tasks

Authors:
Joachim Koehler, Siemens AG, (Germany)

Volume 1, page 417, paper no. 1941

Abstract:
This paper presents our new results on multilingual phone modeling and adaptation into a new target language which is not included in the trained multilingual models. The experiments were carried out with the SpeechDat(M) and MacroPhone databases including the languages French, German, Italian, Portuguese, Spanish and American English. First, we constructed language-dependent and multilingual phone models. The recognition rate for an isolated word task decreased in average only by 3.2% using 95 multilingual instead of 232 language-dependent models. Second, we investigated adaptation techniques for cross-language transfer and showed that only 100 utterances from a new language were needed for adaptation. Using the MAP algorithm the recognition rate was improved from 79.9% to 84.3%. Finally, we defined a phonetic based dissimilarity measure between 2 languages and compared language-dependent and multilingual models for the purpose of cross-language transfer.
ICASSP98 Paper Abstract
Advances in Alpha Digit Recognition Using Syllables

Authors:
Jonathan Hamaker, Mississippi State University, (U.S.A.)
Aravind Ganapathiraju, Mississippi State University, (U.S.A.)
Joseph Picone, Mississippi State University, (U.S.A.)
John J Godfrey, PSL, Texas Instruments Inc., (U.S.A.)

Volume 1, page 421, paper no. 2044

Abstract:
In this paper, we present a set of experiments which explore the use of syllables for recognition of continuous alphadigit utterances. In this system, syllables are used as the primary unit of recognition. This work was motivated by our need to verify and isolate phenomena seen when performing syllable-based experiments on the SWITCHBOARD corpus. The performance of our base syllable system is better than a crossword triphone system while requiring a small portion of the resources necessary for triphone systems. All experiments were performed on the OGI Alphadigits corpus, which consists of telephone-bandwidth alphadigit strings. The WER of the best syllable system (context-independent syllables) reported here is 11.1\% compared to 12.2\% for a crossword triphone system.
LVCSR Rescoring with Modified Loss Functions: A Decision Theoretic Perspective

Authors:
Vaibhava Goel, The Johns Hopkins University, (U.S.A.)
William J. Byrne, The Johns Hopkins University, (U.S.A.)
Sanjeev P. Khudanpur, The Johns Hopkins University, (U.S.A.)

Abstract:
The problem of speech decoding is considered here in a Decision Theoretic framework and a modified speech decoding procedure to minimize the expected risk under a general loss function is formulated. A specific word error rate loss function is considered and an implementation in an N-best list rescoring procedure is presented. Methods for estimation of the parameters of the resulting decision rule are provided for both supervised and unsupervised training. Preliminary experiments on an LVCSR task show a small but statistically significant error rate improvements.
ICASSP98 Paper Abstract


Authors:
Udo Bub, Siemens AG, (Germany)
Harald Hoege, Siemens AG, (Germany)

Volume 1, page 429, paper no. 1961

Abstract:
The research described in this paper focuses on possibilities to avoid the tedious training of Hidden-Markov-Models when setting up a new recognition task. A major speaker independent cause for the decrease of recognition accuracy is a mismatch of the phonetic contexts between training and testing data. To overcome this problem, we introduced in previous work the idea of an update of task independent acoustic models by means of Bayesian learning. In this paper we introduce the new approach of adaptively splitting the probability density functions (pdfs) of a continuous density HMM. The goal is to model the appropriate state pdfs better so that they can more accurately match new contexts that are observed while the system is in service. Splitting AND Bayesian adaptation yields a remarkable reduction of word error rate compared to Bayesian adaptation only.
ICASSP98 Paper Abstract

Improvements in Children’s Speech Recognition Performance

Authors:
Subrata Das, IBM T.J. Watson Research Center, (U.S.A.)
Don Nix, IBM T.J. Watson Research Center, (U.S.A.)
Michael Picheny, IBM T.J. Watson Research Center, (U.S.A.)

Volume 1, page 433, paper no. 2172

Abstract:
There are several reasons why conventional speech recognition systems modeled on adult data fail to perform satisfactorily on children’s speech input. For instance, children’s vocal characteristics differ significantly from those of adults. In addition, their choices of vocabulary and sentence construction modalities usually do not follow adult patterns. We describe comparative studies demonstrating the performance gain realized by adopting to children’s acoustic and language model data to construct a children’s speech recognition system.
ICASSP98 Paper Abstract

Speaker Normalized Acoustic Modeling Based on 3-D Viterbi Decoding

Authors:
Toshiaki Fukada, ATR-ITL, (Japan)
Yoshinori Sagisaka, ATR-ITL, (Japan)

Volume 1, page 437, paper no. 1924

Abstract:
This paper describes a novel method for speaker normalization based on a frequency warping approach to reduce variations due to speaker-induced factors such as the vocal tract length. In our approach, a speaker normalized acoustic model is trained using time-varying (i.e., state, phoneme or word dependent) warping factors, while in the conventional approaches, the frequency warping factor is fixed for each speaker. These time-varying frequency warping factors are determined by a 3-dimensional (i.e., input frames, HMM states and warping factors) Viterbi decoding procedure. Experimental results on Japanese spontaneous speech recognition show that the proposed method yields a 9.7% improvement in speech recognition accuracy compared to the conventional speaker-independent model.
ICASSP98 Paper Abstract

Adaptive Heterodyne Filters (AHF) for Detection and Attenuation of Narrow Band Signals

Authors:
Karl E Nelson, University of California, (U.S.A.)
Michael A Soderstrand, University of California, (U.S.A.)

Volume 1, page 441, paper no. 2264

Abstract:
A fixed filter may be converted into an adaptive filter with a single adaptation parameter through the use of a new Adaptive Heterodyne Filter (AHF) concept in which the frequency of the heterodyne signal is adjusted thereby translating the entire filter transfer function in frequency. If the fixed filter is selected to be a very narrow-band band-pass filter, the new AHF concept can be used very effectively in the elimination of narrow band interference in wide-band communications or control systems. A specific example of the removal of a slow-moving time-varying mechanical resonance from the control signal for a flight control system demonstrates the power of the new AHF concept.
ICASSP98 Paper Abstract

Online Tool Wear Monitoring in Turning Using Time-Delay Neural Networks

Authors:
Bernhard Sick, University of Passau, (Germany)

Abstract:
Wear monitoring systems often use neural networks for a sensor fusion with multiple input patterns. Systems for a continuous online supervision of wear have to process pattern sequences. Therefore recurrent neural networks have been investigated in the past. However, in most cases where only noisy input or even noisy output patterns are available for a supervised learning, success is not forthcoming. That is, recurrent networks don’t perform noticeably better than non-recurrent networks processing only the current input pattern like multilayer perceptrons. This paper demonstrates on the basis of an application example (online tool wear monitoring in turning) that results can be improved significantly with special non-recurrent networks. This approach uses time-delay neural networks which consider the position of a single pattern in a pattern sequence by means of delay elements at the synapses. In the mentioned application example the average error in the estimation of a characteristic wear parameter could be reduced by about 24% compared with multilayer perceptrons.
ICASSP98 Paper Abstract

Discriminative Training of Hidden Markov Models Using a Classification Measure Criterion

Authors:
Cristina Chesta, Politecnico di Torino, (Italy)
Aldo Girardi, Politecnico di Torino, (Italy)
Pietro Laface, Politecnico di Torino, (Italy)
Mario Nigra, CSELT Torino, (Italy)

Volume 1, page 449, paper no. 2063

Abstract:
This paper proposes the optimization of a non standard objective function in the framework of Maximum Mutual Information Estimation (MMIE). In contrast with the classical MMIE estimation, where only misrecognized training utterances contribute to the optimization process, the contributions of near-miss classifications are naturally embedded in the maximization of the proposed function because it takes into account a non linear combination of the probabilities of the competing models that can be tuned by means of a single parameter. This corrective training procedure has been applied to an Isolated Word Recognition task leading to significant performance improvements with respect to Maximum Likelihood Estimation and MMIE.
ICASSP98 Paper Abstract

A Discriminant Measure for Model Complexity Adaptation

Authors:
Lalit R. Bahl, IBM, (U.S.A.)
Mukund Padmanabhan, IBM, (U.S.A.)

Volume 1, page 453, paper no. 1187

Abstract:
We present a new discriminant measure that can be used to determine the "goodness" of acoustic models in speech recognition system, and identify shortcomings in the model. We have used this measure to adapt the complexity of the acoustic model. In general, speech recognition systems model phones or sub-phonetic units with mixtures of Gaussians where the number of components in the mixture are chosen using some simple rule-of-thumb. The new measure is used to select the number of mixture components in a more objective fashion, and provides improvements in the error performance.
ICASSP98 Paper Abstract

Natural Number Recognition Using MCE Trained Inter-Word Context Dependent Acoustic Models

Authors:
Malan B Gandhi, Lucent Technologies, (U.S.A.)
John Jacob, Lucent Technologies, (U.S.A.)

Volume 1, page 457, paper no. 2312

Abstract:
Among applications that require number recognition, the focus has largely been on connected digit recognizers. In this paper, we introduce an acoustic model topology for natural number recognition by using minimum classification error (MCE) training of inter-word context dependent models of the head-body-tail (HBT) type. Experimental results on natural number applications involving dollar amounts and U.S. telephone numbers show that using HBT models for natural number data reduces string error rates by as much as 25% over context independent whole-word models. In addition, for speech input which is strictly of connected digit type, the increase in string error rates is negligible when a natural number telephone grammar is used instead of a connected digit telephone grammar. This will enable natural number speech recognition systems to be more widely accepted because recognition accuracy is maintained while permitting a more natural and flexible user interface.
Abstract:
We attack the general problem of HMM-based speech recognizer design, and in particular, the problem of isolated letter recognition in the presence of background noise. The standard design method based on maximum likelihood (ML) is known to perform poorly when applied to isolated letter recognition. The more recent minimum classification error (MCE) approach directly targets the ultimate design criterion and offers substantial improvements over the ML method. However, the standard MCE method relies on gradient descent optimization which is susceptible to shallow local minima traps. In this paper, we propose to overcome this difficulty with a powerful optimization method based on deterministic annealing (DA). The DA method minimizes a randomized MCE cost subject to a constraint on the level of entropy which is gradually relaxed. It may be derived based on information-theoretic or statistical physics principles. DA has a low implementation complexity and outperforms both standard ML and the gradient descent based MCE algorithm by a factor of 1.5 to 2.0 on the benchmark CSLU spoken letter database. Further, the gains are maintained under a variety of background noise conditions.
ICASSP98 Paper Abstract
Speaker Adaptation for Hybrid MMI/Connectionist Speech Recognition Systems

Authors:
Jörg Rottland, Gerhard-Mercator-University Duisburg, (Germany)
Christoph Neukirchen, Gerhard-Mercator-University Duisburg, (Germany)
Gerhard Rigoll, Gerhard-Mercator-University Duisburg, (Germany)

Abstract:
In this paper we present a new adaptation technique for our hybrid large vocabulary continuous speech recognition system. In most adaptation approaches the HMM parameters are reestimated. In our approach, however, we train a speaker independent continuous speech recognizer, then we keep the HMM parameters fixed and we train a second network, which transforms the features of the adaptation data to fit the HMM parameters. Thus, less parameters have to be estimated, and therefore this approach performs well even for a small number of adaptation data. With this approach we achieve relative improvements in recognition rates on the Wall Street Journal (WSJ) task of 16.5%.
ICASSP98 Paper Abstract

Maximum Mutual Information Based Reduction Strategies for Cross-Correlation Based Joint Distributional Modeling

Authors:
Jeff A Bilmes, ICSI University of California Berkeley, (U.S.A.)

Volume 1, page 469, paper no. 2509

Abstract:
In maximum-likelihood based speech recognition systems, it is important to accurately estimate the joint distribution of feature vectors given a particular acoustic model. In previous work, we showed we can boost accuracy in this task by modeling the joint distribution of time-localized feature vectors along with statistics relating those feature vectors to their surrounding context. In this work, we evaluate information preserving reduction strategies for those statistics. We claim that those statistics corresponding to spectro-temporal loci in speech with relatively large mutual information are most useful in estimating the information contained in the feature-vector joint distribution. Furthermore, we claim that such statistics are most likely to generalize. Using an EM algorithm to compute mutual information between pairs of points in the time-frequency grid, we verify these hypotheses using both overlap plots and speech recognition word error results.
ICASSP98 Paper Abstract
Experiments of HMM Adaptation for Hands-free Connected Digit Recognition

Authors:
Diego Giuliani, ITC-IRST, (Italy)
Marco Matassoni, ITC-IRST, (Italy)
Maurizio Omologo, ITC-IRST, (Italy)
Piergiorgio Svaizer, ITC-IRST, (Italy)

Volume 1, page 473, paper no. 1985

Abstract:
A scenario concerning hands-free connected digit recognition in a noisy office environment is investigated. An array of six omnidirectional microphones and a corresponding time delay compensation module are used to provide a beamformed signal as input to a Hidden Markov Model (HMM) based recognizer. Two different techniques of phone HMM adaptation have been considered, to reduce the mismatch between training and test conditions. Adaptation material and test material were collected in two different sessions. Results show that a digit accuracy close to 98% can be achieved when the talker is at 1.5 m distance from the array. This result has to be compared with 99.5% accuracy obtained by using a close-talk microphone.
ICASSP98 Paper Abstract
Task Independent Minimum Confusibility Training for Continuous Speech Recognition

Authors:
Albino Nogueiras-Rodríguez, Universitat Politecnica de Catalunya, (Spain)
José B Mariño, Universitat Politecnica de Catalunya, (Spain)

Volume 1, page 477, paper no. 2512

Abstract:
In this paper, a task independent discriminative training framework for subword units based continuous speech recognition is presented. Instead of aiming at the optimisation of any task independent figure, say the phone classification or recognition rates, we focus our attention to the reduction of the number of errors committed by the system when a task is defined. This consideration leads to the use of a segmental approach based on the minimisation of the confusibility over short chains of subword units. Using this framework, a reduction of 32% in the string error rate may be achieved in the recognition of unknown length digit strings using task independent phone like units.
Discriminative Model Combination

Authors:
Peter Beyerlein, Philips Research Labs, (Germany)

Abstract:
Discriminative model combination is a new approach in the field of automatic speech recognition, which aims at an optimal integration of all given (acoustic and language) models into one log-linear posterior probability distribution. As opposed to the maximum entropy approach, the coefficients of the log-linear combination are optimized on training samples using discriminative methods, to obtain an optimal classifier. Three methods are discussed to find coefficients, which minimize the empirical word error rate on given training data: the well-known GPD-based minimum error rate training, a minimization of the mean distance between the discriminant function of the model combination and an “ideal” discriminant function and a minimization of a smoothed error count measure. Latter two methods lead to closed-form solutions for the coefficients of the model combination. The accuracy of a large vocabulary continuous speech recognition could be increased by the new approach.
ICASSP98 Paper Abstract

Joint MCE Estimation of VQ and HMM Parameters for Gaussian Mixture Selection

Authors:
Shawn M Herman, Lucent Technologies, (U.S.A.)
Rafid A. Sukkar, Lucent Technologies, (U.S.A.)

Abstract:
Vector Quantization (VQ) has been explored in the past as a means of reducing likelihood computation in speech recognizers which use hidden Markov models (HMMs) containing Gaussian output densities. Although this approach has proved successful, there is an extent beyond which further reduction in likelihood computation substantially degrades recognition accuracy. Since the components of the VQ frontend are typically designed after model training is complete, this degradation can be attributed to the fact that VQ and HMM parameters are not jointly estimated. In order to restore the accuracy of a recognizer using VQ to aggressively reduce computation, joint estimation is necessary. In this paper, we propose a technique which couples VQ frontend design with Minimum Classification Error training. We demonstrate on a large vocabulary subword task that in certain cases, our joint training algorithm can reduce the string error rate by 79% compared to that of VQ mixture selection alone.
ICASSP98 Paper Abstract

Multilevel Discriminative Training for Spelled Word Recognition

Authors:
Luca Rigazio, Panasonic Technologies Inc., (U.S.A.)
Jean-Claude Junqua, Panasonic Technologies Inc., (U.S.A.)
Michael Galler, Panasonic Technologies Inc., (U.S.A.)

Volume 1, page 489, paper no. 1800

Abstract:
Discriminative training is effective in enhancing robustness for recognition tasks characterized by high confusion rates. In this paper, we apply discriminative training to different components of a spelled word recognizer to improve recognition accuracy among confusable letters. First we weighted the HMM states to emphasize the letters’ discriminant part. The training achieved a 17% decrease in unit (letter) error rate when the search was performed with an unconstrained grammar. Then we designed a new algorithm that relies on discriminative training to adapt the grammar transition probabilities and the language weight. This method uses acoustic information to provide a tight coupling between the acoustic and language models. Experimental results showed the state weighting followed by the adaptation of a bigram language model reduced by 11% the total unit errors and by 12% the unit errors among the E-Set of the English alphabet.
ICASSP98 Paper Abstract
Comparison of Discriminative Training Criteria

Authors:
Ralf Schlüter, RWTH Aachen, (Germany)
Wolfgang Macherey, RWTH Aachen, (Germany)

Volume 1, page 493, paper no. 2062

Abstract:
In this paper, a formally unifying approach for a class of discriminative training criteria including Maximum Mutual Information (MMI) and Minimum Classification Error (MCE) criterion is presented including, the optimization methods gradient descent (GD) and extended Baum-Welch (EB) algorithm. Comparisons are discussed for the MMI and the MCE criterion including, the determination of the sets of word sequence hypotheses for discrimination using word graphs. Experiments have been carried out on the SieTill corpus for telephone line recorded German continuous digit strings. Using several approaches for acoustic modeling, the word error rates obtained by MMI training using single densities always were better than those for Maximum Likelihood (ML) using mixture densities. Finally, results obtained for corrective training (CT), i.e. using only the best recognized word sequence in addition to the spoken word sequence, could not be improved by using the word-graph based discriminative training.
ICASSP98 Paper Abstract

Improved Neural Network Training of Inter-Word Context Units for Connected Digit Recognition

Authors:
Wei Wei, Oregon Graduate Institute of Science and Technology, (U.S.A.)
Sarel Van Vuuren, Oregon Graduate Institute of Science and Technology, (U.S.A.)

Abstract:
For connected digit recognition the relative frequency of occurrence (prior) for context-dependent phonetic units at inter-word boundaries in training data tends to be much lower than the prior expected for a single test utterance. A problem in using a neural network to model context-dependent phonetic units is that it learns the prior of the training data and not that expected of a test utterance. We show how to compensate for the problem by roughly flattening the class prior for infrequently occurring context units by a suitable weighting of the neural network cost function - based entirely on the training set prior. We show that this leads to improved recognition performance. We give results for telephone speech on the OGI Numbers Corpus. Our method gives a 12.37% reduction of the sentence-level error rate (to 14.76%) and a 9.93% reduction of the word-level error rate (to 3.81%) compared to not doing compensation.
ICASSP98 Paper Abstract
Context Modeling in Hybrid Segment-Based/Neural Network Recognition Systems

Authors:
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Jean-Pierre Martens, ELIS, University of Ghent, (Belgium)

Volume 1, page 501, paper no. 2088

Abstract:
In this paper, we describe the incorporation of context-dependent models in hybrid Segment-Based/Neural Network speech recognition systems. We present alternative probabilistic frameworks and evaluate them by performing speaker-independent phone recognition experiments on the TIMIT corpus. We compare their recognition performances with that of a context-independent hybrid SB/NN system and with the best published performances on this task.
ICASSP98 Paper Abstract

ACID/HNN: Clustering Hierarchies of Neural Networks for Context-Dependent Connectionist Acoustic Modeling

Authors:
Juergen Fritsch, University of Karlsruhe, (Germany)
Michael Finke, University of Karlsruhe, (Germany)

Volume 1, page 505, paper no. 1968

Abstract:
We present the ACID/HNN framework, a principled approach to hierarchical connectionist acoustic modeling in large vocabulary conversational speech recognition (LVCSR). Our approach consists of an Agglomerative Clustering algorithm based on Information Divergence (ACID) to automatically design and robustly estimate Hierarchies of Neural Networks (HNN) for arbitrarily large sets of context-dependent decision tree clustered HMM states. We argue that a hierarchical approach is crucial in applying locally discriminative connectionist models to the typically very large state spaces observed in LVCSR systems. Furthermore, we focus on the benefits of the proposed connectionist acoustic model, namely exploiting the hierarchical structure for speaker adaptation and decoding speed-up algorithms.
ICASSP98 Paper Abstract

New Feedback Method of Hybrid HMM/ANN Methods for Continuous Speech Recognition

Authors:
Tranzai Lee, Chinese Academy of Sciences, (China)
Daowen Chen, Chinese Academy of Sciences, (China)

Volume 1, page 509, paper no. 1120

Abstract:
In the continuous speech recognition, the co-pronunciation between two successive phonemes seriously disturbs recognition effect. It is difficult for pure hidden Markov model (HMM) methods to cope with the co-pronunciation, because HMM methods consider that two successive frames of speech are independant. The hybrid HMM and artificial neural networks (ANN) methods with feedback MLP[1,3] provide the ability to cope with the co-pronunciation by means of the feedback input. In this paper, we propose a new feedback method for feedback hybrid HMM/ANN methods on the basis of the original methods[1,3]. New feedback method provides the more information of co-pronunciation to feedback ANN. As a result, new feedback method falls the error rate 20.4%. Additionally, By means of our previous work, the hybrid methods HMM/ANN with the feedback double MLP structure, we discuss the method that reduces the computation of the feedback MLP during the recognition.
ICASSP98 Paper Abstract

Use of the Pitch Synchronous Wavelet Transform as a New Decomposition Method for WI

Authors:
Nicola R Chong, University of Wollongong, (Australia)
Ian S Burnett, University of Wollongong, (Australia)
Joe F Chicharo, University of Wollongong, (Australia)
Mark M Thomson, Motorola Australian Research Centre, (Australia)

Abstract:
A new characteristic waveform decomposition method based on wavelets is proposed for the Waveform Interpolation (WI) paradigm. In WI, pitch-cycle waveforms are filtered in the evolution domain to decompose the signal into two waveform surfaces, one characterising voiced speech and a second representing unvoiced speech. The slow roll-off of FIR filters leads, however, to a significant inter-relationship between the decomposed surfaces. Here we present the Pitch Synchronous Wavelet Transform (PSWT) as an alternative decomposition mechanism. Filtering is again performed in the evolutionary waveform domain, producing characteristic surfaces at several resolutions. This multi-scale characterisation leads to more flexible quantisation of parameters, especially at higher rates than WI's 2.4kb/s. FIR filters are replaced in the Wavelet filter bank by causal, stable IIR filters which achieve significant delay reductions over their FIR counterparts. Furthermore, IIR filters track the dynamic aspects of the evolutionary surfaces faster, overcoming problems existing in the current WI decomposition.
ICASSP98 Paper Abstract

A 2.4 KBPS Variable Bit Rate ADP-CELP Speech Coder

Authors:
Masahiro Oshikiri, Kansai Research Laboratories, Toshiba Corporation, (Japan)
Masami Akamine, Kansai Research Laboratories, Toshiba Corporation, (Japan)

Volume 1, page 517, paper no. 1549

Abstract:
This paper presents a variable bit rate ADP-CELP (Adaptive Density Pulse Code Excited Linear Prediction) coder that selects one of four kinds of coding structure in each frame based on short time speech characteristics. To improve speech quality and reduce the average bit rate, we have developed a speech/non-speech classification method using spectrum envelope variation, which is robust for background noise. In addition, we propose an efficient pitch lag coding technique. The technique interpolates consecutive frame pitch lags and quantizes a vector of relative pitch lags consisting of variation between an estimated pitch lag and a target pitch lag in plural subframes. The average bit rate of the proposed coder was approximately 2.4 kbps for speech sources with activity factor of 60%. Our subjective testing indicates the quality of the proposed coder exceeds that of the Japanese digital cellular standard with rate of 3.45 kbps.
ICASSP98 Paper Abstract
Multiple Source MOS Evaluation of a Flexible Low-Rate Vocoder

Authors:
Richard L Zinser, GE Corporate Research and Development, (U.S.A.)
Mark I Grabb, GE Corporate Research and Development, (U.S.A.)
Steven R Koch, GE Corporate Research and Development, (U.S.A.)

Volume 1, page 521, paper no. 2059

Abstract:
This paper describes the design and MOS performance of a family of low rate, low complexity speech coding algorithms known as Time Domain Voicing Cutoff (TDVC). TDVC is a predictive coding algorithm that employs a single transition frequency dividing voiced and unvoiced excitation. It provides the voicing flexibility of a frequency domain algorithm with lower complexity and rate overhead. A number of algorithm variants were MOS tested using three distinct sets of source material. The results are discussed in terms of performance for each of the three sources, and demonstrate that choice of source material has a great impact on both vocoder scoring and ranking.
Techniques for Improving Sinusoidal Transform Vocoders

Authors:
Wen-Wei Chang, National Chiao-Tung University, (Taiwan)
De-Yu Wang, National Chiao-Tung University, (Taiwan)

Abstract:
This paper presents quality enhancement of sinusoidal transform coders (STC) via the development of new parametric models. First explored are the benefits of Bark spectrum for use in the design of perceptual coding of the sine-wave amplitudes. According to our results, the proposed approach provides a uniform perceptual fit across the spectrum. To enhance the accuracy of phase representation, noncausal all-pole modeling of the vocal system is also discussed. Experimental results indicate that the use of new parametric models allows the STC to improve the phase accuracy as well as the synthetic speech quality.
ICASSP98 Paper Abstract
Pitch-Synchronous Subband Representation of the Linear-Prediction Residual of Speech

Authors:
Huimin Yang, Tsinghua University, (China)
W. Bastiaan Kleijn, KTH, Royal Institute of Technology, (Sweden)

Volume 1, page 529, paper no. 1992

Abstract:
In this paper, the characteristic waveform (CW) used in the waveform interpolation (WI) speech coder is interpreted as a pitch-synchronous subband representation (PSSR) of the speech. The inconsistency of the method, using the Gabor transform or the cosine modulated lapped transform. Perfect reconstruction of the speech is then guaranteed. Instead of using a time-varying transform, the speech signal is time-warped and pitch-synchronized operation is achieved by a time-invariant transform. Since the PSSR has the same physical meaning as that of the CW used in the WI speech coder, the coding efficiency can be expected to be similar at low rates, while the exact reconstruction property will lead to better quality at higher rates.
ICASSP98 Paper Abstract
Robust Voicing Estimation with Dynamic Time Warping

Authors:
Tian Wang, University of California, Santa Barbara, (U.S.A.)
Vladimir Cuperman, University of California, Santa Barbara, (U.S.A.)

Volume 1, page 533, paper no. 2224

Abstract:
This paper presents a robust voicing estimation algorithm for low bit rate harmonic speech coding. The algorithm is based on waveform time-warping followed by spectral matching based on voiced and unvoiced local spectral models. The objective of time warping is to reduce the effect of pitch variations on the voicing decision. Several adaptive techniques are used to improve the flexibility and robustness of the conventional spectral matching algorithm. An objective evaluation of the new voicing algorithm is obtained by comparing to manually estimated voicing values. Subjective tests of a sinusoidal coder using the new voicing algorithm show significantly better performance than the standard spectral matching algorithm under both clean and noisy environments.
ICASSP98 Paper Abstract

A Simplified Version of the ITU Algorithm for Objective Measurement of Speech Codec Quality

Authors:
Stephen D Voran, Institute for Telecommunication Sciences, (U.S.A.)
Volume 1, page 537, paper no. 1764

Abstract:
ITU-T Recommendation P.861 describes an objective speech quality assessment algorithm for speech codecs. This algorithm transforms codec input and output speech signals into a perceptual domain, compares them, and generates a noise disturbance value, which can be used to estimate perceived speech quality. The performance of this algorithm can be judged by the correlation between those estimates and actual listener opinions from formal subjective listening tests. We show that significant simplifications can be made to the P.861 algorithm with very minimal effect on its performance. Specifically, for the portions of the algorithm under study here, 64% of the floating point operations can be eliminated with only a 3.5% decrease in average correlation to listener opinions. The resulting simplified algorithm may offer a practical new objective function to drive parameter selections, excitation searches, and bit-allocations in speech and audio coders.
ICASSP98 Paper Abstract

Performance of the Modified Bark Spectral Distortion as an Objective Speech Quality Measure

Authors:
Wonho Yang, Temple University, (U.S.A.)
Majid Benbouchta, Temple University, (U.S.A.)
Robert Yantorno, Temple University, (U.S.A.)

Volume 1, page 541, paper no. 2461

Abstract:
The Modified Bark Spectral Distortion (MBSD), used for an objective speech quality measure, was presented previously. The MBSD measure takes into account the noise masking threshold in order to use only audible distortions in the calculation of the distortion measure. Preliminary simulation results have shown improvement of the MBSD over the conventional BSD. In this paper, performance of the MBSD is reported in terms of frame sizes, speech classes, and spectral regions. The performance of the MBSD is not very sensitive to the frame size. The performance of the MBSD for voiced speech is almost the same as for non-silent speech. The high frequency region appears to play an important role in human perception of speech quality.
ICASSP98 Paper Abstract

Application of Meddis’ Inner Hair-Cell Model to The Prediction of Subjective Speech-Quality

Authors:
Markus Hauenstein, University of Kiel, (Germany)

Abstract:
This paper demonstrates how an instrumental speech-quality measure based on the comparison of auditory-nerve firing-patterns can be constructed. Four available subjective tests prove that the mean opinion scores (MOS) estimated by the objective measure are in good agreement with the subjectively obtained results.
ICASSP98 Paper Abstract

Classification of Speech Under Stress Based on Features Derived from the Nonlinear Teager Energy Operator

Authors:
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James F. Kaiser, Duke University, (U.S.A.)

Volume 1, page 549, paper no. 2284

Abstract:
Studies have shown that distortion introduced by stress or emotion can severely reduce speech recognition accuracy. Techniques for detecting or assessing the presence of stress could help neutralize stressed speech and improve robustness of speech recognition systems. Although some acoustic variables derived from linear speech production theory have been investigated as indicators of stress, they are not consistent. In this paper, three new features derived from the nonlinear Teager Energy Operator (TEO) are investigated for stress assessment and classification. It is believed that TEO based features are better able to reflect the nonlinear airflow structure of speech production under adverse stressful conditions. The proposed features outperform stress classification using traditional pitch by +22.5% for the Normalized TEO Autocorrelation Envelope Area feature (TEO-Auto-Env), and by +28.8% for TEO based Pitch feature (TEO-Pitch). Overall neutral/stress classification rates are more consistent for TEO based features (TEO-Auto-Env: standard deviation = 5.15, TEO-Pitch: standard deviation = 7.83) vs. (Pitch: standard deviation = 23.40). Also, evaluation results using actual emergency aircraft cockpit stressed speech from NATO show that TEO-Auto-Env works best for stress assessment.
Improved Robustness for Speech Recognition Under Noisy Conditions Using Correlated Parallel Model Combination

Authors:
Jeih-weih Hung, Institute of Information Science, Academia Sinica, (Taiwan)
Jia-lin Shen, Institute of Information Science, Academia Sinica, (Taiwan)
Lin-Shan Lee, Institute of Information Science, Academia Sinica, (Taiwan)

Abstract:
The parallel model combination (PMC) technique has been shown to achieve very good performance for speech recognition under noisy conditions. In this approach, the speech signal and the noise are assumed uncorrelated during modeling. In this paper, a new correlated PMC is proposed by properly estimating and modeling the nonzero correlation between the speech signal and the noise. Preliminary experimental results show that this correlated PMC can provide significant improvements over the original PMC in terms of both the model differences and the recognition accuracies. Error rate reduction on the order of 14% can be achieved.
ICASSP98 Paper Abstract

Multi-Resolution Cepstral Features for Phoneme Recognition Across Speech Sub-Bands

Authors:
Paul M McCourt, The Queens University of Belfast, (Northern Ireland)
Saeed Vaseghi, The Queens University of Belfast, (Northern Ireland)
Naomi Harte, The Queens University of Belfast, (Northern Ireland)

Abstract:
Multi-resolution sub-band cepstral features strive to exploit discriminative cues in localised regions of the spectral domain by supplementing the full bandwith cepstral features with sub-band cepstral features derived from several levels of sub-band decomposition. Multiresolution feature vectors, formed by concatenation of the subband cepstral features into an extended feature vector, are shown to yield better performance than conventional MFCCs for phoneme recognition on the TIMIT database. Possible strategies for the recombination of partial recognition scores from independent multi-resolution sub-band models are explored. By exploiting the sub-band variations in signal to noise ratio for linearly weighted recombination of the log likelihood probabilities we obtained improved phoneme recognition performance in broadband noise compared to MFCC features. This is an advantage over a purely sub-band approach using non linear recombination which is robust only to narrow band noise.
ICASSP98 Paper Abstract

Improved Model Parameter Compensation Methods for Noise-Robust Speech Recognition

Authors:
YukHyun Chang, LGIC, (Korea)
YongJoo Chung, LGIC, (Korea)
SungHyun Park, LGIC, (Korea)

Abstract:
In this paper we study model parameter compensation methods for noise-robust speech recognition based on CDHMM. First, we proposed a modified PMC method where adjustment term in the model parameter adaptation is varied depending on mixture components of HMM to obtain more reliable model. A state-dependent association factor that controls the average parameter variability of Gaussian mixtures and the variability of the respective mixtures is used to find the final optimum model parameters. Second, we propose a re-estimation solution of environmental variables with additive noise and spectral tilt based on expectation-maximization(EM) algorithm in the cepstral domain. The approach is based on the vector Taylor series(VTS) approximation. In our experiments on a speaker independent isolated Korean word recognition, the modified PMC show better performance than the Gales' PMC and the proposed VTS is superior to both of them.
ICASSP98 Paper Abstract

A Fuzzy Logic-Based Speech Detection Algorithm for Communications in Noisy Environments

Authors:
Alfredo Cavallaro, University of Catania, (Italy)
Francesco Beritelli, University of Catania, (Italy)
Salvatore Casale, University of Catania, (Italy)

Abstract:
In the field of mobile communications correct Voice Activity Detection (VAD) is a crucial point for the perceived speech quality, the reduction of co-channel interference, the power consumption in portable equipment. This paper shows that a valid alternative to deal with the problem of activity decision is to use methodologies like fuzzy logic, which are suitable for problems requiring approximate rather than exact solutions, and which can be presented through descriptive or qualitative expressions. The Fuzzy Voice Activity Detector (FVAD) proposed uses the same set of parameters adopted by the VAD in Annex B to ITU-T G.729 and a set of six fuzzy rules automatically extracted through supervised learning. Objective and listening tests confirm a significative improvement respect the traditional methods above all for low signal-to-noise ratios.
ICASSP98 Paper Abstract
Subband Based Classification of Speech under Stress

Authors:
Ruhí Sarikaya, *Clemson University, (U.S.A.)*
John N. Gowdy, *Clemson University, (U.S.A.)*

Volume 1, page 569, paper no. 2112

Abstract:
This study proposes a new set of feature parameters based on subband analysis of the speech signal for classification of speech under stress. The new speech features are Scale Energy (SE), Autocorrelation-Scale-Energy (ACSE), Subband based cepstral parameters (SC), and Autocorrelation-SC (ACSC). The parameters’ ability to capture different stress types is compared to widely used Mel-scale cepstrum based representations: Mel-frequency cepstral coefficients (MFCC) and Autocorrelation-Mel-scale (AC-Mel). Next, a feedforward neural network is formulated for speaker-dependent stress classification of 10 stress conditions: Angry, Clear, Cond50/70, Fast, Loud, Lombard, Neutral, Question, Slow, and Soft. The classification algorithm is evaluated using a previously established stressed speech database (SUSAS)[4]. Subband based features are shown to achieve +7.3% and +9.1% increase in the classification rates over the MFCC based parameters for ungrouped and grouped stress closed vocabulary test scenarios, respectively. Moreover, the average scores across the simulations of new features are +8.6% and +13.6% higher than MFCC based features for the ungrouped and grouped stress test scenarios respectively.
ICASSP98 Paper Abstract
Separation of Non-Spontaneous and Spontaneous Speech

Authors:
Owen P. Kenny, Defense Science & Technology Organization, (Australia)
Douglas J. Nelson, Department of Defense, Fort Meade, MD, (U.S.A.)
John S. Bodenschatz, Department of Defense, Fort Meade, MD, (U.S.A.)
Heather A. McMonagle, Department of Defense, Fort Meade, MD, (U.S.A.)

Volume 1, page 573, paper no. 5057

Abstract:
This paper outlines and compares three methods for automatically classifying spontaneous and non-spontaneous speech and presents experimental results comparing the performance of the methods. All three methods are based on an analysis of the probability distributions of prosodic features extracted from the speech signal. The first method uses an expansion of the probability distribution in terms of the statistical moments. The second method is an application of a modified Hellinger's method, and the third method is based on a measure of the non-Gaussianity of the data.
ICASSP98 Paper Abstract

Robust Features Derived from Temporal Trajectory Filtering for Speech Recognition under the Corruption of Additive and Convolutional Noises

Authors:
Kuo-Hwei Yuo, National Tsing Hua University, (Taiwan)
Hsiao-Chuan Wang, National Tsing Hua University, (Taiwan)

Volume 1, page 577, paper no. 1526

Abstract:
This paper presents a novel method using robust features for speech recognition when the speech signal is corrupted by additive and convolutional noises. This method is conceptually simple and easy to be implemented. The additive noise and the convolutional noise are removed by temporal trajectory filtering in autocorrelation domain and crpstral domain, respectively. A task of multi-speaker isolated digit recognition is conducted to demonstrate the effectiveness of using these robust features. The case of channel filtered speech signal corrupted by additive white noise and color noise are tested. Experimental results show that significant improvements can be achieved as comparing with some traditional features.
ICASSP98 Paper Abstract

Enhanced Harmonic Coding of Speech with Frequency Domain Transition Modeling

Authors:
Chunyan Li, University of California, Santa Barbara, (U.S.A.)
Vladimir Cuperman, University of California, Santa Barbara, (U.S.A.)

Volume 2, page 581, paper no. 2204

Abstract:
A major source of audible distortion in current low-bit-rate harmonic speech coding algorithms is the ineffective modeling of the transitional speech signals such as onsets, plosives etc. A new method of modeling transitional speech based on a frequency domain approach is introduced in this paper. The approach uses a modified harmonic model able to produce non-periodic pulse sequences in conjunction with a closed-loop analysis-by-synthesis scheme for parameter estimation and quantization. The structure of a speech coding system based on this model is outlined. The proposed approach is shown to give better performance than transition encoding based on a standard CELP algorithm at rates of 4-8kb/s.
ICASSP98 Paper Abstract
Combined Harmonic and Waveform Coding of Speech at Low Bit Rates

Authors:
Eyal Shlomot, University of California, Santa Barbara, (U.S.A.)
Vladimir Cuperman, University of California, Santa Barbara, (U.S.A.)
Allen Gersho, University of California, Santa Barbara, (U.S.A.)

Abstract:
In this paper we present a new approach for speech coding, which combines frequency-domain harmonic coding for periodic and "noise like" unvoiced segments of speech with a time-domain waveform coder for transition signals. This hybrid coder requires special handling of the boundary between voiced and transition segments. We outline the details of a 4kbps hybrid coder and present subjective quality test results of this coder.
ICASSP98 Paper Abstract

A Mixed Sinusoidally Excited Linear Prediction Coder at 4 Kb/s and Below

Authors:
Suat Yeldener, COMSAT, (U.S.A.)
Juan Carlos De Martin, Texas Instruments, (U.S.A.)
Vishu Viswanathan, Texas Instruments, (U.S.A.)

Abstract:
There is currently a great deal of interest in the development of speech coding algorithms capable of delivering toll quality at 4 kb/s and below. For synthesizing high quality speech, accurate representation of the voiced portions of speech is essential. For bit rates of 4 kb/s and below, conventional Code Excited Linear Prediction (CELP) may likely not provide the appropriate degree of periodicity. It has been shown that good quality low bit rate speech coding can be obtained by frequency domain techniques such as Sinusoidal Transform Coding (STC), Multi Band Excitation (MBE), Mixed Excitation Linear Prediction (MELP), and Multi-Band LPC (MB-LPC) vocoders. In this paper, a speech coding algorithm based on an improved version of MB-LPC is presented. Main features of this algorithm include a multi-stage time/frequency pitch estimation and an improved mixed voicing representation. An efficient quantization scheme for the spectral amplitudes of the excitation, called Formant Weighted Vector Quantization, is also used. This improved coder, called Mixed Sinusoidally Excited Linear Prediction (MSELP), yields an unquantized model with speech quality better than the 32 kb/s ADPCM quality. Initial efforts towards a fully quantized 4 kb/s coder, although not yet successful in achieving the toll quality goal, have produced good output speech quality.
ICASSP98 Paper Abstract

A 1.7 KB/S MELP Coder with Improved Analysis and Quantization

Authors:
Alan V. McCree, Texas Instruments, (U.S.A.)
Juan Carlos De Martin, Texas Instruments, (U.S.A.)

Volume 2, page 593, paper no. 2565

Abstract:
This paper describes our new Mixed Excitation Linear Predictive (MELP) coder designed for very low bit rate applications. This new coder, through algorithmic improvements and enhanced quantization techniques, produces better speech quality at 1.7 kb/s than the new U.S. Federal Standard MELP coder at 2.4 kb/s. Key features of the coder are an improved pitch estimation algorithm and a Line Spectral Frequencies (LSF) quantization scheme that requires only 21 bits per frame. With channel coding, this new MELP coder is capable of maintaining good speech quality even in severely degraded channels, at a total bit rate of only 3 kb/s.
ICASSP98 Paper Abstract

A New Approach to Modeling Excitation in Very Low-Rate Speech Coding

Authors:
Shahrokh Ghaemmaghami, Queensland University of Technology, (Australia)
Mohamed Deriche, Queensland University of Technology, (Australia)

Abstract:
A new method for two-band approximation of excitation signals in an LPC model, to improve speech naturalness in very low rate coding, is proposed. Based on a simplified model of Multi-Band Excitation, the method accurately determines the degree of periodicity, using the concept of Instantaneous Frequency (IF) estimation in frequency domain. The harmonic structure in the spectrum of LPC residual, within individual bands, is identified based on flatness of the IF as a criterion for pitch and voicing detection. On this basis, the excitation is modelled by combining a predefined periodic signal in the lower band and a random signal in the higher band. It is shown that this improves considerably the naturalness of reconstructed speech in very low rate coding in comparison with that obtained using traditional binary excitation [1]. The performance of the technique is also given in Temporal Decomposition (TD) based coding at 800 b/s.
ICASSP98 Paper Abstract

A Spectrally Mixed Excitation Vocoder (SMX) with Robust Parameter Determination

Authors:
Yong Duk Cho, Samsung Advanced Institute of Technology, (Korea)
Moo Young Kim, Samsung Advanced Institute of Technology, (Korea)
Sang Ryong Kim, Samsung Advanced Institute of Technology, (Korea)

Abstract:
Sinusoidal speech coders have been widely studied for low-bit rate coding around 4 kbit/s. However, the estimation error of the sinusoidal model parameters would seriously degrade the speech quality. In general, the estimation errors are caused by the effects of various types of speech signal or background noise. In this paper we propose a sinusoidal speech coder with robust parameter determination methods. They consist of spectro-temporal autocorrelation method for robust pitch determination, frequency shifting method for robust voicing level measurement, and residual-spectrum magnitude coding method for spectral magnitude compensation. From the experimental results, we can find the robustnesses of the proposed techniques. In addition, informal listening test of the synthesized speech confirms the effectiveness of the incorporated schemes.
ICASSP98 Paper Abstract
Segmental Vocoder - Going Beyond the Phonetic Approach

Authors:
Jan Cernocky, Technical University of Brno, (Czech Republic)
Genevieve Baudoin, ESIEE Paris, (France)
Gerard Chollet, ENST Paris, (France)

Volume 2, page 605, paper no. 2048

Abstract:
In our paper, the problem of very low bit rate segmental speech coding is addressed. The basic units are found automatically in the training database using temporal decomposition, vector quantization and multigrams. They are modelled by HMMs. The coding is based on recognition and synthesis. In single speaker tests, we obtained intelligible and naturally sounding speech at mean rate of 211.2 b/s. In the end, future extensions of our scheme (diphone-like synthesis and speaker adaptation) as well as possible use of automatically derived units in recognition are discussed.
ICASSP98 Paper Abstract

A Very Low Bit Rate Speech Coder Using HMM-Based Speech Recognition/Synthesis Techniques

Authors:
Keiichi Tokuda, Nagoya Institute of Technology, (Japan)
Takashi Masuko, Tokyo Institute of Technology, (Japan)
Jun Hiroi, Nagoya Institute of Technology, (Japan)
Takao Kobayashi, Tokyo Institute of Technology, (Japan)
Tadashi Kitamura, Nagoya Institute of Technology, (Japan)

Volume 2, page 609, paper no. 2281

Abstract:
This paper presents a very low bit rate speech coder based on HMM (Hidden Markov Model). The encoder carries out phoneme recognition, and transmits phoneme indexes, state durations and pitch information to the decoder. In the decoder, phoneme HMMs are concatenated according to the phoneme indexes, and a sequence of mel-cepstral coefficient vectors is generated from the concatenated HMM by using an ML-based speech parameter generation technique. Finally we obtain synthetic speech by exciting the MLSA (Mel Log Spectrum Approximation) filter, whose coefficients are given by mel-cepstral coefficients, according to the pitch information. A subjective listening test shows that the performance of the proposed coder at about 150 bit/s (for the test data including 26% silence region) is comparable to a VQ-based vocoder at 400 bit/s (= 8 bit/frame x 50 frame/s) without pitch quantization for both coders.
ICASSP98 Paper Abstract

On Properties of Modulation Spectrum for Robust Automatic Speech Recognition

Authors:
Noboru Kanedera, Ishikawa National College of Technology, (Japan)
Hynek Hermansky, Oregon Graduate Institute of Science and Technology, (U.S.A.)
Takayuki Arai, International Computer Science Institute, (U.S.A.)

Volume 2, page 613, paper no. 1448

Abstract:
We report on the effect of band-pass filtering of the time trajectories of spectral envelopes on speech recognition. Several types of filter (linear-phase FIR, DCT, and DFT) are studied. Results indicate the relative importance of different components of the modulation spectrum of speech for ASR. General conclusions are: (1) most of the useful linguistic information is in modulation frequency components from the range between 1 and 16 Hz, with the dominant component at around 4 Hz, (2) it is important to preserve the phase information in modulation frequency domain, (3) The features which include components at around 4 Hz in modulation spectrum outperform the conventional delta features, (4) The features which represent the several modulation frequency bands with appropriate center frequency and band width increaserecognition performance.
ICASSP98 Paper Abstract
Spectral Subband Centroid Features for Speech Recognition

Authors:
Kuldip K. Paliwal, Griffith University, (Australia)

Abstract:
Cepstral coefficients derived either through linear prediction analysis or from filter bank are perhaps the most commonly used features in currently available speech recognition systems. In this paper, we propose spectral subband centroids as new features and use them as supplement to cepstral features for speech recognition. We show that these features have properties similar to formant frequencies and they are quite robust to noise. Recognition results are reported in the paper justifying the usefulness of these features as supplementary features.
ICASSP98 Paper Abstract

Spectral Weighting of SBCOR for Noise Robust Speech Recognition

Authors:
Shoji Kajita, Nagoya University, (Japan)
Kazuya Takeda, Nagoya University, (Japan)
Fumitada Itakura, Nagoya University, (Japan)

Abstract:
Subband-autocorrelation (SBCOR) analysis is a noise robust acoustic analysis based on filter bank and autocorrelation analysis, and aims to extract periodicities associated with the inverse of the center frequency in a subband. In this paper, it is derived that SBCOR results in the lateral inhibitive weighting (LIW) processing of power spectrum, and shown that the LIW is significantly effective for noise robust acoustic analysis using a DTW word recognizer. An interpretation of LIW is also described. In the second half of this paper, a flattening technique of noise spectral envelope using LPC inverse filter is applied to speech degraded with noise, and DTW word recognition is performed. The idea of this inverse filtering technique comes from weakening the strong periodic components included in noise. The experimental results using 32th order LPC inverse filter show that the recognition performance of SBCOR (or LIW) is improved for computer room noise.
ICASSP98 Paper Abstract

Robust Word Recognition Using Threaded Spectral Peaks

Authors:
Brian P Strope, University of California, Los Angeles, (U.S.A.)
Abeer Alwan, University of California, Los Angeles, (U.S.A.)

Volume 2, page 625, paper no. 2166

Abstract:
A novel technique which characterizes the position and motion of dominant spectral peaks in speech, significantly reduces the error-rate of an HMM-based word-recognition system. The technique includes approximate auditory filtering, temporal adaptation, identification of local spectral peaks in each frame, grouping of neighboring peaks into threads, estimation of frequency derivatives, and slowly updating approximations of the threads and their derivatives. This processing provides a frame-based speech representation which is both dependent on perceptually salient aspects of the frame's immediate context, and well-suited to segmentally-stationary statistical characterization. In noise, the representation reduces the error-rate obtained with standard Mel-filter-based feature vectors by as much as a factor of 4, and provides improvements over other common feature-vector manipulations.
ICASSP98 Paper Abstract

A Novel Robust Feature of Speech Signal Based on the Mellin Transform for Speaker-independent Speech Recognition

Authors:
Jingdong Chen, National Laboratory of Pattern Recognition, (China)
Bo Xu, National Laboratory of Pattern Recognition, (China)
Taiyi Huang, National Laboratory of Pattern Recognition, (China)

Abstract:
This paper presents a novel kind of speech feature which is the modified Mellin transform of the log-spectrum of the speech signal (short for MMTLS). Because of the scale invariance property of the modified Mellin transform, the new feature is insensitive to the variation of the vocal tract length among individual speakers, and thus it is more appropriate for speaker-independent speech recognition than the popular used cepstrum. The preliminary experiments show that the performance of the MMTLS-based method is much better in comparison with those of the LPC- and MFC-based methods. Moreover, the error rate of this method is very consistent for different outlier speakers.
ICASSP98 Paper Abstract

Stochastic Features for Noise Robust Speech Recognition

Authors:
Naoto Iwahashi, Sony Corporation, (Japan)
Hongchang Pao, Sony Corporation, (Japan)
Hitoshi Honda, Sony Corporation, (Japan)
Katsuki Minamino, Sony Corporation, (Japan)
Masanori Omote, Sony Corporation, (Japan)

Volume 2, page 633, paper no. 2542

Abstract:
This paper describes a novel technique for noise robust speech recognition, which can incorporate the characteristics of noise distribution directly in features. The feature itself of each analysis frame has a stochastic form, which can represent the probability density function of the estimated speech component in the noisy speech. Using the sequence of the probability density functions of the estimated speech components and Hidden Markov Modelling of clean speech, the observation probability of the noisy speech is calculated. In the whole process of the technique, the explicit information on SNR is not used. The technique is evaluated by large vocabulary isolated word recognition under car noise environment, and is found to have clearly outperformed nonlinear spectral subtraction (between 13% and 44% reduction in recognition errors).
ICASSP98 Paper Abstract
Improved Scale-Cepstral Analysis in Speech

Authors:
Srinivasan Umesh, IIT, (India)
Leon Cohen, Hunter College of CUNY, (U.S.A.)

Volume 2, page 637, paper no. 5058

Abstract:
In this paper, we present improvements over the original scale-cepstrum proposed by us. The scale-cepstrum is motivated by a desire to normalize the first-order effects of differences in vocal-tract lengths for a given vowel. Our subsequent work (in ICSLP'96), has shown that a more appropriate frequency-warping than the log-warping used originally is necessary to account for the frequency-dependency of the scale-factor. Using this more appropriate frequency-warping and a modified method of computing the scale-cepstrum we have obtained improved features that provide better separability between vowels than before, and are also robust to noise.
ICASSP98 Paper Abstract
Multi-Band Speech Recognition in Noisy Environments

Authors:
Shigeki Okawa, AT&T Labs, (U.S.A.)
Enrico L. Bocchieri, AT&T Labs, (U.S.A.)
Alexandros Potamianos, AT&T Labs, (U.S.A.)

Volume 2, page 641, paper no. 1784

Abstract:
This paper presents a new approach for multi-band based automatic speech recognition (ASR). Recent work by Bourlard and Hermansky suggests that multi-band ASR gives more accurate recognition, especially in noisy acoustic environments, by combining the likelihoods of different frequency bands. Here we evaluate this likelihood recombination (LC) approach to multi-band ASR, and propose an alternative method, namely feature recombination (FC). In the FC system, after different acoustic analyzers are applied to each sub-band individually, a vector is composed by combining the sub-band features. The speech classifier then calculates the likelihood from the single vector. Thus, band-limited noise affects only few of the feature components, as in multi-band LC system, but, at the same time, all feature components are jointly modeled, as in conventional ASR. The experimental results show that the FC system can yield better performance than both the conventional ASR and the LC strategy for noisy speech.
ICASSP98 Paper Abstract
Clustering via the Bayesian Information Criterion with Applications in Speech Recognition

Authors:
Scott S. Chen, IBM, (U.S.A.)
Ponani S. Gopalakrishnan, IBM, (U.S.A.)

Volume 2, page 645, paper no. 2203

Abstract:
One difficult problem we are often faced with in clustering analysis is how to choose the number of clusters. In this paper, we propose to choose the number of clusters by optimizing the Bayesian information criterion (BIC), a model selection criterion in the statistics literature. We develop a termination criterion for the hierarchical clustering methods which optimizes the BIC criterion in a greedy fashion. The resulting algorithms are fully automatic. Our experiments on Gaussian mixture modeling and speaker clustering demonstrate that the BIC criterion is able to choose the number of clusters according to the intrinsic complexity present in the data.
Restructuring Gaussian Mixture Density Functions in Speaker-Independent Acoustic Model

Authors:
Atsushi Nakamura, ATR Interpreting Telecommunications Research Labs, (Japan)

Volume 2, page 649, paper no. 1943

Abstract:
In continuous speech recognition featuring hidden Markov model (HMM), word N-gram and time-synchronous beam search, a local modeling mismatch in the HMM will often cause the recognition performance to degrade. To cope with this problem, this paper proposes a method of restructuring Gaussian mixture pdfs in a pre-trained speaker-independent HMM based on speech data. In this method, mixture components are copied and shared among multiple mixture pdfs with the tendency of local errors taken into account. The tendency is given by comparing the pre-trained HMM and speech data which was used in the pre-training. Experimental results prove that the proposed method can effectively restore local modeling mismatches and improve the recognition performance.
ICASSP98 Paper Abstract

Using Aggregation to Improve the Performance of Mixture Gaussian Acoustic Models

Authors:
Timothy J Hazen, *MIT, (U.S.A.)*
Andrew K Halberstadt, *MIT, (U.S.A.)*

Volume 2, page 653, paper no. 1799

Abstract:
This paper investigates the use of aggregation as a means of improving the performance and robustness of mixture Gaussian models. This technique produces models that are more accurate and more robust to different test sets than traditional cross-validation using a development set. A theoretical justification for this technique is presented along with experimental results in phonetic classification, phonetic recognition, and word recognition tasks on the TIMIT and Resource Management corpora. In speech classification and recognition tasks error rate reductions of up to 12% were observed using this technique. A method for utilizing tree-structured density functions for the purpose of pruning the aggregated models is also presented.
ICASSP98 Paper Abstract
Semi-Tied Covariance Matrices

Authors:
Mark J.F. Gales, IBM T.J. Watson Research, (U.S.A.)
Volume 2, page 657, paper no. 2154

Abstract:
A standard problem in many classification tasks is how to model feature vectors whose elements are highly correlated. If multi-variate Gaussian distributions are used to model the data then they must have full covariance matrices to do so. This requires a large number of parameters per distribution which restricts the number of distributions that may be robustly estimated, particularly when high dimensional feature vectors are required. This paper describes an alternative to full covariance matrices in these situations. An approximate full covariance matrix is used. The covariance matrix is now split into two elements, one full and one diagonal, which may be tied at completely separate levels. Typically the full elements are extensively tie, resulting in only a small increase in the number of parameters compared to the diagonal case. Thus, dramatically increasing the number of distributions that may be robustly estimated. Simple re-estimation formulae for all the parameters within the standard EM framework are presented. On a large vocabulary speech recognition task a 10% reduction in word error rate over a standard system was achieved.
ICASSP98 Paper Abstract

Maximum Likelihood Modeling with Gaussian Distributions for Classification

Authors:
Ramesh A Gopinath, IBM, (U.S.A.)

Volume 2, page 661, paper no. 2265

Abstract:

Maximum Likelihood (ML) modeling of multiclass data for classification often suffers from the following problems: a) data insufficiency implying overtrained or unreliable models b) large storage requirement c) large computational requirement and/or d) ML is not discriminating between classes. Sharing parameters across classes (or constraining the parameters) clearly tends to alleviate the first three problems. In this paper we show that in some cases it can also lead to better discrimination (as evidenced by reduced misclassification error). The parameters considered are the means and variances of the gaussians and linear transformations of the feature space (or equivalently the gaussian means). Some constraints on the parameters are shown to lead to Linear Discrimination Analysis (a well known result) while others are shown to lead to optimal feature spaces (a relatively new result). Applications of some of these ideas to the speech recognition problem are also given.
ICASSP98 Paper Abstract

Full Expansion of Context-Dependent Networks in Large Vocabulary Speech Recognition

Authors:
Merhyar Mohri, AT&T Labs - Research, ( U.S.A.)
Michael D. Riley, AT&T Labs - Research, ( U.S.A.)
Donald Hindle, AT&T Labs - Research, ( U.S.A.)
Andrej Ljolje, AT&T Labs - Research, ( U.S.A.)
Fernando C Pereira, AT&T Labs - Research, ( U.S.A.)

Volume 2, page 665, paper no. 2474

Abstract:
We combine our earlier approach to context-dependent network representation with our algorithm for determinizing weighted networks to build optimized networks for large-vocabulary speech recognition combining an n-gram language model, a pronunciation dictionary and context-dependency modeling. While fully expanded networks have been used before in restrictive settings (medium vocabulary or no cross-word contexts), we demonstrate that our network determinization method makes it practical to use fully expanded networks also in large-vocabulary recognition with full cross-word context modeling. For the DARPA North American Business News task (NAB), we give network sizes and recognition speeds and accuracies using bigram and trigram grammars with vocabulary sizes ranging from 10,000 to 160,000 words. With our construction, the fully expanded NAB context-dependent networks contain only about twice as many arcs as the corresponding language models. Interestingly, we also find that, with these networks, real-time word accuracy is improved by increasing vocabulary size and n-gram order.
ICASSP98 Paper Abstract

Dynamically Configurable Acoustic Models for Speech Recognition

Authors:
Mei-Yuh Hwang, Microsoft Research, (U.S.A.)
Xuedong Huang, Microsoft Research, (U.S.A.)

Volume 2, page 669, paper no. 2568

Abstract:
Senones were introduced to share Hidden Markov model (HMM) parameters at a sub-phonetic level in 1992 and decision trees were incorporated to predict unseen phonetic contexts in 1993 by Hwang. In this paper, we will describe two applications of the senonic decision tree in (1) dynamically downsizing a speech recognition system for small platforms and in (2) sharing the Gaussian covariances of continuous density HMMs (CHMMs). We experimented how to balance different parameters that can offer the best trade off between recognition accuracy and system size. The dynamically downsized system, without retraining, performed even better than the regular Baum-Welch trained system. The shared covariance model provided as good a performance as the unshared full model and thus gave us the freedom to increase the number of Gaussian means to increase the accuracy of the model. Combining the downsizing and covariance sharing algorithms, a total of 8% error reduction was achieved over the Baum-Welch trained system with approximately the same parameter size.
ICASSP98 Paper Abstract
Training of Subspace Distribution Clustering Hidden Markov Model

Authors:
Brian K Mak, AT&T Labs - Research, (U.S.A.)
Enrico L. Bocchieri, AT&T Labs - Research, (U.S.A.)

Abstract:
Recently we presented our novel subspace distribution clustering hidden Markov models (SDCHMMs) which can be converted from continuous density hidden Markov models (CDHMMs) by clustering subspace Gaussians in each stream over all models. The model conversion has two drawbacks: (1) it does not take advantage of the fewer model parameters in SDCHMMs — theoretically SDCHMMs may be trained with smaller amount of data; and, (2) it involves two separate optimization steps (first training CDHMMs, then clustering subspace Gaussians) and the resulting SDCHMMs are not guaranteed to be optimal. In this paper, we show how SDCHMMs may be trained directly from less speech data if we have a priori knowledge of their architecture. On the ATIS task, a context-independent 20-stream SDCHMM system trained using our novel SDCHMM reestimation algorithm with only 8 minutes of speech performs as well as a CDHMM system trained with 105 minutes of speech.
Exploiting Both Local and Global Constraints for Multi-Span Statistical Language Modeling

Authors:
Jerome R Bellegarda, Apple Computer, (U.S.A.)

Volume 2, page 677, paper no. 1164

Abstract:
A new framework is proposed to integrate the various constraints, both local and global, that are present in the language. Local constraints are captured via n-gram language modeling, while global constraints are taken into account through the use of latent semantic analysis. An integrative formulation is derived for the combination of these two paradigms, resulting in several families of multi-span language models for large vocabulary speech recognition. Because of the inherent complementarity in the two types of constraints, the performance of the integrated language models, as measured by perplexity, compares favorably with the corresponding n-gram performance.
ICASSP98 Paper Abstract

Topic Adaptation for Language Modeling Using Unnormalized Exponential Models

Authors:
Stanley F Chen, Carnegie Mellon University, (U.S.A.)
Kristie Seymore, Carnegie Mellon University, (U.S.A.)
Ronald Rosenfeld, Carnegie Mellon University, (U.S.A.)

Volume 2, page 681, paper no. 2357

Abstract:
In this paper, we present novel techniques for performing topic adaptation on an n-gram language model. Given training text labeled with topic information, we automatically identify the most relevant topics for new text. We adapt our language model toward these topics using an exponential model, by adjusting probabilities in our model to agree with those found in the topical subset of the training data. For efficiency, we do not normalize the model; that is, we do not require that the "probabilities" in the language model sum to 1. With these techniques, we were able to achieve a modest reduction in speech recognition word-error rate in the Broadcast News domain.
ICASSP98 Paper Abstract
Shrinking Language Models by Robust Approximation

Authors:
Adam L Buchsbaum, AT&T Labs, (U.S.A.)
Raffaele Giancarlo, University of Palermo, (Italy)
Jeffery R Westbrook, AT&T Labs, (U.S.A.)

Volume 2, page 685, paper no. 1792

Abstract:
We study the problem of reducing the size of a language model while preserving recognition performance (accuracy and speed). A successful approach has been to represent language models by weighted finite-state automata (WFAs). Analogues of classical automata determinization and minimization algorithms then provide a general method to produce smaller but equivalent WFAs. We extend this approach by introducing the notion of approximate determinization. We provide an algorithm that, when applied to language models for the North American Business task, achieves 25-35% size reduction compared to previous techniques, with negligible effects on recognition time and accuracy.
ICASSP98 Paper Abstract
Cyberpunc: A Lightweight Punctuation Annotation System for Speech

Authors:
Doug Beeferman, Carnegie Mellon University, (U.S.A.)
Adam Berger, Carnegie Mellon University, (U.S.A.)
John Lafferty, Carnegie Mellon University, (U.S.A.)

Volume 2, page 689, paper no. 2392

Abstract:
This paper describes a lightweight method for the automatic insertion of intra-sentence punctuation into text. Despite the intuition that pauses in an acoustic stream are a positive indicator for some types of punctuation, this work will demonstrate the feasibility of a system which relies solely on lexical information. Besides its potential role in a speech recognition system, such a system could serve equally well in non-speech applications such as automatic grammar correction in a word processor and parsing of spoken text. After describing the design of a punctuation-restoration system, which relies on a trigram language model and a straightforward application of the Viterbi algorithm, we summarize results, both quantitative and subjective, of the performance and behavior of a prototype system.
ICASSP98 Paper Abstract

Sub-Sentence Discourse Models for Conversational Speech Recognition

Authors:
Kristine W. Ma, GTE/BBN Technologies, (U.S.A.)
George Zavaliagkos, GTE/BBN Technologies, (U.S.A.)
Marie Meteer, GTE/BBN Technologies, (U.S.A.)

Volume 2, page 693, paper no. 2020

Abstract:
According to discourse theories in linguistics, conversational utterances possess an informational structure that partitions each sentence into two portions: a "given" and "new". In this work, we explore this idea by building sub-sentence discourse language models for conversational speech recognition. The internal sentence structure is captured in statistical language modeling by training multiple n-gram models using the Expectation-Maximization algorithm on the Switchboard corpus. The resulting model contributes to a 30% reduction in language model perplexity and a small gain in word error rate.
ICASSP98 Paper Abstract

Two-Step Generation of Variable-Word-Length Language Model Integrating Local and Global Constraints

Authors:
Shoichi Matsunaga, NTT Human Interface Laboratories, (Japan)
Shigeki Sagayama, NTT Human Interface Laboratories, (Japan)

Volume 2, page 697, paper no. 1940

Abstract:
This paper proposes two-step generation of a variable-length class-based language model that integrates local and global constraints. In the first step, an initial class set is recursively designed using local constraints. Word elements for each class are determined using Kullback divergence and total entropy. In the second step, the word classes are recursively and words are iteratively recreated, by grouping consecutive words to generate longer units and by splitting the initial classes into finer classes. These operations in the second step are carried out selectively, taking into account local and global constraints on the basis of a minimum entropy criterion. Experiments showed that the perplexity of the proposed initial class set is superior to that of the conventional part-of-speech class, and the perplexity of the variable-word-length model consequently becomes lower. Furthermore, this two-step model generation approach greatly reduces the training time.
ICASSP98 Paper Abstract
Language-Model Optimization by Mapping of Corpora

Authors:
Dietrich G Klakow, Philips GmbH Forschungslaboratorien, (Germany)

Volume 2, page 701, paper no. 2287

Abstract:
It is questionable whether words are really the best basic units for the estimation of stochastic language models - grouping frequent word sequences to phrases can improve language models. More generally, we have investigated various coding schemes for a corpus. In this paper, they are applied to optimize the perplexity of n-gram language models. In tests on two large corpora (WSJ and BNA) the bigram perplexity was reduced by up to 29%. Furthermore, this approach allows to tackle the problem of an open vocabulary with no unknown word.
ICASSP98 Paper Abstract
Just-In-Time Language Modelling

Authors:
Adam Berger, Carnegie Mellon University, (U.S.A.)
Robert C Miller, Carnegie Mellon University, (U.S.A.)

Abstract:
Traditional approaches to language modelling have relied on a fixed corpus of text to inform the parameters of a probability distribution over word sequences. Increasing the corpus size often leads to better-performing language models, but no matter how large, the corpus is a static entity, unable to reflect information about events which postdate it. In these pages we introduce an online paradigm which interleaves the estimation and application of a language model. We present a Bayesian approach to online language modelling, in which the marginal probabilities of a static trigram model are dynamically updated to match the topic being dictated to the system. We also describe the architecture of a prototype we have implemented which uses the World Wide Web (WWW) as a source of information, and the results of some initial proof of concept experiments.
ICASSP98 Paper Abstract

Weighted Viterbi Algorithm and State Duration Modelling for Speech Recognition in Noise

Authors:
Nestor Becerra Yoma, University of Edinburgh, Scotland, (U.K.)
Fergus R. McInnes, University of Edinburgh, Scotland, (U.K.)
Mervyn A. Jack, University of Edinburgh, Scotland, (U.K.)

Abstract:
A weighted Viterbi algorithm (HMM) is proposed and applied in combination with spectral subtraction and Cepstral Mean Normalization to cancel both additive and convolutional noises in speech recognition. The weighted Viterbi approach is compared and used in combination with state duration modelling. The results presented in this paper show that a proper weight on the information provided by static parameters can substantially reduce the error rate, and that the weighting procedure improves better the robustness of the Viterbi algorithm than the introduction of temporal constraints with a low computational load. Finally, it is shown that the weighted Viterbi algorithm in combination with temporal constraints leads to a high recognition accuracy at moderate SNR's without the need of an accurate noise model.
ICASSP98 Paper Abstract

Transmissions and Transitions: A Study of Two Common Assumptions in Multi-Band ASR

Abstract:
Is multi-band ASR inherently inferior to a full-band approach because phonetic information is lost due to the division of the frequency space into sub-bands? Do the phonetic transitions in sub-bands occur at different times? The first statement is a common objection of the critics of multi-band ASR, and the second, a common assumption by multi-band researchers. This paper is dedicated to finding answers to both these questions. To study the first point, we calculate phonetic feature transmission for sub-bands. Not only do we fail to substantiate the above objection, but we observe the contrary. We confirm the second hypothesis by analyzing the phonetic transition lags in each sub-band. These results reinforce our view that multi-band speech analysis provides useful information for ASR, particularly when band merging takes place at the end state for a phonetic or syllabic model, allowing sub-bands to be independently time-aligned within the model.
ICASSP98 Paper Abstract

A Recombination Model for Multi-Band Speech Recognition

Authors:
Christophe Cerisara, Loria, (France)
Jean-Paul Haton, Loria, (France)
Jean-Francois Mari, Loria, (France)
Dominique Fohr, Loria, (France)

Volume 2, page 717, paper no. 1979

Abstract:
In this paper, we describe a continuous speech recognition system that uses the multi-band paradigm. This principle is based on the recombination of several independent sub-recognizers, each one assigned to a specific frequency band. The major issue of such systems consists of deciding at which time the recombination must be done. Our algorithm lets each band totally independent from the others, and uses the different solutions to resegment the initial sentence. Finally, the bands are synchronously merged together, according to this new segmentation. The whole system is too complex to be entirely described here, and, in this paper, we will concentrate on the synchronous recombination part, which is achieved by a classifier. The system has been tested in clean and noisy environments, and proved to be especially robust to noise.
ICASSP98 Paper Abstract

Incorporating Information from Syllable-length Time Scales into Automatic Speech Recognition

Authors:
Su-Lin Wu, UC Berkeley & ICSI, (U.S.A.)
Brian E. D. Kingsbury, UC Berkeley & ICSI, (U.S.A.)
Nelson Morgan, UC Berkeley & ICSI, (U.S.A.)
Steven Greenberg, UC Berkeley & ICSI, (U.S.A.)

Abstract:
Including information distributed over intervals of syllabic duration (100-250 ms) may greatly improve the performance of automatic speech recognition (ASR) systems. ASR systems primarily use representations and recognition units covering phonetic durations (40-100 ms). Humans certainly use information at phonetic time scales, but results from psychoacoustics and psycholinguistics highlight the crucial role of the syllable, and syllable-length intervals, in speech perception. We compare the performance of three ASR systems: a baseline system that uses phone-scale representations and units, an experimental system that uses a syllable-oriented front-end representation and syllabic units for recognition, and a third system that combines the phone-scale and syllable-scale recognizers by merging and rescoring N-best lists. Using the combined recognition system, we observed an improvement in word error rate for telephone-bandwidth, continuous numbers from 6.8% to 5.5% on a clean test set, and from 27.8% to 19.6% on a reverberant test set, over the baseline phone-based system.
ICASSP98 Paper Abstract
Towards Speech Rate Independence in Large Vocabulary Continuous Speech Recognition

Authors:
Fernando Martínez, Telefónica Investigación y Desarrollo, (Spain)
Daniel Tapias, Telefónica Investigación y Desarrollo, (Spain)
Jorge Alvarez, Telefónica Investigación y Desarrollo, (Spain)

Abstract:
In this paper we present a new speech rate classifier (SRC) which is directly based on the dynamic coefficients of the feature vectors and it is suitable to be used in real time. We also report the study that has been carried out to determine what parameters of speech are the best regarding the speech rate classification problem. In this study we analyse the correlation between several speech parameters and the average speech rate of the utterance. Finally, we report a compensation technique which is used together with the SRC. This technique provides with a word error rate (WER) reduction of a 64.1% for slow speech rate and a 32% reduction of the average WER.
ICASSP98 Paper Abstract
Combining Multiple Estimators of Speaking Rate

Authors:
Nelson Morgan, ICSI/UC Berkeley, (U.S.A.)
Eric Fosler-Lussier, ICSI/UC Berkeley, (U.S.A.)

Volume 2, page 729, paper no. 2497

Abstract:
We report progress in the development of a measure of speaking rate that is computed from the acoustic signal. The newest form of our analysis incorporates multiple estimates of rate; besides the spectral moment for a full-band energy envelope that we have previously reported, we also used pointwise correlation between pairs of compressed sub-band energy envelopes. The complete measure, called mrate, has been compared to a reference syllable rate derived from a manually transcribed subset of the Switchboard database. The correlation with transcribed syllable rate is significantly higher than our earlier measure; estimates are typically within 1-2 syllables/second of the reference syllable rate. We conclude by assessing the use of mrate as a detector for rapid speech.
ICASSP98 Paper Abstract

A Recursive Feature Vector Normalization Approach for Robust Speech Recognition in Noise

Authors:
Olli Viikki, Nokia Research Center, (Finland)
David Bye, Nokia Mobile Phones, (Finland)
Kari Laurila, Nokia Research Center, (Finland)

Abstract:
The acoustic mismatch between testing and training conditions is known to severely degrade the performance of speech recognition systems. Segmental feature vector normalization (8) was found to improve the noise robustness of MFCC feature vectors and to outperform other state-of-the-art noise compensation techniques in speaker-dependent recognition. The objective of feature vector normalization is to provide environment-independent parameter statistics in all noise conditions. In this paper, we propose a more efficient implementation approach for feature vector normalization where the normalization coefficients are computed in a recursive way. Speaker-dependent recognition experiments show that the recursive normalization approach obtains over 60%, the segmental method approx. 50%, and Parallel Model Combination 14% overall error rate reduction, respectively. Moreover, in the recursive case, this performance gain is obtained with the smallest implementation costs. Also in speaker-independent connected digit recognition, over 16% error rate reduction is obtained with the proposed feature vector normalization approach.
ICASSP98 Paper Abstract

Some Solutions to the Missing Feature Problem in Data Classification, with Application to Noise-Robust ASR

Authors:
Andrew C Morris, University of Sheffield, ( U.K.)
Martin P Cooke, University of Sheffield, ( U.K.)
Phil D Green, University of Sheffield, ( U.K.)

Volume 2, page 737, paper no. 2024

Abstract:
We address the theoretical and practical issues involved in ASR when some of the observation data for the target signal is masked by other signals. Techniques discussed range from simple missing data imputation to Bayesian optimal classification. The Bayesian approach allows prior knowledge to be incorporated naturally into the recognition process, thereby permitting us to go beyond the simple "integrate over missing data" or "marginals" approach reported elsewhere, which we show to be inadequate for dealing with realistic patterns of missing data. These techniques are formulated in the context of an HMM based CSR system. This scheme is evaluated under both random and more realistic patterns of missing data, with speech from the DARPA RM corpus and noise from NOISEX. We find that a key problem in real world recognition with missing data is that efficient ASR requires data vector components to be independent, and incomplete data cannot be orthogonalised in the usual way by projection. We show that the use of spectral peaks only can provide an effective solution to this problem.
ICASSP98 Paper Abstract

A Study of Prior Sensitivity For Bayesian Predictive Classification Based Robust Speech Recognition

Authors:
Qiang Huo, University of Hong Kong, (Hong Kong)
Chin-Hui Lee, Bell Labs, (U.S.A.)

Volume 2, page 741, paper no. 1534

Abstract:
We previously introduced a new Bayesian predictive classification (BPC) approach to robust speech recognition and showed that BPC is capable of coping with many types of distortions. We also learned that the efficacy of the BPC algorithm is influenced by the appropriateness of the prior distribution for the mismatch being compensated. If the prior distribution fails to characterize the variability reflected in the model parameters, then the BPC will not help much. In this paper, we show how the knowledge and/or experience of the interaction between speech signal and the possible mismatch guide us to obtain a better prior distribution which improves the performance of the BPC approach.
ICASSP98 Paper Abstract

Magnitude-Only Estimation of Handset Nonlinearity with Application to Speaker Recognition

Authors:
Thomas F Quatieri, MIT (U.S.A.)
Douglas A. Reynolds, MIT (U.S.A.)
Gerald C O'Leary, MIT (U.S.A.)

Volume 2, page 745, paper no. 1087

Abstract:
A method is described for estimating telephone handset nonlinearity by matching the spectral magnitude of the distorted signal to the output of a nonlinear channel model, driven by an undistorted reference. This "magnitude-only" representation allows the model to directly match unwanted speech formants that arise over nonlinear channels and that are a potential source of degradation in speaker and speech recognition algorithms. As such, the method is particularly suited to algorithms that use only spectral magnitude information. The distortion model consists of a memoryless polynomial nonlinearity sandwiched between two finite-length linear filters. Minimization of a mean-squared spectral magnitude error, with respect to model parameters, relies on iterative estimation via a gradient descent technique, using a Jacobian in the iterative correction term with gradients calculated by finite-element approximation. Initial work has demonstrated the algorithm's usefulness in speaker recognition over telephone channels by reducing mismatch between high- and low-quality handset conditions.
ICASSP98 Paper Abstract

Non-Parametric Estimation and Correction of Non-Linear Distortion in Speech Systems

Authors:
Rajesh Balchandran, Rutgers University, (U.S.A.)
Richard J Mammone, Rutgers University, (U.S.A.)

Volume 2, page 749, paper no. 1400

Abstract:
The performance of speech systems such as speaker recognition degrades drastically when there is mismatch between training and testing conditions, caused by non-linear distortion. This paper describes a technique to estimate and correct such non-linear distortion in speech. The focus is on constrained restoration of degraded speech, that is distortion in the test speech is undone relative to the training speech. Restoration is a two step process - estimation followed by inversion. The non-linearity is estimated in the form of a look-up table by a process of statistical matching using a reference speech template. This statistical matching technique provides a very good estimate of the true non-linear characteristic, and the process is robust, computationally efficient, and universally applicable. Speaker-ID experiments, using artificially corrupted test speech, showed significant improvement in performance after the test speech was 'cleaned' using this technique. The restoration process itself does not introduce appreciable distortion.
ICASSP98 Paper Abstract

A Distance Measure between Collections of Distributions and its Application to Speaker Recognition

Authors:
Homayoon S.M Beigi, IBM Research, (U.S.A.)
Stephane H Maes, IBM Research, (U.S.A.)
Jeffrey S Sorensen, IBM Research, (U.S.A.)

Volume 2, page 753, paper no. 2520

Abstract:
This paper presents a distance measure for evaluating the closeness of two sets of distributions. The job of finding the distance between two distributions has been addressed with many solutions present in the literature. To cluster speakers using the pre-computed models of their speech, a need arises for computing a distance between these models which are normally built of a collection of distributions such as Gaussians. The definition of this distance measure creates many possibilities for speaker verification, speaker adaptation, speaker segmentation and many other related applications. A distance measure is presented for evaluating the closeness of a collection of distributions and several applications with some results are presented using this distance measure.
ICASSP98 Paper Abstract
Clustering Speakers by their Voices

Authors:
Alex Solomonoff, GTE/BBN, (U.S.A.)
Angela Mielke, GTE/BBN, (U.S.A.)
Michael Schmidt, GTE/BBN, (U.S.A.)
Herbert Gish, GTE/BBN, (U.S.A.)

Volume 2, page 757, paper no. 2122

Abstract:
The problem of clustering speakers by their voices is addressed. With the mushrooming of available speech data from television broadcasts to voice mail, automatic systems for archive retrieval, organizing and labeling by speaker are necessary. Clustering conversations by speaker is a solution to all three of the above tasks. Another application for speaker clustering is to group utterances together for speaker adaptation in speech recognition. Metrics based on purity and completeness of clusters are introduced. Next our approach to speaker clustering is described and finally experimental results on a subset of the Switchboard corpus are presented.
ICASSP98 Paper Abstract
GMM Based Speaker Identification Using Training-Time-Dependent Number of Mixtures

Authors:
Chakib Tadj, Ecole de Technologie Superieure, (Canada)
Pierre Dumouchel, Centre de Recherche Informatique de Montreal, (Canada)
Pierre Ouellet, Ecole de Technologie Superieure, (Canada)

Volume 2, page 761, paper no. 1193

Abstract:
In this paper, we present the study of the performance of our standard GMM speaker identification system in a limited amount of training data context. We explore the use of different mixture components for different speakers/models. Different approaches are presented: (a) a nonlinear transformation of speech duration vs. number of mixtures is proposed in order to set correctly the appropriate number of model mixtures for each speaker according to the available training data. (b) From exhaustive experiments, the appropriate linear transformation is deduced. The resulting transformation offers several advantages: (a) each speaker is well modelized (b) the performance is improved by more than 6% on the SPIDRE corpus and finally (c) the number of mixtures is reduced and thus leads to a faster system response.
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 27% identification rate improvement on TIMIT, and to 17% on NTIMIT.
ICASSP98 Paper Abstract
Feature Selection for a DTW-based Speaker Verification System

Authors:
Medha Pandit, University of Surrey, (U.K.)
Josef Kittler, University of Surrey, (U.K.)

Volume 2, page 769, paper no. 1995

Abstract:
Speaker verification systems, in general, require 20 to 30 features as input for satisfactory verification. We show that this feature set can be optimised by appropriately choosing proper feature subset from the input feature set. This paper proposes a technique for optimisation of the feature sets, in an Dynamic Time Warping based text-dependent speaker verification system, to improve false acceptance rate. The optimisation technique is based on l-r algorithm. The proposed scheme is applied to study cepstrum coefficients and their first order orthogonal polynomial coefficients. Experiments are conducted on two data bases: French and Spanish. The results indicate that with the optimised feature set the performance of the system may improve but it is never degraded. Moreover, the speed of verification is significantly increased.
ICASSP98 Paper Abstract

AHUMADA: A Large Speech Corpus in Spanish for Speaker Identification and Verification

Authors:
Javier Ortega-Garcia, Univ. Politecnica de Madrid, (Spain)
Joaquin Gonzalez-Rodriguez, Univ. Politecnica de Madrid, (Spain)
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Juan Jesus Diaz-Gomez, Servicio de Policia Judicial, (Spain)
Ramon Garcia-Jimenez, Servicio de Policia Judicial, (Spain)
Jose Lucena-Molina, Servicio de Policia Judicial, (Spain)
Jose Antonio G. Sanchez-Molero, Servicio de Policia Judicial, (Spain)

Volume 2, page 773, paper no. 1558

Abstract:
Speaker Recognition is a major task when security applications through speech input are needed. Regarding speaker identity, several factors of variability must be considered: a) Factors concerning peculiar intra-speaker variability (manner of speaking, inter-session variability, dialectal variations, emotional condition, etc.) or forced intra-speaker variability (Lombard effect, cocktail-party effect). b) Factors depending on external influences (kind of microphone, channel effects, noise, reverberation, etc). To cope with all these variability sources, a specific speech database called AHUMADA has been designed and collected for speaker recognition tasks in Castilian Spanish. AHUMADA incorporates six different recording sessions, including both in situ and telephone speech recordings. A total of 104 male speakers uttered isolated digits, digit strings, phonologically balanced short utterances, phonologically and syllabically balanced read text and more than one minute of spontaneous speech, so about 15 GB of speech material is available. Speaker verification results, concerning the available variability sources are also presented.
ICASSP98 Paper Abstract

Text-Prompted Speaker Verification Experiments with Phoneme Specific MLPs

Authors:
Dijana Petrovska-Delacretaz, CIRC-EPFL, (Switzerland)
Jean Hennebert, CIRC-EPFL, (Switzerland)

Volume 2, page 777, paper no. 2383

Abstract:
The aims of the study described in this paper are (1) to assess the relative speaker discriminant properties of phonemes and (2) to investigate the importance of the temporal frame-to-frame information for speaker modelling in the framework of a text-prompted speaker verification system using Hidden Markov Models (HMMs) and Multi Layer Perceptrons (MLPs). It is shown that, with similar experimental conditions, nasals, fricatives and vowels convey more speaker specific informations than plosives and liquids. Regarding the influence of the frame-to-frame temporal information, significant improvements are reported from the inclusion of several acoustic frames at the input of the MLPs. Results tend also to show that each phoneme has its optimal MLP context size giving the best Equal Error Rate (EER).
ICASSP98 Paper Abstract

An Efficient Phonotactic-Acoustic System for Language Identification

Authors:
Jiri Navratil, Technical University of Ilmenau, (Germany)
Werner Zuehlke, Technical University of Ilmenau, (Germany)

Volume 2, page 781, paper no. 1122

Abstract:
This paper presents a combined two-component system for language identification based on phonotactic and acoustic features. The phonotactic part consisting of a multilingual phone-recognizer with a double bigram-decoding architecture and a phonetic-context mapping is supported by a second part with pronunciation modeling of the recognized phone-sequence using Gaussian density models. Both parts are post-processed by a neural-based final classifier. Measured on the NIST’95 evaluation set, the described system outperforms state-of-the-art components and, at the same time, requires considerably less computational expense, as compared to implicit phonotactic-acoustic modeling and parallel recognizer architectures.
ICASSP98 Paper Abstract

Fast Robust Inverse Transform Speaker Adapted Training Using Diagonal Transformations

Authors:
Hubert Jin, BBN Technologies, (U.S.A.)
Spyros Matsoukas, Northeastern University, (U.S.A.)
Richard Schwartz, BBN Technologies, (U.S.A.)
Francis Kubala, BBN Technologies, (U.S.A.)

Volume 2, page 785, paper no. 2533

Abstract:
We present a new method of Speaker Adapted Training (SAT) that is more robust, faster, and results in lower error rate than the previous methods. The method, called Inverse Transform SAT (ITSAT) is based on removing the differences between speakers before training, rather than modeling the differences during training. We develop several methods to avoid the problems associated with inverting the transformation. In one method, we interpolate the transformation matrix with an identity or diagonal transformation. We also apply constraints to the matrix to avoid estimation problems. Finally, we show that the resulting method is much faster, requires much less disk space, and results in higher accuracy than the original SAT method.
ICASSP98 Paper Abstract

Instantaneous Environment Adaptation Techniques Based on Fast PMC and MAP-CMS Methods

Authors:
Tetsuo Kosaka, Canon Inc., (Japan)
Hiroki Yamamoto, Canon Inc., (Japan)
Masayuki Yamada, Canon Inc., (Japan)
Yasuhiro Komori, Canon Inc., (Japan)

Volume 2, page 789, paper no. 1454

Abstract:
This paper proposes instantaneous environment adaptation techniques for both additive noise and channel distortion based on the fast PMC (FPMC) and the MAP-CMS methods. The instantaneous adaptation techniques enable a recognizer to improve recognition on a single sentence that is used for the adaptation in real-time. The key innovations enabling the system to achieve the instantaneous adaptation are: 1) a cepstral mean subtraction method based on maximum a posteriori estimation (MAP-CMS), 2) real-time implementation of the fast PMC that we proposed previously, 3) utilization of multi-pass search, and 4) a new combination method of MAP-CMS and FPMC to solve the problem of both channel distortion and additive noise. Experiment results showed that the proposed methods enabled the system to perform recognition and adaptation simultaneously nearly in real-time and obtained good improvements in performance.
Unsupervised Adaptation Using Structural Bayes Approach

Authors:
Koichi Shinoda, NEC Corporation, (Japan)
Chin-Hui Lee, Bell Labs, Lucent Technologies, (U.S.A.)

Volume 2, page 793, paper no. 1761

Abstract:
It is well-known that the performance of recognition systems is often largely degraded when there is a mismatch between the training and testing environment. It is desirable to compensate for the mismatch when the system is in operation without any supervised learning. Recently, a structural maximum a posteriori (SMAP) adaptation approach, in which a hierarchical structure in the parameter space is assumed, was proposed. In this paper, this SMAP method is applied to unsupervised adaptation. A novel normalization technique is also introduced as a front end for the adaptation process. The recognition results showed that the proposed method was effective even when only one utterance from a new speaker was used for adaptation. Furthermore, an effective way to combine the supervised adaptation and the unsupervised adaptation was investigated to reduce the need for a large amount of supervised learning data.
ICASSP98 Paper Abstract

A Study on Speaker Normalization Using Vocal Tract Normalization and Speaker Adaptive Training

Authors:
Lutz Welling, University of Technology, Aachen, (Germany)
Reinhold Haeb-Umbach, Philips GmbH Forschungslaboratorien, (Germany)
Xavier Aubert, Philips GmbH Forschungslaboratorien, (Germany)
Nils Haberland, University of Technology, Aachen, (Germany)

Volume 2, page 797, paper no. 1978

Abstract:
Although speaker normalization is attempted in very different manners, vocal tract normalization (VTN) and speaker adaptive training (SAT) share many common properties. We show that both lead to more compact representations of the phonetically relevant variations of the training data and that both achieve improved error rate performance only if a complementary normalization or adaptation operation is conducted on the test data. Algorithms for fast test speaker enrollment are presented for both normalization methods: in the framework of SAT, a pre-transformation step is proposed, which alone, i.e. without subsequent unsupervised MLLR adaptation, reduces the error rate by almost 10% on the WSJ 5k test sets. For VTN, the use of a Gaussian mixture model makes obsolete a first recognition pass to obtain a preliminary transcription of the test utterance at hardly any loss in performance.
Decision Tree State Tying Based on Segmental Clustering for Acoustic Modeling

Abstract:
In this paper, a fast segmental clustering approach to decision tree tying based acoustic modeling is proposed for large vocabulary speech recognition. It is based on a two level clustering scheme for robust decision tree state clustering. This approach extends the conventional segmental K-means approach to phonetic decision tree state tying based acoustic modeling. It achieves high recognition performances while reducing the model training time from days to hours comparing to the approaches based on Baum-Welch training. Experimental results on standard Resource Management and Wall Street Journal tasks are presented which demonstrate the robustness and efficacy of this approach.
ICASSP98 Paper Abstract

Automatic Question Generation for Decision Tree Based State Tying

Authors:
Klaus Beulen, RWTH Aachen, University of Technology, (Germany)
Hermann Ney, RWTH Aachen, University of Technology, (Germany)

Abstract:
Decision tree based state tying uses so-called phonetic questions to assign triphone states to reasonable acoustic models. These phonetic questions are in fact phonetic categories such as vowels, plosives or fricatives. The assumption behind this is that context phonemes which belong to the same phonetic class have a similar influence on the pronunciation of a phoneme. For a new phoneme set, which has to be used e.g. when switching to a different corpus, a phonetic expert is needed to define proper phonetic questions. In this paper a new method is presented which automatically defines good phonetic questions for a phoneme set. This method uses the intermediate clusters from a phoneme clustering algorithm which are reduced to an appropriate number afterwards. Recognition results on the Wall Street Journal data for within-word and across-word phoneme models show competitive performance of the automatically generated questions with our best handcrafted question set.
ICASSP98 Paper Abstract
Scaled Random Segmental Models

Authors:
Jacob Goldberger, Tel Aviv University, (Israel)
David Burshtein, Tel Aviv University, (Israel)

Volume 2, page 809, paper no. 1030

Abstract:
We present the concept of a scaled random segmental model, which aims to overcome the modeling problem created by the fact that segment realizations of the same phonetic unit differ in length. In the scaled model the variance of the random mean trajectory is inversely proportional to the segment length. The scaled model enables a Baum-Welch type parameter reestimation, unlike the previously suggested, non-scaled models, that require more complicated iterative estimation procedures. In experiments we have conducted with phoneme classification, it was found that the scaled model shows improved performance compared to the non-scaled model.
ICASSP98 Paper Abstract
Factorial HMMS for Acoustic Modeling

Authors:
Beth Logan, University of Cambridge, (U.K.)
Pedro J Moreno, Digital Equipment Corporation, (U.S.A.)

Volume 2, page 813, paper no. 2453

Abstract:
Recently in the machine learning research field several extensions of hidden Markov models (HMMs) have been proposed. In this paper we study their possibilities and potential benefits for the field of acoustic modeling. We describe preliminary experiments using an alternative modeling approach knowns as factorial hidden Markov Models (FHMMs). We present these models as extensions of HMMs and detail a modification to the original formulation which seems to allow a more natural fit to speech. We present experimental results on the phonetically balanced TIMIT database comparing the performance of FHMMs with HMMs. We also study alternative feature representations that might be more suited to FHMMs.
ICASSP98 Paper Abstract

Improved Lexical Tree Search for Large Vocabulary Speech Recognition

Authors:
Stefan Ortmanns, RWTH-Aachen, (Germany)
Andreas Eiden, RWTH-Aachen, (Germany)
Hermann Ney, RWTH-Aachen, (Germany)

Volume 2, page 817, paper no. 1953

Abstract:
This paper describes some extensions to the language model (LM) look-ahead pruning approach which is integrated into the time-synchronous beam search algorithm. The search algorithm is based on a lexical prefix tree in combination with a word-conditioned dynamic search space organization for handling trigram language models in a one-pass strategy. In particular, we study several LM look-ahead pruning techniques. Further, we improve the efficiency of this look-ahead technique by exploiting subtree dominance. This method avoids the computation of redundant subtrees within the copies of the lexical prefix tree and thus reduces the memory requirements of the search algorithm. In addition, we present a pruning criterion depending on the state index. The experimental results on the 20000-word NAB'94 task (ARPA North American Business Corpus) indicate that the computational effort can be reduced to 3.3 times real time on a ALPHA 5000 PC without a significant loss in the recognition accuracy.
ICASSP98 Paper Abstract

Efficient Search with Posterior Probability Estimates in HMM-Based Speech Recognition

Authors:
Daniel Willett, Gerhard-Mercator-University Duisburg, (Germany)
Christoph Neukirchen, Gerhard-Mercator-University Duisburg, (Germany)
Gerhard Rigoll, Gerhard-Mercator-University Duisburg, (Germany)

Volume 2, page 821, paper no. 1736

Abstract:
In this paper we present the methods we developed to estimate posterior probabilities for HMM states in continuous and discrete HMM-based speech recognition systems and several ways to speed up decoding by using these posterior probability estimates. The proposed pruning techniques are State Deactivation Pruning (SDP), similar to an approach proposed for hybrid recognition systems, and a novel posteriori-based lookahead technique, Posteriori Lookahead Pruning (PLP), that evaluates future posteriors in order to exclude unlikely HMM states as early as possible during search. By applying the proposed methods we managed to vastly reduce the decoding time consumed by our time-synchronous Viterbi-decoder for recognition systems based on the Verbmobil and the Wall Street Journal database with hardly any additional search error.
ICASSP98 Paper Abstract

Improved Search Strategy for Large Vocabulary Continuous Mandarin Speech Recognition

Authors:
Tai-Hsuan Ho, National Taiwan University, (Taiwan)
Kae-Cherng Yang, National Taiwan University, (Taiwan)
Kuo-Hsun Huang, National Taiwan University, (Taiwan)
Lin-Shan Lee, Institute of Information Science, Academia Sinica, (Taiwan)

Volume 2, page 825, paper no. 2381

Abstract:
This paper presents a new search strategy for large vocabulary continuous Mandarin speech recognition considering the special structure of Chinese language. This strategy is composed of a forward and a backward passes, between which a high-quality syllable lattice is generated to bridge the syllable-level and word-level decoding processes. In the forward pass, considering the small number of syllables in Chinese language, a frame-synchronous stack decoder is used to integrate the high-order syllable N-Gram language model, so as to generate a very accurate and compact syllable lattice. In the backward pass, considering the special monosyllabic wording structure in Chinese language, the search space for the word-level decoding is expanded dynamically from the syllable lattice, and the best word sequence is extracted based on the knowledge provided by the word pronunciation lexicon and the word N-Gram language model. In the preliminary experiments, it was found that, with this strategy, the character error rate can be reduced by more than 20% as compared with a previous system using syllable-aligned lattice approach on a speaker-adaptive continuous speech recognition task.
ICASSP98 Paper Abstract
Time-First Search for Large Vocabulary Speech Recognition

Authors:
Tony Robinson, SoftSound, (U.K.)
James Christie, Cambridge University, (U.K.)

Volume 2, page 829, paper no. 1743

Abstract:
This paper describes a new search technique for large vocabulary speech recognition based on a stack decoder. Considerable memory savings are achieved with the combination of a tree based lexicon and a new search technique. The search proceeds time-first, that is partial path hypotheses are extended into the future in the inner loop and a tree walk over the lexicon is performed as an outer loop. Partial word hypotheses are grouped based on language model state. The stack maintains information about groups of hypotheses and whole groups are extended by one word to form new stack entries. An implementation is described of a one-pass decoder employing a 65,000 word lexicon and a disk-based trigram language model. Real time operation is achieved with a small search error, a search space of about 5 Mbyte and a total memory usage of about 35 Mbyte.
ICASSP98 Paper Abstract

Improving Vocabulary Independent HMM Decoding Results by Using the Dynamically Expanding Context

Authors:
Mikko Kurimo, Helsinki University of Technology, (Finland)

Volume 2, page 833, paper no. 1860

Abstract:
A method is presented to correct phoneme strings produced by a vocabulary independent speech recognizer. The method first extracts the N best matching result strings using mixture density hidden Markov models (HMMs) trained by neural networks. Then the strings are corrected by the rules generated automatically by the Dynamically Expanding Context (DEC). Finally, the corrected string candidates and the extra alternatives proposed by the DEC are ranked according to the likelihood score of the best HMM path to generate the obtained string. The experiments show that N need not be very large and the method is able to decrease recognition errors from a test data that even has no common words with the training data of the speech recognizer.
ICASSP98 Paper Abstract

Development of Robust Speech Recognition Middleware on Microprocessor

Authors:
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Hiroaki Kokubo, Central Research laboratory, Hitachi Ltd., (Japan)
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Akio Amano, Central Research laboratory, Hitachi Ltd., (Japan)

Volume 2, page 837, paper no. 2537

Abstract:
We have developed speech recognition middleware on a RISC microprocessor which has robust processing functions against environmental noise and speaker differences. The speech recognition middleware enables developers and users to use a speech recognition process for many possible speech applications, such as car navigation systems and handheld PCs. In this paper, we report implementation issues of speech recognition process in middleware of microprocessors and propose robust noise handling functions using ANC(Adaptive Noise Cancellation) and noise adaptive models. We also propose a new speaker adaptation algorithm, in which the relationships among HMMs(Hidden Markov Models) transfer vectors are provided as a set of pre-trained interpolation coefficients. Experimental evaluations on 1000-word vocabulary speech recognition showed promising results for both robust processing functions of the proposed noise handling methods and the proposed speaker adaptation method.
ICASSP98 Paper Abstract

Mandarin Telephone Speech Recognition for Automatic Telephone Number Directory Service

Authors:
Yih-Ru Wang, National Chaio-Tung University, (Taiwan)
Sin-Horng Chen, National Chaio-Tung University, (Taiwan)

Volume 2, page 841, paper no. 1353

Abstract:
This paper discusses an HMM-based Mandarin telephone speech recognition method for implementing a prototype system of automatic telephone number directory service. It adopted the GPD/MCE training algorithm to train the HMM models for 100 final-dependent syllable initials and 40 syllable finals. The SBR method was used to compensate the speaker and channel effects. Besides, an RNN-based pre-classification scheme was employed to speed up the recognition search. A syllable recognition rate of 53.7% was achieved. This method was then used to implement an isolated-word recognizer for the prototype system to discriminate 1922 names of bank and insurance companies. Word recognition rates of 94.8% for top-1 and 97.9% for top-3 were achieved.
ICASSP98 Paper Abstract

Design and Implementation of an Auto-attendant System for the T.U.C. Campus Using Speech Recognition

Authors:
Kostas Koumpis, Technical University of Crete, (Greece)
Vassilios Digalakis, Technical University of Crete, (Greece)
Hy Murveit, Nuance Communications, (U.S.A.)

Volume 2, page 845, paper no. 2163

Abstract:
We present an auto-attendant system, which is based on a statistical speech recognizer and has been developed for the Technical University of Crete (TUC) campus. Auto-attendants allow remote callers to reach a person of department by simply speaking an appropriate name. This is the first speech-recognition system in Greece operating in continuous speech and speaker-independent modes, and we describe our approaches for solving several special phenomena specific to the Greek language. The high recognition accuracy of the engine supports several hundred of names. Evaluation on our database yielded more than 97.5% name retrieval for a dictionary of 350 names of persons and services.
Name Dialing Using Final User Defined Vocabularies in Mobile (GSM & TACS) and Fixed Telephone Networks

Authors:
Jose M Elvira, Telefonica I + D, (Spain)
Juan C Torrecilla, Telefonica I + D, (Spain)

Volume 2, page 849, paper no. 1787

Abstract:
This work presents the results obtained on the evaluation of a new approach for generation of phonetic transcriptions for name dialing applications in different telephone networks and with temporal variations. In this kind of applications on-line construction of user vocabularies is mandatory. The proposed method allows adaptive selection of new transcriptions requiring much less speech utterances for system training than other approaches. The new approach is evaluated using data from different telephone networks (PSTN, GSM and TACS networks) and from different temporal utterances (recordings done in a period of two months).
ICASSP98 Paper Abstract
The RWTH Large Vocabulary Continuous Speech Recognition System

Authors:
Hermann Ney, RWTH Aachen, (Germany)
Lutz Welling, RWTH Aachen, (Germany)
Stefan Ortmanns, RWTH Aachen, (Germany)
Klaus Beulen, RWTH Aachen, (Germany)
Frank Wessel, RWTH Aachen, (Germany)

Volume 2, page 853, paper no. 2442

Abstract:
In this paper, we present an overview of the RWTH Aachen large vocabulary continuous speech recognizer. The recognizer is based on continuous density hidden Markov models and a time-synchronous left-to-right beam search strategy. Experimental results on the ARPA Wall Street Journal (WSJ) corpus verify the effects of several system components, namely linear discriminant analysis, vocal tract normalization, pronunciation lexicon and cross-word triphones, on the recognition performance.
Determining Polarity of Speech Signals Based on Gradient of Spurious Glottal Waveforms

Abstract:
Speech polarity is crucial in many speech processing fields. We present a novel method to determine polarity of speech signals from gradient of spurious glottal waveforms. We use the iterative adaptive LPC inverse filtering to cancel effect of vocal tract transfer function while maintaining the most properties of source excitation. Then we take the first-derivative (gradient component) of spurious glottal waveforms to capture the sharp gradient near the glottal closure instant. By using the gradient components of the spurious glottal waveforms, we detect speech polarity, i.e., the polarity of glottal waveforms, by finding whether the glottal closure instants are located above or below the zero-line. Furthermore, a frame-based decision technique is applied to get robust results. Experimental results with a wide variety of speech utterances reveal a high performance and the computation complexity is much more less than a previously proposed method.
ICASSP98 Paper Abstract

Efficient Representation of Short-time Phase Based on Group Delay

Authors:
Hideki Banno, Nara Institute of Science and Technology, (Japan)
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Satoshi Nakamura, Nara Institute of Science and Technology, (Japan)
Kiyohiro Shikano, Nara Institute of Science and Technology, (Japan)
Hideki Kawahara, Wakayama University, (Japan)

Volume 2, page 861, paper no. 1874

Abstract:
An efficient representation of short-time phase characteristics of speech sounds is proposed, based on recent findings which suggest perceptual importance of phase characteristics. Subjective tests indicated that the synthesized speech sounds by the proposed method are indistinguishable from the original speech sounds with a moderate data compression. The proposed representation uses lower-order coefficients of inverse Fourier transform of the group delay of speech. It also alleviates the voiced-unvoiced decision, which is an indispensable part in conventional speech coding algorithms. These features make our method potentially very useful in many applications like speech morphing.
ICASSP98 Paper Abstract

Polynomial Quasi-Harmonic Models for Speech Analysis and Synthesis

Authors:
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Olivier Cappé, ENST, (France)
Frederic Bimbot, IRISA, (France)

Volume 2, page 865, paper no. 2039

Abstract:
Harmonic plus noise models have been successfully applied to a broad range of speech processing applications, including, among others, low bit-rate speech coding, and speech restoration and transformation. In conventional methods, the frequencies, the relative phases and the amplitudes of the pitch-harmonic components are assumed to be piecewise constants over an analysis frame. This assumption is inadequate in segments where fast variations of these parameters may occur, e.g. phoneme-to-phoneme boundaries or speech onsets. In this contribution, a time-varying models of the pitch-harmonic parameter is presented; it is based on a basis expansion technique, consisting in representing the time-varying functions as a linear combination of fixed basis function. An estimation procedure for the parameters of this expansion is presented. Results are provided to demonstrate the effectiveness of this approach.
ICASSP98 Paper Abstract
Perceptual Relevance of Objectively Measured Descriptors for Speaker Characterization

Authors:
Burhan F Necioglu, Georgia Institute of Technology, (U.S.A.)
Mark A. Clements, Georgia Institute of Technology, (U.S.A.)
Thomas P Barnwell III, Georgia Institute of Technology, (U.S.A.)
Astrid Schmidt-Nielsen, Naval Research Laboratory, (U.S.A.)

Abstract:
Subjective testing of speaker recognizability is an intricate, time consuming and very expensive process, but using objectively measurable descriptors to augment the subjective speaker recognizability tests could result in increased efficiency and reliability. This paper describes our investigation into the relevancy of a set of objective descriptors to human perception of speaker identity through multidimensional scaling (MDS) of subjective speaker pair similarity judgments. The evaluated objective descriptors can achieve same/different detection error rates as low as 4.13% for male speaker pairs, and 8.17% for female speaker pairs, with only 3 seconds of speech. Five descriptors related to glottal, vocal tract and prosodic features were found to have significant correlations with the perceptual dimensions of the MDS solutions.
ICASSP98 Paper Abstract
The Spectral Relevance of Glottal-Pulse Parameters

Authors:
Raymond N.J. Veldhuis, IPO, (The Netherlands)

Abstract:
The paper analyses how variations of the parameters of the Liljencrants-Fant(LF) model of glottal flow influence the speech spectrum, in order to determine the spectral relevance of these parameters. The effects of small parameter variations are described analytically. This analysis also gives an indication to what extent the LF parameters can be estimated reliably from the speech spectrum. The effects of larger parameter variations are discussed with the help of figures. Results are presented for a number of sets of estimated glottal-pulse parameters that were taken from the literature. The conclusion is that the LF model, which, given the fundamental period, is a three-parameter model, actually operates as a one- or a two-parameter model.
ICASSP98 Paper Abstract
Speech Synthesis Using Warped Linear Prediction and Neural Networks

Authors:
Matti Karjalainen, Helsinki University of Technology, (Finland)
Toomas Altosaar, Helsinki University of Technology, (Finland)
Martti Vainio, University of Helsinki, (Finland)

Abstract:
A text-to-speech synthesis technique, based on warped linear prediction (WLP) and neural networks, is presented for high-quality individual sounding synthetic speech. Warped linear prediction is used as a speech production model with wide audio bandwidth yet with highly compressed control parameter data. An excitation codebook, inverse filtered from a target speaker's voice, is applied to obtain individual tone quality. A set of neural networks, specialized to yield synthesis control parameters from phonemic input in specific contexts, generate the detailed parametric controls of WLP. Neural nets are also used successfully to compute the prosodic parameters. We have applied this approach in prototyping highly improved text-to-speech synthesis for the Finnish language.
ICASSP98 Paper Abstract
Source-Filter Models for Time-Scale Pitch-Scale Modification of Speech

Authors:
Alex Acero, Microsoft Research, (U.S.A.)

Abstract:
This paper presents two time-scale pitch-scale modification techniques to be used in speech synthesis systems. They have been applied to Microsoft’s Whistler system, which is based on concatenative synthesis. Both methods are based on a source-filter model, one of them using LPC parameters and the other one using cepstral parameters. The proposed methods achieve high quality prosody modification, retain the characteristics of the donor speaker, allow for spectral manipulation (to reduce spectral discontinuities at unit boundaries), and yield compact acoustic inventories.
ICASSP98 Paper Abstract
Speaker-Specific Pitch Contour Modeling and Modification

Authors:
David T. Chappell, Duke University, (U.S.A.)
John H.L. Hansen, Duke University, (U.S.A.)

Abstract:
This paper describes new techniques for modeling and generating speaker-dependent pitch contours for sentences. Speech synthesis applications could generally benefit from such speaker-specific pitch contours. The proposed algorithms begin with an existing pitch contour for an utterance and use data from training utterances to modify the contour to be appropriate for a second speaker. One approach modifies the original pitch values to statistically match the desired speaker at each point in time. A second novel approach uses dynamic time warping (DTW) to select a new pitch contour from a pre-determined code book and time-align the chosen contour to the original sentence. Such contour mapping can transfer one speaker's natural pitch characteristics to another person's speech. Informal listener evaluations suggest that while shifting the frequency range of the original pitch contour yields some improvement, better results are obtained by applying DTW techniques to time warp the contour from an existing sentence produced by the desired speaker.
ICASSP98 Paper Abstract

Articulatory Synthesis of Formant Targeted Sounds with Parameters Derived from the Inverse Solution of Speech Production

Authors:
Zhenli Yu, Hanzhou University, (China)
P.C. Ching, Chinese University of Hong Kong, (Hong Kong)

Abstract:
A new approach to produce high fidelity speech sound by applying both the inverse solution of speech production and the pitch-synchronous articulatory synthesis techniques is presented. Given a formant trace target, the dynamic vocal-tract area function together with time variant VT length are estimated using an inverse solution of speech production. The improved Kelly-Lochbaum filter of the synthesizer, with multi-rate system sampling and dynamic scattering wave adjustment, is employed to deal with the variable VT length and VT area function. A distinguished feature of this method is that artificially specified formant traces can be precisely obtained. Experimental results show that the formant targets can be precisely matched by the synthetic sound. A potential application of this method for text-to-speech conversion is discussed.
ICASSP98 Paper Abstract
Corpus-Based Mandarin Speech Synthesis with Contextual Syllabic Units Based on Phonetic Properties

Authors:
Fu-chiang Chou, National Taiwan University, (Taiwan)
Chiu-yu Tseng, Institute of Linguistics, Academia Sinica, (Taiwan)

Volume 2, page 893, paper no. 1798

Abstract:
This paper describes an improved concatenative synthesis module for a Chinese text-to-speech system. The concatenated segments are on-line selected from a designed speech corpus that is precisely segmented with an improved version of HMM models. The selection criteria are the prosodic and contextual similarities between the units and the desire targets from the previous module of the TTS system. The TD-PSOLA modifies the prosodic parameters of the selected units, and three methods for unit concatenation are performed according to the types of the syllabic junctures. These types are classified with the knowledge from the phonetic observations of large amounts of speech data. The output speech is remarkably fluent and natural because the coarticulation effects cross syllabic boundaries are well modeled and less prosodic modification is needed for the TD-PSOLA.
ICASSP98 Paper Abstract

Serbo-Croatian LVCSR on the Dictation and Broadcast News Domain

Authors:
Peter Scheytt, University of Karlsruhe, (Germany)
Petra Geutner, University of Karlsruhe, (Germany)
Alex Waibel, University of Karlsruhe, (Germany)

Volume 2, page 897, paper no. 2307

Abstract:
This paper describes the development of a Serbo-Croatian dictation and broadcast news speech recognizer. The intention is to generate an automatic text transcription of a news show, which will be submitted to a multilingual Informedia database. We outline the complete system development process using the JRTk, beginning with data collection, design and training of parameters, tuning and evaluation. We report on general recognition techniques like segmentation, adaptation and language model interpolation, as well as language specific problems, e.g. high OOV rate due to inflected word forms. We show that even a low amount of acoustic training data, combined with Web based interpolated language models, is sufficient to build up a fairly reliable automatic news transcription system, which yields a performance of 36.0% WE.
ICASSP98 Paper Abstract

Transcription of Broadcast News - Some Recent Improvements to IBM’s LVCSR System

Authors:
Lazaros Polymenakos, IBM, (U.S.A.)
Peder Olsen, IBM, (U.S.A.)
Dimitri Kanevsky, IBM, (U.S.A.)
Ramesh A Gopinath, IBM, (U.S.A.)
Ponani S. Gopalakrishnan, IBM, (U.S.A.)
Scott S. Chen, IBM, (U.S.A.)

Volume 2, page 901, paper no. 2262

Abstract:
This paper describes extensions and improvements to IBM's large vocabulary continuous speech recognition (LVCSR) system for transcription of broadcast news. The recognizer uses an additional 35 hours of training data over the one used in the 1996 Hub4 evaluation (7). It includes a number of new features: optimal feature space for acoustic modeling (in training and/or testing), filler-word modeling, Bayesian Information Criterion (BIC) based segment clustering, an improved implementation of iterative MLLR and 4-gram language models. Results using the 1996 DARPA Hub4 evaluation data set are presented.
ICASSP98 Paper Abstract

The BBN Byblos 1997 Large Vocabulary Conversational Speech Recognition System

Authors:
George Zavaliagkos, GTE/BBN Technologies, (U.S.A.)
John McDonough, GTE/BBN Technologies, (U.S.A.)
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Manhung Siu, GTE/BBN Technologies, (U.S.A.)
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Volume 2, page 905, paper no. 1740

Abstract:
This paper presents the 1997 BBN Byblos Large Vocabulary Speech Recognition (LVCSR) system. We give an outline of the algorithms and procedures used to train the system, describe the recognizer configuration and present the major technological innovations that lead to performance improvements. The major testbed we present our results for is the Switchboard Corpus, where current word error rates vary from 27% to 34% depending on the test set. In addition, we present results on the CallHome Spanish and Arabic tests, where we demonstrate that technology developed on English Corpora is very much portable to other problems and languages.
ICASSP98 Paper Abstract
Experiments in Broadcast News Transcription

Authors:
Philip C. Woodland, Cambridge University, (U.K.)
Thomas Hain, Cambridge University, (U.K.)
Sue E Johnson, Cambridge University, (U.K.)
Thomas R. Niesler, Cambridge University, (U.K.)
Andreas Tuerk, Cambridge University, (U.K.)
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Volume 2, page 909, paper no. 1988

Abstract:
This paper presents the recent development of the HTK broadcast news transcription system. Previously we have used data type specific modeling based on adapted Wall Street Journal trained HMMs. However, we are now experimenting with data for which no manual pre-classification or segmentation is available and therefore automatic techniques are required and suitable acoustic modeling strategies adopted. An approach for automatic audio segmentation and classification is described and evaluated as well as extensions to our previous work on segment clustering. A number of recognition experiments are presented that compare data-type specific and non-specific models; differing amounts of training data; the use of gender-dependent modeling; and the effects of automatic data-type classification. It is shown that robust segmentation into a small number of audio types is possible and that models trained on a wide variety of data types can yield good performance.
ICASSP98 Paper Abstract

Speech Recognition Performance on a Voicemail Transcription Task

Authors:
Mukund Padmanabhan, IBM, (U.S.A.)
Ellen Eide, IBM, (U.S.A.)
Bhuvana Ramabhadran, IBM, (U.S.A.)
Ganesh N. Ramaswamy, IBM, (U.S.A.)
Lalit R. Bahl, IBM, (U.S.A.)

Abstract:
The paper describes the collection of a novel database of voicemail messages (telephone bandwidth large-vocabulary conversational speech) where the speech data represents interaction between a human and a machine, and is consequently quite different from existing databases (for eg. Switchboard or CallHome). We present an analysis of this data and several novel techniques to improve the recognition performance on it. In particular the use of a new discriminant measure for improving the acoustic models, an automated technique for cleaning up transcriptions to provide cleaner acoustic models, the use of compound words to model crossword coarticulation effects, the use of class language models, etc., are shown to improve the baseline recognition rate on the task.
ICASSP98 Paper Abstract

Transcribing Broadcast News with the 1997 ABBOT System

Authors:
Gary D Cook, Cambridge University, (U.K.)
Tony J Robinson, Cambridge University, (U.K.)

Abstract:
Recent DARPA CSR evaluations have focused on the transcription of broadcast news from both television and radio programmes. This is a challenging task because the data includes a variety of speaking styles and channel conditions. This paper describes the development of a connectionist-hidden Markov model (HMM) system, and the enhancements designed to improve performance on broadcast news data. Both multilayer perceptron (MLP) and recurrent neural network acoustic models have been investigated. We assess the effect of using gender-dependent acoustic models, and the impact on performance of varying both the number of parameters and the amount of training data used for acoustic modelling. The use of context-dependent phone models is described, and the effect of the number of context classes is investigated. We also describe a method for incorporating syllable boundary information during search. Results are reported on the 1997 DARPA Hub-4 development test set.
ICASSP98 Paper Abstract
Experiments in Automatic Meeting Transcription Using JRTk

Authors:
Hua Yu, Carnegie Mellon University, (U.S.A.)
Cortis Clark, Carnegie Mellon University, (U.S.A.)
Robert Malkin, Carnegie Mellon University, (U.S.A.)
Alex Waibel, Carnegie Mellon University, (U.S.A.)

Volume 2, page 921, paper no. 2293

Abstract:
In this paper we describe our early exploration of automatic recognition of conversational speech in meetings for use in automatic summarizers and browsers to produce meeting minutes effectively and rapidly. To achieve optimal performance we started from two different baseline English recognizers adapted to meeting conditions and tested resulting performance. The data were found to be highly disfluent (conversational human to human speech), noisy (due to lapel microphones and environment), and overlapped with background noise, resulting in error rates comparable so far to those on the CallHome-conversational database (40-50% WER). A meeting browser is presented that allows the user to search and skim through highlights from a meeting efficiently despite the recognition errors.
ICASSP98 Paper Abstract

Adaptive Vocabularies for Transcribing Multilingual Broadcast News

Authors:
Petra Geutner, University of Karlsruhe, (Germany)
Michael Finke, Carnegie Mellon University, (U.S.A.)
Peter Scheytt, Carnegie Mellon University, (U.S.A.)

Abstract:
One of the most prevailing problems of large-vocabulary speech recognition systems is the large number of out-of-vocabulary words. This is especially the case for automatically transcribing broadcast news in languages other than English, that have a large number of inflections and compound words. We introduce a set of techniques to decrease the number of out-of-vocabulary words during recognition by using linguistic knowledge about morphology and a two-pass recognition approach, where the first pass only serves to dynamically adapt the recognition dictionary to the speech segment to be recognized. A second recognition run is then carried out on the adapted vocabulary. With the proposed techniques we were able to reduce the OOV-rate by more than 40% thereby also improving recognition results by an absolute 5.8% from 64% word accuracy to 69.8%.
ICASSP98 Paper Abstract

A Variational Approach for Estimating Vocal Tract Shapes from the Speech Signal

Authors:
Yves Laprie, LORIA, (France)
Bruno Mathieu, LORIA, (France)

Volume 2, page 929, paper no. 1392

Abstract:
This paper presents a novel approach to recovering articulatory trajectories from the speech signal using a variational calculus method and Maeda’s articulatory model. The acoustic-to-articulatory mapping is generally assessed by a double criterion: the acoustic proximity of results to acoustic data and the smoothness of articulatory trajectories. Most of the existing methods are unable to exploit the two criteria simultaneously or at least at the same level. On the other hand, our variational calculus approach combines the two criteria simultaneously and ensures the global acoustic and articulatory consistency without further optimization. This method gives rise to an iterative process which optimizes a startup solution given by an improved lookup algorithm. Codebooks generated with an articulatory model show nonuniform sampling of the acoustic space due to nonlinearities of the acoustic-to-articulatory mapping. We therefore designed an improved lookup algorithm building realistic articulatory trajectories which are not necessarily defined throughout the speech signal.
ICASSP98 Paper Abstract

Speech Intelligibility in the Presence of Cross-Channel Spectral Asynchrony

Authors:
Takayuki Arai, International Computer Science Institute, (U.S.A.)
Steven Greenberg, International Computer Science Institute, (U.S.A.)

Volume 2, page 933, paper no. 2353

Abstract:
The spectrum of spoken sentences was partitioned into quarter-octave channels and the onset of each channel shifted in time relative to the others so as to desynchronize spectral information across the frequency axis. Human listeners are remarkably tolerant of cross-channel spectral asynchrony induced in this fashion. Speech intelligibility remains relatively unimpaired until the average asynchrony spans three or more phonetic segments. Such perceptual robustness is correlated with the magnitude of the low-frequency (3-6 Hz) modulation spectrum and thus highlights the importance of syllabic segmentation and analysis for robust processing of spoken language. High-frequency channels (>1.5 kHz) play a particularly important role when the spectral asynchrony is sufficiently large as to significantly reduce the power in the low-frequency modulation spectrum (analogous to acoustic reverberation) and may thereby account for the deterioration of speech intelligibility among the hearing impaired under conditions of acoustic interference (such as background noise and reverberation) characteristic of the real world.
ICASSP98 Paper Abstract

Acoustic Breathiness Measures in the Description of Pathologic Voices

Authors:
Matthias Fröhlich, *Drittes Physikalisches Institut Göttingen, (Germany)*
Dirk Michaelis, *Drittes Physikalisches Institut Göttingen, (Germany)*
Hans Werner Strube, *Drittes Physikalisches Institut Göttingen, (Germany)*

Abstract:
One important perceptual attribute of voice quality is breathiness. Since breathiness is generally regarded to be caused by glottal air leakage, acoustic measures related to breathiness may be used to distinguish between different physiological phonation conditions for pathological voices. Seven “breathiness features” described in the literature plus one self-developed measure (the glottal to noise excitation ratio, GNE) are compared for their distinguishing properties between different well-defined pathological phonation mechanisms. It is found that only GNE allows a distinction between all the pathological groups and both the normal and aphonic reference group. Furthermore, GNE is among the measures showing the most significant distinctions between the different pathologic phonation mechanism groups. Therefore GNE should be given preference over the other features in the independent assessment of glottal air leakage or “breathiness” for moderately or highly disturbed voices.
ICASSP98 Paper Abstract

Automatic Estimation of Formant and Voice Source Parameters Using a Subspace Based Algorithm

Authors:
Chang-Sheng Yang, Utsunomiya University, (Japan)
Hideki Kasuya, Utsunomiya University, (Japan)

Abstract:
An automatic method is proposed to estimate jointly formant and voice source parameters from a speech signal. A Rosenberg-Klatt model is used to approximate a voicing source waveform for voiced speech, whereas a white noise signal is assumed for the unvoiced. The vocal tract characteristic is represented by an IIR filter. The formant and anti-formant values are calculated from the IIR filter coefficients which are estimated by using the subspace-based system identification algorithm, while an exhaustive search procedure is applied to obtain the optimal source parameter values, where an error criterion is introduced in the frequency domain. An experiment has been performed to examine performance of the proposed method with natural speech. The results show that the source parameters such as open and closure instants estimated by the method is in good agreement with those defined on the electro-glottograph signals and the formant values estimated are also accurate.
ICASSP98 Paper Abstract
Estimating the Speaking Rate by Vowel Detection

Authors:
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Guenther Ruske, Technical University of Munich, (Germany)

Volume 2, page 945, paper no. 1189

Abstract:
We present a new feature-based method for estimating the speaking rate by detecting vowels in continuous speech. The features used are the modified loudness and the zerocrossing rate which are both calculated in the standard preprocessing unit of our speech recognition system. As vowels in general correspond to syllable nuclei, the feature-based vowel rate is comparable to an estimate of the lexically-based syllable rate. The vowel detector presented is tested on the spontaneously spoken German Verbmobil task and is evaluated using manually transcribed data. The lowest vowel error rate (including insertions) on the defined test set is 22.72% on average over all vowels. Additionally correlation coefficients between our estimates and reference rates are calculated. These coefficients reach up to 0.796 and therefore are comparable to those for lexically-based measures (like the phone rate) on other tasks. The accuracy is sufficient to use our measurement for speaking rate adaptation.
ICASSP98 Paper Abstract

A New Algorithm for Incorporating Acoustic Constraints into the Inverse Speech Problem

Authors:
James DeLucia, Center for Communications Research, (U.S.A.)
Fred Kochman, Center for Communications Research, (U.S.A.)

Volume 2, page 949, paper no. 5004

Abstract:
We describe a new noniterative algorithm that generates the unique area function determined by the vocal tract length, the lip radius, and the spectral pair consisting of poles of the transfer function and zeros of the input impedance function. Our analysis is restricted to the class of piecewise-constant area functions defined on an even number of equal length intervals. The resulting algorithm involves fewer floating point operations per evaluation than the analogous method of Paige and Zue [4]. A method which uses a corpus of X-ray data is discussed for setting the higher order unobservable pole/zero frequencies.
ICASSP98 Paper Abstract
Cascade Recursive Least Squares with Subsection Adaptation for AR Parameter Estimation

Authors:
Gaguk Zakaria, Virginia Tech, (U.S.A.)
Louis A Beex, Virginia Tech, (U.S.A.)

Volume 2, page 953, paper no. 1621

Abstract:
We propose the adaptive cascade recursive least squares (CRLS-SA) algorithm for the estimation of linear prediction, or AR model, coefficients. The CRLS-SA algorithm features low computational complexity since each section is adapted independently from the other sections. It is shown here that the CRLS-SA algorithm can yield AR coefficient estimates closer to the true values, for some known signals, than the widely used autocorrelation method. CRLS-SA converges faster to the true values of the model, which is critically important for estimation from short data records. While the computational effort of CRLS-SA is a factor of 3 to 4 higher than that for the autocorrelation method, the improvement in performance yields a viable alternative for a number of applications.
ICASSP98 Paper Abstract
Spectral Stability Based Event Localizing Temporal Decomposition

Authors:
Athaudage CR Nandasena, Advanced Institute of Science and Technology, (Japan)
Masato Akagi, Advanced Institute of Science and Technology, (Japan)

Volume 2, page 957, paper no. 1148

Abstract:
In this paper a new approach to temporal decomposition (TD) of speech, called “Spectral Stability Based Event Localizing Temporal Decomposition”, abbreviated SBEL-TD, is presented. The original method of TD proposed by Atal is known to have the drawbacks of high computational cost, and the instability of the number and locations of events [1]. In SBEL-TD, the event localization is performed based on a maximum spectral stability criterion. This overcomes the instability problem of events of the Atal’s method. Also, SBEL-TD avoids the use of the computationally costly singular value decomposition routine used in the Atal’s method, thus resulting in a computationally simpler algorithm of TD. Simulation results show that an average spectral distortion of about 1.5 dB can be achieved with LSF as the spectral parameter. Also, we have shown that the temporal pattern of the speech excitation parameters can also be well described using the SBEL-TD technique.
ICASSP98 Paper Abstract

An Acoustic-Phonetic Feature-Based System for the Automatic Recognition of Fricative Consonants

Authors:
Ahmed M. Abdelatty Ali, University of Pennsylvania, (U.S.A.)
Jan Van der Spiegel, University of Pennsylvania, (U.S.A.)
Paul Mueller, Corticon Inc., (U.S.A.)

Volume 2, page 961, paper no. 1086

Abstract:
In this paper, the acoustic-phonetic characteristics and the automatic recognition of the American English fricatives are investigated. The acoustic features that exist in the literature are evaluated and new features are proposed. To test the value of the extracted features, a knowledge-based acoustic-phonetic system for the automatic recognition of fricatives, in speaker independent continuous speech, is proposed. The system uses an auditory-based front-end processing and incorporates new algorithms for the extraction and manipulation of the acoustic-phonetic features that proved to be rich in their information content. Several features, which describe the relative amplitude, location of the most dominant peak, spectral shape and duration of unvoiced portion, are combined in the recognition process. Recognition accuracy of 95% for voicing detection and 93% for place of articulation detection are obtained for TIMIT database continuous speech of 22 speakers from 5 different dialect regions.
ICASSP98 Paper Abstract

On Second Order Statistics and Linear Estimation of Cepstral Coefficients

Authors:
Yariv Ephraim, George Mason University, (U.S.A.)
Mazin Rahim, AT&T Labs, (U.S.A.)

Abstract:
Explicit expressions for the second order statistics of cepstral components representing clean and noisy signal waveforms are derived. The noise is assumed additive to the signal, and the spectral components of each process are assumed statistically independent complex Gaussian random variables. The key result developed here is an explicit expression for the cross-covariance between the log-spectra of the clean and noisy signals. In the absence of noise, this expression is used to show that the covariance matrix of cepstral components representing a vector of N signal samples, approaches a fixed, signal independent, diagonal matrix at a rate of 1/N^2. In addition, the cross-covariance expression is used to develop an explicit linear minimum mean square error estimator for the clean cepstral components given noisy cepstral components. Recognition results on the ten English digits using the fixed covariance and linear estimator are presented.
ICASSP98 Paper Abstract

An Algorithm for Robust Signal Modelling in Speech Recognition

Authors:
Rivarol Vergin, CML Technologies, (Canada)

Volume 2, page 969, paper no. 1345

Abstract:
The most popular set of parameters used in recognition systems is the mel frequency cepstral coefficients. While giving generally good results, it remains that the filtering process, as used in the evaluation of these parameters, reduces the signal resolution in the frequency domain which can have some impact in discriminating between phonemes. This paper presents a new parameterization approach that preserves most of the characteristics of mel frequency cepstral coefficients while maintaining the initial frequency resolution obtained from the fast Fourier transform. It is shown, by results obtained, that this technique can significantly increase the performance of a recognition system.
Automatic Speech Recognition Based on Cepstral Coefficients and a MEL-Based Discrete Energy Operator

Authors:
Hesham Tolba, INRS-Telecommunications, (Canada)
Douglas O'Shaughnessy, INRS-Telecommunications, (Canada)

Abstract:
In this paper, a novel feature vector based on both Mel Frequency Cepstral Coefficients (MFCCs) and a Mel-based onlinear Discrete-time Energy Operator (MDEO) is proposed to be used as the input of an HMM-based Automatic Continuous Speech Recognition (ACSR) system. Our goal is to improve the performance of such a recognizer using the new feature vector. Experiments show that the use of the new feature vector increases the recognition rate of the ACSR system. The HTK Hidden Markov Model Toolkit was used throughout. Experiments were done on both the TIMIT and NTIMIT databases. For the TIMIT database, when the MDEO was included in the feature vector to test a multi-speaker ACSR system, we found that the error rate decreased by about 9.51%. On the other hand, for NTIMIT, the MDEO deteriorates the performance of the recognizer. That is, the new feature vector is useful for clean speech but not for telephone speech.
ICASSP98 Paper Abstract

Compression of Acoustic Features for Speech Recognition in Network Environments

Authors:
Ganesh N. Ramaswamy, IBM, (U.S.A.)
Ponani S. Gopalakrishnan, IBM, (U.S.A.)

Volume 2, page 977, paper no. 1619

Abstract:
In this paper, we describe a new compression algorithm for encoding acoustic features used in typical speech recognition systems. The proposed algorithm uses a combination of simple techniques, such as linear prediction and multi-stage vector quantization, and the current version of the algorithm encodes the acoustic features at a fixed-rate of 4.0 Kbit/s. The compression algorithm can be used very effectively for speech recognition in network environments, such as those employing a client-server model, or to reduce storage in general speech recognition applications. The algorithm has also been tuned for practical implementations, so that the computational complexity and memory requirements are modest. We have successfully tested the compression algorithm against many test sets from several different languages, and the algorithm performed very well, with no significant change in the recognition accuracy due to compression.
ICASSP98 Paper Abstract

Speaker Clustering for Speech Recognition Using the Parameters Characterizing Vocal-Tract Dimensions

Authors:
Masaki Naito, ATR-ITL, (Japan)
Li Deng, ATR-ITL, (Japan)
Yoshinori Sagisaka, ATR-ITL, (Japan)

Volume 2, page 981, paper no. 1889

Abstract:
We propose speaker clustering methods based on the vocal-tract-size related articulatory parameters associated with individual speakers. Two parameters characterizing gross vocal-tract dimensions are first derived from formants of speaker-specific Japanese vowels, and are then used to cluster a total of 148 male Japanese speakers. The resultant speaker clusters are found to be significantly different from the speaker clusters obtained by conventional acoustic criteria. Japanese phoneme recognition experiments are carried out using speaker-clustered tied-state HMMs (HM Nets) trained for each cluster. Compared with the baseline gender dependent model, 5.7% of recognition error reduction has been achieved based on the clustering method using vocal-tract parameters.
Baby Ears: A Recognition System for Affective Vocalizations

Authors:
Malcolm G. Slaney, Interval Research Corporation, (U.S.A.)
Gerald McRoberts, Lehigh University, (U.S.A.)

Abstract:
We collected more than 500 utterances from adults talking to their infants. We automatically classified 65% of the strongest utterances correctly as approval, attentional bids, or prohibition. We used several pitch and formant measures, and a multidimensional Gaussian mixture-model discriminator to perform this task. As previous studies have shown, changes in pitch are an important cue for affective messages; we found that timbre or cepstral coefficients are also important. The utterances of female speakers, in this test, were easier to classify than were those of male speakers. We hope this research will allow us to build machines that sense the "emotional state" of a user.
ICASSP98 Paper Abstract

Quantization of Cepstral Parameters for Speech Recognition over the World Wide Web

Authors:
Vassilios Digalakis, Technical University of Crete, (Greece)
Leonardo G Neumeyer, SRI International, (U.S.A.)
Manolis Perakakis, Technical University of Crete, (Greece)

Abstract:
We examine alternative architectures for a client-server model of speech-enabled applications over the World Wide Web. We compare a server-only processing model, where the client encodes and transmits the speech signal to the server, to a model where the recognition front end, implemented as a Java applet, runs locally at the client and encodes and transmits the cepstral coefficients to the recognition server over the Internet. We follow a novel encoding paradigm, trying to maximize recognition performance instead of perceptual reproduction, and we find that by transmitting the cepstral coefficients we can achieve significantly higher recognition performance at a fraction of the bit rate required when encoding the speech signal directly.
ICASSP98 Paper Abstract

Speech Enhancement Using Spectral Envelope Side Information

Authors:
Richard J Barron, MIT, (U.S.A.)
Charles K Sestok, MIT, (U.S.A.)
Alan V. Oppenheim, MIT, (U.S.A.)

Volume 2, page 993, paper no. 2156

Abstract:
This paper proposes several methods for noise reduction using deterministic side information about the desired signal as a constraint on the reconstruction. Two forms of side information are considered separately: short-time linear predictive coefficients, and short-time zero-phase impulse response coefficients. We derive general expressions for the ML, MAP and MMSE estimators, and develop algorithms that yield the ML estimators with the above side information for speech corrupted by additive white Gaussian noise. We also explore the use of these methods in the traditional noise reduction problem with no side information.
ICASSP98 Paper Abstract

Kalman Filtering of Colored Noise for Speech Enhancement

Authors:
Dimitrie C Popescu, Rutgers University, (U.S.A.)
Ilija Zeljkovic, AT&T Labs, (U.S.A.)

Abstract:
A method for applying Kalman filtering to speech signals corrupted by colored noise is presented. Both speech and colored noise are modeled as autoregressive (AR) processes using speech and silence regions determined by an automatic end-point detector. Due to the non-stationary nature of the speech signal, non-stationary Kalman filter is used. Experiments indicate that non-stationary Kalman filtering outperforms the stationary case, the average SNR improvement increasing from 0.53 dB to 2.3 dB. Even better results are obtained if noise is considered also non-stationary, in addition to being colored, achieving an average of 7.14 dB SNR improvement.
ICASSP98 Paper Abstract
Noise Reduction by Paired-Microphones Using Spectral Subtraction

Authors:
Mitsunori Mizumachi, Advanced Institute of Science and Technology, (Japan)
Masato Akagi, Advanced Institute of Science and Technology, (Japan)

Volume 2, page 1001, paper no. 1333

Abstract:
This paper proposes a method of noise reduction by paired microphones as a front-end processor for speech recognition systems. This method estimates noises using a subtractive microphone array and subtracts them from the noisy speech signal using the Spectral Subtraction (SS). Since this method can estimate noises analytically and frame by frame, it is easy to estimate noises not depending on these acoustic properties. Therefore, this method can also reduce non stationary noises, for example sudden noises when a door has just closed, which can not be reduced by other SS methods. The results of computer simulations and experiments in a real environment show that this method can reduce LPC log spectral envelope distortions.
ICASSP98 Paper Abstract

Fusion of Auditory and Visual Information for Noisy Speech Enhancement: A Preliminary Study of Vowel Transitions

Authors:
Laurent Girin, ICP, Grenoble, (France)
Gang Feng, ICP, Grenoble, (France)
Jean-Luc Schwartz, ICP, Grenoble, (France)

Volume 2, page 1005, paper no. 1785

Abstract:
This paper deals with a noisy speech enhancement technique based on the fusion of auditory and visual information. We first present the global structure of the system, and then we focus on the tool we used to melt both sources of information. The whole noise reduction system is implemented in the context of vowel transitions corrupted with white noise. A complete evaluation of the system in this context is presented, including distance measures, gaussian classification scores, and a perceptive test. The results are very promising.
ICASSP98 Paper Abstract

Dynamic Adjustment of the Forgetting Factor in Adaptive Filters for Non-Stationary Noise Cancellation in Speech

Authors:
Rafael Martinez, Universidad Politecnica de Madrid, (Spain)
Pedro Gomez, Universidad Politecnica de Madrid, (Spain)
Agustin Alvarez, Universidad Politecnica de Madrid, (Spain)
Victor Nieto, Universidad Politecnica de Madrid, (Spain)
Victoria Rodellar, Universidad Politecnica de Madrid, (Spain)
Manuel Rubio, Universidad Politecnica de Madrid, (Spain)
Mercedes Perez, Universidad Politecnica de Madrid, (Spain)

Volume 2, page 1009, paper no. 1957

Abstract:
Speech Recognition in Noisy Environments is critical for applications in the domain of Communications, Automation, Avionics, etc., where common recognizers fail to meet acceptable standards due to the noise picked during the recording of the Speech Trace. In this case Adaptive Filtering has been traditionally used with acceptable success in noise cancellation using two-microphone schemes. Some problems related with this technique are due to the non-stationary behavior of Speech and Noise, as the sudden changes in their relative energy levels are the cause of misadjustments and un-locking in the Adaptive Algorithms. A method based on the dynamic adjustment of the adaptation factor by a 3-state automaton driven by the continuous tracking of the Energy Differences between Speech and Noise is presented here. The paper discusses the proposed method and presents some results from practical conditions. These show good stability and a special ability for Word-Boundary Detection under Highly Non-Stationary Conditions.
ICASSP98 Paper Abstract

Markov Chain Monte Carlo Methods for Speech Enhancement

Authors:
Jaco Vermaak, Cambridge University, (U.K.)
Mahesan Niranjan, Cambridge University, (U.K.)

Volume 2, page 1013, paper no. 1993

Abstract:
This paper investigates a Bayesian approach to the enhancement of speech signals corrupted by additive white Gaussian noise. Parametric models for the speech and noise processes are constructed, leading to a posterior distribution for the model parameters and uncorrupted speech samples given the observed noisy speech samples. Being analytically intractable, inferences concerning these variables are performed using Markov Chain Monte Carlo (MCMC) methods. The efficiency of the sampling scheme within this framework is further improved by employing state space techniques based on the Kalman filter.
ICASSP98 Paper Abstract

Speech Enhancement in Noise and within Face Mask (Microphone Array Approach)

Authors:
George S Kang, Naval Research Laboratory, (U.S.A.)
Thomas M Moran, Naval Research Laboratory, (U.S.A.)

Volume 2, page 1017, paper no. 1032

Abstract:
In certain communication environments, digital speech transmission systems must work in severe acoustic environments where the noise levels exceed 110 dB. In other environments, speakers must use an oxygen face mask. In both situations, the intelligibility of encoded speech falls below an acceptable level. We have developed a technique for improving speech quality in these situations. Previous speech improvement methods have focused on processing the corrupted signal after it has been induced by the microphone. These methods have not performed adequately. In our technique, speech anomalies are attenuated by a microphone array before speech and noise become mixed into a signal. Our microphone array prototype has shown excellent performance. In an example of speech taken aboard an E2C aircraft, this noise-canceling microphone array improved the speech-to-noise ratio by as much as 18 dB. When the same technique is used in a face mask, muffled speech was almost completely restored to high quality speech.
ICASSP98 Paper Abstract
Voicing State Determination of Co-Channel Speech

Authors:
Daniel S Benincasa, Air Force Research Laboratory, (U.S.A.)
Michael I Savic, Rensselaer Polytechnic Institute, (U.S.A.)

Volume 2, page 1021, paper no. 2045

Abstract:
This paper presents a voicing state determination algorithm (VSDA) that is used to simultaneously estimate the voicing state of two speakers present in a segment of co-channel speech. Supervised learning trains a Bayesian classifier to predict the voicing states. The possible voicing states are silence, voiced/voiced, voiced/unvoiced, unvoiced/voiced and unvoiced/unvoiced. We have assumed the silent state as a subset of the unvoiced class, except when both speakers are silent. We have chosen a binary tree decision structure. Our feature set is a projection of a 37 dimensional feature vector onto a single dimension applied at each branch of the decision tree, using the Fisher linear discriminant. We have produced co-channel speech from the TIMIT database which is used for training and testing. Preliminary results, at signal to interference ratio of 0 dB, have produced classification accuracy of 82.6%, 73.45%, and 68.24% on male/female, male/male and female/female mixtures respectively.
ICASSP98 Paper Abstract

Improvements on Co-Channel Speech Separation using ADF: Low Complexity, Fast Convergence, and Generalization

Authors:
Kuan-Chieh Yen, University of Illinois, Urbana-Champaign, (U.S.A.)
Yunxin Zhao, University of Illinois, Urbana-Champaign, (U.S.A.)

Volume 2, page 1025, paper no. 2194

Abstract:
Three modifications on the adaptive decorrelation filtering (ADF) algorithm are proposed to improve the performance of a co-channel speech separation system. Firstly, a simplified ADF (SADF) is suggested to reduce the computational complexity of ADF from $O(N^2)$ to $O(N)$ per sample, where $N$ is the filter length used in the channel estimation. Secondly, a transform-domain ADF (TDADF) is developed to accelerate the convergence of the filter estimates while maintaining computational complexity at $O(N)$. Thirdly, a generalized ADF (GADF) is derived to handle the noncausal filter estimation problem often encountered in co-channel speech separation. Experimental results showed that when the average signal-to-interferenceratios (SIRs) in the co-channel signals were 6.15 and 5.38 dB, respectively, both the SADF and TDADF improved the SIRs to around 18 to 19 dB, and the GADF further improved the SIRs to around 19 to 24 dB.
Abstract:
We present a computationally efficient method of separating mixed speech signals. The method uses a recursive adaptive gradient descent technique with the cost function designed to maximize the kurtosis of the output (separated) signals. The choice of kurtosis maximization as an objective function (which acts as a measure of separation) is supported by investigation and analysis of a class of random processes which are regarded as excellent models for speech signal statistics. Such processes are identified as spherically invariant random processes (SIRP's). Development and analysis of the adaptive algorithm is presented. Simulation examples using actual voice signals are presented.
ICASSP98 Paper Abstract

Machine Learning and Automatic Linguistic Analysis: The Next Step

Authors:
Eric Brill, Johns Hopkins University, (U.S.A.)

Volume 2, page 1033, paper no. 5260

Abstract:
In order to continue building systems with progressively more complex natural language capabilities, it is crucial that great strides are made toward solving the core linguistic analysis problems for complex and possibly unrestricted domains. A great deal of progress has been made by applying machine learning techniques to automatically train systems from manually annotated corpora to provide detailed linguistic analyses to sentences. This paper examines a number of issues within this paradigm of automatic linguistic knowledge acquisition and how they relate to pushing progress in the field of natural language processing over the next decade.
ICASSP98 Paper Abstract

Accessible Technology for Interactive Systems: A New Approach to Spoken Language Research

Authors:
Ronald A. Cole, Oregon Graduate Institute of Science and Technology, (U.S.A.)
Stephen Sutton, Oregon Graduate Institute of Science and Technology, (U.S.A.)
Yonghong Yan, Oregon Graduate Institute of Science and Technology, (U.S.A.)
Pieter Vermeulen, Oregon Graduate Institute of Science and Technology, (U.S.A.)
Mark Fanty, Oregon Graduate Institute of Science and Technology, (U.S.A.)

Abstract:
In this paper, we argue for a paradigm shift in spoken language technology, from transcription tasks to interactive systems. The current paradigm evaluates speech recognition accuracy on large vocabulary transcription tasks, such as telephone conversations or media broadcasts. Systems are evaluated in international competitions, with strict rules for participation and well-defined evaluation metrics. Participation in these competitions is limited to a few elite laboratories that have the resources to develop and field systems. We propose a new, more productive and more accessible paradigm for spoken language research, in which research advances are evaluated in the context of interactive systems that allow people to perform useful tasks, such as accessing information from the World Wide Web, while driving a car. These systems are made available for daily use by ordinary citizens through telephone networks or placement in easily accessible kiosks in public institutions. It is argued [1,2,3] that this new paradigm, which focuses on the goal of universal access to information for all people, better serves the needs of the research community, as well as the welfare of our citizens. We discuss the challenges and rewards of an interactive system approach to spoken language research, and discuss our initial attempts to stimulate a paradigm shift and engage a large community of researchers through free distribution of the CSLU Toolkit.
Recognition in a New Key - Towards a Science of Spoken Language

Authors:
Steven Greenberg, *International Computer Science Institute, (U.S.A.)*

Abstract:
Automatic speech recognition in the twenty-first century will strive to emulate many properties of human speech understanding that currently lie beyond the capability of present-day systems. Such future-generation recognition will require massive amounts of empirical data in order to derive the organizational principles underlying the generation and decoding of spoken language. Such data can be efficiently collected through systematic computational experimentation designed to identify the important building blocks of speech and delineate the nature of the structural interactions among linguistic tiers associated with the extraction of semantic information.
ICASSP98 Paper Abstract

The Challenge of Domain-Independent Speech Understanding

Authors:
Robert C. Moore, SRI International, (U.S.A.)

Volume 2, page 1045, paper no. 5263

Abstract:
To achieve widespread acceptance, speech understanding technology needs to be domain independent. Deep understanding, however, appears to require knowledge that is domain specific. Speech understanding technology, therefore, must be partitioned into domain-independent and domain-specific components. Development of domain-independent components could be promoted by creation of semantically annotated corpora. Any such corpus, however, would be difficult to produce and would necessarily be controversial because of lack of widespread agreement on principles of semantic analysis. The use of such a corpus for performance evaluation should therefore be left largely up to the research community rather than being imposed by funding agencies.
ICASSP98 Paper Abstract
Understanding Speech Understanding

Authors:
R. K. Moore, DERA Speech Research Unit, (U.K.)

Volume 2, page 1049, paper no. 5264

Abstract:
Despite the significant theoretical and practical advances that have been made in automatic speech recognition in recent years, relatively little effort has been devoted to the evaluation of speech in an interactive multi-modal application interface. This paper introduces a general methodology for assessing speech-based systems and concludes with a proposal for a test scenario which focuses on the understanding component of a spoken language system.
Evaluating Dialog Systems Used in the Real World

Authors:
Harald Aust, Philips Speech Processing, (Germany)
Hermann Ney, RWTH Aachen, University of Technology, (Germany)

Abstract:
An important aspect of creating high performance natural language dialog systems is the question of how they are evaluated. While a universally accepted method for doing so for pure speech recognition exists, this is not clear for speech understanding or dialog systems. We describe the methods we typically use for our systems and argue that it is not sufficient to evaluate their constituents separately. Instead, a measure for a system in its entity is needed.
ICASSP98 Paper Abstract

Next Major Application Systems and Key Techniques in Speech Recognition Technology

Authors:
Kazuyo Tanaka, Electrotechnical Laboratory, (Japan)

Abstract:
In this paper, we discuss several major speech recognition applications which will contribute to some human activities in a decade. At first, recent Japanese speech-related national projects directed toward future intelligent systems are briefly reviewed. The we discuss three systems as the next major speech applications: substantially robust systems, multimodal interaction systems and multilingual dialogue systems. Evaluation of the performance of these systems is separately discussed in view of both total systems and specific technologies. We suggest that the degree of the difficulty of some kinds of specific tasks can be even more precisely measured, while the total system performance evaluation will become more difficult in future complex systems. Last, we take up phrase spotting, distance calculation for phonetic symbol sequences, adaptation/learning, and software modularization/multi-agents as the key techniques in constructing the above applications.
ICASSP98 Paper Abstract

Hidden Markov Modelling for SAR Automatic Target Recognition

Authors:
Chanin - Nilubol, Georgia Institute of Technology, (U.S.A.)
Quoc H. Pham, Georgia Institute of Technology, (U.S.A.)
Russell M. Mersereau, Georgia Institute of Technology, (U.S.A.)
Mark J. T. Smith, Georgia Institute of Technology, (U.S.A.)
Mark A. Clements, Georgia Institute of Technology, (U.S.A.)

Volume 2, page 1061, paper no. 1489

Abstract:
This paper discusses the application of Hidden Markov Models (HMMs) to solve the Translational and Rotational Invariant Automatic Target Recognition (TRIATR) problem associated with SAR imagery. This approach is based on a cascade of these stages: preprocessing, feature extraction and selection, and classification. Preprocessing and feature extraction and selection involve successive applications of extraction operations from measurements of the Radon transform of target chips. The features which are invariant to changes in rotation, position and shifts, although not to changes in scale are optimized through the use of feature selection techniques. The classification stage successively takes as its inputs the multidimensional multiple observation sequences, parameterizes them statistically using continuous density models to capture target and background appearance variability, and thus results in the TRIATR-HMMs. Experimental results have demonstrated that the recognition rate is as high as 99% over both the training set and the testing set.
ICASSP98 Paper Abstract

A Background-thinning Based Algorithm for Separating Connected Handwritten Digit Strings

Authors:
Zhongkang Lu, The Hong Kong Polytechnic University, (Hong Kong)
Zheru Chi, The Hong Kong Polytechnic University, (Hong Kong)
Pengfei Shi, Shanghai Jiaotong University, (China)

Volume 2, page 1065, paper no. 1535

Abstract:
Most algorithms for segmenting connected handwritten digit strings are based on the analysis of the foreground pixel distributions and the features on the upper/lower contours of the image. In this paper, a new approach is presented to segment connected handwritten two-digit strings based on the thinning of background regions. The algorithm first locates several feature points on the background skeleton of the digit image. Possible segmentation paths are then constructed by matching these feature points. With geometric property measures, these segmentation paths are ranked using fuzzy rules generated from a decision-tree approach. Finally, the top ranked segmentation paths are tested one by one by an optimized nearest neighbour classifier until one of these candidates is accepted based on an acceptance criterion. Experimental results on NIST special database 3 show that our approach can achieve a correct classification rate of 92.4% with only 4.7% of digit strings rejected, which compares favorably with the other techniques tested.
ICASSP98 Paper Abstract

A Novel Learning Method by Structural Reduction of DAGS for On-Line OCR Applications

Authors:
I-Jong Lin, Princeton University, (U.S.A.)
Sun-Yuan Kung, Princeton University, (U.S.A.)

Volume 2, page 1069, paper no. 1902

Abstract:
This paper introduces a learning algorithm for a neural structure, Directed Acyclic Graphs (DAGs) that is structurally based, i.e., reduction and manipulation of internal structure are directly linked to learning. This paper extends the concepts of DAG template matching to a neural structure with capabilities for generalization. DAG-Learning is derived from concepts in Finite State Transducers, Hidden Markov Models, and Dynamic Time Warping to form an algorithmic framework within which many adaptive signal techniques such as Vector Quantization, K-Means, Approximation Networks, etc., may be extended to temporal recognition. The paper provides a concept of path-based learning to allow comparison among Hidden Markov Models (HMMs), Finite State Transducers (FSTs) and DAG-Learning. The paper also outlines the DAG-Learning process and provides results from the DAG-Learning algorithm over a test set of isolated cursive handwriting characters.
ICASSP98 Paper Abstract

A Comparison of Ligature and Contextual Models for Hidden Markov Model Based On-Line Handwriting Recognition

Authors:
J.G.A. Dolfing, Philips GmbH Forschungslaboratorien, (Germany)

Volume 2, page 1073, paper no. 1937

Abstract:
This paper addresses the problem of on-line, writer-independent, unconstrained handwriting recognition. Based on hidden Markov models (HMM), we focus on the construction and use of word models which are robust towards contextual character shape variations and variations due to ligatures and diacritics with the objective of an improved word error rate. We compare the performance and complexity of contextual hidden Markov models with a ‘pause’ model for ligatures. While the common contextual models lead to a word error rate reduction of 12.7%-38% at the cost of almost six times more character models, the pause model improves the word error rate by 15%-25% and adds only a single model to the recognition system. The results for a mixed-style word recognition task on two test sets with vocabularies of 200 (up to 98% correct words) and 20,000 words (up to 88.6% correct words) are given.
ICASSP98 Paper Abstract
FANN-Based Video Chrominance Subsampling

Authors:
Adriana Dumitras, University of British Columbia, Vancouver, (Canada)
Faouzi Kossentini, University of British Columbia, Vancouver, (Canada)

Volume 2, page 1077, paper no. 2394

Abstract:
In this paper, we present a video chrominance subsampling method using feedforward neural networks. Experimental results show that our method outperforms spatial subsampling obtained via lowpass filtering and decimation both objectively and subjectively. Other advantages of our algorithm are computational efficiency and low memory requirements. Moreover, no pre- or post-processing is required by our method.
ICASSP98 Paper Abstract

Spatial and Temporal Stability of Vision Chips Including Parasitic Inductances and Capacitances

Authors:
Haruo Kobayashi, Gunma University, (Japan)
Takashi Matsumoto, Waseda University, (Japan)

Volume 2, page 1081, paper no. 5123

Abstract:
There are two dynamics issues in vision chips: (i) The temporal dynamics issue due to the parasitic capacitors in a CMOS chip, and (ii) the spatial dynamics issue due to the regular array of processing elements in a chip. These issues has already been discussed previously for the resistor network with only associated parasitic capacitances. However, in this paper we consider also parasitic inductances as well as parasitic capacitances for a more precise network dynamics model. We show that in some cases the temporal stability condition for the network with parasitic inductances and capacitances is equivalent to that for the network with only parasitic capacitances, but in general they are not equivalent. We also show that the spatial stability conditions are equivalent in both cases.
Speeding Up Fractal Image Coding by Combined DCT and Kohonen Neural Net Method

Authors:
Joan-Maria Mas Ribés, U.C.L.-Laboratoire de Télécommunications, (Belgium)
Benoît Simon, Global One sa, (Belgium)
Benoît Macq, U.C.L.-Laboratoire de Télécommunications, (Belgium)

Abstract:
Iterated Transformation Theory (ITT) coding, also known as Fractal Coding, in its original form, allows fast decoding but suffers from long encoding times. During the encoding step, a large number of block best-matching searches have to be performed which leads to a computationally expensive process. We present in this paper a new method that significantly reduces the computational load of ITT based image coding. Both domain and range blocks of the image are transformed into the frequency domain. Domain blocks are then used to train a two dimensional Kohonen Neural Network (KNN) forming a codebook similar to Vector Quantization coding. The property of KNN (and Self-Organizing Feature Maps in general) which maintains the input space topology allows to perform a neighboring search to find the piecewise transformation between domain and range blocks.
ICASSP98 Paper Abstract

Using Recursive Least Square Learning Method for Principal and Minor Components Analysis

Authors:
A. S. Y. Wong, City University of Hong Kong, (Hong Kong)
K. W. Wong, City University of Hong Kong, (Hong Kong)
C. S. Leung, University of Wollongong, (Australia)

Abstract:
In combining principal and minor components analysis, a parallel extraction method based on recursive least square algorithm is suggested to extract the principal components of the input vectors. After the extraction, the error covariance matrix obtained in the learning process is used to perform minor components analysis. The minor components found are then pruned so as to achieve a higher compression ratio. Simulation results show that both the convergent speed and the compression ratio are improved, which in turn indicate that our method effectively combines the extraction of the principal components and pruning of the minor components.
ICASSP98 Paper Abstract

A Novel, Batch Modular Learning Approach for ECG Beat Classification

Authors:
Vijay P Mani, University of Wisconsin-Madison, (U.S.A.)
Yu Hen Hu, University of Wisconsin-Madison, (U.S.A.)
Surekha Palreddy, University of Wisconsin-Madison, (U.S.A.)

Volume 2, page 1093, paper no. 2094

Abstract:
In this paper, we investigate a modular architecture for ECG beat classification. The feature space is divided into distinct regions and individual classifiers are developed for each region. We compare different combination strategies, and feature space partition strategies. We also describe a novel, batch modular learning method that can be used to incrementally improve the performance of the modular network.
ICASSP98 Paper Abstract

Nonlinear Restoration of Spatially Varying Blurred Images Using Self-Organizing Neural Network

Authors:
Hyo-Kyung Sung, Kyungpook National University, (Korea)
Heung-Moon Choi, Kyungpook National University, (Korea)

Volume 2, page 1097, paper no. 2360

Abstract:
An efficient nonlinear restoration of spatially varying blurred images with noise is presented using a self-organizing neural network (SONN). The proposed method can effectively restore the blurred images by using the region classification and learning property of SONN adapted for the blur sensitivity of the receptive field. Receptive fields are adaptively overlapped to eliminate the block effect within the restored images. The proposed method eliminates the need to calculate the gradient, gradient step size, or Hessian of error surface, which affect the performance of the least squares method or of the constraint optimization. Simulation results for the space-variant blurred pepper image show the performance improvement of about 4.86 dB or 3.57 dB, as compared to that of the Richardson-Lucy algorithm or that of conventional neural networks, respectively.
ICASSP98 Paper Abstract
Speech Coding with Nonlinear Local Prediction Model

Authors:
Ni Ma, South China University of Technology, (China)
Gang Wei, South China University of Technology, (China)

Volume 2, page 1101, paper no. 1179

Abstract:
A new signal process based on a nonlinear local prediction model (NLLP) is presented and applied to speech coding. With the same implementation, the speech coding based on the NLLP gives improved performance compared to reference versions of the standard ITU-T G.728 and linear local scheme. The computational efforts for the NLLP analysis do not increase over the conventional linear prediction (LP), and the NLLP supplies better prediction performance over the LP and linear local prediction.
ICASSP98 Paper Abstract
Parametric Subspace Modelling of Speech Transitions

Authors:
Klaus Reinhard, *Cambridge University, (U.K.)*
Mahesan Niranjan, *Cambridge University, (U.K.)*

Volume 2, page 1105, paper no. 1999

Abstract:
In this paper we report on attempting to capture segmental transition information for speech recognition tasks. The slowly varying dynamics of spectral trajectories carries much discriminant information that is very crudely modelled by traditional approaches such as HMMs. In attempts such as recurrent neural networks there is the hope, but not convincing demonstration, that such transitional information could be captured. We start from the very different position of explicitly capturing the trajectory of short time spectral parameter vectors on a subspace in which the temporal sequence information is preserved (Time Constrained Principal Component Analysis). On this subspace, we attempt a parametric modelling of the trajectory, compute a distance metric to perform classification of diphones. Much of the discriminant information is still retained in this subspace. This is illustrated on the isolated transitions /bee/, /dee/ and /gee/. 


ICASSP98 Paper Abstract

A Neural Architecture for Computing Acoustic-Phonetic Invariants

Authors:
Elaine Tsiang, Monowave Corporation, (U.S.A.)

Abstract:
The proposed neural architecture consists of an analytic lower net, and a synthetic upper net. This paper focuses on the upper net. The lower net performs a 2D multiresolution wavelet decomposition of an initial spectral representation to yield a multichannel representation of local frequency modulations at multiple scales. From this representation, the upper net synthesizes increasingly complex features, resulting in a set of acoustic observables at the top layer with multiscale context dependence. The upper net also provides for invariance under frequency shifts, dilatations in tone intervals and time intervals, by building these transformations into the architecture. Application of this architecture to the recognition of gross and fine phonetic categories from continuous speech of diverse speakers shows that it provides high accuracy and strong generalization from modest amounts of training data.
ICASSP98 Paper Abstract

Simplified Neural Network Architectures for a Hybrid Speech Recognition System with Small Vocabulary Size

Authors:
Hossein Sedarat, Stanford University, (U.S.A.)
Rasool Khadem, Stanford University, (U.S.A.)
Horacio Franco, SRI International, (U.S.A.)

Volume 2, page 1113, paper no. 2147

Abstract:
Recent studies suggest that a hybrid speech recognition system based on a hidden Markov model (HMM) with a neural network (NN) subsystem as the estimator of the state conditional observation probability may have some advantages over the conventional HMMs with Gaussian mixture models for the observation probabilities. The HMM and NN modules are typically treated as separate entities in a hybrid system. This paper, however, suggests that the a priori knowledge of HMM structure can be beneficial in the design of the NN subsystem. A case of isolated word recognition is studied to demonstrate that a substantially simplified NN can be achieved in a structured HMM by applying a Bayesian factorization and pre-classification. The results indicate a similar performance to that obtained with the classical approach with much less complexity in NN structure.
ICASSP98 Paper Abstract

Hidden Neural Networks: Application to Speech Recognition

Authors:
Søren Kamaric Riis, Technical University of Denmark, (Denmark)

Volume 2, page 1117, paper no. 1482

Abstract:
In this paper we evaluate the Hidden Neural Network HMM/NN hybrid presented at last year's ICASSP on two speech recognition benchmark tasks; 1) task independent isolated word recognition on the PHONE-BOOK database, and 2) recognition of broad phoneme classes in continuous speech from the TIMIT database. It is shown how Hidden Neural Networks (HNNs) with much fewer parameters than conventional HMMs and other hybrids can obtain comparable performance, and for the broad class task it is illustrated how the HNN can be applied as a purely transition based system, where acoustic context dependent transition probabilities are estimated by neural networks.
ICASSP98 Paper Abstract
An MRNN-Based Method for Continuous Mandarin Speech Recognition

Authors:
Yuan-Fu Liao, National Chaio-Tung University, (Taiwan)
Sin-Horng Chen, National Chaio-Tung University, (Taiwan)

Volume 2, page 1121, paper no. 1360

Abstract:
A new MRNN-based method for continuous Mandarin speech recognition is proposed. The system uses five RNNs to accomplish many subtasks separately and then combine them to integrally solve the problem. They include two RNNs for the discriminations of the two sub-syllable groups of 100 RFD initials and 39 CI finals, two RNNs for the generations of dynamic weighting functions for sub-syllable*s integrations, and one RNN for syllable boundary detection. All RNN modules are combined using a delay-decision Viterbi search. The method differs from the ANN/HMM hybrid approach on using ANNs to perform not only sub-syllables discrimination but also temporal structure modeling of speech signal. The system is trained using a three-stage training method embedding with the MCE/GPD algorithms. Besides, fast recognition method using multi-level pruning is also proposed. Experimental results showed that it outperforms the HMM method on both the recognition accuracy and the computational complexity.
ICASSP98 Paper Abstract

An Off-Line Working Speech Recognition System Employing a Compound Neural Network and Fuzzy Logic

Authors:
Liqing Zhou, Beijing University of Posts & Telecommunications, (China)

Volume 2, page 1125, paper no. 5007

Abstract:
This paper introduces an off-line working speech recognition hardware system. A new compound structure of neural networks is proposed and fuzzy logic is adopted to implement the system. So the system is able to perform speaker-independent real time speech recognition in actual environments where there are heavier noises.
ICASSP98 Paper Abstract

An Analysis of Data Fusion Methods for Speaker Verification

Authors:
Kevin R. Farrell, T-NETIX Inc., (U.S.A.)
Ravi P Ramachandran, Rowan University, (U.S.A.)
Richard J Mammone, Rutgers University, (U.S.A.)

Volume 2, page 1129, paper no. 2506

Abstract:
In this paper, we analyze the diversity of information as provided by several modeling approaches for speaker verification. This information is used to facilitate the fusion of the individual results into an overall result that provides advantages in accuracy over the individual models. The modeling methods that are evaluated consist of the neural tree network (NTN), Gaussian mixture model (GMM), hidden Markov model (HMM), and dynamic time warping (DTW). With the exception of DTW, all methods utilize subword-based approaches. The phrase-level scores for each modeling approach are used for combination. Several data fusion methods are evaluated for combining the model results, including the linear and log opinion pool approaches along with voting. The results of the above analysis have been integrated into a system that has been tested with several databases collected within landline and cellular environments. We have found the linear and log opinion pool methods to consistently reduce the error rate from that obtained when the models are used individually.
ICASSP98 Paper Abstract
Equivariant Algorithms for Selective Transmission

Authors:
Scott C. Douglas, University of Utah, (U.S.A.)

Abstract:
In this paper, we consider the problem of selective transmission—the dual of the blind source separation task—in which a set of independent source signals are adaptively premixed prior to a non-dispersive physical mixing process so that each source can be independently monitored in the far field. We derive a stochastic gradient algorithm for iteratively-estimating the premixing matrix in the selective transmission problem, and through a simple modification, we obtain a second algorithm whose performance is equivariant with respect to the channel’s mixing characteristics. We also describe an approximate version of the equivariant algorithm and other implementation issues. Simulations indicate the useful behavior of the premixing algorithms for selective transmission.
ICASSP98 Paper Abstract
Recognition of Music Types

Authors:
Hagen Soltau, University of Karlsruhe, (Germany)
Tanja Schultz, University of Karlsruhe, (Germany)
Martin Westphal, University of Karlsruhe, (Germany)
Alex Waibel, University of Karlsruhe, (Germany)

Volume 2, page 1137, paper no. 2286

Abstract:
This paper describes a music type recognition system that can be used to index and search in multimedia databases. A new approach to temporal structure modeling is supposed. The so called ETM-NN (Explicit Time Modeling with Neural Network) method uses abstraction of acoustical events to the hidden units of a neural network. This new set of abstract features representing temporal structures, can be then learned via a traditional neural networks to discriminate between different types of music. The experiments show that this method outperforms HMMs significantly.
ICASSP98 Paper Abstract

A Neural Solution for Multitarget Tracking Based on a Maximum Likelihood Approach

Authors:
Michel Winter, I3S CNRS UNSA, (France)
Gerard Favier, I3S CNRS UNSA, (France)

Volume 2, page 1141, paper no. 1959

Abstract:
This paper presents a new neural solution for multitarget tracking based on a maximum likelihood approach. In the radar tracking context, neural networks are generally used to decide which plot can be assigned to each predetected track, in taking into account only the plots received during the last scan. A neural approach is proposed to determine which particular combinations of the plots received during the k latest scans are likely to represent true target tracks. This data association problem is viewed as a multiple hypothesis test that can be solved in maximizing a likelihood function by means of an Hopfield neural network. Some simulation results are presented to illustrate the behaviour of the proposed neural tracking solution.
ICASSP98 Paper Abstract

Exploring the Time-Frequency Microstructure of Speech for Blind Source Separation

Authors:
Hsiao-Chun Wu, University of Florida, (U.S.A.)
Jose C. Principe, University of Florida, (U.S.A.)
Dongxin Xu, University of Florida, (U.S.A.)

Volume 2, page 1145, paper no. 2161

Abstract:
The blind source separation of linear time invariant mixture for nonstationary signals can be deemed as the learning of the linear feature extraction recently. Instead of the previous prevalent model-based approaches, we try to exploit the tempo-frequency microstructure to identify the mixing matrix. With the short-time subband analysis, we can use one-pass method (without resending the signal over and over again like the competitive learning) to estimate the column vectors of the linear mixture. Simulation results show our proposed approach remarkably outperform the existing competitive learning in the identification of the mixing matrix for both sensor-sufficient (as many sensors as sources) and sensor-deficient (less sensors than sources) cases.
ICASSP98 Paper Abstract
Intrinsically Stable IIR Filters and IIR-MLP Neural Networks for Signal Processing

Authors:
Paolo Campolucci, University di Ancona, (Italy)
Francesco Piazza, University di Ancona, (Italy)

Volume 2, page 1149, paper no. 2283

Abstract:
This paper presents a new technique to control stability of IIR adaptive filters based on the idea of intrinsically stable operations that makes possible to continually adapt the coefficients with no need of stability test or poles projection. The coefficients are adapted in a way that intrinsically assures the poles to be in the unit circle. This makes possible to use an higher step size (also named learning rate here) potentially improving the fastness of adaptation with respect to methods that employ a bound on the learning rate or methods that simply do not control stability. 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.
ICASSP98 Paper Abstract

Adaptive RBF Net Algorithms for Nonlinear Signal Learning with Applications to Financial Prediction and Investment

Authors:
Lei Xu, The Chinese University of Hong Kong, (Hong Kong)

Volume 2, page 1153, paper no. 2377

Abstract:
A smoothed variant of the EM algorithm is given for simultaneous training the first layer and the output layer globally in the Normalized Radial Basis Function (NRBF) nets and Extended Normalized RBF nets (ENRBF), together with a BYY learning criterion for the selection of number of basis function. Moreover, a hard-cut fast implementation and an adaptive algorithm have also been proposed for speeding up the training and to handling time varying in the real time nonlinear signal learning and processing. A number of experiments are made on foreign exchange prediction and trading investment.
ICASSP98 Paper Abstract

Genetic Algorithm Optimisation for Maximum Likelihood Joint Channel and Data Estimation

Authors:
Sheng Chen, University of Portsmouth, (U.K.)
Yan Wu, University of Portsmouth, (U.K.)

Volume 2, page 1157, paper no. 1041

Abstract:
A novel blind equalisation scheme is developed based on maximum likelihood (ML) joint channel and data estimation. In this scheme, the joint ML optimisation is decomposed into a two-level optimisation loop. An efficient version of genetic algorithms (GAs), known as a micro GA, is employed at the upper level to identify the unknown channel model and the Viterbi algorithm (VA) is used at the lower level to provide the maximum likelihood sequence estimation of the transmitted data sequence. The proposed GA based algorithm is accurate and robust, and has a fast convergence rate, as is demonstrated in simulation.
ICASSP98 Paper Abstract

A Novel Measure for Independent Component Analysis (ICA)

Authors:
Dongxin Xu, University of Florida, (U.S.A.)
Jose C. Principe, University of Florida, (U.S.A.)
John Fisher III, MIT, (U.S.A.)
Hsiao-Chun Wu, University of Florida, (U.S.A.)

Volume 2, page 1161, paper no. 2175

Abstract:
Measures of independence (and dependence) are fundamental in many areas of engineering and signal processing. Shannon introduced the idea of Information Entropy which has a sound theoretical foundation but sometimes is not easy to implement in engineering applications. In this paper, Renyi’s Entropy is used and a novel independence measure is proposed. When integrated with a nonparametric estimator of the probability density function (Parzen Window), the measure can be related to the 'potential energy of the samples' which is easy to understand and implement. The experimental results on Blind Source Separation confirm the theory. Although the work is preliminary, the 'potential energy' method is rather general and will have many applications.
ICASSP98 Paper Abstract
GCMAC-Based Equalizer for Nonlinear Channels

Authors:
Francisco J Gonzalez-Serrano, Universidad de Vigo, (Spain)
Anibal R Figueiras-Vidal, Universidad Carlos III de Madrid, (Spain)
Antonio Artes-Rodriguez, Universidad Politecnica de Madrid, (Spain)

Volume 2, page 1165, paper no. 1948

Abstract:
This paper deals with the compensation for the nonlinear distortion introduced by power-efficient amplifiers on linear modulations by means of equalization. We propose a new equalizer based on a reduced-complexity network called GCMAC. The GCMAC-based equalizer is compared with other well-known structures such as the Volterra filter and the Multi-layer Perceptron. Extensive computer simulations have been carried out. The obtained results show the effectiveness of the proposed structure to compensate for strong nonlinearities.
ICASSP98 Paper Abstract
Remote Sensing Segmentation Through a Filter Bank Based on Gabor Functions

Authors:
Jesús García-Consuegra, Universidad de Castilla-La Mancha, (Spain)
Guillermo Cisneros, Universidad Politecnica de Madrid, (Spain)
Jorge Ballesteros, Universidad de Castilla-La Mancha, (Spain)
Rafael Molina, Universidad de Castilla-La Mancha, (Spain)

Abstract:
One of the most critical activities of remote sensing consists of the identification of different coverages (types of crops) on land surface. This task is complicated when the entire plot is not covered by the crop (e.g. almond and olive fields, vineyards, etc.). In this paper, the segregation of these crops is accomplished by using a multi-channel (Gabor functions) filtering approach in remote sensing imagery, in this case applied to aerial photograph.
ICASSP98 Paper Abstract

Intelligent Query and Browsing Information Retrieval (IQBIR) Agent

Authors:
Jong-Min Park, University of Wisconsin-Madison, (U.S.A.)

Abstract:
Reported in this paper is an intelligent agent that aids users to conduct efficient Internet Web information retrieval through query formulation, information collection, information clustering, and analysis. The underlying mechanism is a probabilistic Sample-at-the-boundary learning algorithm for clustering the search results and learning and matching the user concept. Kohonen's "windowed" Learning Vector Quantization algorithm is shown to be related to this Sample-at-the-boundary learning algorithm. A prototype system has been developed and evaluation has been conducted.
ICASSP98 Paper Abstract

Feature Extraction Method for High Impedance Ground Fault Localization in Radial Power Distribution Networks

Authors:
Kaare J Jensen, Technical University of Denmark, (Denmark)
Steen M Munk, NESA A/S, (Denmark)
John A Sorensen, Technical University of Denmark, (Denmark)

Volume 2, page 1177, paper no. 1746

Abstract:
A new approach to the localization of high impedance ground faults in compensated radial power distribution networks is presented. The total size of such networks is often very large and a major part of the monitoring of these is carried out manually. The increasing complexity of industrial processes and communication systems lead to demands for improved monitoring of power distribution networks so that the quality of power delivery can be kept at a controlled level. The ground fault localization method for each feeder in a network, is based on the centralized frequency broad band measurement of three phase voltages and currents. The method consists of a feature extractor, based on a grid description of the feeder by impulse responses, and a neural network for ground fault localization. The emphasis of this paper is the feature extractor, and the detection of the time instance of a ground fault.
ICASSP98 Paper Abstract
Retina Implant Adjustment with Reinforcement Learning

Authors:
Michael Becker, University of Bonn, (Germany)
Mikio Braun, University of Bonn, (Germany)
Rolf Eckmiller, University of Bonn, (Germany)

Abstract:
A tuning method with reinforcement learning (RL) for the Retina Encoder (RE) of a Retina Implant (RI) as a visual prosthesis for blind subjects with retinal degenerations is proposed. RE simulates retinal information processing in real time by means of spatio-temporal receptive field (RF) filters and generates electrical signals for stimulation of several hundreds of ganglion cells (GC) to regain a modest amount of vision. For each contacted GC, RE has to be optimized with regard to the patient's perception. The patient’s (for the present simulated) evaluative feedback is applied here in a dialog module as a reinforcement signal to train several RL agents in a neural network learning process (see also http://www.nero.uni-bonn.de).
ICASSP98 Paper Abstract

An Introduction to Multiscale Defined Systems: Self-Organising IFS Fractal Networks

Authors:
Graham C Freeland, University of Strathclyde, Scotland, (U.K.)
Tariq S Durrani, University of Strathclyde, Scotland, (U.K.)

Abstract:
Deterministic multiscale defined representational forms have found a significant role in the theory and application of signal processing over the last decade. With little argument the most widely important for signal and system modelling is likely to be multiscale defined wavelets. Another class of multiscale representation which has attracted consistent interest over the same period is the group of signal models defined in terms of Iterated Function Systems (IFS). This paper is concerned with widening the IFS application to include system modelling, particularly of neural network-like structures. We introduce an interpolating IFS model as a form of self-organising map with global fractal constraints. Symbolic addressing is employed to discretize the attractor into pseudo network nodes. We present in detail an online gradient based algorithm for training. This particular model is intended for efficient pattern recognition in complex environments, for example, with multifractal sources such as those seen in network traffic and general turbulence.
ICASSP98 Paper Abstract

An Experimental Comparison of the Bayesian Ying-Yang Criteria and Cross Validation for Selection on Number of Hidden Units in Feedforward Networks

Authors:
Wing-kai Lam, The Chinese University of Hong Kong, (Hong Kong)
Lei Xu, The Chinese University of Hong Kong, (Hong Kong)

Volume 2, page 1189, paper no. 2372

Abstract:
Optimizing the number of hidden units in feedforward neural networks is an important issue in learning. Recently, a new criteria on selecting the number of hidden units in feedforward neural networks is proposed by one of the present author, based on the so-called Bayesian Ying-Yang (BYY) learning theory. The new criteria can be simply computed during the implementation of backpropagation training. In this paper, the criteria is experimentally studied and compared with the well-known Cross Validation approach. Simulation results show that obtained number of hidden units by the BYY criteria is highly consistent to the minimal generalization error and outperforms the Cross Validation approach.
ICASSP98 Paper Abstract

Minimum Detection Error Training for Acoustic Signal Monitoring

Authors:
Hideyuki Watanabe, ATR Human Information Processing Labs, (Japan)
Yuji Matsumoto, ATR Human Information Processing Labs, (Japan)
Shigeru Katagiri, ATR Human Information Processing Labs, (Japan)

Volume 2, page 1193, paper no. 1533

Abstract:
In this paper we propose a novel approach to the detection of acoustic irregular signals using Minimum Detection Error (MDE) training. The MDE training is based on the Generalized Probabilistic Descent for discriminative pattern recognizer design. We demonstrate its fundamental utility by experiments in which several acoustic events are detected in a noisy environment.
ICASSP98 Paper Abstract

Fast Subspace Tracking and Neural Network Learning by a Novel Information Criterion

Authors:
Yongfeng Miao, University of Melbourne, (Australia)
Yingbo Hua, University of Melbourne, (Australia)

Volume 2, page 1197, paper no. 1548

Abstract:
We introduce a novel information criterion (NIC) for searching for the optimum weights of a two-layer linear neural network (NN). The NIC exhibits a single global maximum attained if and only if the weights span the (desired) principal subspace of a covariance matrix. The other stationary points of the NIC are (unstable) saddle points. We develop an adaptive algorithm based on the NIC for estimating and tracking the principal subspace of a vector sequence. The NIC algorithm provides a fast on-line learning of the optimum weights for the two-layer linear NN. The NIC algorithm has several key advantages such as faster convergence which is illustrated through analysis and simulation.
ICASSP98 Paper Abstract

Adaptive Regularization of Neural Networks Using Conjugate Gradient

Authors:
Cyril Goutte, Technical University of Denmark, (Denmark)
Jan Larsen, Technical University of Denmark, (Denmark)

Volume 2, page 1201, paper no. 1933

Abstract:
Recently we suggested a regularization scheme which iteratively adapts regularization parameters by minimizing validation error using simple gradient descent. In this contribution we present an improved algorithm based on the conjugate gradient technique. Numerical experiments with feed-forward neural networks successfully demonstrate improved generalization ability and lower computational cost.
ICASSP98 Paper Abstract
Design of Robust Neural Network Classifiers

Authors:
Jan Larsen, Technical University of Denmark, (Denmark)
Lars Nonboe Andersen, Technical University of Denmark, (Denmark)
Mads Hintz-Madsen, Technical University of Denmark, (Denmark)
Lars Kai Hansen, Technical University of Denmark, (Denmark)

Abstract:
This paper addresses a new framework for designing robust neural network classifiers. The network is optimized using the maximum a posteriori technique, i.e., the cost function is the sum of the log-likelihood and a regularization term (prior). In order to perform robust classification, we present a modified likelihood function which incorporate the potential risk of outliers in the data. This leads to introduction of a new parameter, the outlier probability. Designing the neural classifier involves optimization of network weights as well as outlier probability and regularization parameters. We suggest to adapt the outlier probability and regularization parameters by minimizing the error on a validation set, and a simple gradient descent scheme is derived. In addition, the framework allows for constructing a simple outlier detector. Experiments with artificial data demonstrates the potential of the suggested framework.
ICASSP98 Paper Abstract

Extraction of Independent Components from Hybrid Mixture: KuicNet Learning Algorithm and Applications

Authors:
Sun-Yuan Kung, Princeton University, (U.S.A.)
Cristina Mejuto, University of La Coruna, (Spain)

Volume 2, page 1209, paper no. 2114

Abstract:
A hybrid mixture is a mixture of supergaussian, gaussian, and subgaussian independent components (ICs). This paper addresses extraction of ICs from a hybrid mixture. There are two (single-output vs. all-outputs) approaches to the design of contrast functions. We advocate the former approach due to its (1) simple and closed-form analysis, and (2) numerical convergence and computational saving. Via this approach, the positive kurtosis (resp. negative kurtosis) can be proved to be a valid contrast function for extracting supergaussian (resp. subgaussian) ICs from any nontrivial hybrid mixture. We shall also develop a network algorithm, Kurtosis-based Independent Component Network (KuicNet), for recursively extracting ICs. Numerical and convergence properties are analyzed and several application examples demonstrated.
ICASSP98 Paper Abstract

Why Natural Gradient?

Authors:
Shun-ichi Amari, RIKEN Brain Science Institute, (Japan)
Scott C. Douglas, University of Utah, (U.S.A.)

Volume 2, page 1213, paper no. 2134

Abstract:
Gradient adaptation is a useful technique for adjusting a set of parameters to minimize a cost function. While often easy to implement, the convergence speed of gradient adaptation can be slow when the slope of the cost function varies widely for small changes in the parameters. In this paper, we outline an alternative technique, termed natural gradient adaptation, that overcomes the poor convergence properties of gradient adaptation in many cases. The natural gradient is based on differential geometry and employs knowledge of the Riemannian structure of the parameter space to adjust the gradient search direction. Unlike Newton's method, natural gradient adaptation does not assume a locally-quadratic cost function. Moreover, for maximum likelihood estimation tasks, natural gradient adaptation is asymptotically Fisher-efficient. A simple example illustrates the desirable properties of natural gradient adaptation.
ICASSP98 Paper Abstract

Neural Network Inversion of Snow Parameters by Fusion of Snow Hydrology Prediction and SSM/I Microwave Satellite Measurements

Authors:
Yuankai Wang, *University of Washington*, (U.S.A.)
Jenq-Neng Hwang, *University of Washington*, (U.S.A.)
Chi-Te Chen, *University of Washington*, (U.S.A.)
Leung Tsang, *University of Washington*, (U.S.A.)
Bart Nijssen, *University of Washington*, (U.S.A.)
Dennis P. Lettenmaier, *University of Washington*, (U.S.A.)

Volume 2, page 1217, paper no. 2246

Abstract:
Inverse remote sensing problems are generally ill-posed. In this paper, we propose an approach, which integrates the dense media radiative transfer (DMRT) model, snow hydrology model, neural networks and SSM/I microwave measurements, to infer the snow depth. Four multilayer perceptrons (MLPs) were trained using the data from DMRT model. With the provision of initial guess from snow hydrology prediction, neural networks effectively invert the snow parameters based on SSM/I measurements. In addition, a prediction neural network is used to achieve adaptive learning rates and good initial estimate of snow depth for inversion. Result shows that our algorithm can effectively and accurately retrieve snow parameters from these highly nonlinear and many-to-one mappings.
ICASSP98 Paper Abstract

A Piecewise Linear Recurrent Neural Network Structure and its Dynamics

Authors:
Xiao Liu, GlobeSpan Technologies Inc., (U.S.A.)
Tulay Adali, University of Maryland, Baltimore, (U.S.A.)
Levent Demirekler, University of Maryland, Baltimore, (U.S.A.)

Volume 2, page 1221, paper no. 5139

Abstract:
We present a piecewise linear recurrent neural network (PL-RNN) structure by combining the canonical piecewise linear function with the autoregressive moving average (ARMA) model such that an augmented input space is partitioned into regions where an ARMA model is used in each. The piecewise linear structure allows for easy implementation, and in training, allows for use of standard linear adaptive filtering techniques based on gradient optimization and description of convergence regions for the step-size. We study the dynamics of PL-RNN and show that it defines a contractive mapping and is bounded input bounded output stable. We introduce application of PL-RNN to channel equalization and show that it closely approximates the performance of the traditional RNN that uses sigmoidal activation functions.
ICASSP98 Paper Abstract

Recovering Depth from Stereo Using ART Neural Networks

Authors:
Stylianos Markogiannakis, Northeastern University, (U.S.A.)
Elias S Manolakos, Northeastern University, (U.S.A.)

Volume 2, page 1225, paper no. 2398

Abstract:
One of the long standing problems in passive stereo vision is that of constructing an accurate range map using only two images, providing two views of the same world schene. It amounts to identifying pairs of corresponding pixels that are associated with the same point of the real world. We are introducing ART-1 neural networks as a primitive for addressing effectively all aspects of the challenging stereo correspondence problem. Using a multi-pass approach it is possible to increase gradually the density of matched points, while at the same time false matches are filtered by requiring close agreement between disparity estimates in a neighborhood. At the end a reasonably dense disparity map is obtained, to the estend that it allows scene reconstruction by interpolation. Our scheme was tested on random dot stereograms, artificial and real world scenes. In all cases scene reconstructions are shown to be quite realistic.
ICASSP98 Paper Abstract

Nonlinear Acoustic Echo Cancellation Using a Hammerstein Model

Authors:
Lester S.H. Ngia, Chalmers University of Technology, (Sweden)
Jonas E Sjöberg, Chalmers University of Technology, (Sweden)

Volume 2, page 1229, paper no. 2455

Abstract:
In hands-free telephone or video conference application, there exists an acoustic feedback coupling between the loudspeaker and microphone, which creates the acoustic echo. Linear acoustic echo cancellers (AECs) are commonly used to remove this echo. However, they are unable to effectively cancel nonlinear distortions. This paper employs a Hammerstein model to describe the acoustic channel of a nonlinear system concatenated with a linear faded echo path. A feed-forward neural network is used to model the static nonlinearity and a Finite Impulse Response (FIR) structure is used to model the linear dynamic system. The formed nonlinear model is applied to real data collected in an anechoic chamber and it performs slightly better than linear models. Although the improvement is small, the results show some interesting insights on the characteristic of a loudspeaker’s nonlinearities and their effect on the performance of an AEC.
ICASSP98 Paper Abstract

Noise Reduction and Speech Enhancement via Temporal Anti-Hebbian Learning

Authors:
Mark A Girolami, University of Paisley, Scotland, (U.K.)

Abstract:
Temporal extensions of both linear and nonlinear anti-Hebbian learning have been shown to be suited to the problem of blind separation of sources from their convolved mixtures. This paper presents a generalized form of anti-Hebbian learning for a partially connected recurrent network based on the maximum likelihood estimation principle. Inspired by features of the binaural unmasking effect the network and associated online adaptation are applied to the enhancement of speech, which is corrupted by interfering noise, competing speech and reverberation. Graded simulations based on speech corrupted with increasingly complex levels of reverberation are reported. It is shown that for high levels of reverberation the proposed method compares favorably with classical adaptive filter approaches to speech enhancement in real acoustic environments.
ICASSP98 Paper Abstract

A High Quality Text-to-Speech System Composed of Multiple Neural Networks

Authors:
Orhan Karaali, Motorola, (U.S.A.)
Gerald E Corrigan, Motorola, (U.S.A.)
Noel S Massey, Motorola, (U.S.A.)
Corey A Miller, Motorola, (U.S.A.)
Otto L Schnurr, Motorola, (U.S.A.)
Andrew W Mackie, Motorola, (U.S.A.)

Volume 2, page 1237, paper no. 2151

Abstract:
While neural networks have been employed to handle several different text-to-speech tasks, ours is the first system to use neural networks throughout, for both linguistic and acoustic processing. We divide the text-to-speech task into three subtasks, a linguistic module mapping from text to a linguistic representation, an acoustic module mapping from the linguistic representation to speech, and a video module mapping from the linguistic representation to animated images. The linguistic module employs a letter-to-sound neural network and a postlexical neural network. The acoustic module employs a duration neural network and a phonetic neural network. The visual neural network is employed in parallel to the acoustic module to drive a talking head. The use of neural networks that can be retrained on the characteristics of different voices and languages affords our system a degree of adaptability and naturalness heretofore unavailable.
ICASSP98 Paper Abstract

Fraud Detection in Communications Networks Using Neural and Probabilistic Methods

Authors:
Michiaki Taniguchi, Siemens AG, Corporate Technology, (Germany)
Michael Haft, Siemens AG, Corporate Technology, (Germany)
Jaakko Hollmen, Siemens AG, Corporate Technology, (Germany)
Volker Tresp, Siemens AG, Corporate Technology, (Germany)

Volume 2, page 1241, paper no. 2466

Abstract:
Fraud detection refers to the attempt to detect illegitimate usage of a communications network. Three methods to detect fraud are presented. Firstly, a feed-forward neural network based on supervised learning is used to learn a discriminative function to classify subscribers using summary statistics. Secondly, Gaussian mixture model is used to model the probability density of subscribers' past behavior so that the probability of current behavior can be calculated to detect any abnormalities from the past behavior. Lastly, Bayesian networks are used to describe the statistics of a particular user and the statistics of different fraud scenarios. The Bayesian networks can be used to infer the probability of fraud given the subscribers' behavior. The data features are derived from toll tickets. The experiments show that the methods detect over 85% of the fraudsters in our testing set without causing false alarms.
ICASSP98 Paper Abstract

Neural Vision System and Applications in Image Processing and Analysis

Authors:
Ling Guan, University of Sydney, (Australia)
Stuart W. Perry, University of Sydney, (Australia)
Raniero Romagnoli, University of Sydney, (Australia)
Hausan Wong, University of Sydney, (Australia)
Haosong Kong, University of Sydney, (Australia)

Volume 2, page 1245, paper no. 1912

Abstract:
We present a computer vision system based on an integrated neural network architecture. In the low level vision subsystem, a network of networks - a biologically inspired network is used to recursively perform filtering, segmentation and edge detection; in the intermediate level and the high level, hierarchically structured arrays of self-organizing tree maps (SOTM) - extension of the popular self-organizing map are utilized to carry out image/feature analysis. The system has been applied to solve a number of real world problems. Some interesting and encouraging results will be reported.
Combining Time-Delayed Decorrelation and ICA: Towards Solving the Cocktail Party Problem

Authors:
Te-Won Lee, The Salk Institute, CNL, (U.S.A.)
Andreas Ziehe, GMD, FIRST, (Germany)
Reinhold Orglmeister, Berlin University of Technology, (Germany)
Terrence Sejnowski, The Salk Institute, CNL, (U.S.A.)

Abstract:
We present methods to separate blindly mixed signals recorded in a room. The learning algorithm is based on the information maximization in a single layer neural network. We focus on the implementation of the learning algorithm and on issues that arise when separating speakers in room recordings. We used an infomax approach in a feedforward neural network implemented in the frequency domain using the polynomial filter matrix algebra technique. Fast convergence speed was achieved by using a time-delayed decorrelation method as a preprocessing step. Under minimum-phase mixing conditions this preprocessing step was sufficient for the separation of signals. These methods successfully separated a recorded voice with music in the background (cocktail party problem). Finally, we discuss problems that arise in real world recordings and their potential solutions.
ICASSP98 Paper Abstract

Energy-Based Effective Length of the Impulse Response of a Recursive Filter

Authors:
Timo I. Laakso, Helsinki University of Technology, (Finland)
Vesa Valimaki, Helsinki University of Technology, (Finland)

Volume 3, page 1253, paper no. 1501

Abstract:
A measure for the effective length of the impulse response of a stable recursive digital filter based on accumulated energy is proposed. A general definition and a simple algorithm for its evaluation are introduced, and closed-form expressions are derived for first-order IIR filters. The effect of zeros on the effective length is analyzed. An upper bound for the effective length of higher-order filters is derived using results for low-order filters. The new measure finds applications in several fields of digital signal processing, including estimation of the extent of attack transients for filters with dynamically varying inputs, elimination of transients in variable recursive filters, and design and implementation of linear-phase IIR systems.
ICASSP98 Paper Abstract

Design of Recursive Filters with Constant Group Delay and Chebyshev Attenuation

Authors:
Guergana S Mollova
Rolf Unbehauen, University of Erlangen-Nurnberg, (Germany)

Volume 3, page 1257, paper no. 1097

Abstract:
This article presents some new results concerning recursive filters design with approximately linear phase and Chebyshev stopband attenuation. The denominator polynomial $D(z)$ of the transfer function $H(z)=N(z)/D(z)$ is used to obtain a maximally flat behavior for the delay in the passband, whereas $N(z)$ describes equiripple amplitude in the stopband. The approach under consideration is based on $z$-domain concepts. At the end, the paper concludes with several detailed examples and graphics showing the efficacy of the proposed technique.
Implementation of Recursive Filters Having Delay Free Loops

Authors:
Aki Härmä, Helsinki University of Technology, (Finland)

Abstract:
Certain types of recursive filters have been considered as non-realizable because they contain delayless recursive loops. Usually the problem is rather technical than theoretical. In this paper a method of implementing such filters is introduced. The general procedure is to split a delay free recursive filter to a non-delay free and a pure delay free structure. As a combination of these, the filter can be implemented directly and efficiently. In addition, following from the same formulation, a generic procedure to convert any such filter to an equivalent directly realizable structure is also given. As an example, a set of frequency warped all-pole filters is considered. The new warped all-pole lattice introduced in this paper completes the family of warped filters.
ICASSP98 Paper Abstract

Design of IIR Eigenfilters with Arbitrary Magnitude Frequency Response

Authors:
Fabrizio Argenti, University of Florence, (Italy)
Enrico Del Re, University of Florence, (Italy)

Volume 3, page 1265, paper no. 5221

Abstract:
In this study, the eigenfilter approach is applied to designing Infinite Impulse Response (IIR) filters having an arbitrary magnitude frequency response. A causal rational transfer function having an arbitrary number of poles and zeros is achieved. The procedure works in the frequency domain. Some numerical examples showing the application of the presented method to the design of multiband filters with different gains and different magnitude shape in each band are presented.
ICASSP98 Paper Abstract
Design of Linear Phase FIR Filters with Recursive Structure and Discrete Coefficients

Authors:
Hai H Dam, Curtin University, (Australia)
Sven Nordebo, Curtin University, (Australia)
Kok L Teo, Curtin University, (Australia)
Antonio Cantoni, Curtin University, (Australia)

Volume 3, page 1269, paper no. 1550

Abstract:
In this paper, we consider a class of FIR filters defined by the first order different routing digital filter (DRDF) structure and sums of two powers of two coefficients. A novel design method is developed for constructing high quality filters with reference to the min-max error criterion. This method is highly efficient in terms of computational time. Simulation studies show a large improvement over existing methods such as quantization. In some cases, the peak ripple magnitude over the stop and pass bands is reduced up to 13 dB over the quantization method. These results are achieved for even small number of delays.
Optimal Cumulant Domain Filtering

Authors:
Roy Chapman, University of Strathclyde, Scotland, (U.K.),
Tariq S Durrani, University of Strathclyde, Scotland, (U.K.)

Abstract:
This paper presents a new technique which exploits constrained optimization methods to derive optimal two dimensional filters in the cumulant domain for processing signals in non Gaussian noise, or signals with corrupting interferences which have non symmetrical probability density functions. The approach proposed here for enhancing signals in such noise is important, as increasingly practical engineering application areas are identifying occasions where the perceived wisdom of modelling signals in additive noise simply does not hold. Since the bispectrum of non Gaussian noise and interference is not zero, it corrupts the bispectrum of the signal. Thus filters that suppress the bispectral component of the noise and enhance the signal bispectrum are required. The two dimensional filters proposed in this paper have the property of concentrating the filter energy into a hexagonal region in the bispectral domain. This leads to an impulse response for these filters which represents a new form of two dimensional discrete prolate spheriodal sequence. The sensitivity of cumulant determination for non Gaussian noise has been noted in the area of array processing. However this paper presents one of the first attempts to remove non Gaussian noise by cumulant filtering.
ICASSP98 Paper Abstract

An Application of Chebechev’s Inequality Theorem in the Design of Optimal Non-Linear Filters

Authors:
Subhash Challa, Queensland University of Technology, (Australia)
Farhan A Faruqi, Queensland University of Technology, (Australia)

Volume 3, page 1277, paper no. 1873

Abstract:
Chebechev’s inequality theorem from the theory of probability and statistics provides an upperbound for the amount of probability in the "tails" of any given probability density function. This theorem has interesting applications in the numerical solution of the Fokker-Planck-Kolmogorov Equation (FPKE) as shown in this paper. Numerical solution of FPKE is an essential component of the design of optimal nonlinear filters. The solution of the FPKE in conjunction with the Bayes’ conditional density lemma provides optimal (minimum variance) state estimates of any general stochastic dynamic system (SDS).
ICASSP98 Paper Abstract

A Numerical Algorithm for Filtering and State Observation

Authors:
Salim Ibrir, Laboratoire des Signaux et Systemes, (France)

Volume 3, page 1281, paper no. 2005

Abstract:
This paper is dealing with a numerical method for data-fitting and estimation of the continuous higher derivatives of a given signal from its non-exact sampled data. The proposed algorithm is a generalization of the algorithm proposed by C. H. Reinsch[1967]. This algorithm is conceived as being a key element in the structure of the numerical observer discussed in our last papers. The presented algorithm seems to be flexible because of the introduction of equivalent conditions of smoothness derived from finite difference methods. Detailed steps of the computational method will be given to evaluate the continuous approximates of higher derivatives of a signal given by its noisy discrete values together with the filtered continuous signal. Satisfactory results have been obtained showing the efficiency of such an algorithm.

Keywords: Spline functions, Numerical differentiation, Observers, Smooth filters.
ICASSP98 Paper Abstract

Optimum Finite-Length LTI Transmit Filters for ISI-Channels

Authors:
Naofal Al-Dhahir, GE Corporate R&D, (U.S.A.)

Abstract:
Optimum FIR transmit filters for symbol-by-symbol transmission on linear dispersive additive-Gaussian-noise channels are derived by maximizing the channel throughput, subject to a fixed input energy constraint. This maximized throughput is compared with that achievable with water-pour and flat transmit filters. The effect of transmit filter optimization on the receiver performance is investigated by considering the popular MMSE-DFE receiver structure.
Continuous-Time Envelope Constrained Filter Design with Input Uncertainty

Authors:
Ba-Ngu Vo, Chinese University of Hong Kong, (Hong Kong)
Antonio Cantoni, ATRI Curtin University of Technology, (Australia)

Abstract:
In an envelope-constrained filtering problem with uncertain input the set of possible inputs and the set of permissible outputs are each defined by envelopes or masks. This paper considers a continuous-time filter which in structure is comprised of an A/D converter, an FIR filter, a D/A converter and an analog post-filter. The object is to design the digital component of the filter structure so as to minimize the noise enhancement whilst satisfying the constraint that every signal in the input envelope evokes a response which stays in the output envelope.
ICASSP98 Paper Abstract
Anisotropic Diffusion and Local Monotonicity

Authors:
Scott T Acton, Oklahoma State University, (U.S.A.)

Abstract:
This paper investigates the relationship between anisotropic diffusion and local monotonicity. A diffusion technique that has locally monotonic root signals is presented. The enhancement algorithm rapidly converges to a locally monotonic signal of the desired degree. It is shown that the diffusion coefficient used here is the only formation that guarantees idempotence for locally monotonic signals. The signals resulting from locally monotonic diffusion are closer to the original signals than the corresponding median root signals. Furthermore, the diffusion algorithm does not have a difficulty with alternating signals, as does the median filter. In contrast to other anisotropic diffusion techniques, the diffusion method given here does not preserve outliers and does not require a gradient magnitude threshold in the diffusion coefficient.
ICASSP98 Paper Abstract

Application of Infinite Dimensional Linear Programming to FIR Filter Design with Time Domain Constraints

Authors:
Sven Nordebo, Curtin University of Technology; (Australia)
Zhuquan Zang, Curtin University of Technology; (Australia)

Volume 3, page 1297, paper no. 1635

Abstract:
Previously the envelope-constrained filtering problem was formulated as designing an FIR filter such that the filter’s L-2 norm is minimized subject to the constraint that its response to a specified input pulse lies within a prescribed envelope. In this paper, we recast this filter design problem as a frequency-domain L-infinity optimization problem with time-domain constraints. Motivations for solving this problem are given. Then recently developed infinite dimensional linear programming techniques are used for the design of the required FIR filter. For illustration, we apply the approach to a numerical example which deals with the design of an equalization filter for a digital transmission channel.
ICASSP98 Paper Abstract
Complex Frequency Response FIR Filter Design

Authors:
Worayot Lertniphonphun, Georgia Institute of Technology, (U.S.A.)
James H. McClellan, Georgia Institute of Technology, (U.S.A.)

Volume 3, page 1301, paper no. 2060

Abstract:
This paper provides an algorithm for designing FIR filters that approximate both magnitude and phase of the frequency response. The new algorithm produces a filter optimized under the weighted Chebyshev norm. The algorithm starts from a first stage unoptimized filter designed by a Remez-like algorithm and then uses shifted Dirichlet kernel functions to reduce large error peaks and converge to an equiripple set of peaks. The error function is modified at each iteration by subtracting a best-fit linear combination of kernel functions due to the large error peak(s). For one length-100 example, the computation of this algorithm was less than that of the complex Remez by two orders of magnitude.
ICASSP98 Paper Abstract
Frequency Sampling Filters with Algebraic Integers

Authors:
Uwe Meyer-Baese, University of Florida, (U.S.A.)
Jon Mellott, University of Florida, (U.S.A.)
Fred Taylor, University of Florida, (U.S.A.)

Volume 3, page 1305, paper no. 2077

Abstract:
Algebraic integers have been proven beneficial to DFT and non-recursive FIR filter designs since algebraic integers can be dense in $\mathbb{C}$, resulting in short word width, high speed designs. This paper uses another property of algebraic integers: algebraic integers can produce exact pole zero cancellation pairs that are used in recursive FIR, frequency sampling filter designs.
ICASSP98 Paper Abstract

Nonrecursive Synthesis of FIR Filters for Approximate Processing

Authors:
Dietmar W Schill, University of Erlangen, (Germany)
Andre Marguinaud, Alcatel Espace, (France)

Volume 3, page 1309, paper no. 2121

Abstract:
In approximate and real time processing, one encounters the problem of making efficient use of the limited processing power available. Furthermore it can be desirable to enable a system to react dynamically to a change in requirements. For digital filters this implies the possibility of calculating filter coefficients on the fly with low complexity algorithms. Such an algorithm is presented for the design of linear phase FIR lowpass filters. It has the additional property that subsets of coefficients of one filter constitute by themselves filters of reduced stop band attenuation and/or lower bandwidth reduction.
ICASSP98 Paper Abstract
Embedded FIR Generalized Eigenfilters Using Test Inputs

Authors:
Jeffrey O Coleman, Naval Research Laboratory, (U.S.A.)
Volume 3, page 1313, paper no. 2146

Abstract:
A systematic approach is proposed for the individual or joint design of FIR filters to meet specifications on either a single filter or an embedding system (possibly multirate). System power gains in response to particular input spectra are optimized using a generalized eigenvector method. Numerical integration is avoided through a time-domain formulation. Real or complex filters with linear or nonlinear phase or N-th band properties are easily handled.
ICASSP98 Paper Abstract

Minimum Phase FIR Filter Design From Linear Phase Systems Using Root Moments

Authors:
Tania Stathaki, Imperial College, (U.K.)
Anthony Constantinides, Imperial College, (U.K.)
Georgios Stathakis, Imperial College, (U.K.)

Volume 3, page 1317, paper no. 5088

Abstract:
In this contribution we propose a method for a minimum phase Finite Impulse Response (FIR) filter design from a given linear phase FIR function with the same amplitude response. We are concentrating on very high degree polynomials for which factorisation procedures for root extraction are unreliable. The approach taken involves the use Cauchy Residue Theorem applied to the logarithmic derivative of the transfer function. This leads into a set of parameters derivable directly from the polynomial coefficients which facilitate the factorisation problem. The concept is developed in a way that leads naturally to the celebrated Newton Identities. In addition to solving the above problem, the results of the proposed design scheme are very encouraging as far as robustness and computational complexity are concerned.
ICASSP98 Paper Abstract
The Connection Between Continuous and Discrete Lattice Filters

Authors:
Paulo J.S.G. Ferreira, Universidade de Aveiro, (Portugal)

Volume 3, page 1321, paper no. 2435

Abstract:
The importance of lattice structures in connection with filtering and prediction has been known for decades. The demand for faster processing has led to steadily increasing sampling rates, and as a result the behavior of the discrete filters as the sampling period tends to zero has become an important theoretical and practical issue. One way of solving the numerical problems that arise in the usual filter structures when the sampling period becomes small compared with the dynamics of the underlying physical processes is to resort to δ operators instead of delay operators. Although the interrelations between the continuous and discrete lattice structures have been rarely studied, it is known that the δ lattice naturally leads to a continuous form as the sampling rate increases. This paper addresses this point and establishes the rate of convergence of the discrete lattice filter to the continuous filter as a function of the sampling period or of the filter order.
A Fast O(N) Algorithm for Adaptive Filter Bank Design

Authors:
Omid S. Jahromi, Shiraz University, (Iran)
M. A. Masnadi-Shirazi, Shiraz University, (Iran)
Minyue Fu, University of Newcastle, (Australia)

Abstract:
Designing optimal filter banks for subband coding applications has recently attracted considerable attention. In particular, the first two authors had developed an adaptive algorithm based on stochastic gradient descent (SGD) that enables one to optimize two-channel paraunitary filter banks in an online fashion [3]. They have also extended the adaptive algorithm for the case of tree-structured filter banks [4]. The computational complexity of the algorithm proposed originally is proportional to the second power of N, where N is the number of stages in the paraunitary lattice. In this paper, we derive a fast algorithm which reduces the amount of computation to O(N). We also show that the new algorithm can be implemented using an IIR lattice structure. Some issues regarding numerical stability of the proposed IIR implementation are also discussed.
ICASSP98 Paper Abstract
On Existence of FIR Principal Component Filter Banks

Authors:
Ahmet Kirac, Caltech, (U.S.A.)
Palghat P. Vaidyanathan, Caltech, (U.S.A.)

Volume 3, page 1329, paper no. 1305

Abstract:
In this paper we have two interesting results. One is of theoretical interest and the other practical. The theoretical result is that the optimum FIR orthonormal filter bank of a fixed finite degree that maximizes the coding gain does not always contain an optimum compaction filter. In other words, in general, there does not exist a principal component filter bank (PCFB) of a given nonzero degree. This is sharply in contrast to the cases of transform coders and ideal subband coders where the existence of PCFB’s are assured by their very construction. The practical result of the paper is that constraining the filter corresponding to the largest subband variance to be a compaction filter does not result in a significant loss of performance for practical input signals. Since there exist very efficient methods to design FIR compaction filters and since the best completion of the filter bank given the first filter is trivially done by a KLT, we see that this is an extremely efficient method despite the fact that it is suboptimum.
ICASSP98 Paper Abstract

Energy Compaction Performance of Paraunitary FIR Filter Banks For Finite-Length Signals

Authors:
Wolfgang Niehsen, IENT, RWTH Aachen, (Germany)

Abstract:
The energy compaction performance of two-channel paraunitary finite impulse response (FIR) filter banks for finite-length signals is investigated. A detailed non-iterative design procedure for boundary filters which are optimal in a weighted mean square error (MSE) sense in the Fourier domain is presented. Simulation results are given for two-channel paraunitary FIR filter banks based on minimum-phase Daubechies filters and least-asymmetric Daubechies filters, respectively.
ICASSP98 Paper Abstract

Jointly Optimal Analysis and Synthesis Filter Banks for Bit Constrained Source Coding

Authors:
Are Hjørungnes, University of California, Santa Barbara, (U.S.A.)
Tor A. Ramstad, University of California, Santa Barbara, (U.S.A.)

Abstract:
A subband coder structure is fully optimized with respect to the minimum block mean squared error between the output and the input signals under a bit constraint. The analysis filter bank structure generates maximally decimated and equal bandwidth subbands. The subband quantizers are modeled as additive noise sources. To simplify the optimization an optimal multiple-input multiple-output system is first derived. Illustrative examples showing the system performance as well as filter transfer functions are given. The performance results are compared to the rate distortion curves.
ICASSP98 Paper Abstract

Uniform Filter Banks with Nonuniform Bands: Post-Processing Design

Authors:
Ricardo L. De Queiroz, Xerox Corporation, (U.S.A.)
Volume 3, page 1341, paper no. 1083

Abstract:
In this paper, uniform, critically decimated filter banks are used to approximate nonuniform filter banks wherein different filters have approximately the same magnitude response, but different phase, thus forming a linear periodically time-varying filter whose characteristics are similar to those of a nonuniform bank. This is done by post-processing a number of selected subbands of a uniform bank using a special synthesis filter bank, which combines the selected bands into one. Design methods for the post-processing stage are discussed and design examples are presented.
ICASSP98 Paper Abstract

Linear/Nonlinear Adaptive Polyphase Subband Decomposition Structures for Image Compression

Authors:
Omer N Gerek, Bilkent University, (Turkey)
A. Enis Cetin, Bilkent University, (Turkey)

Volume 3, page 1345, paper no. 1248

Abstract:
Subband decomposition techniques have been extensively used for data coding and analysis. In most filter banks, the goal is to obtain subsampled signals corresponding to different spectral bands of the original data. However, this approach leads to various artifacts in images containing text, subtitles, or sharp edges. In this paper, adaptive filter banks with perfect reconstruction property are presented for such images. The filters of the decomposition structure vary according to the nature of the signal. This leads to higher compression ratios for images containing subtitles compared to fixed filter banks. Simulation examples are presented.
ICASSP98 Paper Abstract

Lossless Image Compression Using Integer Coefficient Filter Banks and Class-wise Arithmetic Coding

Authors:
Ilangko Balasingham, Norwegian University of Science and Technology, (Norway)
John M. Lervik, Fast Internet Transfer AS, (Norway)
Tor A. Ramstad, University of California, Santa Barbara, (U.S.A.)

Volume 3, page 1349, paper no. 1280

Abstract:
A novel way of constructing integer coefficient 2-channel filter banks is proposed. A set of relationships among the filter coefficients is established in order to satisfy linear phase, perfect reconstruction, and FIR properties. The remaining degrees of freedom are used to obtain integer coefficient values by maximizing a performance evaluation function, namely subband coding gain. The number of bits required to represent the subband samples is kept low through efficient nonlinear implementation techniques. An octave-band frequency partitioning where the number of stages is determined according to the image size is employed. The subband samples are then classified into one out of a finite number of classes, and each class is coded by an arithmetic coder. The obtained compression ratios are encouraging compared to the “best” results reported so far in the literature.
ICASSP98 Paper Abstract

Globally Optimal Two Channel FIR Orthonormal Filter Banks Adapted to the Input Signal Statistics

Authors:
Jamal Tuqan, California Institute of Technology, (U.S.A.)
Palghat P. Vaidyanathan, California Institute of Technology, (U.S.A.)

Volume 3, page 1353, paper no. 1846

Abstract:
We introduce a new approach to adapt a two-channel FIR orthonormal filter bank to the input second order statistics. The problem is equivalent to optimizing the magnitude squared response of one of the subband filters for maximum energy compaction under the constraint that it is Nyquist(2). The novel algorithm enjoys important advantages that are not present in previous work. First, we can ensure the positivity of the resulting magnitude squared response over all frequencies simultaneously with the Nyquist constraint. Second, for a fixed input power spectrum, the resulting magnitude squared response is guaranteed to be a global optimum due to the convexity of the new formulation. The optimization problem is expressed as a multi-objective semi definite programming problem which can be solved efficiently and with great accuracy using recently developed interior point methods. Third, the new algorithm is extremely general in the sense that it works for any arbitrary filter order N and any given input power spectrum. Finally, obtaining the subband filter from its magnitude squared response does not require an additional spectral factorization step.
ICASSP98 Paper Abstract
Fast, Accurate Subspace Tracking Using Operator Restriction Analysis

Authors:
Craig S MacInnes, NUWC, (U.S.A.)
Volume 3, page 1357, paper no. 1081

Abstract:
A new noniterative subspace tracking method is presented. This method is called operator restriction analysis (OPERA) and it can be used whenever an update to the principal components of an EVD or SVD of a rank-one update of a given matrix are needed. The updating algorithms are based on the technique of restricting a linear operator to a subspace, and the two concepts of invariant subspace updating and its generalization, singular subspace updating. The algorithm is shown to be as accurate as an EVD or SVD. An application is made to bearing estimation of highly nonstationary sources. Similarities and differences with other subspace tracking algorithms are discussed. Flop counts, tracking accuracy and signal subspace accuracy for OPERA are compared with other fast algorithms and with the EVD.
ICASSP98 Paper Abstract

Recursive Methods for Estimating Multiple Missing Values of a Multivariate Stationary Process

Authors:
Pascal Bondon, CNRS, (France)
Diego P. Ruiz, Universidad de Granada, (Spain)
Antolino Gallego, Universidad de Granada, (Spain)

Volume 3, page 1361, paper no. 1705

Abstract:
Existing methods for estimating linearly s future values of a m-variate stationary random process using a record of p vectors from the past consist in first solving the one-step prediction problem and then all the h-step prediction problems for $h > 1$ independently. When the Levinson algorithm is used, each prediction problem is solved with a numerical complexity proportional to $p^2$. In this paper, we propose new methods to solve the h-step prediction problems for $h > 1$ with a numerical complexity proportional to p.
ICASSP98 Paper Abstract

A Generalized Schur Algorithm in the Krein Space and its Application to H-Infinity Filtering

Authors:
Kyungsup Kim, KAIST, (Korea)
Joohwan Chun, KAIST, (Korea)

Volume 3, page 1365, paper no. 1996

Abstract:
This paper introduces a generalized Schur algorithm in the Krein space with an indefinite inner product. Concepts such as Carathéodory classes and Schur classes used in the classical Schur algorithm cannot be applied in the Krein space since the positive-definiteness corresponds merely to the nonsingularity in the Krein space. We note also that these problems appear when fast algorithms for suboptimal $H^\infty$ filtering are implemented. We shall derive the extended Chandrasekhar algorithm which is a fast implementation of $H^\infty$ filtering, and explain the connection between the generalized Schur algorithm and the Chandrasekhar algorithm. Using this result it is possible to derive a fast algorithm for suboptimal $H^\infty$ filtering.
ICASSP98 Paper Abstract

Matrix Sign Algorithm for Sinusoidal Frequency and DOA Estimation Problems

Authors:
Mohammed A. Hasan, University of Minnesota, Duluth, (U.S.A.)
Jawad A. K. Hasan, University of Baghdad, (Iraq)

Volume 3, page 1369, paper no. 2022

Abstract:
Fast algorithms based on the matrix sign function are developed to estimate the signal and noise subspaces of the samole correlation matrices. These subspaces are then utilized to develop high resolution methods such as MUSIC and ESPRIT for sinusoidal frequency and direction of arrival problems. The main feature of these algorithms is that they generate subspaces that are parameterized by the signal-to-noise ratio. Significant computational saving will be obtained due to the fast convergence of these higher order iterations and to the fact that subspaces rather than individual eigenvectors are actually computed. Simulations showing the performance of these methods were also presented.
A Delta Least Squares Lattice Algorithm for Fast Sampling

Authors:
Parthapratim De, University of Cincinnati, (U.S.A.)
Howard Fan, University of Cincinnati, (U.S.A.)

Volume 3, page 1373, paper no. 1828

Abstract:
Most shift operator-based adaptive algorithms exhibit poor numerical behavior when the input discrete time process is obtained from a continuous time process by fast sampling. This includes the shift operator based least squares lattice algorithm. In this paper, we develop a delta least squares lattice algorithm. This algorithm has low computational complexity compared to the delta Levinson RLS algorithm and shows better numerical properties compared to the shift least squares lattice algorithm. Computer simulations show that the new algorithm also outperforms an existing delta least squares lattice algorithm.
ICASSP98 Paper Abstract
A New DCT Algorithm Based on Encoding Algebraic Integers

Authors:
Vassil S Dimitrov, University of Windsor, (Canada)
Graham A Jullien, University of Windsor, (Canada)
William C Miller, University of Windsor, (Canada)

Volume 3, page 1377, paper no. 1906

Abstract:
In this paper we introduce an algebraic integer encoding scheme for the basis matrix elements of 8X8 DCTs and IDCTs. In particular, we encode the function cos(pi/16) and generate the other matrix elements using standard trigonometric identities. This encoding technique eliminates the requirement to approximate the matrix elements; rather we use algebraic 'placeholders' for them. Using this encoding scheme we are able to produce a multiplication free implementation of the Feig-Winograd algorithm.
ICASSP98 Paper Abstract

FFTW: An Adaptive Software Architecture for the FFT

Authors:
Matteo Frigo, MIT LCS, (U.S.A.)
Steven G Johnson, MIT, (U.S.A.)

Volume 3, page 1381, paper no. 2136

Abstract:
FFT literature has been mostly concerned with minimizing the number of floating-point operations performed by an algorithm. Unfortunately, on present-day microprocessors this measure is far less important than it used to be, and interactions with the processor pipeline and the memory hierarchy have a larger impact on performance. Consequently, one must know the details of a computer architecture in order to design a fast algorithm. In this paper, we propose an adaptive FFT program that tunes the computation automatically for any particular hardware. We compared our program, called FFTW, with over 40 implementations of the FFT on 7 machines. Our tests show that FFTW’s self-optimizing approach usually yields significantly better performance than all other publicly available software. FFTW also compares favorably with machine-specific, vendor-optimized libraries.
ICASSP98 Paper Abstract
Mapped Inverse Discrete Wavelet Transform for Data Compression

Authors:
Haitao Guo, Chromatic Research, ( U.S.A.)

Volume 3, page 1385, paper no. 2317

Abstract:
The Discrete Wavelet Transform (DWT) has been applied to data compression to decorrelate the data and concentrate the energy in a small portion of the coefficients. Compression can be achieved since most of the quantized wavelet coefficients are zeros. For the decoder, the traditional inverse discrete wavelet transform (IDWT) has a complexity proportional to the size of the data. In this paper, we propose a mapped inverse discrete wavelet transform algorithm (MIDWT) that takes advantage of the sparsity of the quantized wavelet coefficients, and significantly lowers the complexity of the IDWT to the level that is proportional to the number of non-zero coefficients. We further generalize the MIDWT to progressive decoding, and propose a realization of progressive IDWT without any run-time multiplication operations. Experiments show that our algorithms outperform the traditional IDWT for sparse coefficients, especially for progressive decompression.
ICASSP98 Paper Abstract
A Fast Orthogonal Matching Pursuit Algorithm

Authors:
Mohammad Gharavi-Alkhansari, University of Illinois, Urbana-Champaign, (U.S.A.)
Thomas S. Huang, University of Illinois, Urbana-Champaign, (U.S.A.)

Volume 3, page 1389, paper no. 2338

Abstract:
The problem of optimal approximation of members of a vector space by a linear combination of members of a large overcomplete library of vectors is of importance in many areas including image and video coding, image analysis, control theory, and statistics. Finding the optimal solution in the general case is mathematically intractable. Matching pursuit, and its orthogonal version, provide greedy solutions to this problem. Orthogonal matching pursuit typically provides significantly better solution compared to the nonorthogonal version, but requires much more computation. This paper presents a fast algorithm for implementation of orthogonal matching pursuit which for many coding applications has computational complexity very close to that of the nonorthogonal version.
ICASSP98 Paper Abstract

Four Step Genetic Search for Block Motion Estimation

Authors:
Man F So, City University of Hong Kong, (Hong Kong)
Angus Wu, City University of Hong Kong, (Hong Kong)

Volume 3, page 1393, paper no. 1886

Abstract:
Genetic Algorithms (GA) is well known for searching global maxima and minima. In general, number of search points required by GA for searching global extreme is much lower than the exhausted search. GA has been applied to Block Matching Algorithm (BMA) and demonstrates positively its capability in BMA. The mean square error (MSE) performance of GA based BMA is close to full search (FS). However, the disadvantage of GA is the computational requirement for practical use. A four-step genetic search algorithm is proposed to BMA. The proposed method takes advantage of GA and 4SS. The simulation result shows the proposed method has similar performance to full-search (FS) in terms of MSE. In addition, number of search points required by the proposed algorithm is approximately equal to 14% of FS and closed to three-step search (3SS). The speed up ratio between proposed algorithm and FS is 5.6 times.
ICASSP98 Paper Abstract
Using Phase Information to Decorrelate the Filtered-X Algorithm

Authors:
Piet C.W. Sommen, Eindhoven University of Technology, (The Netherlands)
John Garas, Eindhoven University of Technology, (The Netherlands)

Volume 3, page 1397, paper no. 1691

Abstract:
A well known algorithm in the field of active noise control is the filtered-x algorithm. As known in literature, the convergence properties of an adaptive algorithm can be improved by decorrelating its input signal. In this paper, the decorrelation needed for the filtered-x algorithm is discussed with the help of block frequency domain adaptive filters. It is shown that decorrelation of not only the input signal but also the amplitude response of the secondary acoustic path is necessary. While the former can be done by dividing the input signal in frequency domain by an estimate of the input signal power, the latter leads to a new method for improving convergence properties without any extra computation; by using only the phase information of the secondary path to calculate the filtered-x signal.
ICASSP98 Paper Abstract

A Recursive Total Least Squares Algorithm for Deconvolution Problems

Authors:
Piet Vandaele, ESAT/SISTA K.U. Leuven, (Belgium)
Marc Moonen, ESAT/SISTA K.U. Leuven, (Belgium)

Volume 3, page 1401, paper no. 1725

Abstract:
Deconvolution problems are encountered in signal processing applications where an unknown input signal can only be observed after propagation through one or more noise corrupted FIR channels. The first step in recovering the input usually entails an estimation of the FIR channels through training based or blind algorithms. The 'standard' procedure then uses least squares estimation to recover the input. A recursive implementation with constant computational cost is based on the Kalman filter. In this paper we focus on a total least squares based approach, which is more appropriate if errors are expected both on the output samples and the estimates of the FIR channels. We will develop a recursive total least squares algorithm (RTLS) which closely approximates the performance of the non-recursive TLS algorithm and this at a much lower computational cost.
ICASSP98 Paper Abstract

Computationally Efficient Algorithms for Third Order Adaptive Volterra Filters

Authors:
Xiaohui Li, University of Illinois, (U.S.A.)
William Kenneth Jenkins, University of Illinois, (U.S.A.)
Charles W. Therrien, Naval Postgraduate School, (U.S.A.)

Volume 3, page 1405, paper no. 5029

Abstract:
The input autocorrelation matrix for a third order (cubic) Volterra adaptive filter for general colored Gaussian input processes is analyzed to determine how to best formulate a computationally efficient fast adaptive algorithm. When the input signal samples are ordered properly within the input data vector, the autocorrelation matrix of the cubic filter inherits a block diagonal structure, with some of the sub-blocks also having diagonal structure. A computationally efficient adaptive algorithm is presented that takes advantage of the sparsity and unique structure of the correlation matrix that results from this formulation.
Numerical Stability Issues of the Conventional Recursive Least Squares Algorithm

Authors:
Athanasios P Liavas, Institut National des Telecommunications, (France)
Phillip A Regalia, Institut National des Telecommunications, (France)

Volume 3, page 1409, paper no. 1376

Abstract:
The continuous use of adaptive algorithms is strongly dependent on their behavior in finite-precision environments. We study the nonlinear round-off error accumulation system of the conventional RLS algorithm and we derive bounds for the relative precision of the computations which guarantee the numerical stability of the finite-precision implementation of the algorithm. The bounds depend on the conditioning of the problem and the exponential forgetting factor. Simulations agree with our theoretical results.
ICASSP98 Paper Abstract

A Combined FDAF/WSAF Algorithm for Stereophonic Acoustic Echo Cancellation

Authors:
Florence Alberge, Ecole Nationale Superieure des Telecommunications, (France)
Pierre Duhamel, Ecole Nationale Superieure des Telecommunications, (France)
Yves Grenier, Ecole Nationale Superieure des Telecommunications, (France)

Volume 3, page 1413, paper no. 2000

Abstract:
Adaptive acoustic echo cancellation in stereophonic teleconferencing is a very demanding application. Characteristics are: very large number of coefficients, non-stationary input (speech), (slowly) time-varying systems to be identified, plus the specific property that both stereo signals are intrinsically very correlated. Basic versions of stochastic gradient algorithms have difficulties to meet these requirements. We show that, in a multi-channel framework, only a combination of techniques can result in an algorithm which convergence is governed by a quasi-diagonal matrix. Simulations with data recorded in a conference room demonstrate the improvement in convergence of our algorithm compared to the LMS.
ICASSP98 Paper Abstract

Bounding the Performance of the LMS Estimator for Cases where Performance Exceeds that of the Finite Wiener Filter

Authors:
Kevin J Quirk, University of California, San Diego, (U.S.A.)
James R Zeidler, University of California, San Diego, (U.S.A.)
Laurence B Milstein, University of California, San Diego, (U.S.A.)

Volume 3, page 1417, paper no. 1620

Abstract:
The least-mean-square (LMS) estimator is a nonlinear estimator with information dependencies spanning the entire set of data fed into it. The traditional analysis techniques which are used to model this estimator obscure this, restricting the estimator to the finite set of data sufficient to span the length of its filter. The finite Wiener filter is thus often considered a bound on the performance of the LMS estimator. Several papers have reported the performance of the LMS filter exceeding that of the finite Wiener filter. In this paper, we will demonstrate a bound on the LMS estimator, which does not exclude the contributions from data outside its filter length, and which demonstrates the ability of the LMS estimator to outperform the finite Wiener filter in certain cases.
ICASSP98 Paper Abstract

Stochastic Analysis of Gradient Adaptive Identification of Nonlinear Systems with Memory

Authors:
Neil J. Bershad, University of California, Irvine, (U.S.A.)
Patrick Celka, EPFL, (Switzerland)
Jean-Marc Vesin, EPFL, (Switzerland)

Volume 3, page 1421, paper no. 1017

Abstract:
This paper analyzes the statistical behavior of a sequential gradient search adaptive algorithm for identifying an unknown nonlinear system comprised of a discrete-time linear system $H$ followed by a zero-memory nonlinearity $g(.)$. The LMS algorithm first estimates $H$. The weights are then frozen. Recursions are derived for the mean and fluctuation behavior of LMS which agree with Monte Carlo simulations. When the nonlinearity is modeled by a scaled error function, the second part of the gradient scheme is shown to correctly learn the scale factor and the error function scale factor. Mean recursions for the scale factors show good agreement with Monte Carlo simulations.
ICASSP98 Paper Abstract

An Unbiased Equation-Error-Based Adaptive IIR Filtering Algorithm

Authors:
Sundar G Sankaran, Virginia Tech, (U.S.A.)
Louis A Beex, Virginia Tech, (U.S.A.)

Volume 3, page 1425, paper no. 1413

Abstract:
We modify the off-line system identification procedure proposed by Regalia into an adaptive IIR filtering algorithm based on the stochastic gradient method. The proposed algorithm aims to minimize equation error, recursively, under a unit-norm constraint on the characteristic polynomial instead of the usual monic constraint. The unit-norm constraint eliminates the bias associated with equation error based estimates, when the additive measurement noise is white. The unit-norm constraint is enforced by adapting the parameters of the characteristic polynomial in (hyper)spherical coordinates. Simulation results indicate that the proposed algorithm provides estimates that are unbiased and that it is a computationally efficient alternative, for the same performance, to FIR adaptive filters.
ICASSP98 Paper Abstract
Design of Causal IIR Perfect Reconstruction Filter Banks

Authors:
Xi Zhang, Nagaoka University of Technology, (Japan)
Toshinori Yoshikawa, Nagaoka University of Technology, (Japan)

Volume 3, page 1429, paper no. 1073

Abstract:
This paper presents a new method for designing two channel biorthogonal IIR filter banks, which satisfy both the perfect reconstruction and causal stable conditions. The proposed method is based on the formulation of a generalized eigenvalue problem by using Remez multiple exchange algorithm. Therefore, the filter coefficients can be computed by solving the eigenvalue problem, and the optimal solution is easily obtained through a few iterations. One example is designed to demonstrate the effectiveness of the proposed method.
ICASSP98 Paper Abstract
Discrete-Coefficient Linear-Phase Prototypes for PR Cosine-Modulated Filter Banks

Authors:
Alfred Mertins, University of Western Australia, (Australia)

Volume 3, page 1433, paper no. 1198

Abstract:
In this paper, a method for the design of perfect reconstruction (PR) linear-phase prototypes for cosine-modulated filter banks with discrete coefficients is presented. Such prototypes are of great interest for efficient hardware implementations. The design procedure is based on a subspace approach that allows to linearly combine PR prototype filters in such a way that the resulting filter also is a PR prototype. Within a given subspace the weights of the optimal linear combination can be easily computed via an eigenanalysis. The filter design is carried out iteratively, while the PR property is guaranteed throughout the design process. No non-linear optimization routine is needed.
ICASSP98 Paper Abstract
Design of Perfect Reconstruction FIR Multifilters

Authors:
C.W. Kok, University of Wisconsin-Madison, (U.S.A.)

Volume 3, page 1437, paper no. 1222

Abstract:
The design of perfect reconstruction FIR multifilters is discussed in this paper. Schur algorithm is applied to factorize the polyphase matrix of multifilters into lattice blocks. The multifilters are characterized by the chain parameters in each lattice block. The complete parameterization of paraunitary multifilters and a class of biorthogonal multifilters are derived. The parameterizations are minimal and result in simple design methods using unconstrained optimization.
ICASSP98 Paper Abstract
The Generalized Lapped Biorthogonal Transform

Authors:
Trac D Tran, University of Wisconsin, (U.S.A.)
Ricardo L De Queiroz, Xerox Corporation, (U.S.A.)
Truong Q Nguyen, Boston University, (U.S.A.)

Volume 3, page 1441, paper no. 1289

Abstract:
A lattice structure based on the singular value decomposition (SVD) is introduced. The lattice can also be proven to use a minimal number of delay elements and to completely span a large class of M-channel linear phase perfect reconstruction filter bank (LPPRFB): all analysis and synthesis filters have the same FIR length of L=KM, sharing the same center of symmetry. The lattice also structurally enforces both linear phase and perfect reconstruction properties, is capable of providing fast and efficient implementation, and avoids the costly matrix inversion problem in the optimization process. From a block transform perspective, the new lattice represents a family of generalized lapped biorthogonal transform (GLBT) with arbitrary integer overlapping factor K. The relaxation of the orthogonal constraint allows the GLBT to have significantly different analysis and synthesis basis functions which can then be tailored appropriately to fit a particular application. Several design examples are presented along with a high-performance GLBT-based progressive image coder to demonstrate the superiority of the new lapped transforms.
ICASSP98 Paper Abstract

Cyclic LTI Systems and the Paraunitary Interpolation Problem

Authors:
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Ahmet Kirac, Caltech, (U.S.A.)

Volume 3, page 1445, paper no. 1298

Abstract:
Cyclic signal processing refers to situations where all the time indices are interpreted modulo some integer L. Since the frequency domain is a uniform discrete grid, there is more freedom in theoretical and design aspects. The basics of cyclic(L) multirate systems and filter banks have already appeared in the literature, and important differences between the cyclic and noncyclic cases are known. Since there is a strong connection between paraunitary filter banks and orthonormal wavelets, some deeper questions pertaining to cyclic(L) paraunitary matrices are addressed in this paper. It is shown that cyclic(L) paraunitary matrices do not in general have noncyclic paraunitary FIR interpolants, though IIR interpolants can always be constructed. It is shown, as a consequence, that cyclic paraunitary systems cannot in general be factored into degree one nonrecursive paraunitary building blocks. The connection to unitariness of the cyclic state space realization is also addressed.
ICASSP98 Paper Abstract

Factorization of Nonuniform Block Orthogonal Transforms

Authors:
Takayuki Nagai, Keio University, (Japan)
Shigeki Obata, Keio University, (Japan)
Masaaki Ikehara, Keio University, (Japan)

Volume 3, page 1449, paper no. 1321

Abstract:
Block orthogonal transforms (BOT's) are commonly used for lots of applications. Conventional BOT's are based on uniform filter banks, however, nonuniform BOT's are often superior to uniform ones. In this research, we investigate the factorization of nonuniform BOT's which does not involve the tree structure. Therefore, optimal nonuniform BOT's are available in the sense of transform coding gain. Some design examples are included to confirm our theory. We also apply the nonuniform BOT to the transform image coding.
ICASSP98 Paper Abstract

Design of Biorthogonal Filter Banks Composed of Linear Phase IIR Filters

Authors:
Masahiro Okuda, Keio University, (Japan)
Masaaki Ikehara, Keio University, (Japan)
Shin-ichi Takahashi, Keio University, (Japan)

Volume 3, page 1453, paper no. 1368

Abstract:
Since IIR filters have lower computational complexity than FIR filters, some design methods for IIR filter banks have been presented in the recent literatures. Smith et al. have proposed a class of linear phase IIR filter banks. However this method restricts the order of the numerator to be odd and ,moreover ,has some drawbacks. In this paper we present two design methods for linear phase IIR filter banks. One is based on Lagrange-Multiplier method, in which optimal IIR filter banks in least squares sense are obtained. In the other approach, IIR filter banks with the maximum number of zeros are derived analytically.
ICASSP98 Paper Abstract

Biorthogonal Cosine-Modulated Filter Banks without DC Leakage

Authors:
Tanja Karp, University of Mannheim, (Germany)
Alfred Mertins, University of Western Australia, (Australia)

Volume 3, page 1457, paper no. 1371

Abstract:
In this paper, we present a structure for implementing the polyphase filters of biorthogonal modulated filter banks that automatically guarantees perfect reconstruction of the filter bank and furthermore allows to specify the values of the filters' frequency responses at certain frequencies. Thus, modulated filter banks without DC leakage can be designed. The new structure is based on lifting schemes for the polyphase filters. DC leakage can be avoided very easily by reducing the number of lifting coefficients that can be freely chosen and used for filter optimization. The great advantage of the new method is that we do not have to take constraints into consideration when optimizing the prototype filter, but PR and specified zeros are structure inherent.
ICASSP98 Paper Abstract

Performance Measures and Lagrange Multiplier Methods to Two-Band PR LP Filter Bank Design

Authors:
Tao Wang, University of Illinois, (U.S.A.)
Benjamin W. Wah, University of Illinois, (U.S.A.)

Volume 3, page 1461, paper no. 5245

Abstract:
In this paper, we apply a new adaptive Lagrangian method for designing QMF (quadrature-mirror-filter) filter banks. We formulate the design problem as a nonlinear constrained optimization problem, using the reconstruction error as the objective, and other performance metrics as constraints. This formulation allows us to search for designs that improve over the best existing designs. We propose to solve the design problem using Lagrangian methods, and study methods to improve the convergence speed of Lagrangian methods without affecting their solution quality. This is done by adjusting dynamically the relative weights between the objective and the Lagrangian part. We show that our adaptive method is able to find better designs within a reasonable amount of time that would not be possible otherwise.
ICASSP98 Paper Abstract

A New Implementation of Arbitrary-Length Cosine-Modulated Filter Bank

Authors:
Xi-Qi Gao, Southeast University, (China)
Zhen-Ya He, Southeast University, (China)
Xiang-Gen Xia, University of Delaware, (U.S.A.)

Volume 3, page 1465, paper no. 5019

Abstract:
In this paper, the fast implementation of cosine-modulated filter bank (CMFB) is revisited. A class of paraunitary CMFBs with arbitrary length is considered. By further reorganizing of the polyphase component matrix and using the linear-phase property of the prototype filter, we obtain a more efficient implementation structure for the CMFB, in which we use 2×2 lossless matrices instead of 2×1 ones. In the new implementation, the number of two-channel lossless lattices is reduced by a factor of two.
ICASSP98 Paper Abstract

A New Adaptive Notch Filter with Constrained Poles and Zeros Using Steiglitz-McBride Method

Authors:
Mu-Huo Cheng, National Chiao-Tung University, (Taiwan)
Jau-Long Tsai, National Chiao-Tung University, (Taiwan)

Volume 3, page 1469, paper no. 1072

Abstract:
In this paper we present a new adaptive notch filter (ANF) using the well-known Steiglitz-McBride method (SMM) for an IIR filter with the constrained poles and zeros. The proposed ANF, termed as SMM-ANF, converges to the unbiased solution, has fast convergence speed, and requires less computational complexity than existing recursive maximum likelihood adaptive notch filters (RML-ANF). In the stationary environments, we analyze SMM-ANF convergence properties using the ordinary differential equation (ODE) technique; we derive conditions for the SMM-ANF convergence solution unbiased. Simulations further display that SMM-ANF has better resolution in identifying frequencies of multiple sine waves than RML-ANF. In the nonstationary environments, we also show that SMM-ANF and RML-ANF have approximately identical tracking performance. Simulations are also done to verify the theoretically derived results.
ICASSP98 Paper Abstract

QRD-Based LSL Interpolators – Part II: A QRD-Based LSL Interpolation Algorithm

Authors:
Ying-Wen Bai, Fu Jen Catholic University, (Taiwan)
Jenq-Tay Yuan, Fu Jen Catholic University, (Taiwan)

Volume 3, page 1473, paper no. 1018

Abstract:
In this paper, we derive a QRD-LSL interpolation algorithm that can be used to construct order-recursive QRD-LSL interpolators based on the exact decoupling property developed in a companion paper. QRD-LSL predictors are well known and use past data samples to predict the present data sample while the QRD-LSL interpolators use both past and future data samples to estimate the present data sample. Except for an overall delay needed for physical realization, QRD-LSL interpolators may achieve much better performance than that of the QRD-LSL predictors.
An Adaptive, High-Order, Notch Filter Using All-Pass Sections

Authors:
Sebastian Torres, *University of Oklahoma*, (U.S.A.)
Victor E De Brunner, *University of Oklahoma*, (U.S.A.)

Abstract:
A fully adaptive infinite impulse response notch filter in cascade form is proposed to detect and track multiple time-varying frequencies in additive white noise. Based on transformations for digital filters in the frequency domain, the filter results in a minimal number of parameters. In addition, a simple adaptive algorithm with good tracking and convergence properties is obtained by using all-pass filters and truncating the gradient. Computer simulations are included to verify the competitive performance of this filter under a wide range of conditions. From this analysis, we conclude that our new design is computationally simple, achieves rapid convergence, and is consequently a good choice in many non-stationary environments.
ICASSP98 Paper Abstract

A Fast Instrumental Variable Affine Projection Algorithm

Authors:
Karim Maouche, Institut Eurecom, (France)
Dirk T.M. Slock, Institut Eurecom, (France)

Volume 3, page 1481, paper no. 2386

Abstract:
We derive a new adaptive filtering algorithm called the Instrumental Variable Affine Projection (IVAP) algorithm and give its fast version (FIVAP 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 FIVAP algorithm whose computational complexity is nearly the same as the one of the FAP algorithm.
ICASSP98 Paper Abstract

A New Definition of Continuous Fractional Hartley Transform

Authors:
Soo-Chang Pei, National Taiwan University, (Taiwan)
Chien-Cheng Tseng, Hwa Hsia College, (Taiwan)
Min-Hung Yeh, National Taiwan University, (Taiwan)
Jian-Jiun Ding, National Taiwan University, (Taiwan)

Volume 3, page 1485, paper no. 1043

Abstract:
This paper is concerned with the definition of the continuous fractional Hartley transform. First, a general theory of linear fractional transform is presented to provide a systematic procedure to define the fractional version of any well-known linear transforms. Then, the results of general theory are used to derive the definitions of fractional Fourier transform (FRFT) and fractional Hartley transform (FRHT) which satisfy the boundary conditions and additivity property simultaneously. Next, an important relationship between FRFT and FRHT is described. Finally, a numerical example is illustrated to demonstrate the transform results of delta function of FRHT.
ICASSP98 Paper Abstract

Distributed Adaptive Algorithms for Large Dimensional MIMO Systems

Authors:
Barry D Van Veen, University of Wisconsin, (U.S.A.)
Olivier Leblond, University of Wisconsin, (U.S.A.)
Vijay P Mani, University of Wisconsin, (U.S.A.)
Daniel J Sebald, University of Wisconsin, (U.S.A.)

Volume 3, page 1489, paper no. 1579

Abstract:
A distributed algorithm for MIMO adaptive filtering is introduced. This algorithm distributes the adaptive computation over a set of linearly connected computational modules. Each module transmits data to and receives data from its nearest neighbor. A back-propagation LMS based algorithm is presented for adapting the parameters in each module. The performance surface is explored to identify upper bounds on each parameter and guidelines for choosing the LMS algorithm step sizes. An example illustrates application of the algorithm.
ICASSP98 Paper Abstract

Log Adaptive Filters - Structures and Analysis for the Scalar Case

Authors:
Michael Rakijas, University of California, Irvine, (U.S.A.)
Neil J. Bershad, University of California, Irvine, (U.S.A.)

Volume 3, page 1493, paper no. 1037

Abstract:
Feed-forward multi-layer neural networks (MLNN’s) are complex nonlinear learning systems which can be trained by well-known rules such as back-propagation (BP). The resulting adaptation procedures are extremely difficult to analyze for stochastic training data. Significant analytic results have been obtained for the single-layer case and for some simple two-layer cases. Recently, a structural simplification has been studied which models each threshold function as a linear device. This linearized MLNN can only create hyperplane decision rules after convergence. However, the multiplicative behavior of the layers may offer some performance advantages over linear adaptive algorithms (LMS or RLS) when used for a linear problem. A new log-domain linear MLNN adaptive structure is proposed and analyzed here. The log operation converts the layer multiplications into additions whereupon linear analysis techniques can be used. The transient and steady-state statistical behavior of the log linear MLNN is analyzed for Gaussian training data. Deterministic recursions are derived for the mean and fluctuation behavior of the new algorithm. These recursion are shown to be in excellent agreement with Monte Carlo simulations.
A New QRD-Based Block Adaptive Algorithm

Authors:
Mounir Bhouri, Universite Rene Descartes, (France)
Madeleine Bonnet, Universite Rene Descartes, (France)
Mamadou Mboup, Universite Rene Descartes, (France)

Abstract:
In this paper we present a new robust adaptive algorithm. It is derived from the standard QR Decomposition based RLS (QRD-RLS) algorithm by introducing a non-orthogonal transform into the update recursion. Instead of updating an upper triangular matrix, as it is the case for the QRD-RLS, we adapt an upper triangular block diagonal matrix. The complexity of the algorithm, thus obtained, varies from $O(N^2)$ to $O(N)$ when the size of the diagonal blocks decreases. Simulations of the new algorithm have shown a better robustness than the standard QRD-based algorithm in the context of multichannel adaptive filtering with highly intercorrelated channels.
ICASSP98 Paper Abstract

Novel Cost Function Adaptation Algorithm for Echo Cancellation

Authors:
Colin F.N. Cowan, *The Queens University of Belfast, (Northern Ireland)*
Corneliu G Rusu, *Tampere University of Technology, (Finland)*

Volume 3, page 1501, paper no. 1898

Abstract:
A new stochastic gradient algorithm for data echo cancellation, based on the cost function adaptation (CFA) is proposed. Qualities of the new adaptation algorithm as compared with that of the least mean square (LMS) and the least mean fourth (LMF) algorithms are demonstrated by means of simulations. Thus it is shown that continuous and automatic, adaptation of the error power yields a more satisfactory result. The cost function adaptation allows an increase in convergence rate and, at the same time, an improvement of residual error. The results were obtained with non-Gaussian binary sequences of data in presence of far-end signals in data echo-cancellers for full duplex digital data transmission over telephone lines.
ICASSP98 Paper Abstract

On the Performance of an Adaptation of Adichie’s Rank Tests for Signal Detection: and its Relationship to the Matched Filter

Authors:
Christopher L Brown, Queensland University of Technology, (Australia)
Abdelhak M Zoubir, Queensland University of Technology, (Australia)
Boualem Boashash, Queensland University of Technology, (Australia)

Volume 3, page 1505, paper no. 1672

Abstract:
The Adichie rank test and signed rank test are adapted for signal detection. We establish a relationship with the correlation between a function of the signal to be detected and the ranks of the observed data. A comparison between the power of these tests and the constant false alarm rate matched filter (CFAR MF) shows that the rank tests perform better when longer observations are available and for the symmetric alpha stable distributions encountered in applications with impulsive interference.
ICASSP98 Paper Abstract

Critically Sampled Wavelet Representations for Multi-dimensional Signals with Arbitrary Regions of Support

Authors:
John G. Apostolopoulos, MIT, (U.S.A.)
Jae S. Lim, MIT, (U.S.A.)

Volume 3, page 1509, paper no. 5227

Abstract:
Transform/subband representations form an important element of many signal processing algorithms and applications. Until recently, representations have typically been designed for signals with convenient supports, e.g. 2-D signals with rectangular supports. However, a number of applications require representations for signals with arbitrary (non-rectangular) regions of support. We present a novel algorithm for creating critically sampled perfect reconstruction wavelet representations for signals defined over arbitrary supports. The proposed algorithm selects a subset of vectors from a convenient superset basis which under appropriate conditions provides a basis over the given arbitrary support. The algorithm can be interpreted as solving a corresponding sampling problem.
ICASSP98 Paper Abstract
Adaptive Wavelet Transforms via Lifting

Authors:
Roger L. Claypoole Jr, Rice University, (U.S.A.)
Richard Baraniuk, Rice University, (U.S.A.)
Robert D. Nowak, Michigan State University, (U.S.A.)

Volume 3, page 1513, paper no. 2177

Abstract:
This paper develops two new adaptive wavelet transforms based on the lifting scheme. The lifting construction exploits a spatial-domain, prediction-error interpretation of the wavelet transform and provides a powerful framework for designing customized transforms. We use the lifting construction to adaptively tune a wavelet transform to a desired signal by optimizing data-based prediction error criteria. The performances of the new transforms are compared to existing wavelet transforms, and applications to signal denoising are investigated.
ICASSP98 Paper Abstract
On Multiscale Wavelet Analysis for Step Estimation

Authors:
Brian M. Sadler, Army Research Laboratory, (U.S.A.)
Ananthram Swami, Army Research Laboratory, (U.S.A.)

Volume 3, page 1517, paper no. 1806

Abstract:
We consider step detection and estimation using a multiscale wavelet analysis, based on the ability of a certain discrete wavelet transform (DWT) to characterize signal steps and edges. This DWT, developed by Mallat and Zhong, estimates the gradient at various smoothing levels without downsampling in time. As first proposed by Rosenfeld for edge sharpening, multiple scales are combined by forming the pointwise product across scales. We show that this approach is a non-linear whitening transformation, and characterize the non-Gaussian pdf of the output. Detection curves are shown for parameterized sigmoidal step change signals. Step location estimation performance is also shown, with comparison to the Cramer-Rao bound in additive white Gaussian noise.
ICASSP98 Paper Abstract

A Memory System Supporting the SIMD Computation of the Two Dimensional DWT

Authors:
Maria A Trenas, University of Malaga, (Spain)
Juan Lopez, University of Malaga, (Spain)
Francisco Arguello, University of Santiago, (Spain)
Emilio L Zapata, University of Malaga, (Spain)

Volume 3, page 1521, paper no. 1775

Abstract:
Real time image processing uses SIMD engines to accelerate the computation of algorithms as DCT, FFT or DWT. So, a good skewing scheme becomes essential for avoiding memory bank conflicts. In this paper a memory system is introduced for the efficient in-place computation of such transforms. It consists of \( M = 2^m \) memory modules, providing parallel access to \( M \) image points whose patterns are a row or a column, the interval in both cases being \( 2^l, l \geq 0 \). The efficiency of our design is proved through the computation of the 2D-DWT.
ICASSP98 Paper Abstract
Multiresolution Sinusoidal Modeling Using Adaptive Segmentation

Authors:
Michael M Goodwin, University of California, Berkeley, (U.S.A.)
Volume 3, page 1525, paper no. 2412

Abstract:
The sinusoidal model has proven useful for representation and modification of speech and audio. One drawback, however, is that a sinusoidal signal model is typically derived using a fixed frame size, which corresponds to a rigid signal segmentation. For nonstationary signals, the resolution limitations that result from this rigidity lead to reconstruction artifacts. It is shown in this paper that such artifacts can be significantly reduced by using a signal-adaptive segmentation derived by a dynamic program. An atomic interpretation of the sinusoidal model is given; this perspective suggests that algorithms for adaptive segmentation can be viewed as methods for adapting the time scales of the constituent atoms so as to improve the model by employing appropriate time-frequency tradeoffs.
ICASSP98 Paper Abstract
High Order Balanced MultiWavelets

Authors:
Jerome Lebrun, Swiss Federal Institute of Technology, Lausanne, (Switzerland)
Martin Vetterli, Swiss Federal Institute of Technology, Lausanne, (Switzerland)

Abstract:
In this paper, we study the issue of regularity for multiwavelets. We generalize here the concept of balancing for higher degree discrete-time polynomial signals and link it to a very natural factorization of the lowpass refinement mask that is the counterpart of the well-known zeros at Pi condition for wavelets. This enables us to clarify the subtle relations between approximation power, smoothness and balancing order. Using these new results, we are also able to construct a family of orthogonal multiwavelets with symmetries and compact support that is indexed by the order of balancing. More details (filters coefficients, drawings of the whole family, frequency responses, etc.) can be obtained on the [WEB] at: http://lcavwww.epfl.ch/lebrun.
ICASSP98 Paper Abstract

Approximate Continuous Wavelet Transform with an Application to Noise Reduction

Authors:
James M Lewis, Rice University, (U.S.A.)
C. Sidney Burrus, Rice University, (U.S.A.)

Volume 3, page 1533, paper no. 1830

Abstract:
We describe a generalized scale-redundant wavelet transform which approximates a dense sampling of the continuous wavelet transform (CWT) in both time and scale. The dyadic scaling requirement of the usual wavelet transform is relaxed in favor of an approximate scaling relationship which in the case of a Gaussian scaling function is known to be asymptotically exact and irrational. This scheme yields an arbitrarily dense sampling of the scale axis in the limit. Similar behavior is observed for other scaling functions with no explicit analytic form. We investigate characteristics of the family of Lagrange interpolating filters (related to the Daubechies family of compactly-supported orthonormal wavelets), and finally present applications of the transform to denoising and edge detection.
ICASSP98 Paper Abstract

A General Approach to the Generation of Biorthogonal Bases of Compactly-Supported Wavelets

Authors:
Mitchell Oslick, Stanford University, (U.S.A.)
Ivan R. Linscott, Stanford University, (U.S.A.)
Snezana Maslakovic, Stanford University, (U.S.A.)
Joseph D. Twicken, Stanford University, (U.S.A.)

Volume 3, page 1537, paper no. 2220

Abstract:
Biorthogonal bases of compactly-supported wavelets are characterized by the FIR perfect-reconstruction filterbanks to which they correspond. In this paper we develop explicit representations of all such filterbanks, allowing us to generate every possible biorthogonal compactly-supported wavelet basis. For these filterbanks, the product of the two lowpass filters must have N zeros at \( z = -1 \), where \( N \) is two or more. There are \( N+1 \) minimal-length filterbanks for each \( N \). The filterbanks associated with standard orthogonal and symmetric biorthogonal wavelet bases are found as a special case by using appropriate factorizations of symmetric product filters with even \( N \); other filterbanks lead to novel biorthogonal bases.
Wavelet Systems with Zero Moments

Authors:
C. Sidney Burrus, Rice University, (U.S.A.)
Jan E. Odegard, Rice University, (U.S.A.)

Volume 3, page 1541, paper no. 2079

Abstract:
The Coifman wavelets created by Daubechies have more zero moments than imposed by specifications. This results in systems with approximately equal numbers of zero scaling function and wavelet moments and gives a partitioning of the systems into three well defined classes. The nonunique solutions are more complex than for Daubechies wavelets.
ICASSP98 Paper Abstract
Image Deblocking by Singularity Detection

Authors:
Tai-Chiu Hsung, The Hong Kong Polytechnic University, (Hong Kong)
Daniel Pak-Kong Lun, The Hong Kong Polytechnic University, (Hong Kong)

Volume 3, page 1545, paper no. 1422

Abstract:
Blocking effect is considered as the most disturbing artifact of JPEG decoded images. Many researchers have suggested various methods to tackle this problem. Recently, the wavelet transform modulus maxima (WTMM) approach was proposed and gives a significant improvement over the previous methods in terms of signal-to-noise ratio and visual quality. However, the WTMM deblocking algorithm is an iterative algorithm that requires a long computation time to reconstruct the processed WTMM to obtain the deblocked image. In this paper, a new wavelet based algorithm for JPEG image deblocking is proposed. The new algorithm is based on the idea that, besides using the WTMM, the singularity of an image can also be detected by computing the sums of the wavelet coefficients inside the so-called "directional cone of influence" in different scales of the image. The new algorithm has the advantage as the WTMM approach that it can effectively identify the edge and the smooth regions of an image irrespective the discontinuities introduced by the blocking effect. It improves over the WTMM approach in that only a simple inverse wavelet transform is required to reconstruct the processed wavelet coefficients to obtain the deblocked image. As the WTMM approach, the new algorithm gives consistent and significant improvement over the previous methods for JPEG image deblocking.
Continuous-Dilation Discrete-Time Self-Similar Signals and Linear Scale-Invariant Systems

Authors:
Wei Zhao, Rochester Institute of Technology, (U.S.A.)
Raghuveer M Rao, Rochester Institute of Technology, (U.S.A.)

Volume 3, page 1549, paper no. 1825

Abstract:
In this paper we present a novel model for purely discrete-time self-similar processes and scale-invariant systems. The results developed are based on a new interpretation of the discrete-time scaling (equivalently dilation or contraction) operation which is defined through a mapping between discrete and continuous time. It is shown that it is possible to have continuous scaling factors through this operation even though the signal itself is discrete-time. We study both deterministic and stochastic discrete-time self-similar signals. We then derive the existence conditions of discrete-time deterministically self-similar signals with respect to some specific mappings. Finally, we discuss the construction of discrete-time linear scale-invariant (LSI) system and present results related to white noise driven system models of stochastic self-similar signals. Unlike continuous-time self-similar signals, it is possible to construct a wide class of non-trivial discrete-time self-similar signals.
ICASSP98 Paper Abstract

Regionally Optimised Kernels for Time-Frequency Distributions

Authors:
Mark J Coates, Cambridge University, (U.K.)
Christophe Molina, Anglia University, (U.K.)
William J. Fitzgerald, Cambridge University, (U.K.)

Volume 3, page 1553, paper no. 1710

Abstract:
Ideally, kernels used to generate bilinear time-frequency distributions (TFD) should be signal-dependent, and optimised independently at every location in the time-frequency (TF) plane. This poses an extremely severe computational burden. A compromise is proposed in this paper: time-varying kernels are optimised for specific regions in the time-frequency plane. The regions, designed to isolate separate components comprising the signal, are determined by modelling the TFD using a finite mixture model of Gaussian distributions. The parameters of the model are estimated using a combination of the expectation-maximisation algorithm and functional merging. The regional optimisation provides improved separation and resolution of closely-spaced components when compared to methods using a solely time-varying kernel, without incurring an overwhelming computational expense.
ICASSP98 Paper Abstract
Optimum Signal Synthesis for Time-Scale Estimation

Authors:
Jean-Philippe Ovarlez, ONERA/DEMR, (France)

Volume 3, page 1557, paper no. 1208

Abstract:
In signal analysis, the joint estimation of the time-scale parameters which can affect a known signal (Doppler effect or scale effect, delay, ...) may be a problem of interest. An important result has shown that, even if the quality of the time delay estimation is classically given by the inverse spread of the signal spectral density, the quality of the scale estimation only depends on the inverse of the signal spread in Mellin space. This spread has a direct interpretation in the time-frequency plane and can be precisely estimated when duration, bandwidth and relative bandwidth of the signal are known. We propose here to develop two methods of optimum signal synthesis which minimize the variance of the estimates given by the Cramer-Rao lower bounds. The first method is based on the stationary phase principle, applied on frequency and Mellin spaces, which allows to construct signals with given autocorrelation functions in scale and time spaces. The second one is devoted to the construction of a frequency phase law depending on the mellin variable with the spreads in frequency and Mellin spaces related to the expected scale and time-delay resolutions.
Algorithm for Decomposing an Analytic Signal into AM and Positive FM Components

Authors:
Ramdas Kumaresan, University of Rhode Island, (U.S.A.)
Ashwin Rao, University of Rhode Island, (U.S.A.)

Volume 3, page 1561, paper no. 1226

Abstract:
An analytic signal permits unambiguous characterization of the phase and envelope of a real signal. But the analytic signal’s phase-derivative i.e., the instantaneous frequency (IF) is typically a wild function and can take on values ranging from negative infinity to positive infinity. Fortunately, any analytic signal can be decomposed into a minimum phase (MinP) signal component and an all-phase (AllP) signal component. While the MinP signal’s log-envelope and its phase form a Hilbert transform pair, the AllP signal has a positive definite instantaneous frequency (PIF), unlike that of the original analytic signal. We propose an elegant computational algorithm that separates the MinP and AllP components of the analytic signal. The envelope of the MinP component corresponds to the AM and the PIF of the AllP component corresponds to the positive FM.
ICASSP98 Paper Abstract
The Gabor Expansion Based Positive Distribution

Authors:
Flemming Pedersen, Aalborg University, (Denmark)

Volume 3, page 1565, paper no. 1255

Abstract:
This paper presents and studies a time frequency distribution obtained from a Gabor expansion of a signal. The distribution is named the Positive Gabor Spectrogram, and is a new positive-like distribution with correct marginal distributions. Two side effects of correct marginals are non-additivity and time frequency fading. These are phenomena of a statistical correct distribution which do not agree with our intuitive expectation of a time frequency representation.
ICASSP98 Paper Abstract

Classification of Transient Time-Varying Signals Using DFT and Wavelet Packet Based Methods

Authors:
Christoph M Delfs, Universität Karlsruhe, (Germany)
Friedrich M Jondral, Universität Karlsruhe, (Germany)

Volume 3, page 1569, paper no. 1205

Abstract:
The classification of transient time-varying signals is important for industrial, biomedical and military applications. The attack phase of piano sounds is used as an example for transient, time-varying signals in a real data application. Discrete Fourier transform and time-invariant wavelet packet based algorithms are used alternatively for feature extraction. The training set is used for determining an appropriate feature selection. A classifier checks whether the generated features are sufficient in order to identify the correct piano. Classification results are presented and discussed.
ICASSP98 Paper Abstract

New Concepts in Narrowband and Wideband Weyl Correspondence Time-Frequency Techniques

Authors:
Byeong-Gwan Iem, University of Rhode Island, (U.S.A.)
Antonia Papandreou-Suppappola, University of Rhode Island, (U.S.A.)
G. Faye Boudreaux-Bartels, University of Rhode Island, (U.S.A.)

Volume 3, page 1573, paper no. 2028

Abstract:
We propose the new Po-Weyl symbol to analyze system induced time shifts and scale changes on the input signal. This new Weyl symbol (WS) is useful in wideband signal analysis. We also propose new WS as tools for analyzing systems which produce dispersive frequency shifts on the signal. We obtain these generalized, frequency-shift covariant WS by warping conventional, narrowband WS. Using the new, generalized WS, we provide a formulation for the Weyl correspondence for linear systems with instantaneous frequency characteristics matched to user specified characteristics. We also propose a new interpretation of linear signal transformations as weighted superpositions of non-linear frequency shifts on the signal. Application examples in signal analysis and detection demonstrate the advantages of our new results.
The Generalization of the Wiener-Khinchin Theorem

Authors:
Leon Cohen, City University of New York, Hunter College, (U.S.A.)

Abstract:
We generalize the Wiener-Khinchin theorem. A full generalization is presented where both the autocorrelation function and power spectral density are defined in terms of a general basis set. In addition, we present a partial generalization where the density is the Fourier transform of the autocorrelation function but the autocorrelation function is defined in terms of an arbitrary basis set. Both the deterministic and random cases are considered.
ICASSP98 Paper Abstract
Distributions in the Discrete Cohen’s Classes

Authors:
Jeffrey C O’Neill, Ecole Normale Superieure de Lyon, (France)
William J Williams, University of Michigan, (U.S.A.)

Abstract:
Cohen's class of time-frequency distributions for continuous signals has recently been to extended to discrete signals using both an axiomatic approach and an operator theory approach. In this paper, we investigate the formulation of several classical time-frequency distributions (Wigner, Rihaczek, Margenau-Hill, Page, Levin, Born-Jordan, spectrogram) in the discrete Cohen’s classes. The main result of this paper concludes that there does not exist a formulation of the Wigner distribution in all of the discrete Cohen's classes.
ICASSP98 Paper Abstract

On the Relationship Between $1/f$ and Alpha-Stable Processes

Authors:
Jijun Yin, Drexel University, (U.S.A.)
Athina P Petropulu, Drexel University, (U.S.A.)

Volume 3, page 1585, paper no. 2241

Abstract:
$1/f^\beta$-type spectral behavior has received considerable attention in the past few years because it arises from a wide range of nature phenomena. By expressing a $1/f^\beta$ process as a fractional integral of white noise, we show that, if $\beta < 1$, the process is stationary and follows an $\alpha$-stable model, while if $\beta > 1$, the process has stationary alpha-stable increments. We also provide closed form expressions for the relationship between $\beta$ and $\alpha$. The theoretical results are verified via real ultrasound data. Ultrasound breast data, or their increments, which appear to be $1/f^\beta$, are shown to follow reasonably well the $\alpha$-stable model.
Nonlinear Adaptive Noise Suppression Based on Wavelet Transform

Authors:
Xiao-Ping Zhang, University of Texas, San Antonio, (U.S.A.)
Mita D Desai, University of Texas, San Antonio, (U.S.A.)

Abstract:
The conventional linear adaptive filters are not effective for discriminating the transient wideband signal components from noise. A recently developed wavelet shrinkage approach is able to maintain the function local regularity while suppressing noise however, it has only been used in function estimation problems. In this paper, a new type of nonlinear filtering method for adaptive noise suppression is presented, based on shrinkage method. A new class of shrinkage functions is also presented. The filtering structure and the learning algorithm are developed. The theoretical analysis proves convergence in certain statistical sense. The numerical results of our system are presented for both the standard and the new shrinkage function and compared with the conventional linear adaptive filter based techniques. Results indicate that both the optimal solution and the learning performance are superior to the conventional methods. It is shown that our new shrinkage function performs better than the standard shrinkage function.
ICASSP98 Paper Abstract

A Comparative Study of Nonlinear Video Rate Control Techniques: Neural Networks and Fuzzy Logic

Authors:
Yoo-Sok Saw, University of Bristol, (U.K.)
Peter M. Grant, University of Edinburgh, Scotland, (U.K.)
John M. Hannah, University of Edinburgh, Scotland, (U.K.)

Abstract:
Data rate management of compressed digital video has been treated mainly from the teletraffic control point of view, i.e. by modelling congestion control via network protocols. Relatively less attention has been focused on video rate management in the source coding side. In this paper we consider that it is more efficient and less costly to control video rate at the video source than handling network congestion (or overloading) due to an extremely large quantity of incoming variable bit rate (VBR) video traffic. Thus this paper investigates effective rate control algorithms for video encoders. Considering the non-stationary nature of video rate derived from scene variations (i.e. the wide band nature of digital video), we adopted and compare the performance of two nonlinear approaches; radial basis function (RBF) estimation using a neural network-based approach and fuzzy logic control as a nonlinear feedback control.
ICASSP98 Paper Abstract
Stable One-Bit Delta-Sigma Modulators Based on Switching Control

Authors:
Takis Zourntos, University of Toronto, (Canada)
David A Johns, University of Toronto, (Canada)

Volume 3, page 1597, paper no. 1299

Abstract:
We present a globally stable arbitrary-order single-bit delta-sigma modulator architecture with continuous-time loop filtering. Using Lyapunov arguments and the method of equivalent control, it is shown that stability is guaranteed for any input signal with peak magnitude less than \( L > 0 \), where \(-L\) and \(+L\) denote the quantization levels. The design augments the conventional delta-sigma modulator with switching feedback and the use of distinct operating modes; the additional circuitry required for the implementation of these stabilizing measures is nominal. For a given noise transfer function and fixed oversampling ratio, the new architecture achieves the same peak signal-to-noise-plus-distortion ratio as a traditional delta-sigma modulator. The proposed design can also yield near-peak performance for inputs which destabilize the conventional delta-sigma data converter. Simulation results are provided for the proposed modulator and a comparable standard interpolative design.
Passivity Analysis for Uncertain Signal Processing Systems

Authors:
Minyue Fu, University of Newcastle, (Australia)
Lihua Xie, Nanyang Technological University, (Singapore)
Huaizhong Li, Laboratoire d’Automatique de Grenoble, (France)

Volume 3, page 1601, paper no. 1626

Abstract:
The problem of passivity analysis finds important applications in many signal processing systems such as digital quantizers, decision feedback equalizers and digital and analog filters. This paper considers the passivity analysis problem for a large class of systems which involve uncertain parameters, time delays, quantization errors, and unmodeled high order dynamics. By characterizing these and many other types of uncertainty using a general tool called integralquadratic constraints (IQC), we present a solution to the problem of robust passivity analysis. More specifically, we determine if a given uncertain system is robustly passive. The solution is given in terms of the feasibility of a linear matrix inequality (LMI) which can be solved efficiently.
ICASSP98 Paper Abstract
A Volterra Model for the High Density Optical Disc

Authors:
Luigi Agarossi, Philips Research Monza, (Italy)
Sandro Bellini, Politecnico di Milano, (Italy)
Alberto Canella, CEFRIEL, (Italy)
Pierangelo Migliorati, University of Brescia, (Italy)

Volume 3, page 1605, paper no. 1713

Abstract:
This paper presents a study aiming to define a nonlinear model, based on the Volterra series, of the high density optical disc read out process. Under high density condition, because of the high linear density and reduced track pitch, the signal read out is not a linear process and suffers from cross talk. To cope with such a problem the identification of a suitable nonlinear model is required. According to the Hopkins analysis, a physical model based on the optical scalar theory was implemented. The results of this analysis have then been used to identify the kernels of a nonlinear model based on the Volterra series. The obtained results show that a second order bidimensional model is sufficient to accurately describe the read out process. The nonlinear Volterra model is a convenient starting point to devise and analyze nonlinear equalization and cross talk cancellation techniques.
ICASSP98 Paper Abstract

Identification of Bilinear Systems Using Bayesian Inference

Authors:
Souad Meddeb, ENSEEIHT/GAPSE, (France)
Jean Yves Tourneret, ENSEEIHT/GAPSE, (France)
Francis Castanie, ENSEEIHT/GAPSE, (France)

Volume 3, page 1609, paper no. 2009

Abstract:
A large class of non-linear phenomena can be described using bilinear systems. Such systems are very attractive since they usually require few parameters, to approximate most non-linearities (compared to other systems). This paper addresses the problems of bilinear system identification using Bayesian inference. The Gibbs sampler is used to estimate the bilinear system parameters, from measurements of the system input and output signals.
ICASSP98 Paper Abstract
Prediction and Estimation for Fractal Processes Using Multiscale State-Space Algorithms

Authors:
Alexander C Wang, MIT RLE, (U.S.A.)
Gregory W. Wornell, MIT RLE, (U.S.A.)

Volume 3, page 1613, paper no. 2111

Abstract:
The 1/f family of fractal processes provides useful models for the extraordinary variety of natural and man-made phenomena that exhibit long-term dependence. Using algorithms based on a multiscale state-space representation, we address the problems of parameter estimation of discrete 1/f signals in white noise, estimation of deterministic signals in 1/f noise, and prediction of discrete 1/f processes. Among other results, distant past data are shown to have a dramatically greater effect on these estimators than when ARMA processes are involved.
Equalization and Linearization of Nonlinear Systems

Authors:
Alberto Carini, University of Trieste, (Italy)
Giovanni L Sicuranza, University of Trieste, (Italy)
V. John Mathews, University of Utah, (U.S.A.)

Abstract:
This paper presents a theory for the exact and the p-th order equalization or linearization of nonlinear systems with known recursive or nonrecursive polynomial input-output relationships. The equalizing and linearizing filters have simple and computationally efficient structures. An experimental result that illustrates the good properties of the technique we propose is also included in this paper.
Signal Restoration with Controlled Piecewise Monotonicity Constraint

Authors:
Jian Lu, Apple Computer, (U.S.A.)

Abstract:
A signal restoration problem can be formulated as a least-squares inversion subject to a constraint that the signal has no more than k piecewise monotonic segments. We refer to the associated constraint as controlled piecewise monotonicity or CPM. We show that this constraint alone is powerful enough to stabilize an ill-posed inversion and enables us to incorporate the knowledge about the waveform geometry of the signal. This leads to a new algorithm for constrained signal restoration. We describe a highly efficient iterative scheme for computing the CPM constrained least-squares restoration. We also present experimental results and discuss issues related to the new algorithm.
ICASSP98 Paper Abstract
The Stability of a Direct Method for Superresolution

Authors:
Jose M.N. Vieira, Universidade de Aveiro, (Portugal)
Paulo J.S.G. Ferreira, Universidade de Aveiro, (Portugal)

Volume 3, page 1625, paper no. 2430

Abstract:
A direct method for superresolution recently proposed by Walsh and Delaney is further analyzed from the point of view of numerical stability. The method is based on a set of linear equations $Ax=b$, where $A$ is $mn$, and $b$ is a subset (of cardinal $n$) of the Fourier transform of the object (which has a total of $N$ samples). We give exact and best possible approximate expressions for the determinant of $A$, when $m=n$. As a corollary, it is shown that the smallest eigenvalue of $A$ in absolute value satisfies, where (which is independent of $N$) is explicitly given. The magnitude of the smallest eigenvalue of $A$ becomes increasingly small as $N$ grows, even when the number of unknowns $n$ remains constant. When $m>n$ the singular values of $A$ are studied, and related to the eigenvalues of the matrix of other direct methods. The connection between the method and the other direct methods is clarified.
ICASSP98 Paper Abstract
Signal Decomposition Using Adaptive Block Transform Packets

Authors:
Jyhchau Horng, Polytechnic University, (U.S.A.)
Richard A Haddad, New Jersey Institute of Technology, (U.S.A.)

Volume 3, page 1629, paper no. 1119

Abstract:
A Block Transform Packet (BTP) is an orthonormal block transform which is constructed from conventional block transform (e.g. DCT) and represents an arbitrary tiling of the time-frequency plane. Unlike the progenitor transforms, the BTP has time-localizabilities and is capable of dealing with non-stationary signals. This paper describes the procedures for signal decomposition using the BTP in an adaptive way. Three examples show the adaptive compression efficiency over DCT.
ICASSP98 Paper Abstract

Sub-Nyquist Sampling of Multiband Signals: Perfect Reconstruction and Bounds on Aliasing Error

Authors:
Raman Venkataramani, University of Illinois, (U.S.A.)
Yoram Bresler, University of Illinois, (U.S.A.)

Volume 3, page 1633, paper no. 2585

Abstract:
We consider the problem of periodic nonuniform sampling of a multiband signal and its reconstruction from the samples. We derive the conditions for exact reconstruction and find an explicit reconstruction formula. Key features of this method are that the sampling rate can be made arbitrarily close to the minimum (Landau) rate and that it can handle classes of multiband signals that are not packable. We compute various bounds on the aliasing error due to mismodeling the spectral support and examine the performance in the presence of additive white sample noise. Finally we provide optimal designs for the reconstruction system.
ICASSP98 Paper Abstract

Use of Selected HOS Information for Low-Variance Estimation of Bandlimited Systems with Short Data Records

Authors:
Haralambos Pozidis, Drexel University, (U.S.A.)
Athina P Petropulu, Drexel University, (U.S.A.)

Abstract:
Although reconstruction of a nonminimum-phase system excited by a stationary non-Gaussian white input is only possible using higher-order statistics (HOS) of the system output, there has been a lot of criticism in the literature against the amount of data required for keeping estimation errors low, and the complexity involved. Recently several attempts for reducing the variance of the HOS estimates have appeared. In the case of bandlimited signals, we have demonstrated via simulations that the estimation variance can be reduced if “good” slices, instead of the whole bispectrum, are used. This suggests a potential reduction of variance in the system estimates, without having to resort to long observations. In this paper we justify theoretically the dependence of the system estimate variance on the bispectrum slice, and the criterion of slice selection. We also present simulation results, where the selected-slices approach appears to result in much lower estimation variance, as compared to other entire-bispectrum based approaches, for data lengths as low as 64 samples.
ICASSP98 Paper Abstract

1-D Continuous Non-Minimum Phase Retrieval Using the Wavelet Transform

Authors:
Amy E Bell, Virginia Tech, (U.S.A.)
Andrew E. Yagle, University of Michigan, (U.S.A.)

Volume 3, page 1641, paper no. 1390

Abstract:
The phase retrieval problem arises when a signal must be reconstructed from only the magnitude of its Fourier transform; if the phase information were also available, the signal could simply be synthesized using the inverse Fourier transform. In continuous phase retrieval, most previous solutions rely on discretizing the problem and then employing an iterative algorithm. We avoid this approximation by using wavelet expansions to transform this uncountably infinite problem into a linear system of equations. The wavelet bases permit a solution by incorporating a priori signal information and they provide a structured system of equations which results in a fast algorithm. Our solutions obviate the stagnation problems associated with iterative algorithms, they are computationally simpler and more stable than previous non-iterative algorithms, and they can accommodate noisy Fourier magnitude information. This paper develops our 1-D continuous, non-zero-minimum phase retrieval algorithm and illustrates its effectiveness with numerical examples.
ICASSP98 Paper Abstract

H-infinity Filtering for Noise Reduction Using a Total Least Squares Estimation Approach

Authors:
Jun'ya Shimizu, University of California, Santa Barbara, (U.S.A.)
Sanjit K. Mitra, University of California, Santa Barbara, (U.S.A.)

Volume 3, page 1645, paper no. 1267

Abstract:
A noise reduction algorithm for signals corrupted by additive unknown L2 white noise is proposed using an H-infinity filtering framework. The proposed algorithm consists of two steps: a signal enhancement step and a parameter estimation step, which are iterated at each instant. To weaken the dependence between the signal enhancement step and the parameter estimation step, a total least squares estimation step for the dynamical model parameters needed in the H-infinity filtering is introduced. The effectiveness of the proposed algorithm under low signal-to-noise ratio environments is demonstrated by simulation.
ICASSP98 Paper Abstract
Detection and Estimation of Superimposed Signals

Authors:
Jean Jacques Fuchs, Universite de Rennes, (France)

Abstract:
The problem of fitting a small number of superimposed signals to noisy observations is addressed. An approach allowing us to evaluate both the number of signals and their characteristics is presented. The idea is to search for a parsimonious representation of the data. The parsimony is insured by adding to a maximum likelihood like criterion a regularization term built upon the l1 norm of the weights. Different equivalent formulations of the criterion that is optimised are presented. They lead to appealing physical interpretations. We analyse the performance of the algorithm that has already been successfully applied to different classes of problems.
ICASSP98 Paper Abstract
Continuous-Time Reconstruction of Nonuniformly Sampled Signals on a Band-Limited Wavelet Basis

Authors:
Livia Nita, SUPELEC, (France)
Jacques Oksman, SUPELEC, (France)

Volume 3, page 1653, paper no. 2030

Abstract:
We propose a reconstruction method of continuous-time random signals by fitting nonuniform samples to a band-limited continuous-time wavelet basis. Based on wavelet analysis, our method uses a windowing technique with variable-sized intervals, taking advantage of the nonuniform signal sampling. This method leads to analytical formulas for the reconstructed continuous-time signal, and as well as for its derivatives. This can be very useful to perform a parametric estimation of so-called continuous-time ARMA models adopted for continuous-time random signal modeling. Several parameters like mother wavelet type, time shift interval between consecutive wavelets and resolution levels number can be adapted, function of nature of nonuniformly sampled signal. In this paper, we describe the principle of the proposed reconstruction method and discuss its performances.
ICASSP98 Paper Abstract

An Improved Sequential Backward Selection Algorithm for Large-Scale Observation Selection Problems

Authors:
Stanley J. Reeves, Auburn University, (U.S.A.)

Volume 3, page 1657, paper no. 2148

Abstract:
Some signal reconstruction problems allow for flexibility in the selection of observations and hence the signal formation equation. In such cases, we have the opportunity to determine the best combination of observations before acquiring the data. We analyze the computational complexity of various forms of sequential backward selection (SBS) to select observations. In light of this analysis, we present a computationally improved algorithm for large-scale observation selection problems.
ICASSP98 Paper Abstract

Analysis of a Delayless Subband Adaptive Filter Structure

Authors:
Paulo S.R. Diniz, COPPE/EE/UFRJ, (Brazil)
Ricardo Merched, COPPE/EE/UFRJ, (Brazil)
Mariane R Petraglia, COPPE/EE/UFRJ, (Brazil)

Volume 3, page 1661, paper no. 1023

Abstract:
In this paper, we present an analysis of the delayless subband adaptive filter structure previously proposed by the authors. We derive a simple expression for the excess MSE of the proposed structure, and show that it requires up to 3.7 less computational complexity than the corresponding fullband LMS structure. Also, we establish a connection between subband block adaptive filtering, where the latter can be interpreted as a special case of the former. Some computer simulations are presented in order to verify the performance of the proposed structure and the theoretical results.
ICASSP98 Paper Abstract

A Comparison of Initialization Schemes for Blind Adaptive Beamforming

Authors:
Thomas E Biedka, Raytheon E-Systems, (U.S.A.)
Volume 3, page 1665, paper no. 2490

Abstract:
Many blind adaptive beamforming algorithms require the selection of one or more non-zero initial weight vectors. Proper selection of the initial weight vectors can speed algorithm convergence and help ensure convergence to the desired solutions. Three alternative initialization approaches are compared here, all of which depend only on second order statistics of the observed data. These methods are based on Gram-Schmidt orthogonalization, eigendecomposition, and QR-decomposition of the observed data covariance matrix. We show through computer simulations that the eigendecomposition approach yields the best performance.
ICASSP98 Paper Abstract

MSE Analysis of the M-Max NLMS Adaptive Algorithm

Authors:
T. Aboulnasr, University of Ottawa, (Canada)
K. Mayyas, University of Science and Technology, (Jordan)

Volume 3, page 1669, paper no. 1737

Abstract:
In this paper, we provide a mean square analysis of the M-Max NLMS (MMNLMS) adaptive algorithm introduced in [1]. The algorithm selects, at each iteration, a specified number of coefficients that provide the largest reduction in the error. It is shown that while the MMNLMS algorithm reduces the complexity of the adaptive filter, it maintains the closest performance to the full update NLMS filter for a given number of updates. The stability of the algorithm is shown to be guaranteed for the extreme case of only one update/iteration. Analysis of the MSE convergence and steady state performance for i.i.d. signals is also provided for that extreme case.
ICASSP98 Paper Abstract

Analysis of the Sign-Sign Algorithm Based on Gaussian Distributed Tap Weights

Authors:
Shin'ichi Koike, NEC Corporation, (Japan)

Volume 3, page 1673, paper no. 5028

Abstract:
In this paper, a new set of difference equations is derived for convergence analysis of adaptive filters using the Sign-Sign Algorithm with Gaussian input reference and additive Gaussian noise. The analysis is based on the assumption that the tap weights are jointly Gaussian distributed. Residual mean squared error after convergence and simpler approximate difference equations are further developed. Results of experiment exhibit good agreement between theoretically calculated convergence and that of simulation for a wide range of parameter values of adaptive filters.
Mean-Squared Error Analysis of the Binormalized Data-Reusing LMS Algorithm Using a Discrete-Angular-Distribution Model for the Input Signal

Authors:
Marcello L.R. De Campos, Instituto Militar de Engenharia, (Brazil)
Jose A Apolinario Jr, COPPE/UFRJ, (Brazil)
Paulo S.R. Diniz, COPPE/UFRJ, (Brazil)

Volume 3, page 1677, paper no. 2441

Abstract:
Providing a quantitative mean-squared error analysis of adaptation algorithms is of great importance for determining their usefulness and for comparison with other algorithms. However, when the algorithm reutilizes previous data, such analysis becomes very involved as the independence assumption cannot be used. In this paper, a thorough mean-squared-error analysis of the binormalized data-reusing LMS algorithm is carried out. The analysis is based on a simplified model for the input-signal vector, assuming independence between the continous radial probability distribution and the discrete angular probability distribution. Throughout the analysis only parallel and orthogonal input-signal vectors are used in order to obtain a closed-form formula for the excess mean-squared error. The formula agrees closely with simulation results even when the input-signal vector is a delay line. Furthermore, the analysis can be readily extended to other algorithms with expected similar accuracy.
ICASSP98 Paper Abstract

On a Perturbation Approach for the Analysis of Stochastic Tracking Algorithms

Authors:
Eric Moulines, Ecole Nationale Superieure des Telecommunications, (France)
Pierre Priouret, Universite de Paris VI, (France)
Rafik Aguech, Universite de Paris VI, (France)

Volume 3, page 1681, paper no. 1618

Abstract:
In this paper, a perturbation expansion technique is introduced to decompose the tracking error of a general adaptive tracking algorithm in a linear regression model. This method allows to obtain tracking error bound but also tight approximate expressions for the moments of the tracking error. These expressions allow to evaluate, both qualitatively and quantitatively, the impact of several factors on the tracking error performance which have been overlooked in previous contributions.
ICASSP98 Paper Abstract

Numerical Properties of the Linearly Constrained QRD-RLS Adaptive Filter

Authors:
Jiaquan Huo, Curtin University of Technology, (Australia)
Yee H Leung, Curtin University of Technology, (Australia)

Volume 3, page 1685, paper no. 1908

Abstract:
Shepherd and McWhirter proposed a QRD-RLS algorithm for adaptive filtering with linear constraints. In this paper, the numerical properties of this algorithm are considered. In particular, it is shown that the computed weight vector satisfies a set of constraints which are perturbed from the original ones, the amount of the perturbation being dependent on the wordlength. The linearly constrained FLS algorithm of Resende et al is also studied. Simulation results show that this algorithm is numerically unstable even in the absence of explosive divergence.
ICASSP98 Paper Abstract

Analysis of the Euclidean Direction Set Adaptive Algorithm

Authors:
Guo Fang Xu, University of Colorado, (U.S.A.)
Tamal Bose, University of Colorado, (U.S.A.)

Volume 3, page 1689, paper no. 2182

Abstract:
A mathematica analysis is performed on a recently reported gradient based adaptive algorithm named the Euclidean Direction Set (EDS) method. It has been shown that the EDS algorithm has a computational complexity of O(N) for each system update and a rate of convergence (based on computer simulations) comparable to the RLS algorithm. In this paper, the stability of the EDS method is studied and it is shown that the algorithm converges to the true solution. It is also proved that the convergence rate of the EDS method is superior to that of the steepest descent method.
ICASSP98 Paper Abstract
Quantifying the Accuracy of Adaptive Tracking Algorithms

Authors:
Brett M. Ninness, University of Newcastle, (Australia)
Juan-Carlos Gomez, University of Newcastle, (Australia)

Volume 3, page 1693, paper no. 1378

Abstract:
The use of adaptive algorithms such as Kalman Filtering, LMS and RLS together with FIR model structures is very common and extensively analysed. In the interests of improved performance an extension of the FIR structure has been proposed in which the fixed poles are not all at the origin, but instead are chosen by prior knowledge to be close to where the true poles are. Existing FIR analysis would indicate that the noise and tracking properties of such a scheme are invariant to the choice of fixed pole location. This paper establishes both numerically and theoretically that in fact this is not the case. Instead, the dependence of fixed pole location is made explicit by deriving frequency domain expressions that are obtained by using new results on generalised Fourier series and generalised Toeplitz matrices.
ICASSP98 Paper Abstract
A Unifying View of Error Nonlinearities in LMS Adaptation

Authors:
Tareq Y Al-Naffouri, Georgia Institute of Technology, (U.S.A.)
Azzedine Zerguine, King Fahd University of Petroleum and Minerals, (Saudi Arabia)
Maamar Bettayeb, King Fahd University of Petroleum and Minerals, (Saudi Arabia)

Abstract:
This paper presents a unifying view of various error nonlinearities that are used in least mean square (LMS) adaptation such as the least mean fourth (LMF) algorithm and its family and the least-mean mixed-norm algorithm. Specifically, it is shown that the LMS algorithm and its error-modified variants are approximations of two newly proposed optimum nonlinearities which are expressed in terms of the additive noise probability density function (pdf). This is demonstrated through an approximation of the optimum nonlinearities by expanding the noise pdf in a Gram-Charlier series. Thus, a link is established between intuitively proposed and theoretically justified variants of the LMS algorithm. The approximation has also a practical advantage in that it provides a trade-off between simplicity and more accurate realization of the optimum nonlinearities.
ICASSP98 Paper Abstract
Filtered Error Adaptive IIR Algorithms and their Application to Active Noise Control

Authors:
Carlos Mosquera, Universidad de Vigo, (Spain)
Jose A Gomez, Universidad de Vigo, (Spain)
Fernando Perez-Gonzalez, Universidad de Vigo, (Spain)

Abstract:
This paper is concerned with systems in which the output error of an adaptive IIR filter is subsequently filtered, e.g., an active noise control system where a transfer function models the path between the injection point of the cancellation noise and the point where the residual error is measured. We propose a family of algorithms suited to this type of scenarios, deriving conditions for their deterministic convergence. The analysis of the convergence is particularized to the Filtered-U Recursive LMS algorithm, a popular scheme whose global convergence has never been proved formally. Finally, some results based on real measurements are also presented.
A Frequency Domain Adaptive Algorithm for Colored Measurement Noise Environment

Authors:
Tõnu Trump, Ericsson Radio Systems AB, (Sweden)

Volume 3, page 1705, paper no. 1286

Abstract:
The problem of incorporating partial knowledge of measurement noise into a frequency domain adaptive filtering scheme is addressed. 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.
ICASSP98 Paper Abstract

Truncated Orthogonal Expansions of Recurrent Signals: Equivalence to a Periodic Time-Variant Filter

Authors:
Salvador Olmos, University of Zaragoza, (Spain)
Jose Garcia, University of Zaragoza, (Spain)
Raimon Jane, Politechnic University of Catalonia, (Spain)
Pablo Laguna, University of Zaragoza, (Spain)

Volume 3, page 1709, paper no. 1559

Abstract:
In this work we show that orthogonal expansions of recurrent signals like electrocardiograms with a reduced number of coefficients can be considered as a periodic time-variant filter. Instantaneous impulse and frequency responses are analyzed for two cases: estimation of the coefficients with inner product and adaptive estimation with LMS algorithm.
ICASSP98 Paper Abstract

A Numerically Stable Fast Newton Type Adaptive Filter Based on Order Update Fast Least Squares Algorithm

Authors:
Youhua Wang, Kanazawa University, (Japan)
Kazushi Ikeda, Kanazawa University, (Japan)
Kenji Nakayama, Kanazawa University, (Japan)

Volume 3, page 1713, paper no. 1919

Abstract:
The numerical property of an adaptive filter algorithm is the most important problem in practical applications. Most fast adaptive filter algorithms have the numerical instability problem and the fast Newton transversal filter (FNTF) algorithms are no exception. In this paper, we propose a numerically stable fast Newton type adaptive the proposed algorithm from the order update fast least squares (FLS) algorithm. This derivation is direct and simple to understand. Second, we give the stability analysis using linear time-variant state-space method. The transition matrix of the proposed algorithm is given. The eigenvalues of the ensemble average of the transition matrix are shown to be asymptotically all less than unity. This results in a much improved numerical performance compared with the FNTF algorithms. The computer simulations implemented by using a finite-precision arithmetic have confirmed the validity of our analysis.
ICASSP98 Paper Abstract
Selective Block Update of NLMS Type Algorithms

Authors:
Thomas Schertler, Darmstadt University of Technology, (Germany)

Abstract:
Adaptive filters for the cancellation of acoustic echoes, as applied in hands-free telephone sets, require about a thousand coefficients and more to get a significant echo reduction. This leads to a very high computational effort and cannot be realized on most low-cost DSPs. One common proposition to decrease the computational load is to update only a portion of the coefficients at a time. This decreases not only the computational load but also the convergence speed. To reduce this drawback, it has been suggested that only the most significant coefficients be updated. This improves the convergence speed considerably. Unfortunately, it requires additional memory of twice the filter length. In our proposal, we present a modified version of the mentioned algorithm which has almost the same adaptation speed but consumes significantly less memory.
ICASSP98 Paper Abstract

An Instrumental Variable Based Subspace Tracking Algorithm Based on Subspace Averaging

Authors:
Tony Gustafsson, Chalmers University of Technology, (Sweden)

Volume 3, page 1721, paper no. 1580

Abstract:
In this paper an instrumental variable (IV) based subspace tracking algorithm is proposed. The basic idea of the algorithm is to reduce the amount of computations using a certain perturbation/approximation strategy. The complexity is reduced to mnn, which should be compared to mll for the SVD, where m,l » n in general (m denotes the number of sensors, l denotes the number of instruments, and n denotes the number of signals). The proposed algorithm turns out to be related to Karasalo's subspace averaging approach. In a series of simulations we demonstrate that the detection-, stationary estimation-, and tracking performance of the proposed algorithm is essentially equivalent to that achieved by the truncated SVD.
ICASSP98 Paper Abstract

Application of Subband Analysis to Adaptive Prediction

Authors:
James R Saffle, Villanova University, (U.S.A.)
Sathyanarayan S Rao, Villanova University, (U.S.A.)

Volume 3, page 1725, paper no. 1527

Abstract:
A closed-loop adaptive subband prediction architecture is presented by employing an adaptive subband filter in the prediction configuration. Some authors have suggested that applying open-loop prediction methods to subband signals can realize increased prediction gain over fullband prediction. Furthermore, the benefits of applying multirate techniques to adaptive filtering are well understood in terms of reduction of computational complexity and increased convergence speed. Thus, the closed-loop subband adaptive predictor is a novel approach that is expected to exhibit these same benefits along with the advantages of backward adaptation. Results show that the new subband predictor can produce a higher prediction gain than a similar fullband adaptive prediction filter. The proposed architecture is implemented in C++ on the Pentium processor.
A Globally Convergent Modified OE IIR Adaptive Filter for Sufficient Modeling

Authors:
K. Mayyas, University of Science and Technology, (Jordan)
T. Aboulnasr, University of Ottawa, (Canada)

Volume 3, page 1729, paper no. 1726

Abstract:
A modification to the OE IIR system structure is proposed to ensure global convergence for sufficient modeling of the unknown system. The proposed structure is effectively equivalent to whitening the input signal before being applied to the original OE setup. This guarantees the unimodality of the error surface for sufficient modeling. An adaptive update scheme for the new structure is derived based on the least mean square (LMS) technique. Examples are provided to demonstrate the effectiveness of the proposed structure under different conditions.
ICASSP98 Paper Abstract

Regressor Based Adaptive Infinite Impulse Response Filtering

Authors:
Emrah Acar, Carnegie Mellon University, (U.S.A.)
Orhan Arikan, Bilkent University, (Turkey)

Volume 3, page 1733, paper no. 1791

Abstract:
To take advantage of fast converging multi-channel recursive least squares algorithms, we propose an adaptive IIR system structure consisting of two parts: a two-channel FIR adaptive filter whose parameters are updated by rotation-based multi-channel least squares lattice (QR-MLSL) algorithm, and an adaptive regressor which provides more reliable estimates to the original system output based on previous values of the adaptive system output and noisy observation of the original system output. Two different regressors are investigated and robust ways of adaptation of the regressor parameters are proposed. Based on extensive set of simulations, it is shown that the proposed algorithms converge faster to more reliable parameter estimates than LMS type algorithms.
ICASSP98 Paper Abstract

Frequency-Domain Realizations of Adaptive Parallel--Cascade Quadratic Filters

Authors:
Linshan Li, Northwestern Polytechnical University, (China)
V. John Mathews, University of Utah, (U.S.A.)

Volume 3, page 1737, paper no. 2096

Abstract:
Parallel-cascade realizations of truncated Volterra systems implement higher-order systems using a parallel connection of multiplicative combinations of lower-order systems. Such realizations are modular and permit efficient approximations of truncated Volterra systems. Frequency-domain realizations of the least-mean-square (LMS) adaptive filter and the normalized LMS adaptive filter that implements the system model using the parallel-cascade structure are presented in this paper. Computational complexity analysis and simulation results show that the normalized frequency-domain, parallel-cascade LMS adaptive quadratic filter has the advantages of computational simplicity and superior performance over direct form realizations.
Abstract:
We illustrate how high-level knowledge from the musical domain may be integrated with sophisticated signal processing algorithms within a system for separating possibly overlapping partial frequency components from polyphonic music. Musical knowledge utilized in our system is in the form of constraints on the time-frequency behaviors of musical signals such as the frequency locations of notes on the western musical scale and the presence or absence of vibrato in each note. For any given signal scenario, these constraints help in appropriately initializing and adjusting a set of algorithms for constant-Q processing, spectral peak picking, and multihypothesis tracking through Kalman filtering. As demonstrated by the evaluation of our system with a variety of signals containing two simultaneously played violin notes, the application of these algorithms results in the accurate separation of individual partials.
ICASSP98 Paper Abstract

An Implementation of a Parallel Ray Tracing Algorithm on Hybrid Parallel Architecture

Authors:
Chang-Geun Kwon, Kyungpook National University, (Korea)
Hyo-Kyung Sung, Kyungpook National University, (Korea)
Heung-Moon Choi, Kyungpook National University, (Korea)

Volume 3, page 1745, paper no. 2362

Abstract:
In this paper, we present a parallel ray tracing algorithm on hybrid parallel architecture with processor farm model to speed up the ray tracing. Hybrid parallel architecture, a hybrid of a tightly- and a loosely-coupled one, is used in which reconfiguration for local and virtual shared memory is made through a crossbar network with local and global bus. The proposed architecture enhances the overall performance of the parallel ray tracing by reducing the data communication time between the processors in dynamic load balancing while maintaining data coherency. The proposed algorithm is implemented on TMS320C80, an MVP (multimedia video processor), which has one master processor and four slave processors. The experimental results show that the proposed algorithm gives almost a linear speedup for parallel ray tracing of a complex image.
ICASSP98 Paper Abstract
Distributed Signal Processing

Authors:
Li Lee, *MIT Research Laboratory of Electronics, (U.S.A.)*
Alan V. Oppenheim, *MIT Research Laboratory of Electronics, (U.S.A.)*

Volume 3, page 1749, paper no. 1875

Abstract:
This paper explores issues arising from designing digital signal processing algorithms for dynamically-varying computing environments such as an unreliable network of processors. We present a language for specifying signal processing algorithms which permits the execution path of the algorithm to be dynamically chosen. The language leads naturally to a graphical representation of the algorithm with interesting interpretations. Finally, we formulate and characterize the solution for the problem of dynamically and optimally choosing the execution path of algorithms to minimize a system-wide cost function such as expected congestion.
ICASSP98 Paper Abstract

A Noise-Robust Echo Canceller on V830 Multimedia RISC Processor Integrated into a Car Navigation System

Authors:
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Akihiro Hirano, NEC Corporation, (Japan)
Masaya Kanazawa, NEC IC Microcomputer Systems Ltd, (Japan)

Volume 3, page 1753, paper no. 1314

Abstract:
This paper presents a noise-robust, fast-convergence echo canceller and its implementation on a multimedia RISC (Reduced Instruction Set Computer). Faster convergence is achieved by introducing an improved noise power estimator for step-size control. This echo canceller has been implemented on V830 multimedia embedded RISC and has been integrated into a car navigation system. V830 provides performance comparable to a digital signal processor (DSP) and extended flexibility while power consumption is lower than that of a DSP. Computer simulations and measurements using a V830 board show fast convergence and robustness against disturbance such as a noise and a double-talk without double-talk detection.
ICASSP98 Paper Abstract

A Framework for the Graphical Specification and Execution of Complex Signal Processing Applications

Authors:
Andreas Sicheneder, University of Passau, (Germany)
Armin Bender, University of Passau, (Germany)
Erich Fuchs, University of Passau, (Germany)
Roland Mandl, University of Passau, (Germany)
Bernhard Sick, University of Passau, (Germany)

Abstract:
A framework with a tool-supported high-level specification technique is very important for the development of complex signal processing applications containing software-intensive parts (e.g. hybrid systems in automated production processes) in order to provide safe and reliable systems. In this paper we present the concept of a framework, which is an object-oriented CASE-tool offering a graphical specification ability to model and validate a given application and to control its execution. A variety of people having different programming skills is able to use this visual specification technique effectively. Especially users not being interested in implementation details can specify their application on a high abstraction level by connecting reusable and reliable components (modules representing basic algorithms). As a result, complex signal graphs representing the dataflow between the modules are created. The tool supports this software specification technique by automatic type-checking for the connections between modules and by changeable module parameters. On the other hand it is easy for software engineers to integrate additional signal processing algorithms into the framework thus building suitable module libraries without considering a specific high-level application.
ICASSP98 Paper Abstract

Computationally Efficient Implementation of Hypercomplex Digital Filters

Authors:
Hisamichi Toyoshima, Kanagawa University, (Japan)

Volume 3, page 1761, paper no. 1199

Abstract:
Hypercomplex digital filters have an attractive advantage of the order reduction, however, also have a drawback that multiplication requires a large amount of computations. This paper proposes a novel implementation of hypercomplex digital filters. By decomposing hypercomplex number multiplication, we show that it can be realized as two parallel complex multiplications. Using this technique, any types of hypercomplex digital filters can be implemented with less than half computations of the direct approach.
ICASSP98 Paper Abstract

A Highly-Scalable FIR Using the RADIX-4 Booth Algorithm

Authors:
Oscal T.C. Chen, National Chung Cheng University, (Taiwan)
Wei-Lung Liu, National Chung Cheng University, (Taiwan)
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Jeng-Yih Wang, Industrial Technology Research Institute, (Taiwan)

Volume 3, page 1765, paper no. 2492

Abstract:
A Highly-scaleable FIR architecture based on the radix-4 Booth algorithm has been designed with scaleable dynamic ranges of input data and filter coefficients. The radix-4 Booth algorithm is demonstrated to have a lower hardware complexity and a fair throughput rate than the other radix approaches. In order to achieve scaleability, the configurable-connection function between latches of input data, and filter taps has been explored. The precision of filter coefficients is adjustable by using a path-control function. Especially, the proposed architecture only employs data-path controls to realize the scaleable issue without changing the word lengths and components of input latches and filter taps. The pre-processing unit for manipulating input data and post-processing unit for computing accumulation results have been realized to support scaleable operations. Based on our architecture in a chip design, the cascaded configuration between chips is also easily accomplished for many industrial applications.
ICASSP98 Paper Abstract
Reversible Discrete Cosine Transform

Authors:
Kunitoshi Komatsu, University of Tokyo, (Japan)
Kaoru Sezaki, University of Tokyo, (Japan)

Volume 3, page 1769, paper no. 1092

Abstract:
In this paper a reversible discrete cosine transform (RDCT) is presented. N-point reversible transform is firstly presented, then 8-point RDCT is obtained by substituting the 2 and 4-point reversible transforms for 2 and 4-point transforms which compose 8-point discrete cosine transform (DCT), respectively. Integer input signal can be losslessly recovered, although the transform coefficients are integer numbers. If floor function is ignored in RDCT, the transform is exactly the same as DCT with determinant = 1. RDCT is also normalized so that we can avoid the problem that dynamic range is nonuniform. Simulation on continuous-tone still images shows that lossless and lossy compression efficiency of RDCT are comparable to those obtained with reversible wavelet transform.
ICASSP98 Paper Abstract

Short-Time Fourier Analysis - A Novel Window Design Procedure

Authors:
Zoran D Cvetkovic, AT&T Labs, (U.S.A.)

Volume 3, page 1773, paper no. 2103

Abstract:
Weyl-Heisenberg frames are the tool for short-time Fourier analysis. These are generated from a prototype window function using translation on a rectangular grid in the time-frequency plane. Particularly appealing Weyl-Heisenberg frames are those which are tight as they allow for signal representations analogous to orthonormal expansions and have good numerical stability properties. Designing the window of a tight Weyl-Heisenberg frame requires optimization of the frequency characteristics of the window, usually some form of frequency selectivity, under a set of nonlinear constraints. For long windows that can be a formidable task, if not infeasible. We propose a new filter design method based on expansions with respect to prolate spheroidal sequences. The advantages of this new method are more and more pronounced as redundancy of the frame increases in reducing computational complexity and allowing for design of good filters which can be specified with few parameters.
ICASSP98 Paper Abstract

A New Signal Adaptive Approach to Positive Time-Frequency Distributions with Suppressed Interference Terms

Authors:
Robert M Nickel, The University of Michigan, (U.S.A.)
Tzu-Hsien Sang, The University of Michigan, (U.S.A.)
William J Williams, The University of Michigan, (U.S.A.)

Volume 3, page 1777, paper no. 2188

Abstract:
Quadratic time varying-spectral analysis methods that achieve a high resolution jointly in time and frequency generally suffer from interference terms that obscure the true location of the auto components in the resulting time-frequency representation. Unfortunately, as of now, there is no general mathematical model available for an exact distinction between cross-terms and autoterms. Consequently an attempt to suppress interferences can only rely on a few qualitative properties which are commonly associated with cross terms. Most of the reduced interference distributions that have been developed so far exploit the fact that cross terms tend to oscillate and can hence be suppressed by a properly chosen two-dimensional low pass filter. Besides the fact that cross-terms oscillate, they are also known to be responsible for all negative density values of a time-frequency distribution. None of the currently existing methods addresses this characteristic. In this paper we introduce an entirely new approach that achieves a significant interference reduction by specifically exploiting the negative density structure of cross-terms.
ICASSP98 Paper Abstract
Adaptive Chirplet Based Signal Approximation

Authors:
Shie Qian, National Instruments, (U.S.A.)
Dapang Chen, National Instruments, (U.S.A.)
Qinye Yin, Xi’an Jiaotong University, (China)

Volume 3, page 1781, paper no. 1034

Abstract:
The chirp function is one of the most fundamental functions in nature. Many natural events can be roughly approximated by a group of chirp functions. In this paper, we present a practical adaptive chirplet based signal approximation algorithm. Unlike the other chirplet decompositions known so far, the elementary chirplet functions employed in this algorithm are adaptive. Therefore, the resulting approximation could better match the underlying signal and uses fewer coefficients. The effectiveness of the algorithm is demonstrated by numerical simulations.
ICASSP98 Paper Abstract
Time-Frequency Derivation of Periodic Wide Band Probing Signals

Authors:
Thomas W Parks, Cornell University, (U.S.A.)
Michael S Richman, Cornell University, (U.S.A.)

Volume 3, page 1785, paper no. 2187

Abstract:
An analysis of a discrete time-frequency distribution yields a new periodic wide band probing signal for use in unknown system identification. The derivation is based on mathematical properties of the discrete Wigner distribution. Like the continuous distribution, the discrete version also satisfies the covariance property, meaning transformations in the time-frequency plane are equivalent to transformations in the time domain. By utilizing this property, the linearly swept frequency measurement is extended to discrete, periodic signals. The resulting probing signal possesses favorable characteristics such as a short illumination time requirement and good resistance to noise. The performance of the proposed probing method is compared with m-sequence methods and chirp signal methods.
ICASSP98 Paper Abstract

Analytical Design of 3-D Wavelet Filter Banks Using the Multivariate Bernstein Polynomial

Authors:
David B.H. Tay, Nanyang Technological University, (Singapore)

Abstract:
The design of 3-D multirate filter banks where the downsampling/upsampling is on the FCO (Face Centered Orthorhombic) lattice is addressed in this paper. With such a sampling lattice, the ideal 3-D subband of the low-pass filter is of the TRO (TRuncated Octahedron) shape. The transformation of variables has been shown previously to be an effective technique for designing M-D filter banks. We present a design technique for the transformation function using the multivariate Bernstein polynomial which provides good approximation to the TRO subband shape. The method is analytically based and does not require any optimization procedure. Closed form expressions are obtained for the filters of any order. Another advantage of this technique is that it yields filters with a flat frequency response at the aliasing frequency. The flatness is important for giving regular Discrete Wavelet Transform systems.
ICASSP98 Paper Abstract
Smooth Orthonormal Wavelet Libraries: Design and Application

Authors:
Snezana Maslakovic, Stanford University, (U.S.A.)
Ivan R. Linscott, Stanford University, (U.S.A.)
Mitchell Oslick, Stanford University, (U.S.A.)
Joseph D. Twicken, Stanford University, (U.S.A.)

Volume 3, page 1793, paper no. 2229

Abstract:
For signal-based design of orthonormal (ON) wavelets, an optimization of a cost function over an N-dimensional angle space is required. However: (1) the N-dim space includes both smooth and non-smooth wavelets; (2) many of the smooth wavelets are similar in shape. A more practical approach for some applications may be to construct a library of smooth ON wavelets in advance—a library that consists of representative wavelet shapes for a given filter length. Existing ON wavelet libraries (Daubechies, nearly-symmetric, Coiflets) provide only one wavelet for each filter length. We construct ON libraries using local variation to determine wavelet smoothness and the discrete inner product to discriminate between wavelet shapes. The relationship between library size and the similarity threshold is investigated for various filter lengths. We apply an entropy-based wavelet selection algorithm to an example signal set, and investigate compactness in the wavelet domain as a function of library size.
ICASSP98 Paper Abstract

Theory and Design of Spectrum-Efficient Bandwidth-on-Demand Multiplexer-Demultiplexer Pairs Based on Wavelet Packet Tree and Polyphase Filter Banks

Authors:
Michael Sablatash, Communications Research Centre, (Canada)
John H. Lodge, Communications Research Centre, (Canada)

Volume 3, page 1797, paper no. 1753

Abstract:
A set of desirable characteristics of a multicarrier spectrum-efficient bandwidth-on-demand multiplexer-demultiplexer pair for use in mobile satellite and personal communication systems is identified and described. New characteristics are the use of single VSB channels, design of multiplexer channels based on wavelet packet trees which have specified stopband attenuation, overlap of the multiplexer channel magnitude frequency responses at the 3-dB points for spectral efficiency, bandwidth on demand, reasonable lengths for the overall equivalent filters for each multiplexer-demultiplexer channel from input to receiver output, and low-complexity receivers. A multirate digital transmultiplexer is proposed consisting of a wavelet packet-based synthesis filter bank tree followed by a DFT polyphase synthesis filter bank at the transmitter, and a matching demultiplexer at the receiver. A simplified receiver for reception of one channel at a time is described. BER performances when there are phase and timing errors are given.
ICASSP98 Paper Abstract
Arbitrary Bandwidth Wavelet Sets

Authors:
Gianpaolo Evangelista, Univ. Federico II, Napoli, (Italy)
Sergio Cavaliere, Univ. Federico II, Napoli, (Italy)

Abstract:
In this paper we consider an extension of the wavelet transform leading to the construction of wavelets with arbitrary bandwidth. The new wavelets are complete, orthonormal and dyadic; nevertheless their bandwidth is not constrained to be one octave, rather it may be designed by selecting a set of parameters. The construction of the new bases starts in the discrete-time domain, exploiting properties of the Laguerre transform. Furthermore, we provide a procedure to define continuous-time warped wavelets. Flexibility of the bandwidth allocation allows for more and improved applications of the wavelet transform, such as signal coding, the design of auditory model based filterbanks and transient detection in pseudoperiodic signals, pointed out in the paper.
ICASSP98 Paper Abstract

Non-Invasive Quantification of Physiological Processes with Dynamic PET using Blind Deconvolution

Authors:
Chi-Hoi Lau, Hong Kong Polytechnic University, (Hong Kong)
Daniel Pak-Kong Lun, Hong Kong Polytechnic University, (Hong Kong)
Dagan Feng, University of Sydney, (Australia)

Abstract:
Dynamic Positron Emission Tomography (PET) has opened the possibility of quantifying physiological processes within the human body. On performing dynamic PET studies, the tracer concentration in blood plasma has to be measured, and acts as the input function for tracer kinetic modelling. In this paper, we propose an approach to estimate physiological parameters for dynamic PET studies without the need of taking blood samples. The proposed approach comprises two major steps. First, a wavelet denoising technique is used to filter the noise appeared in the projections. The denoised projections are then used to reconstruct the dynamic images using filtered backprojection. Second, an eigen-vector based blind deconvolution technique is applied to the reconstructed dynamic images to estimate the physiological parameters. To demonstrate the performance of the proposed approach, we carried out a Monte Carlo simulation using the fluoro-deoxy-2-glucose model, as applied to tomographic studies of human brain. The results demonstrate that the proposed approach can estimate the physiological parameters with an accuracy comparable to that of invasive approach which requires the tracer concentration in plasma to be measured.
ICASSP98 Paper Abstract

Compressing ECG Signals by Piecewise Polynomial Approximation

Authors:
Ranveig M. Nygaard, Stavanger College, (Norway)
Dag Haugland, Stavanger College, (Norway)

Volume 3, page 1809, paper no. 1742

Abstract:
Compression of digital ElectroCardioGram (ECG) signals has traditionally been tackled by heuristical approaches. Recently, it has been demonstrated that exact optimization algorithms outclass these heuristical approaches by wide margin with respect to reconstruction error. As opposed to traditional time-domain algorithms, where some heuristic is used to extract representative signal samples from the original signal, the exact optimization algorithm presented here formulates the sample selection problem as a graph theory problem. Thus well known optimization theory can be applied in order to yield optimal compression. This paper generalizes an existing exact optimization algorithm such that reconstruction can be made by second order polynomial interpolation in the extracted signal samples. The polynomials are fitted in a way that guarantees minimal reconstruction error, and the method proves good performance compared to the case where linear interpolation is used in reconstruction of the signal.
ICASSP98 Paper Abstract

Characterization of Autonomic Release Sites Using The Time-Frequency Analysis of Junction Potentials in Smooth Muscle

Authors:
Priya J Godbole, Institute of Technology, Bombay, (India)
Rohit Manchanda, Institute of Technology, Bombay, (India)
Uday B Desai, Institute of Technology, Bombay, (India)
Krisheti Venkateswarlu, Institute of Technology, Bombay, (India)

Volume 3, page 1813, paper no. 1104

Abstract:
The determination of the probability of neurotransmitter release from neuronal release sites and their electrical characterization is an issue of central interest in neuropsychology. For autonomic nerves, this can be done by analysing the inflexions in the rising phases of the evoked junction potentials (EJPs) recorded from smooth muscle. Since these inflexions contain time-varying frequency information, we have applied recent methods of time-frequency analysis, based upon wavelet transforms, on EJPs to characterize autonomic neuronal function. We find that these methods allow accurate and convenient characterization of individual release sites, and that their probability of release falls between 0.002 and 0.003. These results are compared with those reported earlier using analogue filtering techniques. The present method is advantageous as regards automation, accuracy and suppression of noise.
Designing Frames for Matching Pursuit Algorithms

Authors:
Kjersti Engan, Stavanger College, (Norway)
Sven Ole Aase, Stavanger College, (Norway)
John Hakon Husøy, Stavanger College, (Norway)

Volume 3, page 1817, paper no. 1731

Abstract:
A technique for designing frames to use with vector selection algorithms, for example Matching Pursuits (MP), is presented. The design algorithm is iterative and requires a training set of signal vectors. An MP algorithm chooses frame vectors to approximate each training vector. Each vector in the frame is then adjusted by using the residuals for the training vectors which used that particular frame vector in their expansion. The frame design algorithm is applied to speech and electrocardiogram (ECG) signals, and the designed frames are tested on signals outside the training sets. Experiments demonstrate that the approximation capabilities, in terms of mean square error (MSE), of the optimized frames are significantly better than those found using frames designed by ad-hoc techniques. Experiments show typical reduction in MSE by 20 - 50%.
ICASSP98 Paper Abstract
Complex Wavelet Packets for Multicarrier Modulation

Authors:
Tushar K. Adhikary, Institute of Science, (India)
Vellenki U. Reddy, Institute of Science, (India)

Abstract:
In this paper, we first discuss two approaches for designing complex wavelet packets which can be used as orthogonal carriers for modulations like QAM and PM, and then compare the performance of the wavelet packet based modulation scheme with that of discrete multitone modulation using DFT bases. The results show that the wavelet packet based scheme yields lower average bit error probability compared to the DFT based scheme. The improved performance of the wavelet packet based scheme is because of the spectrally contained nature of the wavelet packet bases which are under the control of the designer.
ICASSP98 Paper Abstract
Transmission of Two Users by Means of Periodic Clock Changes

Authors:
Alban Duverdier, ENSEEIHT/SIC, (France)
Bernard Lacaze, ENSEEIHT/SIC, (France)

Volume 3, page 1825, paper no. 1947

Abstract:
In modern telecommunications, it is often necessary to transmit several informations at the same time. It corresponds to multi-user transmission. In this paper, we present a new multi-user method by means of linear periodic time-varying filters. For two users, it is seen that the use of periodic clock changes simplifies the reconstruction. We apply this method to transmission of two stationary binary signals. Simulations show that perfect reconstruction is possible.
ICASSP98 Paper Abstract
Asymptotically Perfect Reconstruction in Hybrid Filter Banks

Authors:
Omid Oliaei, ENST-ELEC, (France)

Volume 3, page 1829, paper no. 1589

Abstract:
A procedure to derive a hybrid filter bank from a digital filter bank is presented. Perfect reconstruction is shown to be possible only asymptotically. The stability of the analog filters is ensured if FIR or IIR stable filters are used in the digital prototype.
ICASSP98 Paper Abstract

Myopic Deconvolution Combining Kalman Filter and Tracking Control

Authors:
Patrick Sarri, Laboratoire d'Automatique Industrielle, INSA Lyon, (France)
Gerard Thomas, Ecole Centrale de Lyon, (France)
Edgar Sekko, Universite Claude Bernard, (France)
Philippe Neveux, Universite Claude Bernard, (France)

Volume 3, page 1833, paper no. 2499

Abstract:
In this paper, we propose a deconvolution method based on discrete-time optimal control. By combining Kalman filtering with optimal control, we state the problem in terms of tracking problem. This leads to solve a set of recurrent equations, including in particular a matrix Riccati equation. We present a method that transforms the solution of these recurrent equations in that of a linear system of equations. Once the linear system has been set up, the deconvolution procedure becomes very fast, and permits on-line deconvolution. It is also possible to use the discrete impulsional response, and perform blind deconvolution. This technique include a Kalman filter. Numerical examples illustrate the robustness of the procedure.
ICASSP98 Paper Abstract

An Irregular Sampling Algorithm Adapted to the Local Frequency Content of Signals and the Corresponding On-Line Reconstruction Algorithm

Authors:
Voicu I Filimon, Daimler Benz Forschungsinstitut, (Germany)

Volume 3, page 1837, paper no. 1134

Abstract:
Description of signals using wavelet transforms leads to useful time-frequency localization and possible signal compression. Based on the Discrete Wavelet Transform (DWT) an adaptive sampling algorithm in the discrete time domain is constructed, by finding an univocal relation between the signal's samples and the non-zero transform coefficients of its DWT. Reconstruction is performed through repeated projections of an approximation of the initial signal based on the arriving samples, into the original signal's subspace, using the Neumann method of inverting bounded operators. Both adaptive sampling and reconstruction are on-line because of the finite support of the analyzing wavelets.
ICASSP98 Paper Abstract

A Poly-Phase Based Blind Deconvolution Technique Using Second-Order Statistics

Authors:
Hiroshi Ochi, University of the Ryukyu, (Japan)
Mirai Oshiro, University of the Ryukyu, (Japan)

Volume 3, page 1841, paper no. 2273

Abstract:
A novel second-order statistics-based blind deconvolution and equalizer technique is proposed in this literature. This technique makes use of a two-channel perfect reconstruction filter bank derived from a two-component poly-phase decomposition of transmission channel in order to make exact system identifications possible. The proposed blind deconvolution algorithm is superior to conventional algorithms in view of simple structure and the uniqueness of solution. In order to verify the effectiveness of this method, several computer simulations including a 256 QAM signal equalizer and a blurred image recovery have been shown.
ICASSP98 Paper Abstract

DSP Instead of Circuits? - Transition to Undergraduate DSP Education at Rose-Hulman

Authors:
Wayne T Padgett, Rose-Hulman Institute of Tecnology, (U.S.A.)
Mark A Yoder, Rose-Hulman Institute of Tecnology, (U.S.A.)

Volume 3, page 1845, paper no. 2230

Abstract:
We assert that digital signal processing (DSP) can and should be taught early (sophomore-junior) in the electrical and computer engineering curricula. This paper looks at the impact this has on the rest of the curriculum, both in electrical and computer engineering and in other engineering curricula. While the early introduction of basic DSP makes it possible to offer better senior electives and graduate courses in DSP, the biggest benefit is the ability to build on DSP core concepts just as we have traditionally built on circuits core concepts in the past. Further, motivational examples in DSP lend themselves to multimedia and are often more familiar to today’s students than basic circuits.
ICASSP98 Paper Abstract

A Java Signal Analysis Tool for Signal Processing Experiments

Authors:
Axel Clausen, Arizona State University, (U.S.A.)
Andreas Spanias, Arizona State University, (U.S.A.)
Anand Xavier, Arizona State University, (U.S.A.)
Maya Tampi, Arizona State University, (U.S.A.)

Volume 3, page 1849, paper no. 2053

Abstract:
In this paper, a program to simulate discrete time linear systems is presented. The software is written as a Java applet and can be accessed on the internet. Object oriented programming allows the user to construct and simulate a variety of systems. The program uses a graphical user interface which is easy to learn and it provides a visualization of the system and signal flow. The software is currently being used at Arizona State University to support an online software laboratory for a senior level digital signal processing (DSP) course. This paper presents our experiences gained by using the program in a class setting and gives examples of possible laboratory problems.
Visualization of Signal Processing Concepts

Authors:
Janna Shaffer, Mississippi State University, (U.S.A.)
Jonathan Hamaker, Mississippi State University, (U.S.A.)
Joseph Picone, Mississippi State University, (U.S.A.)

Abstract:
One of the key difficulties in a Signals and Systems course is the visualization of mathematically complex concepts presented. Thus, there is a need for graphical tools which enhance the students' comprehension of these difficult concepts by allowing interactive learning. In this paper we present a software package to assist in the explanation and visualization of signal processing concepts for an educational environment. We provide a set of Java-based tools for understanding the concepts of convolution, spectral analysis, and pole/zero system response. The wide availability and platform-independence provided by Java make this tool highly portable and easily accessible to a broader audience of students than comparable systems based on Matlab or other commercial software. The software described in this paper is available in the public domain at our website: http://isip.msstate.edu/.
An Experimental Architecture for Interactive Web-based DSP Education

Authors:
Martti Rahkila, Helsinki University of Technology, (Finland)
Matti Karjalainen, Helsinki University of Technology, (Finland)

Abstract:
This paper describes an experimental architecture for interactive education of DSP in the World Wide Web environment. The architecture is based on a client-server model providing means of distributing resources. The major design goal has been to combine interactivity and computational resources. The architecture itself is open in a sense that it does not specify implementation and it can be used for a variety of applications. Computer Based Education (CBE) of DSP is one area where it can be beneficially applied. An example of implementation for that purpose is presented. With our implementation, special attention has been paid to minimize the requirements of additional software for the user because of educational reasons. Thus only a Java-capable browser and audio support are needed at the user end.
ICASSP98 Paper Abstract
Signal Processing with the Sparseness Constraint

Authors:
Bhaskar D. Rao, University of California, San Diego, (U.S.A.)

Volume 3, page 1861, paper no. 2199

Abstract:
An overview is given of the role of the sparseness constraint in signal processing problems. It is shown that this is a fundamental problem deserving of attention. This is illustrated by describing several applications where sparseness of solution is desired. Lastly, a review is given of some of the algorithms that are currently available for computing sparse solutions.
ICASSP98 Paper Abstract
Application of Basis Pursuit in Spectrum Estimation

Authors:
Scott Shaobing Chen, IBM, (U.S.A.)
David L. Donoho, Stanford University, (U.S.A.)

Volume 3, page 1865, paper no. 5259

Abstract:
In this paper, we apply Basis Pursuit, an atomic decomposition technique, for spectrum estimation. Compared with several modern time series methods, our approach can greatly reduce the problem of power leakage: it is able to superresolve; moreover, it works well with noisy and unevenly sampled signals. We present experiments on bizarrely spaced radial velocity data from one of the newly-discovered extra planetary systems.
ICASSP98 Paper Abstract
Parsimony and Wavelet Methods for Denoising

Authors:
Hamid Krim, MIT, (U.S.A.)
Jean-Christophe Pesquet, University Paris Sud, (France)
Irvin C Schick, GTE Internetworking and Harvard University, (U.S.A.)

Volume 3, page 1869, paper no. 2465

Abstract:
Some wavelet-based methods for signal estimation in the presence of noise are reviewed in the context of the parsimonious representation of the underlying signal. Three approaches are considered. The first is based on the application of the MDL principle. The robustness of this method is improved in the second approach, by relaxing the assumption of known noise distribution following Huber's work. In the third approach, a Bayesian strategy is adopted in order to incorporate prior information pertaining to the signal of interest; this method is especially useful at low signal-to-noise ratios.
ICASSP98 Paper Abstract
Parsimonious Side Propagation

Authors:
Paul S Bradley, *University of Wisconsin-Madison, (U.S.A.)*
Olvi L Mangasarian, *University of Wisconsin-Madison, (U.S.A.)*

Volume 3, page 1873, paper no. 1221

Abstract:
A fast parsimonious linear-programming-based algorithm for training neural networks is proposed that suppresses redundant features while using a minimal number of hidden units. This is achieved by propagating sideways to newly added hidden units the task of separating successive groups of unclassified points. Computational results show an improvement of 26.53 % and 19.76 % in tenfold cross-validation test correctness over a parsimonious perceptron on two publicly available datasets.
ICASSP98 Paper Abstract

Fast Optimal and Suboptimal Algorithms for Sparse Solutions to Linear Inverse Problems

Authors:
Gopal Harikumar, Tellabs Research, (U.S.A.)
Christophe Couvreur, University of Illinois, Urbana-Champaign, (U.S.A.)
Yoram Bresler, University of Illinois, Urbana-Champaign, (U.S.A.)

Abstract:
We present two ""fast"" approaches to the NP-hard problem of computing a maximally sparse approximate solution to linear inverse problems, also known as best subset selection. The first approach, a heuristic, is an iterative algorithm globally convergent to sparse elements of any given convex, compact set. We demonstrate its effectiveness in bandlimited extrapolation and in sparse filter design. The second approach is a polynomial-time greedy sequential backward elimination algorithm. We show that if A has full column rank and e is small enough, then the algorithm will find the sparsest x satisfying \( \|Ax - b\| < \epsilon \) (or equal to) e, if such exists.
ICASSP98 Paper Abstract
Measures and Algorithms for Best Basis Selection

Authors:
Kenneth Kreutz-Delgado, University of California, San Diego, (U.S.A.)
Bhaskar D. Rao, University of California, San Diego, (U.S.A.)

Volume 3, page 1881, paper no. 2190

Abstract:
A general framework based on majorization, Schur-concavity, and concavity is given that facilitates the analysis of algorithm performance and clarifies the relationships between existing proposed diversity measures useful for best basis selection. Admissible sparsity measures are given by the Schur-concave functions, which are the class of functions consistent with the partial ordering on vectors known as majorization. Concave functions form an important subclass of the Schur-concave functions which attain their minima at sparse solutions to the basis selection problem. Based on a particular functional factorization of the gradient, we give a general affine scaling optimization algorithm that converges to a sparse solution for measures chosen from within this subclass.
ICASSP98 Paper Abstract
Sparse Inverse Solution Methods for Signal and Image Processing Applications

Authors:
Brian D Jeffs, Brigham Young University, (U.S.A.)
Volume 3, page 1885, paper no. 2309

Abstract:
This paper addresses image and signal processing problems where the result most consistent with prior knowledge is the minimum order, or "maximally sparse" solution. These problems arise in such diverse areas as astronomical star image deblurring, neuromagnetic image reconstruction, seismic deconvolution, and thinned array beamformer design. An optimization theoretic formulation for sparse solutions is presented, and its relationship to the MUSIC algorithm is discussed. Two algorithms for sparse inverse problems are introduced, and examples of their application to beamforming array design and star image deblurring are presented.
ICASSP98 Paper Abstract

Image Denoising Using Multiple Compaction Domains

Authors:
Prakash Ishwar, University of Illinois, Urbana-Champaign, (U.S.A.)
Krishna C. Ratakonda, University of Illinois, Urbana-Champaign, (U.S.A.)
Pierre Moulin, University of Illinois, Urbana-Champaign, (U.S.A.)
Narendra Ahuja, University of Illinois, Urbana-Champaign, (U.S.A.)

Volume 3, page 1889, paper no. 5249

Abstract:
We present a novel framework for denoising signals from their compact representation in multiple domains. Each domain captures, uniquely, certain signal characteristics better than others. We define confidence sets around data in each domain and find sparse estimates that lie in the intersection of these sets, using a POCS algorithm. Simulations demonstrate the superior nature of the reconstruction (both in terms of mean-square error and perceptual quality) in comparison to the adaptive Wiener filter.
ICASSP98 Paper Abstract
A Different First Course in Electrical Engineering

Authors:
Don H. Johnson, Rice University, (U.S.A.)
J.D. Wise Jr, Rice University, (U.S.A.)

Volume 3, page 1893, paper no. 2183

Abstract:
Traditional introductory courses in electrical engineering are typically circuit theory courses, and may include both analog and digital hardware and possibly software. Recent alternatives have focused on how to teach (using discrete-time signals rather than analog) than on what to teach. We have developed a top-down course sequence that uses as its underlying principle the transmission and manipulation of information. Students are given a broad perspective of both analog and digital approaches, with a goal of helping students appreciate electrical and computer engineering and framing a context for their advanced courses. Laboratories stress construction of analog systems and analysis with signal processing tools.
ICASSP98 Paper Abstract

Integrating Engineering Design, Signal Processing, and Community Service in the EPICS Program

Authors:
Leah H. Jamieson, Purdue University, (U.S.A.)
Edward J. Coyle, Purdue University, (U.S.A.)
Mary P. Harper, Purdue University, (U.S.A.)
Edward J. Delp, Purdue University, (U.S.A.)

Volume 3, page 1897, paper no. 2534

Abstract:
One of the most challenging problems in engineering - and signal processing - education is providing realistic and meaningful design experience. In the Engineering Projects in Community Service (EPICS) program, teams of engineering undergraduates earn academic credit for multi-year projects that solve technology-based problems for community organizations. Key features of EPICS include the long-term nature of the projects; the emphasis on "real-world" start-to-finish design; the learning experience embodied in solving ambitious engineering problems; the vertical, multidisciplinary teams; the development of teamwork and communication skills; and the use of engineering to help the community. We describe the EPICS program and highlight three EPICS signal processing projects: a real-time system to measure speaking rate for Purdue’s speech clinic; voice-controlled interactive software to encourage speech in developmentally delayed children; and an image and video archiving system for the Tippecanoe County Historical Association. Full program and project descriptions are at www.ecn.purdue.edu/epics.
ICASSP98 Paper Abstract

An Interactive Learning Environment for Adaptive Systems

Authors:
Jose C. Principe, University of Florida, (U.S.A.)
Neil Euliano, University of Florida, (U.S.A.)
Curt Lefebvre, Neurodimension, (U.S.A.)

Volume 3, page 1901, paper no. 2158

Abstract:
We developed a new computer based learning environment to teach adaptive systems to EE undergraduate students. The electronic book is composed of an hypertext document linked with a software simulator which runs examples on-line, and is fully controlled by the student.
ICASSP98 Paper Abstract
Interactive DSP Education Using Java

Authors:
Yves Cheneval, Swiss Federal Institute of Technology, (Switzerland)
Laurent Balmelli, Swiss Federal Institute of Technology, (Switzerland)
Paolo Prandoni, Swiss Federal Institute of Technology, (Switzerland)
Jelena Kovacevic, Bell Labs Lucent Technologies, (U.S.A.)
Martin Vetterli, Swiss Federal Institute of Technology, (Switzerland)

Volume 3, page 1905, paper no. 5257

Abstract:
In this paper, we argue that Java is a natural language to develop interactive teaching material that can be shared and distributed widely. Unlike any other programming language or platform we know, Java development is justified because of its almost universal acceptance. We develop a Block Diagram (BD) based approach that allows to develop interactive and downloadable signal processing laboratories. As an example, we show how specific experiments for a DSP class, as well as for an advanced course on wavelets have been developed. The article first explains why the Java language has been chosen, and then describes what has been realized today. Finally, we show how the BD representation can be efficiently used for the development of a wavelet theory course. It is shown that only a few simple blocks are sufficient for creating many didactic programs. This can be seen as an a posteriori justification of the BD model.
Multi-Platform CBI Tools Using Linux and Java-Based Solutions for Distance Learning

Authors:
Eric K. Patterson, Clemson University, (U.S.A.)
Duanpei Wu, Sony Corporation, (U.S.A.)
John N. Gowdy, Clemson University, (U.S.A.)

Abstract:
This paper presents a set of software tools developed to enhance local and distance learning for speech and signal processing classes. The multi-platform paradigm is stressed in both computer-based instruction (CBI) and World Wide Web (WWW), Java-based tools. The CBI speech tools are a package developed to enhance a graduate course in digital speech processing; the tools may be run on various platforms and will benefit local or distance learners taking the course via teleconferencing. Students may gain "hands-on" experience using this package on their own personal computer (PC) or a university computer. The WWW, Java-based tools have been designed to be used for interactive homework in signal processing classes but also have a wide range of applicability. JavaGram is the name for the currently developed application that allows students to easily "turn-in" homework diagrams, such as signal-flow charts, via the WWW. This is an improvement over past systems that use only forms in WWW homework for remote classes. These tools have been found to greatly enhance local and remote offerings of a course on speech signal processing.
ICASSP98 Paper Abstract
Exploiting Structure in Positioning of Non-Symmetric Signals

Authors:
Amir A. Ghazanfarian, Stanford University, (U.S.A.)
Xun Chen, Stanford University, (U.S.A.)
Thomas Kailath, Stanford University, (U.S.A.)
Mark A. McCord, Stanford University, (U.S.A.)
Fabian W. Pease, Stanford University, (U.S.A.)

Volume 4, page 1913, paper no. 1007

Abstract:
One of the most crucial emerging challenges in lithography is achieving rapid and accurate alignment under a wide variety of conditions brought about by different processing steps. Current alignment algorithms assume symmetric alignment signals. In this paper, we propose a new algorithm based on subspace decomposition of alignment signals. We assume that the process-induced asymmetries are small enough so that only linear effects need to be considered. We first find the subspace of alignment signals using a set of signals with pre-known positions. The position of a new signal is calculated considering that, if shifted correctly, it will lie in the same subspace of previous signals. Since this method exploits the structure of the signals, it results in more accurate measurement of the position. Simulation results show that the alignment error is about an order of magnitude smaller than that achieved with conventional Maximum Likelihood or phase-fitting approaches.
ICASSP98 Paper Abstract

On-line Detection/Estimation Scheme in Laser Doppler Anemometry

Authors:
Frederic Galtier, ENSICA, (France)
Olivier Besson, ENSICA, (France)

Volume 4, page 1917, paper no. 1020

Abstract:
Laser anemometers have become a promising technique for estimating velocities in a flow. In this paper, we study their use for on-board aircraft speed of flight estimation. More specifically, this paper addresses the problem of simultaneous detection of the arrival of aerosol particles in a laser anemometer and estimation of their velocity. A joint detection-estimation scheme is proposed. A Likelihood Ratio Test is presented and considerations about the specificities of the problem are used to calculate the threshold. Computationally efficient algorithms for estimating the parameters of interest are derived and on-line implementation issues are addressed. Numerical examples attest for the performance of the method, on both simulated and real data recorded during a flight test.
ICASSP98 Paper Abstract

Tripulse: An Accurate Orientation/Attitude Estimation System for Satellite Borne Phased Arrays

Authors:
Seth D Silverstein, GE Corporate Research and Development, (U.S.A.)
Jeffrey M Ashe, GE Corporate Research and Development, (U.S.A.)
Gregory M Kautz, GE Corporate Research and Development, (U.S.A.)
Frederick W Wheeler, GE Corporate Research and Development, (U.S.A.)

Volume 4, page 1921, paper no. 1162

Abstract:
Tripulse is a novel orientation/attitude estimation system that is designed to accurately estimate the orientation of a satellite borne phased array relative to one or more earth stations. This system has an accuracy potential that is significantly better than conventional Earth-Sun-Moon attitude sensors. Tripulse has conceptual similarities to amplitude comparison onopulse systems used in tracking radars. Detailed Tripulse statistical performance analyses for noise, beamforming quantization errors, and hardware failures are presented.
ICASSP98 Paper Abstract

Automatic Detection System of Venous Air Embolism Employing Signal Processing Methods

Authors:
Ralf Schlag, Universität Kaiserslautern, (Germany)
Ulrich Korell, Klinikum Kaiserslautern, (Germany)
Bernd Siegmund, Universität Kaiserslautern, (Germany)
Martin Pfeiffer, Universität Kaiserslautern, (Germany)
Madhukar Pandit, Universität Kaiserslautern, (Germany)

Volume 4, page 1925, paper no. 1237

Abstract:
Methods of signal processing which have been developed and tested for the detection of venous air embolism using ultrasound Doppler systems are presented. The detection scheme developed is based on a time-frequency characterization of the Doppler signals obtained with a suitable transducer placed on the cartoid vein. The developed scheme has been implemented and tested for the automatic signaling of an embolism which can occur in the course of a surgical operation.
ICASSP98 Paper Abstract

Undersampling for Parameter Estimation with Application to Time of Arrival Estimation

Authors:
Hagit Messer, Tel Aviv University, (Israel)

Volume 4, page 1929, paper no. 1537

Abstract:
This paper deals with the effect of sampling the continuous observations on parameter estimation errors. In particular, we study the problem of estimating the time of arrival (TOA) of a continuous, deterministic signal in noise. For this problem, the sampling procedure transforms the continuous parameter space into a discrete one, resulting in inherent estimation errors. We introduce a general tool for evaluating the achievable performance for any parameter estimation problem at a given sampling rate. For TOA estimation with a Gaussian-shaped signal, we show that one can undersample with a factor up to 3 times the Nyquist rate with average TOA estimation performance reduction of less than 3dB.
ICASSP98 Paper Abstract

Nonlinear System Identification of Hydraulic Actuator Friction Dynamics Using a Hammerstein Model

Authors:
Byung-Jae Kwak, University of Michigan, (U.S.A.)
Andrew E. Yagle, University of Michigan, (U.S.A.)
Joel A Levitt, The Ford Motor Company, (U.S.A.)

Abstract:
We present two Hammerstein-type models for parametric system identification of the lip seal friction process in a hydraulic actuator. Adaptive algorithms with least-squares criteria are derived, and the performances of the two models are evaluated using experimental results.
Information-Theoretic Analysis of Neural Coding

Authors:
Don H. Johnson, Rice University, (U.S.A.)
Charlotte M Gruner, Rice University, (U.S.A.)

Abstract:
We describe a family of new techniques for analyzing single- and multi-unit discharge patterns. These techniques are based on information theoretic distance measures and on empirical theories derived from work on universal signal processing. They are capable of determining transneuron statistical dependencies even when time-varying responses occur. The response portion contributing most to information coding can be identified and the coding fidelity can be quantified regardless of the neural coding mechanisms—be it timing, rate or transneural correlations.
ICASSP98 Paper Abstract
Multidimensional Independent Component Analysis

Authors:
Jean-Francois Cardoso, CNRS & ENST, (France)

Abstract:
This discussion paper proposes to generalize the notion of Independent Component Analysis (ICA) to the notion of Multidimensional Independent Component Analysis (MICA). We start from the ICA or blind source separation (BSS) model and show that it can be uniquely identified provided it is properly parameterized in terms of one-dimensional subspaces. From this standpoint, the BSS/ICA model is generalized to multidimensional components. We discuss how ICA standard algorithms can be adapted to MICA decomposition. The relevance of these ideas is illustrated by a MICA decomposition of ECG signals.
ICASSP98 Paper Abstract
Subspace Domain Forwards-Backwards Averaging

Authors:
Michael A Zatman, MIT Lincoln Lab, (U.S.A.)

Volume 4, page 1945, paper no. 1060

Abstract:
In this paper a procedure which filters out roughly half of the array manifold errors for approximately centro-symmetric arrays is described. The procedure - subspace domain forwards-backwards averaging - improves the performance of subspace based direction finding algorithms such as MUSIC and ESPRIT. Experimental data from the Mountaintop system are used to confirm the theoretical results.
ICASSP98 Paper Abstract

Closed-Form Direction-Finding with Arbitrarily Spaced Electromagnetic Vector-Sensors at Unknown Locations

Authors:
Kainam Thomas Wong, Nanyang Technological University, (Singapore)
Michael D. Zoltowski, Purdue University, (U.S.A.)

Volume 4, page 1949, paper no. 1184

Abstract:
This paper introduces a novel closed-form ESPRIT-based algorithm for multi-source direction finding using arbitrarily spaced electromagnetic vector-sensors whose locations need not be known. The electromagnetic vector-sensor, already commercially available, consists of six co-located but diversely polarized antennas separately measuring all six electromagnetic-field components of an incident wavefield. In this novel algorithm, ESPRIT exploits the non-spatial inter-relations among the six unknown electromagnetic-field components of each source and produces from the measured data a set of eigenvalues, from which the source’s electromagnetic-field vector may be estimated to within a complex scalar. Application of a vector cross-product operation to this ambiguous electromagnetic-field vector estimate produces an unambiguous estimate of that source’s normalized Poynting-vector, which contains as its components the source’s Cartesian direction-cosines. Monte Carlo simulation results verify the efficacy and versatility of this innovative scheme.
ICASSP98 Paper Abstract
Subspace Tracking using a Constrained Hyperbolic URV Decomposition

Authors:
Alle-Jan Van Der Veen, Delft University of Technology, (The Netherlands)
Volume 4, page 1953, paper no. 1211

Abstract:
The class of Schur subspace estimators provides a parametrization of all minimal-rank matrix approximants that lie within a specified distance of a given matrix, and in particular gives expressions for the column spans of these approximants. In this paper, we derive an updating algorithm for an interesting member of the class, making use of a constrained hyperbolic URV-like decomposition.
ICASSP98 Paper Abstract

Joint Angle-Frequency Estimation Using Multi-Resolution Estrip

Authors:
Aweke N Lemma, Delft University of Technology, (The Netherlands)
Alle-Jan Van Der Veen, Delft University of Technology, (The Netherlands)
Ed F. Deprettere, Delft University of Technology, (The Netherlands)

Volume 4, page 1957, paper no. 1258

Abstract:
Multi-resolution ESPRIT is an extension of the ESPRIT direction finding algorithm to antenna arrays with multiple baselines. A short (half wavelength) baseline is necessary to avoid aliasing, a long baseline is preferred for accuracy. The MR-ESPRIT algorithm allows to combine both estimates. The same algorithm can be used for multi-resolution frequency estimation using two different sampling frequencies. We show how this can be used to construct a joint angle-frequency estimator.
ICASSP98 Paper Abstract

Robust Weighted Subspace Fitting in the Presence of Array Model Errors

Authors:
Magnus Jansson, Royal Institute of Technology, (Sweden)
Arnold Lee Swindlehurst, Brigham Young University, (U.S.A.)
Björn Ottersten, Royal Institute of Technology, (Sweden)

Volume 4, page 1961, paper no. 1322

Abstract:
Model error sensitivity is an issue common to all high resolution direction of arrival estimators. Much attention has been directed to the design of algorithms for minimum variance estimation taking only finite sample errors into account. Approaches to reduce the sensitivity due to array calibration errors have also appeared in the literature. Herein, a weighted subspace fitting method for a wide class of array perturbation models is derived. This method provides minimum variance estimates under the assumption that the prior distribution of the perturbation model is known. Interestingly enough, the method reduces to the WSF (MODE) estimator in no model errors are present. On the other hand, when model errors dominate, the proposed method turns out to be equivalent to the "Model-errors-only subspace fitting method". Unlike previous techniques for model errors, the estimator can be implemented using a two-step procedure if the nominal array is uniform and linear, and it is also consistent even if the signals are fully correlated.
ICASSP98 Paper Abstract

Robust Adaptive Subspace Detectors for Space Time Processing

Authors:
Ariela Zeira, Signal Processing Technology Ltd., (U.S.A.)
Benjamin Friedlander, University of California, Davis, (U.S.A.)

Volume 4, page 1965, paper no. 1851

Abstract:
In this paper we consider the problem of detecting a subspace signal when there is uncertainty in the subspace. Such uncertainty usually causes a mismatch between the detector and the signal to be detected, which may lead to significant loss in performance. To improve the robustness of the detection procedure we apply robust adaptive subspace detectors based on extending the dimension of the signal subspace. We consider two types of adaptive constant false alarm rate (CFAR) detector structures: CFAR generalized likelihood ratio detector (CFAR GLR) and CFAR matched subspace detector (CFAR MSD). Using Monte-carlo simulations, we study the performance of the robust adaptive subspace detectors for space-time processing.
ICASSP98 Paper Abstract

A Geometrical Framework for the Determination of Ambiguous Directions in Subspace Methods

Authors:
Anne Flieller, LESiR, (France)
Pascal Larzabal, LESiR, (France)
Henri Clergeot, LESiR, (France)

Volume 4, page 1969, paper no. 1971

Abstract:
In subspace parameter estimation techniques, like MUCSIC, degradations may occur due to parasite peaks in the pectrum, which may be connected to high sidelobes in the beam pattern or to ambiguities themselves. The aim of this paper is to study the presence of ambiguities in an array of given planar geometry. We propose a general framework for the analysis and therefore we obtain a generalisation of results given in recent publications for rank one and two ambiguities. For rank k greater than three ambiguities, the study is restricted to linear arrays, for which we derive original and synthetic results. We present a geometrical construction able to determine all the ambiguous directions which can appear for a given linear array. The method allows determination of any rank ambiguities and for each ambiguous direction set, the rank of ambiguity is obtained. The search is exhaustive. Application of the method requires no assumption for the linear array and is easy to implement. An example is detailed for a non uniform linear array.
Weighted Subspace Fitting Using Subspace Perturbation Expansions

Authors:
Richard J. Vaccaro, University of Rhode Island, (U.S.A.)

Abstract:
This paper presents a new approach to deriving statistically optimal weights for weighted subspace fitting (WSF) algorithms. The approach uses a formula called a “subspace perturbation expansion,” which shows how the subspaces of a matrix change when the matrix elements are perturbed. The perturbation expansion is used to derive an optimal WSF algorithm for estimating directions of arrival in array signal processing.
ICASSP98 Paper Abstract

Case Study of Principal Component Inverse and Cross Spectral Metric for Low Rank Interference Adaptation

Authors:
Brian E Freburger, University of Rhode Island, (U.S.A.)
Donald W Tufts, University of Rhode Island, (U.S.A.)

Volume 4, page 1977, paper no. 2086

Abstract:
This paper presents a review of the Principal Component Inverse method of rapidly adaptive signal detection and contrasts the use of Principal Components with the more recent Cross Spectral Metric method for the Generalized Sidelobe Canceller. The CSM method is optimal with known statistics and has been shown to outperform the PCI method in many cases of unknown covariance. This paper describes a scenario which represents a class of covariances where the PCI method can be expected to outperform the CSM method. The choice of method is therefore more subtle than previously thought.
ICASSP98 Paper Abstract

On Sources Covariance Matrix Singularities and High-Resolution Active Wideband Source Localization

Authors:
Daniel Goncalves, CEPHAG, (France)
Patrick Gounon, CEPHAG, (France)

Volume 4, page 1981, paper no. 2279

Abstract:
High resolution eigenstructure-based techniques for signal source localization are known to be ineffective when the source covariance matrix is not of full rank. We present here two techniques to circumvent this problem in the context of wideband active source localization. An extension is made to show how eigenstructure methods can be applied even when there is only one snapshot available to estimate the wideband spectral matrices.
ICASSP98 Paper Abstract

Processing of Experimental Seismic Array Data Using 2-D Wideband Interpolated Root-MUSIC

Authors:
Dmitri V. Sidorovich, Ruhr University of Bochum, (Germany)
Alex B. Gershman, Ruhr University of Bochum, (Germany)
Johann F. Bohme, Ruhr University of Bochum, (Germany)

Volume 4, page 1985, paper no. 1047

Abstract:
2-D elementspace and beamspace extensions of Friedlander's wideband interpolated root-MUSIC are applied to azimuth-velocity source location using real seismic data from the GERESS array (Germany). We demonstrate that 2-D interpolated root-MUSIC is able to estimate the parameters of a typical seismic source with a good accuracy. The use of interpolated root-MUSIC and its beamspace modification is motivated by the significant reduction of processing time allowing on-line implementation.
ICASSP98 Paper Abstract

Nonstationary Array Signal Detection Using Time-Frequency and Time-Scale Representations

Authors:
Anil M Rao, University of Illinois, Urbana-Champaign, (U.S.A.)
Douglas L Jones, University of Illinois, Urbana-Champaign, (U.S.A.)

Volume 4, page 1989, paper no. 1105

Abstract:
Quadratic time-frequency representations (TFRs) and time-scale representations (TSRs) have been shown to be very useful for detecting nonstationary signals in the presence of nonstationary noise. The theory developed thus far is only for the single observation case; however, in many situations involving signal detection, there are advantages in using an array of receiving sensors. Sensor arrays allow for target or source localization and can provide a large gain in the SNR. We show that time-frequency and time-scale representations provide a natural structure for the detection of a large class of nonstationary signals in the presence of nonstationary noise using an array of sensors. That is, time-frequency and time-scale provide a detection structure that is both optimal and allows for efficient implementation. In developing the TFR/TSR-based optimal quadratic array processor, we consider several types of array environments including those with full, partial, and no coherence.
ICASSP98 Paper Abstract
The Source Number Estimation Based on Gerschgorin Radii

Authors:
Olivier Caspary, CRAN-CNRS, (France)
Patrice Nus, CRAN-CNRS, (France)
Thierry Cecchin, CRAN-CNRS, (France)

Volume 4, page 1993, paper no. 1951

Abstract:
The GDE criterion is based on the estimation of the Gerschgorin disks’ radii where the disks are separated in two distinct sets: one associated to the signal sources, the other related to the noise. We aim at modifying that criterion into a new one called SGDE by using the sum of the disks’ radii. Besides, the SGDE criterion is modified with a simple deflation on the sum of the Gerschgorin radii to obtain a better estimation with sources of different power. We also suggest applying a deflation method to the covariance matrix before using the criteria based on the Gerschgorin radii. The transformed Gerschgorin radii can be connected to the Least-Squares through the transformed cross-correlation vector. So, two new criteria are put forward on the same principle as the SGDE criterion. These criteria can be applied in many situations: coloured or white noise, sources of different power.
ICASSP98 Paper Abstract

Array Processing in Non-Gaussian Noise with the EM Algorithm

Authors:
Richard J Kozick, Bucknell University, (U.S.A.)
Brian M. Sadler, Army Research Laboratory, (U.S.A.)
Rick S Blum, Lehigh University, (U.S.A.)


Abstract:
A central problem in sensor array processing is the localization of multiple sources and the reception of the signals emitted by those sources. Many approaches have been studied for this problem when the additive noise in the sensor array data is modeled with a Gaussian distribution. However, the schemes designed for Gaussian noise typically perform very poorly when the noise is non-Gaussian. An algorithm is presented in this paper for array processing in non-Gaussian noise. The algorithm is based on modeling the noise with a Gaussian mixture distribution. The expectation-maximization (EM) algorithm is then used to derive an iterative processing structure that estimates the source locations, estimates the source waveforms, and adapts the processing to match the characteristics of the noise. Simulation examples are presented to illustrate the performance of the algorithm.
ICASSP98 Paper Abstract

Multistage Cancellation of Terrain Scattered Jamming and Conventional Clutter

Authors:
Daniel J Rabideau, MIT Lincoln Lab, (U.S.A.)

Volume 4, page 2001, paper no. 2097

Abstract:
This paper addresses the problem of adaptively canceling both conventional clutter and terrain-scattered jamming (TSJ) in airborne radar systems. Existing algorithms for this type of interference adapt first in space/fast-time to cancel the TSJ, then in space/slow-time to cancel the conventional clutter. Unfortunately, the rapid weight updating required to cancel the nonstationary TSJ will modulate the clutter and targets, making the cancellation of conventional clutter extremely difficult and reducing the accuracy of the reported target locations. This paper proposes a multi-stage beamformer that prevents modulated clutter from degrading cancellation performance. The processor is formulated and its properties are described. The application of this beamformer to site-specific simulated data sets is used to illustrate its performance.
ICASSP98 Paper Abstract

Analysis of an Adaptive Detection Algorithm for Non-Homogeneous Environments

Authors:
Christ D Richmond, MIT Lincoln Lab, (U.S.A.)

Volume 4, page 2005, paper no. 2129

Abstract:
The adaptive matched filter (AMF) detector is known to be highly vulnerable to jammers and clutter discretes on which it has not trained. A vulnerability often leading to impractical false alarm rates in non-homogeneous environments. Sequentially following the AMF test with the adaptive cosine estimator (ACE) detector was proposed as a method of regulating the AMF’s high false alarm rate. The overall detection algorithm, called the adaptive sidelobe blanker (ASB), is two dimensional and has exhibited significant potential in experimental settings of inhomogeneous environments. The goal of this paper is to theoretically examine the potential of this algorithm for application in non-homogeneous environments.
ICASSP98 Paper Abstract

Error Reduction of Range Estimates in Multipath Environments

Abstract:
A reduction in error is obtained for estimates of range computed in a multipath environment. A good estimation technique for this problem typically involves exploiting the multipath interference, that is, analyzing each received signal reflection to improve the estimate. Under certain conditions, however, the errors in multipath estimates can be quite large. We compute the Cramér-Rao bounds for an example satisfying these conditions to demonstrate this phenomenon. We propose a modification to the original signal model which better represents the received signals. While the modified model does introduce a bias error, we provide a suitable estimate of the bias error so that a complete error analysis of the modified model is possible. The total error (noise error plus bias error) for the modified model is computed for an example and compares favorably with the error obtained via the original signal model.
ICASSP98 Paper Abstract

Design of Chemical Sensor Arrays for Monitoring Disposal Sites on the Ocean Floor

Authors:
Aleksandar Jeremic, University of Illinois, Chicago, (U.S.A.)
Arye Nehorai, University of Illinois, Chicago, (U.S.A.)

Volume 4, page 2013, paper no. 2256

Abstract:
We develop detection methods for automatic environmental monitoring of disposal sites on the deep ocean floor using chemical sensor arrays. Such sites have been proposed for the relocation of dredge materials from harbors and shipping channels; the monitoring is used to detect possible release of pollutants at the site. We model the underwater transport of the pollutants as a diffusion process, and obtain a measurement model by exploiting the spatial and temporal evolution of the associated concentration distribution. The detection problem is defined by a one side hypothesis test for the case of multiple sources. We derive two detectors, the generalized likelihood ratio (GLR) test and the mean detector, and determine their performance in terms of the probabilities of false alarm and detection. The results are applied to the design of chemical sensor arrays satisfying criteria specified in terms of these probabilities, and to optimally select numbers of sensors and time samples. Numerical examples are used to demonstrate the applicability of our results.
ICASSP98 Paper Abstract

Eigenstructure Beamspace Root Estimator Bank with Interpolated Array

Authors:
Alex B. Gershman, Ruhr University of Bochum, (Germany)
Johann F. Bohme, Ruhr University of Bochum, (Germany)

Volume 4, page 2017, paper no. 1057

Abstract:
A beamspace root modification of PseudoRandom Joint Estimation Strategy (PR-JES) [1] is developed. The essence of PR-JES is to generate the eigenstructure-based estimator bank for given sample covariance or data matrix. Combining the results of “parallel” underlying estimators, PR-JES removes the outliers and improves the threshold performance. In the case of non-uniform array, the interpolated array approach is used to enable the application of root underlying estimators. Simulations and results of real ultrasonic data processing show that the proposed beamspace root implementation significantly outperforms spectral elementspace PR-JES and achieves the performance similar or better than that of stochastic ML method.
ICASSP98 Paper Abstract

Minimum-Noise-Variance Beamformer with an Electromagnetic Vector Sensor

Authors:
Arye Nehorai, The University of Illinois, Chicago, (U.S.A.)
Kwok-Chiang Ho, Centre for Signal Processing, (Singapore)
B.T.G. Tan, National University of Singapore, (Singapore)

Volume 4, page 2021, paper no. 1202

Abstract:
We develop a minimum-noise-variance beamformer employing one electromagnetic vector sensor, capable of measuring the complete electric and magnetic fields induced by electromagnetic signals at one point. Two types of signals are considered: one carries a single message, and the other carries two independent messages simultaneously. The state of polarization of the interference under consideration ranges from completely polarized to unpolarized. To analyze the performance, we first obtain explicit expressions for the signal to interference-plus-noise ratio (SINR) in terms of the parameters of the desired signal, interference and noise. Then we discuss some physical implications associated with the SINR expressions. Our SINR expressions provide a basis for effective interference suppression, as well as generation of dual-message signals of which the two message signals have minimum interference effect on one another.
ICASSP98 Paper Abstract
Performance of OTH Radar Array Calibration

Authors:
Ishan S.D. Solomon, Defence Science and Technology Organisation, (Australia)
Yuri I Abramovich, CRC for Sensor Signal & Information Processing, (Australia)
Douglas A Gray, CRC for Sensor Signal & Information Processing, (Australia)
Stuart J. Anderson, Defence Science and Technology Organisation, (Australia)

Volume 4, page 2025, paper no. 1363

Abstract:
Array calibration algorithms for over-the-horizon (OTH) radar arrays have been recently proposed in the literature. These algorithms perform array calibration by using echoes from ionized meteor trails, and estimate sensor position errors and mutual coupling. In this paper we derive the Cramer-Rao performance bound for this array calibration problem and then investigate the performance bound. We obtain insight on the achievable accuracy as a function of the signal-to-noise ratio, number of snapshots and number of sources. Further, we consider the advantage of using sources with known bearings, as opposed to unknown bearings, and consider the identifiability of the array calibration problem.
ICASSP98 Paper Abstract

Broadband Frequency Invariant Beamforming Method with Low Computational Cost

Authors:
Tetsuki Taniguchi

Volume 4, page 2029, paper no. 1426

Abstract:
This paper presents a method for reducing the computational complexity required for farfield broadband beamforming. First, the propagating wave received by a linear or planar array of sensors is sampled efficiently on the multidimensional frequency plane in spatio-temporal sense so that the so-called non-physical area where no spectrum exists is filled with aliasing component. Next, the derived time-space series is upsampled and processed by a multidimensional filter to derive the target beampattern. Through some considerations and examples, it is shown that this method has some restrictions on the beamwidth and/or maximum beam center angle, but it can reduce the operations to about half the number of conventional one.
ICASSP98 Paper Abstract

An Optimum Space-Time MTI Processor for Airborne Radar

Authors:
Ha Jong Sung, Yonsei University, (Korea)
Young Cheol Park, Samsung Biomedical Research Institute, (Korea)
Dae Hee Youn, Yonsei University, (Korea)
V. John Mathews, University of Utah, (U.S.A.)

Volume 4, page 2033, paper no. 1518

Abstract:
This paper presents an optimum space-time moving target indication (MTI) processor for the airborne radar. The optimization is based on a stochastic target model, rather than deterministic target models adopted in most space-time MTI processor designs. The optimum solution that maximizes the improvement factor yielded by the processor is shown to be the generalized eigenvector corresponding to the smallest generalized eigenvalue of the signal and clutter covariance matrices. A suboptimal, but computationally simpler solution to this problem is also derived. This approach requires the solution of a linearly constrained minimum variance (LCMV) problem. Unlike typical LCMV problems, our solution also calculates the response vector specifying the frequency response along the look direction. Experimental results demonstrating the usefulness of our methods are included in the paper. The results indicate that the suboptimal solution does not suffer from significant performance loss.
ICASSP98 Paper Abstract

An Adaptive Monopulse Processor for Angle Estimation in a Mainbeam Jamming and Coherent Interference Scenario

Authors:
Yaron Seliktar, Georgia Institute of Technology, (U.S.A.)
E. Jeff Holder, Georgia Institute of Technology, (U.S.A.)
Douglas B Williams, Georgia Institute of Technology, (U.S.A.)

Abstract:
Mainbeam jamming poses a particularly difficult challenge for conventional monopulse radars. In such cases spatially adaptive processing provides some interference suppression when the target and jammer are not exactly co-aligned, but the resulting array pattern is too distorted to be suitable for monopulse processing. The presence of coherent multipath in the form of terrain scattered interference (TSI) is normally considered a nuisance source of interference. However, it can also be exploited to suppress mainbeam jamming with space-time processing. Here we present a method for incorporating space-time processing into monopulse processing to yield a space-time monopulse processor with distortionless spatial array patterns that can achieve far better mainbeam jamming cancelation and target angle estimation than has been previously possible. Performance results for the monopulse processor are obtained for Mountaintop data containing a jammer and TSI, that demonstrate a dramatic improvement in performance over conventional monopulse and spatially adaptive monopulse.
ICASSP98 Paper Abstract

Broadband Beamforming Using Elementary Shape Invariant Beampatterns

Authors:
Thushara D Abhayapala, Australian National University, (Australia)
Rodney A Kennedy, Australian National University, (Australia)
Robert C Williamson, Australian National University, (Australia)

Volume 4, page 2041, paper no. 1638

Abstract:
This paper presents a new method of designing a beamformer having a desired beampattern with focusing capability to operate at any radial distance from the array origin. An important consequence of our result is that the beamformer can be factored in to three levels of filtering: (i) beampattern independent elementary beamformers; (ii) beampattern shape dependent filters; and (iii) radial zooming filters where a single parameter can be adjusted to zoom-focus the array to a desired radial distance from the array origin. As an illustration the method is applied to the problem of producing a practical array design that achieves a frequency invariant beampattern over the frequency range of 1:10.
ICASSP98 Paper Abstract
Near Field Superdirectivity (NFSD)

Authors:
Wolfgang Tager, CNET Telecom, (France)

Volume 4, page 2045, paper no. 1990

Abstract:
In some array applications, the source of interest is close to the array, so that we have to use a near field model. Almost always the near field is considered as an additional difficulty. We contradict this point of view and show that if the desired source is in the near field and the other sources are in the far field, then even a small array can be at the same time highly directive and comparatively robust. Instead of relying on small phase differences for low frequencies, we fully exploit the fact that the amplitude vector of the source of interest is different from that of any other source. The array geometry should be chosen to enhance this effect. Unlike far field superdirectivity, we can steer the main lobe to arbitrary directions without prohibitive loss of performance. We applied our method to microphone array sound pick up for workstations. Simulation results and measurements of a real time implementation on a fixed point DSP are provided.
ICASSP98 Paper Abstract

Harmonic Phase Coupling for Battlefield Acoustic Target Identification

Authors:
Douglas E Lake, Army Research Laboratory, (U.S.A.)

Volume 4, page 2049, paper no. 2095

Abstract:
Target identification using battlefield acoustic sensor arrays is an important problem for the Army. Acoustic signatures of targets of interest, such as tanks and trucks, exhibit time-varying patterns of harmonic amplitudes that facilitate target ID. In this paper, harmonic phase coupling is shown to also be present in these acoustic signatures though issues remain to fully exploit these relationships for target ID. With all else being equal, frequencies with simple harmonic relationships, such as 2 to 1 or 3 to 2, are preferred to produce stable features. Also, naive use of FFT approaches to estimate phase leads to erratic estimates. Cramer-Rao bounds (CRB) are presented and provide insights into the limitations of the accuracy of phase coupling estimates and suggests the value of developing nonstationary methods.
ICASSP98 Paper Abstract

DOA Estimation with Hexagonal Arrays

Authors:
Zhi Tian, George Mason University, (U.S.A.)
Harry L. Van Trees, George Mason University, (U.S.A.)

Volume 4, page 2053, paper no. 2251

Abstract:
Hexagonal arrays are widely used in practice but have received less attention in the optimum array processing literature. In this paper, we show how unitary ESPRIT can be applied to hexagonal arrays for direction-of-arrival (DOA) estimation. The resulting estimates exhibit good threshold behavior, and are close to the Cramer-Rao bound above threshold. We also show how to use spatial smoothing in hexagonal arrays for DOA estimation in the presence of coherent signals.
ICASSP98 Paper Abstract

A Polynomial Rooting Approach to the Localization of Coherently Scattered Sources

Authors:
Jason Goldberg, Tel Aviv University, (Israel)
Hagit Messer, Tel Aviv University, (Israel)

Volume 4, page 2057, paper no. 1028

Abstract:
The problem of passive localization of coherently scattered sources with an array of sensors is considered. The spatial extent of such a source is typically characterized by an angular mean and an angular spreading parameter. The maximum likelihood estimator for this problem requires a complicated search of dimension equal to twice the number of sources. However, a previously reported sub-optimal MUSIC type method reduces the search dimension to two. In this paper, the search over the angular mean parameter in the above MUSIC type technique is replaced by a potentially more efficient polynomial rooting procedure. Computer simulations verify that for sufficiently high polynomial order and sufficiently high SNR, the proposed method achieves the Cramer-Rao Bound.
ICASSP98 Paper Abstract

Generalised Spatial Smoothing Using Partial Arrays for Multiple Target Bearing Estimation

Authors:
Yuri I Abramovich, CSSIP, (Australia)
Nicholas K Spencer, CSSIP, (Australia)

Volume 4, page 2061, paper no. 1366

Abstract:
This paper considers the problem of bearing estimation for a small number of radar targets which cannot be resolved in range or Doppler frequency. Bearing estimation for non-fluctuating targets involves a single "snapshot" resulting from a multi-channel optimum (matched) filtering process. The standard spatial smoothing technique may be applied to this single-snapshot model, but only for uniform linear antenna arrays. Here we introduce a special class of nonuniform geometry with embedded "partial arrays" and a corresponding "generalised spatial smoothing" (GSS) algorithm. The partial array characteristics determine the resulting bearing estimation accuracy. A two-stage bearing estimation procedure is proposed. The initialization stage involves spatial averaging over all suitable partial arrays. The refinement stage uses a local maximum-likelihood search. Typical radar model simulations and Cramer-Rao bound calculations demonstrate the efficiency of this approach compared with standard spatial smoothing using a uniform linear array.
ICASSP98 Paper Abstract


Authors:
Gonzalo Seco Granados, Universitat Politecnica de Catalunya, (Spain)
Juan A. Fernandez Rubio, Universitat Politecnica de Catalunya, (Spain)

Volume 4, page 2065, paper no. 1596

Abstract:
The problem of estimating the propagation-delay of a desired signal in the presence of interferences and multipath propagation is addressed. This paper presents the maximum likelihood (ML) propagation-delay estimator for a signal arriving at a sensor array. The novel characteristic herein is that the desired signal impinges on the array with a known steering vector. This fact allows to assume an unknown and arbitrary spatially colored noise. The Cramer-Rao bound (CRB) for the problem at hand is derived and numerically compared with the variance of the MLE. The MLE is applied to the Global Navigation Satellite Systems, in order to reduce the serious performance deterioration that the interferences and the multipath propagation produce. We show that in the presence of coherent reflections of the desired signal the presented estimator is no longer the MLE and becomes biased. However, its bias is much lower than that of other conventional estimators.
ICASSP98 Paper Abstract

Modified IQML and a Statistically Efficient Method for Direction Estimation without Eigendecomposition

Authors:
Martin Kristensson, Royal Institute of Technology, (Sweden)
Magnus Jansson, Royal Institute of Technology, (Sweden)
Björn Ottersten, Royal Institute of Technology, (Sweden)

Volume 4, page 2069, paper no. 1599

Abstract:
This paper deals with direction estimation of signals impinging on a uniform linear sensor array. A well known algorithm for this problem is IQML. Unfortunately, the IQML estimates are in general biased, especially in noisy scenarios. We propose a modification of IQML (MIQML) that gives consistent estimates at approximately the same computational cost. In addition, an algorithm with an estimation error covariance which is asymptotically identical to the asymptotic Cramer-Rao lower bound is presented. The optimal algorithm resembles weighted subspace fitting or MODE, but achieves optimal performance without having to compute an eigendecomposition of the sample covariance matrix.
ICASSP98 Paper Abstract

Linear Constrained Reduced Rank and Polynomial Order Methods

Authors:
Hong Guan, The University of Texas, Dallas, (U.S.A.)
Ronald D De Groat, The University of Texas, Dallas, (U.S.A.)
Eric M Dowling, The University of Texas, Dallas, (U.S.A.)
Darel A Linebarger, The University of Texas, Dallas, (U.S.A.)

Abstract:
The Subspace-based Reduced Rank and Polynomial Order (RRPO) methods were proposed recently, which estimate a reduced order linear prediction polynomial whose roots are the desired "signal roots". In this paper, we describe how to extend the RRPO methods to include constraints involving known signal information. The usefulness of the proposed constrained RRPO methods is demonstrated by an application to DOA findings over a wide range of scenarios. Simulation results indicate that by incorporating known signal information such as source direction angle, the estimation of unknown source directions can be significantly improved, especially when the unknown source is weak, closely spaced and highly coherent with the known source.
ICASSP98 Paper Abstract

Identification of Closed-Loop Linear Systems via Cyclic Spectral Analysis: An Equation-Error Formulation

Authors:
Channarong Tontiruttananon, Auburn University, (U.S.A.)
Jitendra K. Tugnait, Auburn University, (U.S.A.)

Volume 4, page 2077, paper no. 1108

Abstract:
The problem of closed-loop system identification given noisy input-output measurements is considered. The closed-loop system operates under an external cyclostationary input which is not measured. Noisy measurements of the (direct) input and output of the plant are assumed to be available. The various disturbances affecting the system are either stationary or cyclostationary with cycle frequencies different from the input cycle frequencies. The closed-loop system must be stable but it is allowed to be unstable in open-loop. A frequency-domain parametric solution is proposed and analyzed using an equation error formulation, and cyclic spectrum and cross-spectrum of the input-output measurements. The parameter estimator is shown to be consistent. A simulation example using an unstable open-loop system is presented to illustrate the proposed approach.
ICASSP98 Paper Abstract

Channel Identification with Doppler and Time Shifts Using Mixed Training Signals

Authors:
Xiang-Gen Xia, University of Delaware, (U.S.A.)

Volume 4, page 2081, paper no. 1394

Abstract:
In this paper, we present a method to identify channels with both Doppler and time shifts using mixed training signals. The training signals we use consist of two parts, where one part is a constant and the other part is a conventional training signal, such as a pseudo-random signal. These two parts may be separated in either the time or the frequency domain. We provide a necessary and sufficient condition on the channel identifiability in terms of the time and Doppler shifts when the mixed training signals are used. It can be shown that the condition holds almost surely in most cases of interests in practice. Some numerical examples are also presented.
Feasibility of Source Separation in Frequency Domain

Authors:
Christine Serviere, CEPHAG-ENSIEG, (France)

Volume 4, page 2085, paper no. 1567

Abstract:
We focus on the feasibility of the source separation in the frequency domain. First, it is linked with
the convergence speed towards gaussianity of signals after L-point discrete Fourier Transform. We
test here a distance to gaussianity thanks to the spectral kurtosis. We analyse the influence of L, of the
duration of the source tricorrelations and of a non linear filtering. We mainly develop the case of QARMA
processes. The second point consists in the reconstruction of the spectra of the estimated sources from
the signals identified at each frequency bin. Indeed, the source associated to the ith identified signal is
not necessarily the same from one frequency bin to another. The algorithm efficiency is then illustrated
on QARMA processes, including the procedures of separation and reconstruction.
ICASSP98 Paper Abstract
Source Separation in Post Nonlinear Mixtures: An Entropy-Based Algorithm

Authors:
Anisse Taleb, INPG-LIS, (France)
Christian Jutten, INPG-LIS, (France)
Serge Olympieff, INPG-LIS, (France)

Volume 4, page 2089, paper no. 1770

Abstract:
This paper proposes a new approach for sources separation in special nonlinear mixtures, called post nonlinear mixtures (PNL). We first explain the nice separability properties of these mixtures: solutions have almost the same indeterminacies than separation of sources in instantaneous linear mixtures. The method proposed in this paper is based on the minimization of the mutual information, which needs the knowledge of source distributions or more exactly of log-derivative of source distributions (the so-called score functions). The algorithm consists of three adaptive blocks: one nonlinear block, devoted to adaptive estimation of source score functions, drives the adaptation of the two other blocks corresponding to estimation of the linear and nonlinear parts of the mixtures. The paper finishes with a few experimental results which prove the efficacy of the algorithm.
ICASSP98 Paper Abstract

A Gauss-Newton Method for Blind Source Separation of Convolutive Mixtures

Authors:
Sergio A. Cruces-Alvarez, University of Seville, (Spain)
Luis Castedo-Ribas, University of A Coruna, (Spain)

Volume 4, page 2093, paper no. 1914

Abstract:
In this paper we present several Gauss-Newton algorithms for blind source separation of convolutive mixtures. The algorithms can be interpreted as generalizations of two previous approaches due to Gerven-Compernolle and Nguyen-Jutten. Since they are of the Gauss-Newton type, they exhibit a fast rate of convergence. Also, we present a stability analysis for two sources and instantaneous mixtures where we show that the algorithms cannot converge to non-separating solutions.
ICASSP98 Paper Abstract

Adaptive Blind Separation of Convolutive Mixtures of Independent Linear Signals

Authors:
Jitendra K. Tugnait, Auburn University, (U.S.A.)

Abstract:
This paper is concerned with the problem of blind separation of independent signals (sources) from their linear convolutive mixtures. The various signals are assumed to be linear non-Gaussian but not necessarily i.i.d. Recently an iterative, normalized higher-order cumulant maximization based approach was developed using the fourth-order normalized cumulants of the “beamformed” data. A byproduct of this approach is a decomposition of the given data at each sensor into its independent signal components. In this paper an adaptive implementation of the above approach is developed using a stochastic gradient approach. Some further enhancements including a Wiener filter implementation for signal separation and adaptive filter reinitialization are also provided. A computer simulation example is presented.
ICASSP98 Paper Abstract

A Least-Squares Interpretation of the Single-Stage Maximization Criterion for Multichannel Blind Deconvolution

Authors:
Shuichi Ohno, Shimane University, (Japan)
Yujiro Inouye, Shimane University, (Japan)

Volume 4, page 2101, paper no. 1348

Abstract:
In order to attain multichannel blind deconvolution of linear time-invariant nonminimum-phase dynamic systems, Inouye and Habe proposed in 1995 a single-stage maximization criterion. The criterion function is the sum of squared forth-order cumulants of the equalizer outputs, and the coefficients of the equalizer are determined at once. On the other hand, one of possible approaches for multichannel blind deconvolution is to construct an equalizer based on the system identified by higher-order cumulant-matching. In this paper, it is shown that the single-stage maximization criterion is equivalent to a least-squares fourth-order cumulant-matching criterion after multichannel pre-whitening of channel outputs. This result provides us with an important interpretation of the single-stage maximization criterion.
ICASSP98 Paper Abstract
Adaptive Minimum Variance Methods for Direct Blind Multichannel Equalization

Authors:
Zhengyuan Xu, Stevens Institute of Technology, (U.S.A.)
Michail K. Tsatsanis, Stevens Institute of Technology, (U.S.A.)

Abstract:
Constrained adaptive optimization techniques are employed in this paper to design direct blind equalizers. The method is based on minimizing the equalizer's output variance subject to appropriate constraints. The constraints are chosen to guarantee no desired signal cancellation and are also jointly and recursively optimized to improve performance. Our method provides adaptive solutions which directly optimize the equalizer's parameters, while its performance compares favorably to that of the linear prediction based approaches. Global convergence is established and comparisons with other blind and trained methods are presented.
ICASSP98 Paper Abstract

On the Convolutive Mixture Source Separation by the Decorrelation Approach

Authors:
Carine Simon, UMLV, (France)
Guy D'Urso, EDF/DER, (France)
Christophe Vignat, UMLV, (France)
Philippe Loubaton, UMLV, (France)
Christian Jutten, UMLV, (France)

Volume 4, page 2109, paper no. 1794

Abstract:
In this paper, we consider the problem of blind separation of causal minimum phase convolutive mixtures of two sources. We study in detail the so-called decorrelation approach. It consists in finding a causal minimum phase filter which, driven by the observations, produces decorrelated outputs. It is well established that this approach allows to separate the sources if the mixing filter is a non static FIR filter. We show that this result is no longer true in the IIR case. We establish that it exists infinitely many causal minimum phase filters producing decorrelated outputs and provide a parameterisation of these filters. This clearly shows that the decorrelation approach is, in practice, non robust. In order to overcome this drawback, we propose an alternative approach based on a linear prediction scheme, which, as the decorrelation approach, uses essentially the second order statistics of the observations.
ICASSP98 Paper Abstract

Bilinear Methods for Blind Channel Equalization: (No) Local Minimum Issue

Authors:
Eric Pite, Ecole Nationale Superieure des Telecommunications, (France)
Pierre Duhamel, Ecole Nationale Superieure des Telecommunications, (France)

Volume 4, page 2113, paper no. 2018

Abstract:
Bilinear methods for jointly estimating the channel coefficients and the symbols emitted through these channels are very appealing. However, they can be trapped by local minima. This paper provides (i) a full characterization of the local minima, (ii) a simple criterion for checking whether the procedure has converged to the global minimum, (iii) a simple algorithm for obtaining this solution, with a proof of convergence.
ICASSP98 Paper Abstract

Blind Identification of Single-Input Multiple-Output Pole-Zero Systems

Authors:
Gopal T Venkatesan, University of Minnesota, (U.S.A.)
Mostafa Kaveh, University of Minnesota, (U.S.A.)
Ahmed H. Tewfik, University of Minnesota, (U.S.A.)
Kevin M. Buckley, Villanova University, (U.S.A.)

Volume 4, page 2117, paper no. 2108

Abstract:
In this paper we present a technique for the blind identification of single-input multiple-output (SIMO) pole-zero (PZ) systems using only the second order statistics of the system output data. The system input is treated as an unknown deterministic sequence, and hence, restrictive i.i.d. assumptions on the input sequence are not required. We estimate the poles and zeros of the channels in two steps: 1) estimate product of all permutations of a numerator and a denominator polynomial from two different channels, and 2) extract individual numerator and denominator polynomials for each channel from above estimate. Our technique performs well even with short records of data.
ICASSP98 Paper Abstract
Blind Channel Estimation by Least Squares Smoothing

Authors:
Lang Tong, University of Connecticut, (U.S.A.)
Qing Zhao, University of Connecticut, (U.S.A.)

Abstract:
A linear least squares smoothing approach is proposed for the blind channel estimation. It is shown that the single-input multiple-output moving average process has the property that the error sequence of the least squares smoother, under certain conditions, uniquely determines the channel impulse response. The relationship among the dimension of the observation space, channel order and smoothing delay is presented. A new algorithm for channel estimation based on the least squares smoothing is developed. The proposed approach has the finite-sample convergence property in the absence of the channel noise. It also has a structure suitable for recursive implementations.
ICASSP98 Paper Abstract
Blind Signal Separation with a Projection Pursuit Index

Authors:
Amir Sarajedini, University of California, San Diego, (U.S.A.)
Paul M. Chau, University of California, San Diego, (U.S.A.)

Volume 4, page 2125, paper no. 2304

Abstract:
Blind Signal Separation (BSS) is a powerful technique for separation of mixed signals with weak assumptions on the incoming signals. The objectives of BSS are analogous to the objectives of Exploratory Projection Pursuit which is widely used in the statistical community for finding structure in high dimensional data sets. In this paper, we adapt Exploratory Projection pursuit for BSS. First, we introduce Exploratory Projection Pursuit and the associated projection pursuit index (PPI). We adapt the PPI for application to BSS. We also investigate the order of approximation required to achieve satisfactory separation using the PPI, and compare its performance to a maximum-likelihood BSS technique using a Gram-Charlier Expansion.
ICASSP98 Paper Abstract

Optimum Subarray Configurations Using Genetic Algorithms

Authors:
Jin Wang, Center for Monitoring Research, SAIC, (U.S.A.)
Hans Israelsson, Center for Monitoring Research, SAIC, (U.S.A.)
Robert G. North, Center for Monitoring Research, SAIC, (U.S.A.)

Abstract:
Subarray configuration is not a trivial problem in array signal processing. A proper subarray configuration is important to improve the detectability of an array. A new searching algorithm, which is based on Genetic Algorithms (GA), for the optimum subarray configuration is proposed in this paper. Our preliminary application to a seismic array has indicated that the new algorithm can search a population of subarrays in a more efficient and robust way. The beamforming gain of the optimum subarray derived by GA is very close to the theoretical gain. Experimental results on signal detections have demonstrated that a beamforming recipe with optimum subarrays can provide further enhanced signal-to-noise ratio (SNR), compared to a recipe without subarray configuration. The approach proposed here can be easily extended to the weight determination problem for the weighted beamforming process by using multi-bit instead of 1-bit representation for each sensor in the chromosome model.
ICASSP98 Paper Abstract

Bayesian Estimation of Abrupt Changes Contaminated by Multiplicative Noise Using MCMC

Authors:
Jean-Yves Tourneret, ENSEEIHT, (France)
Michel Doisy, ENSEEIHT, (France)
Manuel Mazzei, ENSEEIHT, (France)

Volume 4, page 2133, paper no. 1100

Abstract:
The paper addresses the estimation of abrupt changes which are contaminated by multiplicative Gaussian noise. The marginal mean a posteriori or marginal maximum a posteriori estimators can be derived for estimating the position of a single abrupt change. However, these estimators have optimization or integration problems for multiple abrupt changes. The paper solves these optimization problems by using Markov Chain Monte Carlo methods.
ICASSP98 Paper Abstract

Locally Optimum Detectors for Deterministic Signals in Multiplicative Noise

Authors:
Mounir Ghogho, Strathclyde University, Scotland, (U.K.)
Asoke K Nandi, Strathclyde University, Scotland, (U.K.)
Bernard Garel, INP-ENSEEIHT, (France)

Volume 4, page 2137, paper no. 1216

Abstract:
This paper addresses the problem of detecting deterministic signals in a multiplicative noise model. The multiplicative noise model is appropriate for modelling coherent imaging systems such as SAR and LASER. Locally Optimum (LO) detectors are derived for any arbitrary multiplicative noise distribution. The gamma and generalized Gaussian distributions are studied in detail. We also introduce an extension of the generalized Gaussian density to include asymmetry. The performance of the LO detectors is studied and compared with that of the linear correlation detector. The paper gives insight into the influence of the tail-length of the noise distribution on the detection performance.
ICASSP98 Paper Abstract

Evaluation of CFAR and Texture Based Target Detection Statistics on SAR Imagery

Authors:
Lance M Kaplan, Clark Atlanta University, (U.S.A.)
Romain Murenzi, Clark Atlanta University, (U.S.A.)

Volume 4, page 2141, paper no. 1266

Abstract:
In this work, we evaluated the effectiveness of synthetic aperture radar (SAR) target detection algorithms that consist of any number of combinations of three statistics which include two-parameter CFAR, variance, and extended fractal features. The performance of these algorithms were tested at various threshold settings over the public domain MSTAR database. This database contains one foot resolution X-band SAR imagery. Receiver-operating-characteristic (ROC) curves were generated for the seven resulting algorithms. The results indicate that the CFAR statistic is the least effective detection statistic.
ICASSP98 Paper Abstract

On the Use of a General Amplitude PDF in Coherent Detectors of Signals in Spherically Invariant Interference

Authors:
D. Robert Iskander, *Queensland University of Technology, (Australia)*

Abstract:
The aspects of using a general amplitude probability density function in coherent detectors are investigated. For this purpose, the recently developed Generalised Bessel function K (GBK) distribution is used. The performance of the optimal detector of signals embedded in GBK-distributed interference is compared to the one of the uniformly most powerful invariant detector using extensive Monte Carlo simulations. The results indicate that for small number of integrated pulses the optimal detector outperforms the uniformly most powerful invariant detector by up to 18 dB. It is shown, that this improvement does not vary significantly with changes in the parameters that control the spherical invariance of the GBK distribution.
ICASSP98 Paper Abstract
Statistical Classification of Chaotic Signals

Authors:
Christophe Couvreur, Faculte Polytechnique de Mons, (Belgium)
Cedric Flamme, Faculte Polytechnique de Mons, (Belgium)
Marc Pirlot, Faculte Polytechnique de Mons, (Belgium)

Volume 4, page 2149, paper no. 1786

Abstract:
The classification of chaotic signals generated by a low-dimensional deterministic models given a dictionary of possible model is considered. The proposed classification methods rely on the concept of "best predictor" of signal. A statistical interpretation of this concept based on the ergodic theory of chaotic system is presented. A sort of "bootstrapping" estimator of the statistical properties is introduced. The method is validated by numerical simulations. Directions for future research are suggested.
Detection of Spectrally Equivalent Parametric Processes Using Higher Order Statistics

Authors:
Martial Coulon, ENSEEIHT/GAPSE, (France)
Jean-Yves Tourneret, ENSEEIHT/GAPSE, (France)
Ananthram Swami, Army Research Laboratory, (U.S.A.)

Abstract:
The paper addresses the problem of detecting two spectrally equivalent parametric processes (SEP’s) : the noisy AR process and the ARMA process. Higher order statistics (HOS) are shown to be effective for detection. Two HOS based detectors are derived and compared. The first detector studies the singularity of a HOS-based Yule-Walker matrix. The second detector filters the data by an AR filter estimated from the data; the residual HOS are then shown to be effective for the SEPP detection problem.
ICASSP98 Paper Abstract

Multiple Hypothesis Modulation Classification Based on Cyclic Cumulants of Different Orders

Authors:
Pierre Marchand, CEPHAG - ENSIEG, Domaine Universitaire, (France)
Christophe Le Martret, CELAR, (France)
Jean-Louis Lacoume, CEPHAG - ENSIEG, (France)

Volume 4, page 2157, paper no. 2290

Abstract:
A multiple hypothesis modulation QAM classification task is addressed in this paper. The classifier is designed within the rigorous 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. No assumption about the power of the signal is made. It is shown that this uncertainty about the power of the signal does not affect the decision rule. As an application, we present simulated performance in the context of 4-QAM vs 16-QAM vs 64-QAM classification.
Optimal Sensor Scheduling for Hidden Markov Models

Authors:
Jamie S Evans, University of Melbourne, (Australia)
Vikram K. Krishnamurthy, University of Melbourne, (Australia)

Volume 4, page 2161, paper no. 2345

Abstract:
Consider the Hidden Markov model where the realization of a single Markov chain is observed by a number of noisy sensors. The sensor scheduling problem for the resulting Hidden Markov model is as follows: Design an optimal algorithm for selecting at each time instant, one measurement provided by one out of the many sensors. Each measurement has an associated measurement cost. The problem is to select an optimal measurement scheduling policy, so as to minimize a cost function of estimation errors and measurement costs. The problem of determining the optimal measurement policy is solved via stochastic dynamic programming. Numerical results are presented.
ICASSP98 Paper Abstract

A General Maximum Likelihood Framework for Modulation Classification

Authors:
Denys Boiteau, Centre d'Etudes de Systemes et Techniques Avancees, (France)
Christophe Le Martret, Centre d'Electronique de l'Armement, (France)

Volume 4, page 2165, paper no. 5091

Abstract:
This paper deals with modulation classification. First, a state-of-the-art is given which is separated in two classes: the pattern recognition approach and the Maximum Likelihood (ML) approach. The we propose a new classifier called the General Maximum Likelihood Classifier (GMLC) based on an approximation of the likelihood function. We derive equations of this classifier in the case of linear modulation classification and apply them to the M PSK / M' PSK problem. We show that the new tests are a generalization of previous ones using the 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 systems that exploit cyclostationary property of modulated signals.
ICASSP98 Paper Abstract

Instantaneous Parameter Estimation Based on Continuous Wavelet Transform and Some Improvements

Authors:
Huafeng Zhang, Northwestern Polytechnical University, (China)
Jianping Zhao, Northwestern Polytechnical University, (China)
Jian Guo Huang, Northwestern Polytechnical University, (China)

Volume 4, page 2169, paper no. 1143

Abstract:
In this paper, an novel method based on the phase information of continuous wavelet transform to estimate the instantaneous parameter of AM-FM signal is introduced, and some strategies, including the determination of initial value in iteration and post-processing to the estimated results, to improve the performance of the algorithm are proposed. Compared to several other instantaneous parameter estimators, such as CDF, Teager-Kaiser energy operator, and some TFR-based estimators, the proposed method has the advantages of noise resistance and accuracy by exploiting the time-scale localization of the wavelet transform. Simulation results testify that the proposed strategies improve the performance of the CWT based iteration algorithm greatly, and the method has excellent performance including robustness and accuracy in noisy condition.
ICASSP98 Paper Abstract

Stabilization of Stationary and Time-Varying Autoregressive Models

Authors:
Marko T Juntunen, University of Kuopio, (Finland)
Jouko Tervo, University of Kuopio, (Finland)
Jari P Kaipio, University of Kuopio, (Finland)

Volume 4, page 2173, paper no. 1380

Abstract:
A method for the stabilization of stationary and time-varying autoregressive model is presented. The method is based on the hyperstability constrained LS-problem with nonlinear constraints. The problems are solved iteratively with Gauss-Newton type algorithm that sequentially linearizes the constraints. The proposed method is applied to simulated data in stationary case and to real EEG data in the time-varying case.
ICASSP98 Paper Abstract
An Algorithm for Tracking a Random Walk with Unknown Drift

Authors:
Jean Luc Le Calvez, IRISA/Universite de Rennes, (France)
Bernard B. Delyon, IRISA/INRIA, (France)
Anatoli A. Juditsky, INRIA Rhone-Alpes, (France)

Volume 4, page 2177, paper no. 1496

Abstract:
In this paper we study the problem of tracking a random walk observed with noise when the variance of the walk increment is unknown. We describe a sequence of estimators of the random walk and we design an algorithm to choose the best estimator among all the sequence. We give also a bound for the mean square error of this estimator. Finally, some simulations are presented and we compare our algorithm with the Kalman filter when the variance of the walk increment is estimated.
ICASSP98 Paper Abstract
Identification of Arbitrarily Time-Variant Systems

Authors:
Yael Steinsaltz, EMC Corporation, (U.S.A.)
Hanoch Lev-Ari, Northeastern University, (U.S.A.)

Abstract:
We introduce a technique for identification of systems with arbitrarily time-variant responses from samples of their input and output signals, and without using any prior information about the dynamics of the unknown system response. Our technique is based on the use of optimized averaging filters for the estimation of time-variant second order moments. We demonstrate the utility of our approach and the quality of the resulting estimates via a numerical example.
ICASSP98 Paper Abstract
Adaptive RLS Filtering under the Cyclostationary Regime

Authors:
Hanoch Lev-Ari, Northeastern University, (U.S.A.)

Volume 4, page 2185, paper no. 1586

Abstract:
We present a methodology for adaptive filtering and system identification under the cyclostationary regime. Our technique is based on a deterministic periodic least-squares criterion, and gives rise to adaptive periodic recursive-least-squares (P-RLS) algorithms. Furthermore, we show that every adaptive RLS algorithm has a P-RLS counterpart, which has exactly the same performance attributes, and differs only in the length of the delay used in its time-update recursion.
ICASSP98 Paper Abstract

Extending the Transfer Function Calculus of Time-Varying Linear Systems: A Generalized Underspread Theory

Authors:
Gerald Matz, Vienna University of Technology, (Austria)
Franz Hlawatsch, Vienna University of Technology, (Austria)

Volume 4, page 2189, paper no. 1717

Abstract:
We extend the approximate transfer function calculus of “underspread” linear time-varying (LTV) systems introduced by W. Kozek. Our extension is based on a new, generalized definition of underspread LTV systems that does not assume finite support of the systems’ spreading function. We establish explicit bounds on various error quantities associated with the transfer function approximation. Our results yield a simple and convenient transfer function calculus for a significantly wider and practically more relevant class of LTV systems than that previously considered.
ICASSP98 Paper Abstract
Dynamic Estimation with Selectable Linear Measurements

Authors:
Dana Sinno, Arizona State University, (U.S.A.)
Douglas Cochran, Arizona State University, (U.S.A.)

Volume 4, page 2193, paper no. 1910

Abstract:
This paper deals with a class of dynamic estimation problems in which the estimator may dynamically select, from among a temporally evolving set of possibilities, the source of the data on which the estimate will be based. After motivating and formulating this class of "attentive estimation" problems, the paper focuses on the case in which the state of a linear discrete-time dynamical system driven by gaussian noise is to be estimated using linear measurements corrupted by additive gaussian noise. This differs from the standard Kalman filtering problem in that the measurement map at each stage is selectable from a pre-determined set of such maps. When the system dynamics and noise statistics are known, a criterion for measurement selection yielding an optimal sequence of output functions can be defined prior to the onset of estimation. When the noise statistics or other parameters are unknown, closed-loop adaptive strategies for measurement selection are necessary.
ICASSP98 Paper Abstract
Interpolation of Nonstationary Fields with Stationary Increments

Authors:
Beatrice Pesquet-Popescu, LESIR, ENS Cachan, (France)
Pascal Larzabal, LESIR, ENS Cachan, (France)

Volume 4, page 2197, paper no. 1960

Abstract:
The problem of the linear interpolation of nonstationary multidimensional processes with stationary increments is studied. The expressions of the interpolation filters and of the estimation error are derived, which generalize the results of the interpolation theory for stationary processes. Both finite and infinite extent interpolation are considered. An application to the interpolation of an underwater depth map is presented.
ICASSP98 Paper Abstract

Multiweight Optimization in OBE Algorithms for Improved Tracking and Adaptive Identification

Authors:
Dale Joachim, Michigan State University, (U.S.A.)
John R Deller Jr, Michigan State University, (U.S.A.)
Majid Nayeri, Michigan State University, (U.S.A.)

Volume 4, page 2201, paper no. 2404

Abstract:
Optimal Bounding Ellipsoid (OBE) algorithms offer an attractive alternative to traditional least squares methods for identifying linear-in-parameters signal and system models due to their low computational efficiency, superior tracking ability, and selective updating that permits processor sharing among tasks. These benefits are further enhanced by multiweight optimization (MWO) which yields improved per-point parameter convergence. This paper introduces the MWO process and describes advances in its implementation including the incorporation of a forgetting factor for improved tracking, a new method for efficient weight computation, and extensions to volume-minimizing OBE algorithms. Simulation studies illustrate the results.
ICASSP98 Paper Abstract
An H-Infinity Approach to Multi-Source Tracking

Authors:
Tharmalingam Ratnarajah, McMaster University, (Canada)
Volume 4, page 2205, paper no. 2482

Abstract:
In this paper, a novel $H^\infty$ approach is proposed for tracking of polarized co-channel sources using an array of tripole antennas. The proposed approach partitions the observation data matrix into two sub-matrices that are used, in conjunction with a new state-space model, to provide an $H^\infty$-type recursive estimation of a linear combiner. The linear combiner then provide estimates of the noise and signal subspaces, from which the directions of the incident signals can be estimated and tracked. The proposed technique is also capable of handling the tracking of appearing / disappearing sources during the observation interval and, furthermore, can accommodate array modeling uncertainties. The difficult problem of tracking the crossing sources can be successfully handled by using diversely polarized array.
ICASSP98 Paper Abstract

Frequency Estimation and Detection for Sinusoidal Signals with Arbitrary Envelope: A Nonlinear Least-Squares Approach

Authors:
Olivier Besson, ENSICA, (France)
Petre Stoica, Uppsala University, (Sweden)

Volume 4, page 2209, paper no. 1015

Abstract:
In this paper, we consider the problem of estimating the frequency of a sinusoidal signal whose amplitude could be either constant or time-varying. We present a nonlinear least-squares (NLS) approach when the envelope is time-varying. We show that the NLS estimator can be efficiently implemented using a FFT. A statistical analysis shows that the NLS frequency estimator is nearly efficient. The problem of detecting amplitude time variations is next addressed. A statistical test is formulated, based on the statistics of the difference between two frequency estimates. The test is computationally efficient and yields as a by-product consistent frequency estimates under either hypothesis (i.e. constant or time-varying amplitude). Numerical examples are included to show the performance in terms of both estimation and detection.
On Minimax Lower Bound for Time-Varying Frequency Estimation of Harmonic Signal

Authors:
Alexander Nazin, Institute of Control Science, (Russia)
Vladimir Katkovnik, University of South Africa, (South Africa)

Abstract:
Estimation of the instantaneous frequency and its derivatives is considered for a harmonic signal with a time-varying phase and time-invariant amplitude. The asymptotic minimax lower bound is derived for the meansquared error of estimation provided that the phase is an arbitrary m-times piece-wise differentiable function of time. It is shown that this lower bound is different only in a constant factor from the upper bound for the mean squared errors of the local polynomial periodogram with optimal window size.
ICASSP98 Paper Abstract

The Effect of Sampling and Quantization on Frequency Estimation

Authors:
Anders Host-Madsen, Kwangju Institute of Science & Technology, (Korea)
Peter Hänel, Royal Institute of Technology, (Sweden)

Volume 4, page 2217, paper no. 1435

Abstract:
The effect of sampling and quantization on frequency estimation for a single sinusoid is investigated. Cramér-Rao bound for 1 bit quantization is derived, and compared with the limit of infinite quantization. It is found that 1 bit quantization gives a slightly worse performance, however, with a dramatic increase of variance at certain frequencies. This can be avoided by using 4 times oversampling. The effect of sampling when using non-ideal anti-aliasing lowpass filters is therefore investigated. Cramér-Rao lower bounds are derived, and the optimal filters and sampling frequencies are found. Finally, fast estimators for 1 bit sampling, in particular correlation based estimators, are derived. The paper is concluded with simulation results for 4 times oversampled 1 bit quantization.
Sinusoids in White Noise: A Quadratic Programming Approach

Authors:
Nicolas N Moal, IRISA/Universite de Rennes, (France)
Jean Jacques Fuchs, IRISA/Universite de Rennes, (France)

Abstract:
We address the problem of the estimation and identification of real sinusoids in white Gaussian noise using a correlation-based method. We estimate a partial covariance sequence from the data and seek a representation of these observations as a superposition of a small number of cosines chosen from a redundant basis and the white noise contribution. We propose to minimize a quadratic program in order to choose a parsimonious decomposition among the many that allow the reconstruction. We develop optimality conditions for the criterion that can be geometrically interpreted and present a dual criterion that has an appealing physical interpretation. Some simulated examples are also presented to show the excellent performance in resolution of the approach.
ICASSP98 Paper Abstract
Detection of Uncertain Multiple Cisioid Models

Authors:
Roland Jonsson, Ericsson Microwave Systems AB, (Sweden)
Jan O Hagberg, Ericsson Microwave Systems AB, (Sweden)

Volume 4, page 2225, paper no. 1564

Abstract:
The problem of detection of multiple complex sinusoids, with uncertain parameters, is addressed in this paper. It is shown that uncertainties in amplitude and small uncertainties in frequency can be handled analytically, while unknown phases must be handled numerically. Robust detectors for some or all of the uncertainties are formulated. Performance in noise, and robustness are evaluated through simulations. Finally the applicability of the detectors for the problem of radar target recognition is discussed, and some results are presented.
ICASSP98 Paper Abstract

Bootstrap Model Selection for Polynomial Phase Signals

Authors:
Abdelhak M Zoubir, Queensland University of Technology, (Australia)
D. Robert Iskander, Queensland University of Technology, (Australia)

Volume 4, page 2229, paper no. 1607

Abstract:
We consider the problem of estimating the order of the phase of a complex valued signal, having a constant amplitude and a polynomial phase, measured in additive noise. A new method based on the bootstrap is introduced. The proposed approach does not require knowledge of the distribution of the noise, is easy to implement, and unlike existing techniques, it achieves high performance when only a small amount of data is available. The proposed technique can be easily extended to non-stationary signals which have a polynomial amplitude and phase, provided a consistent estimator for the parameters can be obtained.
ICASSP98 Paper Abstract

Frequency Weighted Generalized Total Least Squares Linear Prediction for Frequency Estimation

Authors:
Shu Hung Leung, City University of Hong Kong, (Hong Kong)
Tin Ho Lee, City University of Hong Kong, (Hong Kong)
Wing Hong Lau, City University of Hong Kong, (Hong Kong)

Volume 4, page 2233, paper no. 1963

Abstract:
This paper presents a frequency weighted generalized total least squares linear prediction for estimating closely spaced sinusoids. In this method, the received data is first processed by a pole-zero prefilter and then a generalized total least squares linear prediction is applied to the prefiltered signal. A procedure of optimizing the generalized solution is introduced. By computer simulations, it is shown that the solution can outperform the existing well known total least squares solutions especially in low signal-to-noise ratio.
ICASSP98 Paper Abstract

Bayesian Analysis for the Fault Detection of Three-phase Induction Machine

Authors:
Michelle Vieira, Universite de Nice-Sophia Antipolis, (France)
Celine Theys, Universite de Nice-Sophia Antipolis, (France)

Volume 4, page 2237, paper no. 2049

Abstract:
One of the most widely used techniques for obtaining information on the health state of three-phase induction machines is based on the processing of stator current. In fact, in the case of steady state operations, anomalous current spectral components, that increase if a fault occurs, allow to diagnose the presence and, in some case, the type of fault. In this paper, a Bayesian approach is proposed using a simulation technique, the Markov chain Monte Carlo (MCMC), to estimate the amplitude of some spectral components modified by machine faults and the slip, a parameter related to the load conditions, with a view to automatically detecting faults. Results on real stator current waveform are given.
ICASSP98 Paper Abstract
On the Concept of Instantaneous Frequency

Authors:
Paulo M Oliveira, Escola Naval, (Portugal)
Victor A.N. Barroso, Instituto Superior Técnico, (Portugal)

Volume 4, page 2241, paper no. 2274

Abstract:
The concept of Instantaneous Frequency is still not clearly defined. Current operational definitions give rise to physical paradoxes, difficulting proper interpretation of the obtained results. In this paper, we discuss why those paradoxes appear, and show how they can be avoided. We introduce a new definition of Instantaneous Frequency, which yields physically consistent results. This is confirmed with the help of several examples.
ICASSP98 Paper Abstract

Joint Bayesian Detection and Estimation of Sinusoids Embedded in Noise

Authors:
Christophe Andrieu, ENSEA-ETIS, (France)
Arnaud Doucet, ENSEA-ETIS, (France)
Patrick Duvaut, ENSEA-ETIS, (France)

Volume 4, page 2245, paper no. 2434

Abstract:
In this paper we address the problem of the joint detection and estimation of sinusoids embedded in noise, from a Bayesian point of view. We first propose an original Bayesian model. A large number of parameters has to be estimated, including the number of sinusoids. No analytical developments can be performed. This lead us to design a new stochastic algorithm relying on reversible jump MCMC (Markov chain Monte Carlo). We obtain very satisfactory results.
Efficient Multiscale Stochastic Realization

Authors:
Austin B Frakt, MIT, (U.S.A.)
Alan S Willsky, MIT, (U.S.A.)

Volume 4, page 2249, paper no. 1195

Abstract:
Few fast statistical signal processing algorithms exist for large problems involving non-stationary processes and irregular measurements. A recently introduced class of multiscale autoregressive models indexed by trees admits signal processing algorithms which can efficiently deal with problems of this type. In this paper we provide a novel and efficient algorithm for translating any second-order prior model to a multiscale autoregressive prior model so that these efficient signal processing algorithms may be applied.
ICASSP98 Paper Abstract

Fast, Non-Iterative Estimation of Hidden Markov Models

Authors:
Hakan Hjalmarsson, KTH, (Sweden)
Brett M. Ninness, University of Newcastle, (Australia)

Abstract:
The solution of many important signal processing problems depends on the estimation of the parameters of a Hidden Markov Model (HMM). Unfortunately, to date the only known methods for performing this estimation have been iterative, and therefore computationally demanding. By way of contrast, this paper presents a new fast and non-iterative method that utilizes certain recent 'state spaced subspace system identification' (4SID) ideas from the control theory literature. A short simulation example presented here indicates this new technique to be almost as accurate as Maximum-Likelihood estimation, but an order of magnitude less computationally demanding than the Baum-Welch (EM) algorithm.
ICASSP98 Paper Abstract

A Reversible Jump Sampler for Autoregressive Time Series

Authors:
Paul T. Troughton, University of Cambridge, (U.K.)
Simon J. Godsill, University of Cambridge, (U.K.)

Volume 4, page 2257, paper no. 1513

Abstract:
We use reversible jump Markov chain Monte Carlo (MCMC) methods to address the problem of model order uncertainty in autoregressive (AR) time series within a Bayesian framework. Efficient model jumping is achieved by proposing model space moves from the full conditional density for the AR parameters, which is obtained analytically. This is compared with an alternative method, for which the moves are cheaper to compute, in which proposals are made only for new parameters in each move. Results are presented for both synthetic and audio time series.
ICASSP98 Paper Abstract

A New Maximum Likelihood Gradient Algorithm for On-Line Hidden Markov Model Identification

Authors:
Iain B Collings, University of Melbourne, (Australia)
Tobias Ryden, Lund University, (Sweden)

Volume 4, page 2261, paper no. 1884

Abstract:
This paper presents a new algorithm for on-line identification of hidden Markov model (HMM) parameters. The scheme is gradient based, and provides parameter estimates which recursively maximise the likelihood function. It is therefore a recursive maximum likelihood (RML) algorithm, and it has optimal asymptotic properties. The only current on-line HMM identification algorithm with anything other than suboptimal rate of convergence is based on a prediction error (PE) cost function. As well as presenting a new algorithm, this paper also highlights and explains a counter-intuitive convergence problem for the current recursive PE (RPE) algorithm, when operating in low noise conditions. Importantly, this problem does not exist for the new RML algorithm. Simulation studies demonstrate the superior performance of the new algorithm, compared to current techniques.
ICASSP98 Paper Abstract

Quasi-Newton Method for Maximum Likelihood Estimation of Hidden Markov Model

Authors:
Olivier Cappé, ENST / CNRS, (France)
Vincent Buchoux, ENST / CNET, (France)
Eric Moulines, ENST / CNRS, (France)

Abstract:
Hidden Markov models (HMMs) are used in many signal processing applications including speech recognition, blind equalization of digital communications channels, etc. The most widely used method for maximum likelihood estimation of HMM parameters is the forward-backward (or Baum-Welch) algorithm which is an early example of application of the Expectation-Maximization (EM) principle. In this contribution, an alternative fast-converging approach for maximum likelihood estimation of HMM parameters is described. This new technique is based on the use of general purpose quasi-Newton optimization methods as well as on an efficient purely recursive algorithm for computing the log-likelihood and its derivative.
ICASSP98 Paper Abstract

Detection and Estimation of Signals by Reversible Jump Markov Chain Monte Carlo Computations

Authors:
Petar M. Djuric, SUNY, Stony Brook, (U.S.A.)
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Volume 4, page 2269, paper no. 2007

Abstract:
Markov Chain Monte Carlo (MCMC) samplers have been a very powerful methodology for estimating signal parameters. With the introduction of the reversible jump MCMC sampler, which is a Metropolis-Hastings method adapted to general state spaces, the potential of the MCMC methods has risen to a new level. Consequently, the MCMC methods currently play a major role in many research activities. In this paper we propose a reversible jump MCMC sampler based on predictive densities obtained by integrating out unwanted parameters. The proposal densities are approximations of the posterior distributions of the remaining parameters obtained by sampling importance resampling (SIR). We apply the method to the problem of signal detection and parameter estimation of signals. To illustrate the procedure, we present an example of sinusoids embedded in noise.
Modeling and Detection in Hyperspectral Imagery

Authors:
Susan M Schweizer, Carnegie Mellon University, (U.S.A.)
José M.F. Moura, Carnegie Mellon University, (U.S.A.)

Abstract:
One aim of using hyperspectral imaging sensors is in discriminating man-made objects from dominant clutter environments. Sensors like Aviris or Hydice simultaneously collect hundreds of contiguous and narrowly spaced spectral band images for the same scene. The challenge lies in processing the corresponding large volume of data that is collected by the sensors. Usual implementations of the Maximum-Likelihood (ML) detector are precluded because they require the inversion of large data covariance matrices. We apply a Gauss-Markov random field (GMRF) model to derive a computationally efficient ML-detector implementation that avoids inversion of the covariance matrix. The paper details the structure of the GMRF model, presents an estimation algorithm to fit the GMRF to the hyperspectral sensor data, and finally, develops the structure of the ML-detector.
ICASSP98 Paper Abstract
Simplified Wavelet-domain Hidden Markov Models Using Contexts

Authors:
Matthew S Crouse, Rice University, (U.S.A.)
Richard G Baraniuk, Rice University, (U.S.A.)

Volume 4, page 2277, paper no. 2092

Abstract:
Wavelet-domain Hidden Markov Models (HMMs) are a potent new tool for modeling the statistical properties of wavelet transforms. In addition to characterizing the statistics of individual wavelet coefficients, HMMs capture the salient interactions between wavelet coefficients. However, as we model an increasing number of wavelet coefficient interactions, HMM-based signal processing becomes increasingly complicated. In this paper, we propose a new approach to HMMs based on the notion of context. By modeling wavelet coefficient inter-dependencies via contexts, we retain the approximation capabilities of HMMs, yet substantially reduce their complexity. To illustrate the power of this approach, we develop new algorithms for signal estimation and for efficient synthesis of non Gaussian, long-range-dependent network traffic.
ICASSP98 Paper Abstract
The Effective Bandwidth of Stable Distributions

Authors:
Stephen Bates, Massana, Dublin, (Ireland)
Steve McLaughlin, University of Edinburgh, Scotland, (U.K.)

Volume 4, page 2281, paper no. 1284

Abstract:
In this paper the effective bandwidths of stable distributions are studied. Effective bandwidths are being heavily promoted as the most appropriate method for call admission control (CAC) and resource allocation within ATM networks. Recent work in teletraffic modelling has suggested that models based on stable distributions provide an efficient mechanism for capturing the long range dependence and infinite variance associated with teletraffic data (the Joseph and Noah effects). This has potentially serious implications for effective bandwidths and we show how the effective bandwidth of such data is theoretically infinite. We then present two approximate methods for estimating the effective bandwidth of data based on stable distribution.
ICASSP98 Paper Abstract

Optimal Selection of Information with Restricted Storage Capacity

Authors:
Luc Pronzato, CNRS, (France)

Volume 4, page 2285, paper no. 1396

Abstract:
We consider the situation where n items have to be selected among a series of N presented sequentially, the information contained in each item being random. The problem is to get a collection of n items with maximal information. We consider the case where the information is additive, and thus need to maximize the sum of n independently identically distributed random variables x(k) observed sequentially in a sequence of length N. This is a stochastic dynamic-programming problem, the optimal solution of which is derived when the distribution of the x(k)'s is known. The asymptotic behaviour of this optimal solution (when N tends to infinity with n fixed) is considered. A (forced) certainty-equivalence policy is proposed for the case where the distribution is unknown and estimated on-line.
ICASSP98 Paper Abstract

Asymptotic Statistical Properties of Autoregressive Model for Mixed Spectrum Estimation

Authors:
Peter J Sherman, Iowa State University, (U.S.A.)
Soon-Seng Lau, Iowa State University, (U.S.A.)

Volume 4, page 2289, paper no. 1624

Abstract:
This work addresses the influence of point spectrum on large sample statistics of the autoregressive spectral estimator. In particular, the asymptotic distributions of the AR coefficients, the innovations variance, and the spectral density estimator of a finite order AR(p) model for a mixed spectrum process are presented. Numerical simulations are performed to verify the analytical results.
ICASSP98 Paper Abstract

Analytic Center Approach to Parameter Estimation: Convergence Analysis

Authors:
Er-wei Bai, University of Iowa, (U.S.A.)
Minyue Fu, University of Newcastle, (Australia)
Roberto Tempo, Politecnico di Torino, (Italy)
Yinyu Ye, University of Iowa, (U.S.A.)

Volume 4, page 2293, paper no. 1628

Abstract:
The so-called analytic center approach to parameter estimation has been proposed recently as an alternative to the well-known least squares approach. This new approach offers a parameter estimate that is consistent with the past data observations, has a simple geometric interpretation, and is computable using linear programming algorithms. In this paper, we study the asymptotic performance of the analytic center approach and show that the resulting estimate converges to the true parameter asymptotically, provided some mild conditions are satisfied. These conditions involve some weak persistent excitation and independence between noise and regressor, similar to the least squares case. This result is used to derive a new parameter estimation approach which offers both good transient and asymptotic performances.
ICASSP98 Paper Abstract

Factorizability of Complex Signals Higher (Even) Order Spectra: A Necessary and Sufficient Condition

Authors:
Joel Le Roux, University of Nice, CNRS, (France)
Cécile Huet, University of Nice, CNRS, (France)

Volume 4, page 2297, paper no. 1715

Abstract:
This paper presents a necessary and sufficient condition for the factorizability of higher order spectra of complex signals. Such a factorizability condition can be used to test if a complex signal can modelize the output of a linear and time invariant system driven by a stationary non gaussian white input. The condition developed here is based on the symmetries of higher order spectra and on an extension of a formula proposed by Marron et al. to unwrap third order spectrum phases. It is an identity between products of six higher order spectra values (which reduces to four values if only phases are considered). Our factorizability test requires no phase unwrapping, unlike existing methods developed in the cepstral domain. Moreover its extension to the N-th order case is direct. Simulations illustrate the deviation to this factorizability condition in a factorizable case (linear system) and a non factorizable case (non linear system).
Nonlinear H-ARMA Models

Authors:
David Declercq, ETIS CNRS, (France)
Patrick Duvaut, ETIS CNRS, (France)


Abstract:
We present, in this contribution, some aspects of nongaussian H-ARMA models. After recalling that an
H-ARMA process is obtained by passing an ARMA process through a Hermite polynomial nonlinearity,
we describe the theoretical analysis of their cumulants and cumulant spectra. The main advantage of
this kind of model is that the cumulant structure of the output can be deduced directly from the input
covariance sequence. We give the analytic forms of these cumulants, together with some comments
on their estimation. Then, we present the problems we are facing concerning the identification of the
model's parameters, and give a first (and naive) method for their estimation. We give some results
obtained on synthetic data and finally conclude with some remarks on this class of processes.
ICASSP98 Paper Abstract

New Higher Order Spectra and Time-Frequency Representations for Dispersive Signal Analysis

Authors:
Robin L Murray, University of Rhode Island, (U.S.A.)
Antonia Papandreou-Suppappola, University of Rhode Island, (U.S.A.)
G. Faye Boudreaux-Bartels, University of Rhode Island, (U.S.A.)

Volume 4, page 2305, paper no. 2091

Abstract:
For analysis of signals with arbitrary dispersive phase laws, we extend the concept of higher order moment functions and define their associated higher order spectra. We propose a new higher order time-frequency representation (TFR), the higher order generalized warped Wigner distribution (HOG-WD). The HOG-WD is obtained by warping the previously proposed higher order Wigner distribution, and is important for analyzing signals with arbitrary time-dependent instantaneous frequency. We discuss links to prior higher order techniques and investigate properties of the HOG-WD. We extend the HOG-WD to a class of higher order, alternating sign, frequency-shift covariant TFRs. Finally, we demonstrate the advantage of using the generalized higher order spectra to detect phase coupled signals with dispersive instantaneous frequency characteristics.
ICASSP98 Paper Abstract

Performance Analysis of Cyclic Estimators for Harmonics in Multiplicative and Additive Noise

Authors:
Ananthram Swami, Army Research Laboratory, (U.S.A.)
Mounir Ghogho, University of Strathclyde, Scotland, (U.K.)

Volume 4, page 2309, paper no. 2098

Abstract:
The problem of interest is the estimation of the parameters of harmonics in the presence of additive and multiplicative noise. Expressions for the asymptotic performance of the cyclic-variance (CV) based method are derived when the multiplicative noise has non-zero mean. We show that the CV-based method may yield more accurate results than methods based on the cyclic mean (CM), depending upon the color of the noise and the intrinsic and local SNRs. Performance is analyzed in detail for several special cases of the multiplicative noise, such as white Gaussian, AR and generalized-Gaussian noise.
ICASSP98 Paper Abstract

On the Fourth-Order Cumulants Estimation for the H0 Blind Separation of Cyclostationary Sources

Authors:
Anne Ferreol, Thomson-CSF, (France)
Pascal Chevalier, Thomson-CSF, (France)

Abstract:
Most of the HO blind source separation methods developed this last decade aim at blindly separating statistically independent sources, assumed stationary and ergodic. Nevertheless, in many situations such as in radiocommunications, the sources are non stationary and very often (quasi)-cyclostationary (digital modulations). In these contexts, it is important to wonder if the performance of these HO blind source separation methods may be affected by the potential non stationarity of the sources. The purpose of this paper is to bring some answers to this question through the behaviour analysis of the classical fourth-order cumulant estimators in the presence of (quasi)-cyclostationary sources.
ICASSP98 Paper Abstract
Kurtosis-Based Criteria for Adaptive Blind Source Separation

Authors:
Constantinos B. Papadias, Lucent Technologies/Bell Laboratories, (U.S.A.)
Volume 4, page 2317, paper no. 1609

Abstract:
We consider the problem of separating adaptively $p$ synchronous user signals that are received by an $m$-element antenna array without the use of training sequences. We establish a set of necessary and sufficient conditions for perfect recovery of all the transmitted signals. Based on these conditions we propose optimization criteria that lead to adaptive algorithms for efficient blind source separation of non-Gaussian signals. Convergence analysis shows important global convergence properties of the proposed techniques. Combined with their low computational complexity, these features make the proposed algorithms good candidates for adaptive source separation.
ICASSP98 Paper Abstract

On Adaptive Local Polynomial Approximation with Varying Bandwidth

Authors:
Vladimir Katkovnik, University of South Africa, (South Africa)

Abstract:
The local polynomial approximation (LPA) of noisy data is considered with the new adaptive procedure for varying bandwidth selection. The algorithm is simple to implement and nearly optimal within ln N factor in the point-wise risk for estimating the function and its derivatives. The adaptive varying bandwidth enables the algorithm to be spatial adaptive over a wide range of the classes of functions in the sense that its quality is close to that which one could achieve if smoothness of the estimated function was known in advance. It is shown that the cross-validation adjustment of the threshold parameter of the algorithm significantly improves its accuracy. In particular, simulation demonstrates that the adaptive algorithm with the adjusted threshold parameter performs better than the wavelet estimators.
ICASSP98 Paper Abstract

Computationally Efficient Maximum-Likelihood Estimation of Structured Covariance Matrices

Authors:
Hongbin Li, University of Florida, (U.S.A.)
Petre Stoica, Uppsala University, (Sweden)
Jian Li, University of Florida, (U.S.A.)

Volume 4, page 2325, paper no. 1038

Abstract:
A computationally efficient method for structured covariance matrix estimation is presented. The proposed method provides an Asymptotic (for large samples) Maximum Likelihood estimate of a structured covariance matrix and is referred to as AML. A closed-form formula for estimating Hermitian Toeplitz covariance matrices is derived which makes AML computationally much simpler than most existing Hermitian Toeplitz matrix estimation algorithms. The AML covariance matrix estimator can be used in a variety of applications. We focus on array processing herein and show that AML enhances the performance of angle estimation algorithms, such as MUSIC, by making them attain the corresponding Cramer-Rao bound (CRB) for uncorrelated signals.
ICASSP98 Paper Abstract
Unbiased Identification of Autoregressive Signals Observed in Colored Noise

Authors:
Wei Xing Zheng, University of Western Sydney, (Australia)

Abstract:
Autoregressive (AR) modeling has played an important role in many signal processing applications. This paper is concerned with identification of AR model parameters using observations corrupted with colored noise. A novel formulation of an auxiliary least-squares estimator is introduced so that the autocovariance functions of the colored observation noise can be estimated in a straightforward manner. With this, the colored-noise-induced estimation bias can be removed to yield the unbiased estimate of the AR parameters. The performance of the proposed unbiased estimation algorithm is illustrated by simulation results. The presented work greatly extends the author’s previous method that was developed for identification of AR signals observed in white noise.
Abstract:
The problem of estimating continuous-time autoregressive process parameters from discrete-time data is considered. The basic approach used here is based on replacing the derivatives in the model by discrete-time differences, forming a linear regression and using the least squares method. It is known, however, that all standard approximations of the highest order derivative give a biased least squares estimate even as the sampling interval tends to zero. Some of our previous approaches to overcome this problem are briefly reviewed. Then two new methods are presented. One of them, termed bias compensation, can be easily implemented efficiently in an order recursive manner. Comparative simulation results are also presented.
ICASSP98 Paper Abstract
Blind Frequency Offset and Delay Estimation of Linearly Modulated Signals Using Second Order Cyclic Statistics

Authors:
Vel B Manimohan, University of Cambridge, (U.K.)
William J. Fitzgerald, University of Cambridge, (U.K.)

Abstract:
A blind (non-data aided), open-loop, joint frequency offset-delay estimation algorithm for a linearly modulated signal in additive stationary noise is developed by exploiting the cyclostationarity of the signal. By considering the sample cyclic autocorrelation function of the received signal and the probability distribution of the estimation error, a general linear model representation of the problem is obtained, from which the parameters are estimated using a Bayesian approach. The algorithm is then extended to a multiple signals of interest scenario. The algorithm is simulated for both single and multiple BPSK signals.
Parameter Estimation Using Volterra Series

Authors:
Mark C.M. Hsieh, University of Cambridge, (U.K.)
Peter J.W. Rayner, University of Cambridge, (U.K.)

Volume 4, page 2341, paper no. 1711

Abstract:
A polynomial approximation to the likelihood function allows for marginalised estimates of model parameters to be obtained in the form of a Volterra series. The series can be applied directly to the observed data vector in an iterative fashion, to converge upon a set of parameter MAP estimates with low computational cost. A sample application towards OCR is used as an illustration.
ICASSP98 Paper Abstract

Wavelet-Domain Modeling and Estimation of Poisson Processes

Authors:
Klaus E Timmermann, Michigan State University, (U.S.A.)
Robert D. Nowak, Michigan State University, (U.S.A.)

Volume 4, page 2345, paper no. 2125

Abstract:
This paper develops a new wavelet-domain Bayesian framework for modeling and estimating the intensity of a Poisson process directly from count observations. A new multiscale, multiplicative innovations model is developed as a prior for the underlying intensity function. The new prior model leads to a simple and efficient close-form estimator that requires order $N$ computations, where $N$ is the dimension of the intensity function. We compare the new method with previously proposed wavelet-based approach to this problem.
ICASSP98 Paper Abstract

Toeplitz and Hankel Matrix Approximation Using Structured Approach

Authors:
Arnab K. Shaw, Wright State University, (U.S.A.)
Srikanth Pokala, Wright State University, (U.S.A.)
Ramdas Kumaresan, University of Rhode Island, (U.S.A.)

Volume 4, page 2349, paper no. 5181

Abstract:
Algorithms are presented for least-squares approximation of Toeplitz and Hankel matrices from noise corrupted of ill-composed matrices, which may not have correct structural or rank properties. Utilizing Caratheodery’s Theorem on complex number representation to model the Toeplitz and Hankel matrices, it is shown that these matrices possess specific row and column structures. The inherent structures of the matrices are exploited to develop a computational algorithm for estimation of the matrices that are closest, in the Frobenius norm sense, to the given noisy or rank-excessive matrices. Simulation studies bear out the effectiveness of the proposed algorithms providing significantly better results than the state-space methods.
ICASSP98 Paper Abstract

Super-Exponential Methods for Multichannel Blind Deconvolution

Authors:
Yujirou Inouye, Shimane University, (Japan)
Kazuaki Tanebe, Osaka University, (Japan)

Abstract:
Multichannel blind deconvolution has received increasing attention during the last decade. Recently, Martone (3, 4) extended the super-exponential method proposed by Shalvi and Weinstein (1, 2) for single-channel blind deconvolution to multichannel blind deconvolution. However, the Martone extension suffers from two types of serious drawbacks. The objective of this paper is to obviate these drawbacks and to propose three approaches to multichannel blind deconvolution. In the first approach, we present a multichannel super-exponential algorithm. In the second approach, we present a super-exponential deflation algorithm. In the third approach, we present a two-stage super-exponential algorithm.
ICASSP98 Paper Abstract

An Efficient Blind Identification Algorithm for Multi-channel FIR Systems Using Linear Prediction

Authors:
Yifeng Zhou, Telexis Corporation, (Canada)
Henry Leung, SurfaceRadar Section, DREO, (Canada)
Patrick C. Yip, McMaster University, (Canada)

Volume 4, page 2357, paper no. 1511

Abstract:
In this paper, an efficient blind identification algorithm for multichannel FIR systems was proposed based on a deterministic model of the channel input. By decoupling the multichannel identification, the proposed method was able to estimate each individual channel responses separately without having to solve for the augmented channel responses. The algorithm was implemented using linear prediction techniques. It was computationally efficient and suitable for real-time applications. Computer simulations were used to demonstrate the effectiveness of the proposed algorithm.
ICASSP98 Paper Abstract

A New Pencil Criterium for Multichannel Blind Deconvolution in Data Communication Systems

Authors:
Santiago Zazo, Universidad Alfonso X El Sabio, (Spain)
Jose Manuel Paez-Borrallo, Universidad Politecnica de Madrid, (Spain)

Volume 4, page 2361, paper no. 1545

Abstract:
It is well known that blind channel deconvolution enables the receiver to equalize the channel simply by analyzing the received digital signal. Much of the work in 1990's faces the challenge presented by multiple-output systems, exploiting cyclostationarity properties and multivariate formulation of the incoming data. Our proposal is twofold: on one hand, we develop a theoretical analysis of a new blind channel deconvolution scheme by the exploitation of some shifting properties of the autocorrelation matrices of the source, in order to propose an appropriate cost function; on the other hand, an efficient programming is considered based on a Generalized Rayleigh Quotient formulation by using a Conjugate Gradient algorithm.
ICASSP98 Paper Abstract

Maximum Likelihood Estimation with Side Information of 1-D Layered Media from Noisy Impulse Reflection Responses

Authors:
Andrew E. Yagle, *University of Michigan*, (U.S.A.)
Rajashri R Joshi, *University of Michigan*, (U.S.A.)

Volume 4, page 2365, paper no. 1861

Abstract:
We consider the problem of computing the maximum likelihood estimates of the reflection coefficients of a discrete 1-D layered medium from noisy observations of its impulse reflection response. We have side information in that a known subset of the reflection coefficients are known to be zero; this knowledge could come from either a priori knowledge of a homogeneous subregion inside the scattering medium, or from a thresholding operation in which noisy reconstructed reflection coefficients with absolute values below a threshold are known to be zero. Our approach is simple, noniterative, and requires only solutions of systems of linear equations. Numerical examples are provided which demonstrate not only the operation of the algorithm, but also that the side information improves the reconstruction of unconstrained reflection coefficients as well as constrained ones, due to the nonlinearity of the problem.
ICASSP98 Paper Abstract

Semi-Blind Identification of Finite Impulse Response Channels

Abstract:
It is a standard result that a finite impulse response channel of length L can be uniquely identified by feeding in a known (and persistently exciting) sequence of 2L-1 consecutive data points. Equivalently, given only 2L-2 consecutive data points, the channel can be uniquely identified up to a multiplicative constant. This paper significantly extends the identifiability criterion to the case when the known inputs are non-consecutively located. It is argued that by introducing 2L-1 non-consecutively spaced zeros into the input stream, for almost all input sequences, the channel can be uniquely identified up to a multiplicative constant. Furthermore, the result can be extended to the case when the known inputs are non-zero, in which case the channel can almost always be identified uniquely. To arrive at these results, general properties of systems of polynomial equations are derived. These properties do not seem to have appeared in the literature before. Key words: Algebraic geometry, Commutative algebra, Polynomial equations, Semi-blind identification, Finite impulse response channels.
ICASSP98 Paper Abstract

Optimal MAP Estimation of Bilinear Systems via the EM Algorithm

Authors:
Vikram K. Krishnamurthy, University of Melbourne, (Australia)
Leigh A Johnston, University of Melbourne, (Australia)
Andrew Logothetis, University of Melbourne, (Australia)

Abstract:
In this paper we present a finite dimensional iterative algorithm for optimal maximum a posteriori (MAP) state estimation of bilinear systems. Bilinear models are appealing in their ability to represent or approximate a broad class of nonlinear systems. We show that several bilinear models previously considered in the literature are special cases of the general bilinear model we propose. Our iterative algorithm for state estimation is based on the Expectation-Maximization (EM) algorithm and outperforms the widely used Extended Kalman filter (EKF). Unlike the EKF, our algorithm is an optimal (in the MAP sense) finite-dimensional solution to the state sequence estimation problem for bilinear models.
ICASSP98 Paper Abstract

Improving Signal Subspace Estimation for Blind Source Separation in the Context of Spatially Correlated Noises

Authors:
Pierre Fabry, INPG-LIS, (France)
Christine Serviere, INPG-LIS, (France)
Jean-Louis Lacoume, INPG-LIS, (France)

Volume 4, page 2377, paper no. 2069

Abstract:
In this paper, we adress the issue of Orthogonal Techniques for Blind Source Separation of periodic signals when the mixtures are corrupted with spatially correlated noises. The noise covariance matrix is assumed to be unknown. This problem is of major interest with experimental signals. We first remind that the Principal Component Analysis (PCA) cannot provide a correct estimate of the signal subspace when the noises are spatially correlated or when their power spectral densities are different. We then introduce a new estimator of the unnoisy spectral matrix using delayed blocks. The only assumption is that the noise correlation and cross-correlation lengths must be shorter than the source correlation lengths. Simulation results show the efficiency of the new method.
ICASSP98 Paper Abstract

Estimation of Blood Pump Parameters for Cardiovascular System Identification

Authors:
Yih-Choung Yu, University of Pittsburgh, (U.S.A.)
Marwan Simaan, University of Pittsburgh, (U.S.A.)
J. Robert Boston, University of Pittsburgh, (U.S.A.)
Philip J. Miller, Novacor, Baxter Healthcare, (U.S.A.)
James F. Antaki, University of Pittsburgh, (U.S.A.)

Volume 4, page 2381, paper no. 2189

Abstract:
This paper describes the use of signal processing techniques to estimate the model parameters of a leftventricular assist device (LVAD). The model consisted of lumped resistance, capacitance, and inductance elements with one time-varying capacitor to estimate the cyclical pressure generation of the device using volume signal from the device. The model parameters were estimated by least squares fit to experimental data obtained in the laboratory. The purpose of this research is to estimate the pressure and flow signals, which are usually measured through invasive physiologic sensors, for an on-line estimator to identify cardiovascular parameters of patients who are under LVAD assist. The success of this development would provide a useful tool to monitor the cardiac function of LVAD patients without indwelling sensors. A computer simulation of the pump with a cardiovascular model was developed to demonstrate the interaction between the LVAD and the cardiovascular system. The simulation results showed agreement with those from an animal experiment and thus the simulation waveforms can be used for testing the cardiovascular estimator.
A Globally Convergent Approach for Blind MIMO Adaptive Deconvolution

Authors:
Azzedine Touzni, ETIS, (France)
Inbar Fijalkow, ETIS, (France)
Micheal G. Larimore, Applied Signal Technology, (U.S.A.)

Abstract:
We address the deconvolution of MIMO linear mixtures. The approach is based on the construction of a hierarchical family of composite criteria involving CM criterion and second order statistics constraint. Although, the criteria are based on fourth order statistics, we give a complete proof of convergence of this structure. We show that each cost function leads to the restoration of one single source. Moreover the approach is naturally robust with respect to the channels order estimation. An adaptive algorithm is derived for the simultaneous estimation of all sources.
ICASSP98 Paper Abstract

Blind Identification and Order Estimation of FIR Communication Channels Using Cyclic Statistics

Authors:
Gianpiero Panci, Dip. INFOCOM, Universita di Roma, (Italy)
Gaetano Scarano, Dip. INFOCOM, Universita di Roma, (Italy)
Giovanni Jacovitti, Dip. INFOCOM, Universita di Roma, (Italy)

Abstract:
In this contribution we address the problem of the blind joint identification and order estimation of a non-minimum phase FIR communication channel by exploiting the cyclostationarity of the received signal sampled at rate greater of the symbol rate. We show that the identification can be formulated as a "subspace fitting" problem; this allows for using the subspace distance as a test statistic to detect the correct channel length (among different hypotheses). Moreover, mimicking estimation procedures proposed in the framework of DOA estimation, an asymptotically efficient procedure is proposed which obtains the channel estimate in two steps: the channel estimate obtained in the first step determines optimal weights which are then employed in the second step to refine the channel estimate. The accuracy of the identification is comparable with other methods described in the literature (e.g. the subspace method [4]) while the test for order detection performs quite well also in the presence of channel disparity outperforming commonly used tests based on the eigenvalues of the covariance matrix.
ICASSP98 Paper Abstract

MRA of Processes Synthesized by Differintegration

Authors:
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Uday B Desai, Institute of Technology, Bombay, (India)

Volume 4, page 2393, paper no. 1102

Abstract:
In this paper a definition of multiresolution analysis (MRA) of Gaussian processes is proposed. The problem, in a natural way, reduces to the MRA of the associated reproducing kernel Hilbert space. We then show that for processes synthesized from Gaussian white process by fractional integration of order greater than 1, this definition is applicable. The MRA results in an orthogonal expansion of these processes. The region of interest is the positive real line. Using this representation then a decomposition of a wider class of Gaussian processes is given. This representation is multiscale in two ways: firstly, the Gaussian process is split into various component processes characterized by the smoothness of their sample paths and secondly, each of these component processes has a MRA as defined in this paper.
ICASSP98 Paper Abstract

Structural Risk Minimization for Reduced-Bias Time-Frequency-Based Detectors Design

Authors:
Cedric Richard, Laboratoire LM2S, UTT, (France)
Regis Lengelle, Laboratoire LM2S, UTT, (France)

Volume 4, page 2397, paper no. 1250

Abstract:
Detectors design requires substantial knowledge of the observation statistical properties, conditionally to the competing hypotheses H0 and H1. However, many applications involve complex phenomena, in which few a priori information is available. Several methods of designing time-frequency-based (TF) receivers from labeled training data have been proposed. Unfortunately, the resulting detectors have large biases, particularly when the number of training samples is small against the data dimension. The method presented here is based on the Structural Risk Minimization principle developed by Vapnik, and consists in locally adjusting the resolution of TF-based detectors to the information carried by each TF location. This operation, controlled by a measure of H0 and H1 separability, allows to advantageously reduce receivers complexity and solutions bias. The resulting reduced-bias TF-based detectors can yield a substantial improvement in detection performances.
ICASSP98 Paper Abstract

Generalized Transfer Function Estimation Using Evolutionary Spectral Deblurring

Authors:
Syed I Shah, University of Pittsburgh, (U.S.A.)
Luis F. Chaparro, University of Pittsburgh, (U.S.A.)
Amro El-Jaroudi, University of Pittsburgh, (U.S.A.)

Volume 4, page 2401, paper no. 1758

Abstract:
We present a new technique for estimating the generalized transfer function (GTF) of a time-varying filter from time-frequency representations (TFRs) of its output. We use the fact that many of these representations can be written as blurred versions of the GTF. The approach consists in minimizing the error between the TFR found from the data and that found by blurring the GTF. The problem as such has many solutions. We, therefore, additionally constrain it to minimize the distance between the GTF-based spectrum and the autoterms of the Wigner distribution, suppressing the cross terms using an appropriate signal dependent mask function. To illustrate the performance of the proposed procedure we apply it to the spectral representation of speech and to signal masking and demonstrate its superior performance over the existing methods.
ICASSP98 Paper Abstract

Time-Varying Spectrum Estimators for Continuous-Time Signals

Authors:
Louis L. Scharf, University of Colorado, Boulder, (U.S.A.)
Benjamin Friedlander, University of California, Davis, (U.S.A.)
C. T. Mullis, University of Colorado, Boulder, (U.S.A.)

Volume 4, page 2405, paper no. 1760

Abstract:
Some quadratic time-frequency representations (TFRs) may be called time-varying spectrum estimators. They are derived from first principles, and they turn out to be time-varying multiwindow spectrum estimators. In special cases they are time-varying spectrograms that may be written as Fourier transforms of lag-windowed, time-varying correlation sequences or as spectrally smoothed time-varying periodograms. These are not ad-hoc variations on stationary ideas to accommodate time variation. Rather, they are the only variations one can obtain for time-varying spectrum analysis.
ICASSP98 Paper Abstract
Multiple Window Non-Linear Time-Varying Spectral Analysis

Authors:
Ferhat Cakrak, University of Pittsburgh, (U.S.A.)
Patrick J Loughlin, University of Pittsburgh, (U.S.A.)

Volume 4, page 2409, paper no. 2036

Abstract:
A non-linear multi-window method for generating a time-varying spectrum of non-stationary signals in noise is presented. The time-varying spectrum is computed from an optimally weighted average of multiple Hermite windowed spectrograms. The weights are determined using linear least squares estimation with respect to a reference time-frequency distribution. A masking operation is also used to reduce extraneous side lobes introduced by higher order Hermite windows. Several examples are provided, with performance criteria measures, to demonstrate and quantify the effectiveness of this new method.
ICASSP98 Paper Abstract
Optimal Pulse Shape for Estimating Positions of Superimposed Pulses

Authors:
Thanh D Nguyen, Auburn University, (U.S.A.)
Stanley J. Reeves, Auburn University, (U.S.A.)
Thomas S Denney Jr, Auburn University, (U.S.A.)

Volume 4, page 2413, paper no. 2149

Abstract:
In this paper, we determine the optimal pulse shape for estimating positions of superimposed pulses by deriving the Cramer-Rao lower bound (CRLB) on the average estimation error variance and optimizing it with respect to pulse shape. Our results show that a significant improvement in estimation error variance can be achieved relative to Gaussian and rectangular pulse shapes.
ICASSP98 Paper Abstract
Resolution of Overlapping Doppler Shifted Echoes

Authors:
Andreas Jakobsson, Uppsala University, (Sweden)
Arnold Lee Swindlehurst, Brigham Young University, (U.S.A.)
Petre Stoica, Uppsala University, (Sweden)

Volume 4, page 2417, paper no. 2240

Abstract:
This paper considers the problem of estimating the time delays and doppler shifts of a known waveform received via several distinct paths by an array of antennas. The general maximum likelihood estimator is presented, and is shown to require a \(2d\)-dimensional non-linear minimization, where \(d\) is the number of received signal reflections. Two alternative solutions based on signal and noise subspace fitting are proposed, requiring only a \(d\)-dimensional minimization. In particular, we show how to decouple the required search into a two-step procedure, where the delays are estimated and the dopplers solved for explicitly. Initial conditions for the time delay search can be obtained by applying generalizations of the MUSIC and ESPRIT algorithms.
ICASSP98 Paper Abstract

Auditory Scene Analysis Based on Time-Frequency Integration of Shared FM and AM

Authors:
Mototsugu Abe, The University of Tokyo, (Japan)
Shigeru Ando, The University of Tokyo, (Japan)

Volume 4, page 2421, paper no. 2552

Abstract:
This paper describes a new method for computational auditory scene analysis which is based on 1) waveform operators to extract instantaneous frequency (IF), frequency change (FM), and amplitude change (AM) from subband signals, and 2) a voting method into a probability distribution to extract coherency (shared fundamental frequency, shared FM, and shared AM) involved in them. We introduce non-parametric Kalman filtering for the time-axis integration. A consistent AM operator which is independent to frequency change is newly defined. Sharpness of the resultant probability distribution is examined with relating to the definition of the operators and subband bandwidth. We evaluate the performance of the algorithm by using several speech sounds.
ICASSP98 Paper Abstract
Nonstationary Spectrum Estimation and Time-Frequency Concentration

Authors:
James W Pitton, MathSoft, (U.S.A.)

Volume 4, page 2425, paper no. 2581

Abstract:
This paper extends Thomson's multitaper spectrum estimation method to nonstationary signals. The method uses a newly-derived set of basis functions which generalize the concentration properties of the prolate spheroidal waveforms to the time-frequency case. We solve for the basis which diagonalizes the nonstationary spectrum generating operator over a finite region of the time-frequency plane. These eigenfunctions are maximally concentrated to and orthogonal over the specified time-frequency region, and are thus doubly orthogonal. Individual spectrograms computed with these eigenfunctions form direct time-frequency spectrum estimates. We next present a multitaper time-frequency spectrum estimation procedure using these time-frequency eigenestimates. Bias and variance expressions are derived, allowing for a statistical characterization of the accuracy of the estimate. The time-frequency concentration property of the basis functions yields an estimator with excellent bias properties, while the variance of the estimate is reduced through the use of multiple orthogonal windows.
ICASSP98 Paper Abstract

Critically Sampled Gabor Transform with Localized Biorthogonal Function

Authors:
Osama A. Ahmed, King Fahd University of Petroleum & Minerals, (Saudi Arabia)
Moustafa M. Fahmy, King Fahd University of Petroleum & Minerals, (Saudi Arabia)

Abstract:
A new implementation of the critically sampled non-periodic real Gabor transform (GT) is presented for non-separable time-frequency (TF) plane sampling. In the proposed implementation, the quincunx sampling is used to sample the TF plane. This leads to a well localized biorthogonal function in both time and frequency. It thus overcomes the main problem of the previous implementations, which is the non-localization of the resultant biorthogonal function. A fast algorithm to compute the derived biorthogonal function is proposed.
ICASSP98 Paper Abstract

Target Classification Near Complex Interfaces Using Time-Frequency Filters

Authors:
Nicole Gache, CPE Lyon, (France)
Patrick Chevret, CPE Lyon, (France)
Veronique Zimpfer, CPE Lyon, (France)

Volume 4, page 2433, paper no. 1583

Abstract:
This paper presents a method for target recognition and classification in shallow water environment. It is based on time-frequency filtering matched to a free field reference target response. The decision strategy lies on the comparison of the reference and the filter output signal. The method is applied to an experimental data base containing target acoustic responses measured in a tank for typical configurations (free field, semi-infinite space and waveguide). First, the recognition of a spherical shell is carried out. The obtained rate of recognition and confusion are more than encouraging. Then, a classification procedure is conducted and a degradation of the mean performances is to be noted in the more general case. However, the classification of 3D targets independently of their attitude gives quite satisfactory results.
ICASSP98 Paper Abstract

An Efficient Array Calibration Method on Underwater High Resolution Direction-Finding

Authors:
Jianfeng Chen, Northwestern Polytechnical University, (China)
Jianguo Huang, Northwestern Polytechnical University, (China)

Volume 4, page 2437, paper no. 1136

Abstract:
In this paper an underwater high-resolution array processing system is described and most of error factors in experiment are analyzed. A novel and practical approach to array calibration is also presented to alleviate the effect of these errors. It is different from other algorithms in that array manifold used in subspace-based methods is obtained through automatical measurement instead of using theoretical value. It can efficiently calibrate errors of gain, phase, mutual-coupling, location, and other reasons generated by the array and sensors even they are direction dependent. Spatial smoothing technique is employed and so the method is effective no matter the sources are correlated or uncorrelated. It is also proved by the test that spatial smoothing is beneficial to reduce some kinds of error. At last, several experimental results are provided to verify the efficiency of the new method.
ICASSP98 Paper Abstract

Robust Matched-Field Processing in Uncertain Shallow-Water Environments using an Lp-Norm Estimator

Authors:
Brian F Harrison, Naval Undersea Warfare Center, (U.S.A.)
Janet I Harrison, Naval Undersea Warfare Center, (U.S.A.)

Volume 4, page 2441, paper no. 1225

Abstract:
An optimal approach to matched-field source localization in the presence of environmental uncertainties is the maximum a posteriori (MAP) estimator. The MAP estimator can be interpreted as an exponentially-weighted average over environmental realizations. In practice, only a finite number of environmental realizations can be included in this average resulting in a suboptimal processor. In this paper, we propose an Lp-norm estimator as a robust alternative to MAP in the presence of finite environmental sampling. We also show, using wavenumber gradients, that accurate localization estimates can be obtained using environmental realizations besides the precise true. Simulation results from a shallow-water environment are presented to illustrate the performance improvement.
Comparison of the Theoretical Performance Bounds for Two Wavefront Curvature Ranging Techniques

Authors:
Anton J Haug, *The MITRE Corporation*, (U.S.A.)
Gary M Jacyna, *The MITRE Corporation*, (U.S.A.)

Volume 4, page 2445, paper no. 1051

Abstract:
The Range Focused Beamformer (RFB) and the Triple Aperture Crosscorrelator (TAC) are the two primary wavefront curvature ranging techniques used in fielded sonar systems. Theoretical performance bounds have been presented in the past for both approaches. This paper develops unified array processing performance bounds where the RFB and the TAC are special cases. Specifically, general Cramer-Rao Lower Bounds (CRLB) on range and bearing estimation for a linear array of directional elements are developed, where the CRLB for the RFB and the TAC are shown to be special cases of the general theory. The ranging performance of the two techniques are then compared.
ICASSP98 Paper Abstract

An Adaptive Beamforming Algorithm for Broadband Active Sonar

Authors:
John M Bourdelais, GTE, (U.S.A.)

Volume 4, page 2449, paper no. 1840

Abstract:
An adaptive beamforming algorithm was developed for broadband active sonar with a convergence time on the order of the pulse duration and effective interference nulling capabilities which enhance desirable echoes. The algorithm is an element based time domain implementation in which beam data that is formed in the direction of each interferer is successively subtracted from the element data using an adaptive FIR filter. The algorithm was applied to impulsive source sonar data received on a bottomed hydrophone array. Collected data contained interfering active returns and noise from nearby shipping as well as desirable echoes from passive reflectors placed near the array. In a representative example, the algorithm adapted fast enough to null out active interference as well as shipping noise, which enhanced the signal to noise ratio of a passive reflector echo by 6 dB.
ICASSP98 Paper Abstract

Two-Stage Kalman Estimator Using Advanced Circular Prediction for Maneuvering Target Tracking

Authors:
Tetsuya Kawase, Keio University, (Japan)
Hideshi Tsurunosono, Keio University, (Japan)
Naoki Ehara, Keio University, (Japan)
Iwao Sasase, Keio University, (Japan)

Volume 4, page 2453, paper no. 1049

Abstract:
Maneuvering targets are difficult to track for the Kalman filter since the target model of tracking filter might not fit the real target trajectory and the statistical characteristic of the target maneuver are unknown in advance. In order to track such a heavy maneuvering target, the estimation of the target turn-direction is necessary. The two-stage estimator using advanced circular prediction which considers the target turn-direction is proposed for maneuvering target tracking. Simulation results are given for a comparison of the performances of our proposed scheme with that of conventional tracking filters.
ICASSP98 Paper Abstract
Target Tracking Using Fuzzy Logic Association

Authors:
Howard A Lazoff, BBN Technologies, (U.S.A.)

Volume 4, page 2457, paper no. 1752

Abstract:
Multistatic active sonar systems are being developed for the mission of tracking enemy submarines in harsh underwater environments. This paper will focus on a revolutionary new method with which to track a target. A fuzzy logic based tracker has been developed which performs remarkably better than a Kalman filter tracker. This is accomplished through the use of additional, non-kinematic information about each of the detections. The fuzzy logic engine combines these additional clues with the kinematic clues. This framework allows for the implementation of user-designed rules which take advantage of the additional information that is provided as input to the tracker. This additional information, effectively handled, yields a significant improvement in tracker performance.
ICASSP98 Paper Abstract

Bottom Backscattering Coefficient Estimation from Wide-band Chirp Sonar Echoes by Chirp Adapted Time-Frequency Representation

Authors:
Ning Ma, Chinese University of Hong Kong, (Hong Kong)
Didier Vray, CREATIS, INSA, (France)

Abstract:
This work is concerned with the estimation of the bottom backscattering coefficient as a function of frequency and of incident angle which is important for bottom imaging with wideband sonar. Generally, the backscattering coefficient is studied for a given frequency and a fixed angle. Meanwhile the backscattering depends not only on the angle, but also on the frequency, the conventional methods cannot be used for the wideband sonar signals analysis. In our work, a wideband chirp sonar (20-140kHz) with -3dB half-angle 12 degree has been used to collect the lacustrine bottom echoes, so the dependence of the backscattering coefficient on the frequency and the incident angle is studied at the same time. An angle-frequency representation is obtained by using the Chirp Adapted time-frequency representation which gives an approximate energy distribution for chirp signals. The proposed method is used for the sand bottom and the pebble bottom echo analysis.
ICASSP98 Paper Abstract

Modeling of Active Reverberation by Time Delay Estimation

Authors:
Robert B. MacLeod, Naval Undersea Warfare Center, (U.S.A.)

Abstract:
This paper explores the statistical properties of underwater reverberation present in active sonar systems. The interference to signal processing which results from reverberation can be extensive, and is particularly acute when the boundaries (surface, bottom) of the water column are nearby. Of particular interest are situations where there may be weak targets masked by reverberation dominating the returning signal. The reverberation will be represented as the output of a linear system with the transmitted signal as an input. The random nature of the reverberation will be accounted for by using random parameters in the linear model, the most important of which are those parameters impacting the spatial distribution of the reverberation. Time delay estimation will be used to analyze reverberant signals obtained from a sonar system operating in a shallow water environment. The statistics of the linear models obtained from these analyses will be computed and discussed.
ICASSP98 Paper Abstract
Long-Range Propagation of a Noise Signal: Arctic Ocean Acoustic Monitoring

Authors:
Vladimir M. Kudryashov, N.N.Andreev Acoustic Institute, (Russia)
Leonid S. Vilentchik, MCB, (Russia)

Volume 4, page 2469, paper no. 1466

Abstract:
Propagation of acoustic signals along a range-dependent track in the Arctic Basin is considered. The structure of the envelope of the temporal correlation function of a narrowband noise signal propagation in a waveguide is investigated. The form of a model pulse is compared with the wave form obtained in acoustic monitoring in the Arctic Basin. Calculations were carried out with a program based on the coupled mode method. This program takes into account the hydroacoustic waveguide parameters affecting the amplitude-time signal structure. This allowance yielded a satisfactory agreement between calculations and the experiment. We considered a method, which operates when the signal-to-noise ratio is less than unity.
ICASSP98 Paper Abstract
Efficient Super Resolution Time Delay Estimation Techniques

Authors:
Jian Li, University of Florida, (U.S.A.)
Renbiao Wu, University of Florida, (U.S.A.)
Zheng-She Liu, University of Florida, (U.S.A.)

Abstract:
In this paper, an efficient Weighted Fourier transform and Relaxation based algorithm (referred to as WRELAX) is first proposed for the well-known time delay estimation problem. WRELAX involves only a sequence of weighted Fourier transforms. Its resolution is much higher than that of the conventional matched filter approach. One disadvantage associated with WRELAX is that it converges slowly when the signals are spaced very closely. To overcome this problem, the well-known high resolution MODE (Method of Direction Estimation) algorithm, which was originally proposed for angle estimation in array processing, is modified and used with WRELAX for super resolution time delay estimation. The latter method is referred to as MODE-WRELAX. MODE-WRELAX provides better accuracy than MODE and higher resolution than WRELAX. Moreover, it applies to both complex- and real-valued signals (including those with highly oscillatory correlation functions). Numerical results show that the MODE-WRELAX estimates can approach the corresponding the Cramer-Rao bounds.
ICASSP98 Paper Abstract

Iterative Gram-Schmidt Orthonormalization for Efficient Parameter Estimation

Authors:
Hongya Ge, New Jersey Institute of Technology, (U.S.A.)

Volume 4, page 2477, paper no. 2401

Abstract:
We present an efficient method for estimating non-linearly entered parameters of a linear signal model corrupted by additive noise. The method uses the Gram-Schmidt orthonormalization procedure in combination with a number of iterations to de-bias and re-balance the coupling between non-orthogonal signal components efficiently. Projection interpretation is provided as rationale of the proposed iterative algorithm. Computer simulations are conducted to show the effectiveness of the algorithm.
ICASSP98 Paper Abstract

The Performance of Maximum Likelihood Over-the-Horizon Radar Coordinate Registration

Authors:
Richard H Anderson, Duke University, (U.S.A.)
Jeffrey L Krolik, Duke University, (U.S.A.)

Volume 4, page 2481, paper no. 1835

Abstract:
A well-known source of target localization errors in over-the-horizon radar is the uncertainty about downrange ionospheric conditions. Maximum likelihood (ML) coordinate registration, using statistical modeling of ionospheric parameters, has recently been proposed as a method which is robust to ionospheric variability. This paper reports ML performance results for real data from a known target using estimates of ionospheric statistics derived from ionosonde measurements. Bootstrap samples derived from these statistics are then used in a hidden Markov model approximation to the ground range likelihood function. Comparison of the ML and conventional methods for over 250 radar dwells indicates the new technique achieves better than a factor of two improvement in ground range accuracy.
ICASSP98 Paper Abstract
A Matrix-Pencil Approach to Blind Separation of Non-White Signals in White Noise

Authors:
Chunqi Chang, University of Hong Kong, (Hong Kong)
Zhi Ding, Hong Kong University of Science & Technology, (Hong Kong)
Sze Fong Yau, Hong Kong University of Science & Technology, (Hong Kong)
Francis H.Y. Chan, University of Hong Kong, (Hong Kong)

Volume 4, page 2485, paper no. 1680

Abstract:
The problem of blind source separation in additive white noise is an important problem in speech, array and acoustic signal processing. In general this problem requires the use of higher order statistics of the received signals. Nonetheless, many signal sources such as speech with distinct, non-white power spectral densities, second order statistics of the received signal mixture can be exploited for signal separation. While previous approaches often assume that additive noise is absent or that the noise correlation matrix is known, we propose a simple and yet effective signal extraction method for signal source separation under unknown white noise. This new and unbiased signal extractor is derived from the matrix pencil formed between output auto-correlation matrices at different delays. Simulation examples are presented.
Exploitation of Signal Structure in Array-Based Blind Copy and Copy-Aided DF Systems

Authors:
Brian G. Agee, Radix Technologies, (U.S.A.)
Stephen P. Bruzzone, Radix Technologies, (U.S.A.)
Matthew C. Bromberg, Radix Technologies, (U.S.A.)

Abstract:
A general approach to array-based copy and DF of structured communication signals is presented that can substantially outperform conventional techniques, by exploiting additional information about the structure of the signals of interest to the reception system. The techniques are derived from optimal parameter estimation concepts that directly incorporate this additional information into the estimation strategy. The resultant algorithms demonstrate strong theoretical, experimental, and implementation advantages over conventional techniques. Results are demonstrated for separation and DF of co-channel FM, CPFSK, DSB-AM, and burst waveforms.
ICASSP98 Paper Abstract
Using Signal Cancellation for Optimum Beamforming in a Cellular CDMA System

Authors:
Tao Luo, Queen’s University, (Canada)
Steven D Blostein, Queen’s University, (Canada)

Volume 4, page 2493, paper no. 2178

Abstract:
We propose a new algorithm for estimating the interference-plus-noise covariance matrix for beamforming in a cellular CDMA system in a fading channel. The method uses direct PN sequence signal cancellation. We show in theory that our method outperforms that of [1,2] for finite input data. The results, confirmed by simulation, show that we get improved DOA estimates and SINR with lower computational requirements.
ICASSP98 Paper Abstract

A Novel Wavelet-Based Generalized Sidelobe Canceller

Authors:
Yi Chu, National Taiwan Univ of Science & Technology, (Taiwan)
Wen-Hsien Fang, National Taiwan Univ of Science & Technology, (Taiwan)
Shun-Hsyung Chang, National Taiwan Ocean University, (Taiwan)

Volume 4, page 2497, paper no. 1683

Abstract:
This paper presents a novel narrowband adaptive beamformer with the generalized sidelobe canceller (GSC) as the underlying structure. The new beamformer employs the regular M-band wavelet filters in the design of the blocking matrix of the GSC, which, as justified analytically, can indeed block the desired signals as required, provided the wavelet filters have sufficiently high regularity. Additionally, the eigenvalue spreads of the covariance matrices of the blocking matrix outputs, as demonstrated in various scenarios, decrease, thus accelerating the convergence speed of the succeeding least mean squares (LMS) algorithm. Also, the new beamformer belongs to a specific type of partially adaptive beamformers, wherein only a portion of weights is utilized in the adaptive processing. Consequently, the computational complexity is substantially reduced as compared with previous approaches. The issues of choosing the parameters involved for superior performance are addressed as well. Simulation results are furnished to justify this new approach.
ICASSP98 Paper Abstract

Generalized Forward/Backward Subaperture Smoothing Techniques for Sample Starved STAP

Authors:
Unnikrishna S Pillai, Polytechnic University, (U.S.A.)
Younglok Kim, Polytechnic University, (U.S.A.)

Abstract:
A major issue in space-time adaptive processing (STAP) for airborne moving target indicator (MTI) radar is the so-called sample support problem. Often, the available sample support for estimating the interference covariance matrix leads to severe rank deficiency, thereby precluding STAP beamforming based on the direct sample matrix inversion (SMI) method. The intrinsic interference subspace removal (ISR) technique, which is a computationally useful form of diagonally loaded SMI method, can handle this case, although the performance is poor in low sample situations. In this context, new subarray-subpulse schemes using forward and backward data vectors are introduced to overcome the data deficiency problem. It is shown here that multiplicative improvement in data samples can be obtained at the expense of negligible loss in space-time aperture of the steering vector.
ICASSP98 Paper Abstract
Optimal Loading Factor for Minimal Sample Support Space-Time Adaptive Radar

Authors:
Younglok Kim, Polytechnic University, (U.S.A.)
Unnikrishna S Pillai, Polytechnic University, (U.S.A.)

Abstract:
A major issue in space-time adaptive processing (STAP) for airborne moving target indicator (MTI) radar is the so-called sample support problem. Often, the available sample support for estimating the interference covariance matrix leads to severe rank deficiency, thereby precluding STAP beamforming based on the direct sample matrix inversion (SMI) method. The intrinsic interference subspace removal (ISR) technique, which is a computationally and analytically useful form of diagonally loaded SMI method, is derived here. It covers from Hung-Turner Projection (HTP) algorithm to matched filter according to the loading factor. Also the optimum loading factor which gives the maximum signal-to-interference-plus-noise ratio (SINR) is derived here from the viewpoint of singular value decomposition of the covariance matrix. The simulation results with synthetic data show that the maximum SINR indeed coincides with the proposed optimum loading factor in various data sample situations.
ICASSP98 Paper Abstract
Spatio-Temporal Coding for Radar Array Processing

Authors:
Philippe Calvary, Thomson-CSF, (France)
Denis Janer, Thomson-CSF, (France)

Volume 4, page 2509, paper no. 1543

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 a single receiver channel. The major advantage is the design of a low cost system providing DBF capabilities.
An Underwater Target Classification Scheme Based on the Acoustic Backscatter Form Function

Authors:
Saman S Abeysekera, Curtin University of Technology, (Australia)
Prabhakar S Naidu, Institute of Science, Bangalore, (India)
Yee-Hong Leung, Curtin University of Technology, (Australia)
Henry Lew, Defence Science and Technology Organisation, (Australia)

Abstract:
By using the acoustic scattering form function, a method for classifying underwater spherical shell targets in an open ocean environment is proposed. The resulting backscatter signal from an incident wideband signal is used to illustrate some of the salient scattering features such as mid-frequency enhancement (MFE). A shell classification technique is then developed. An affine transformation of a template form function is used in the classification scheme. It is shown that the proposed scheme is robust against uncertainties in the material properties and noise.
ICASSP98 Paper Abstract

Post Processing of Sonar Imagery Using Recursive High Order Correlation Method

Authors:
Mahmood R. Azimi-Sadjadi, Colorado State University, (U.S.A.)
Chunhua Yuan, Colorado State University, (U.S.A.)
JoEllen Wilbur, Coastal Systems Stations, (U.S.A.)
Robert J. McDonald, Coastal Systems Stations, (U.S.A.)

Volume 4, page 2517, paper no. 2340

Abstract:
In processing of sonar data, beamforming process plays a central role in reducing the effects of the surface and bottom reverberation. In shallow water environments where the reverberation is dominant, target detection from the beamformed results is not effective and may lead to significantly high false alarm rate. This paper presents a novel approach for post-processing sonar beamformed imagery in order to improve the detectability of the targets while substantially reducing the occurrence of the false detection. This is done using the recursive high order correlation (RHOC) method which exploits the spatial-temporal correlation between consecutive pings of the beamformed images. Test results on several sets of sonar data show the great efficiency and power of the proposed method especially in very high cluttered environment.
ICASSP98 Paper Abstract
Chirp Sounding the Shallow Water Acoustic Channel

Authors:
Gareth J Cook, University of Western Australia, (Australia)
Anthony Zaknich, University of Western Australia, (Australia)

Volume 4, page 2521, paper no. 1453

Abstract:
Characterisation of the shallow water acoustic communications channel involves the analysis of sounding data. Chirp signals have many properties which make them an attractive choice for channel sounding. They are easily generated and channel responses can be processed in the time or frequency domain for channel estimation. In the rapidly varying shallow water environment time domain techniques are most appropriate. In this case weighting windows can be used to reduce clutter in the estimate. A channel sounding experiment is described which employs very simple hardware to generate and record chirp responses for offline processing.
ICASSP98 Paper Abstract

Adaptive Separation of Unknown Narrowband and Broadband Time Series

Authors:
Ivars P Kirsteins, Naval Undersea Warfare Center, (U.S.A.)
Sanjay K Mehta, Naval Undersea Warfare Center, (U.S.A.)
John Fay, Naval Undersea Warfare Center, (U.S.A.)

Volume 4, page 2525, paper no. 1744

Abstract:
Motivated by Thomson’s multiple taper spectral estimation technique, we derive a new, robust procedure for automatically separating time series data into its constituent narrowband and broadband components. The new procedure avoids the pitfalls of adaptive notch filters, PCI method, or other similar algorithms, of mistaking and filtering local spectral peaks of the broadband component as narrowband components by decomposing the data vector into local subbands by a bank of matrix filters. Then in piecewise fashion, the narrowband components are estimated and filtered from each subband using the Principal Component Inverse method. Finally, the filtered components are coherently recombined to obtain the narrowband and broadband time series estimates. Computer simulation results show that the new procedure works well and can have performance close to the clairvoyant Wiener filter.
ICASSP98 Paper Abstract
A New Sequential Detector for Short Duration Signals

Authors:
Peter K Willett, University of Connecticut, (U.S.A.)
Biao Chen, University of Connecticut, (U.S.A.)

Volume 4, page 2529, paper no. 1292

Abstract:
For quickest detection of a permanent change in distribution of otherwise iid observations, Page's test provides the optimal processor. Page's test has also been applied to the detection of transient (i.e. temporary) changes in distribution; it is easy to implement and has reliable performance, but as applied to the transient problem its optimality is questionable. In this paper we offer an alternative to the Page procedure which we call the iterated generalized sequential probability ratio test, or IGSPRT. While Page's test is itself an IGSPRT, its form and performance are constrained by its reliance on constant thresholds and biases. We demonstrate that with these time-varying, markedly increased detection probabilities are possible. The IGSPRT is easiest to understand and motivate in the Gaussian shift-in-mean problem, and we discuss this in detail; but since that problem is of limited practical interest, we also examine the effect of the IGSPRT in a more realistic situation.
ICASSP98 Paper Abstract

Multi-Dwell Matched-Field Altitude Estimation for Over-The-Horizon Radar

Authors:
Michael Papazoglou, Duke University, (U.S.A.)
Jeffrey L. Krolik, Duke University, (U.S.A.)

Abstract:
In previous work, electromagnetic matched-field processing was proposed for estimating aircraft altitude with over-the-horizon radar using a single radar dwell. Although this approach exploits the altitude dependence of unresolved multipath returns in complex delay-Doppler space, its performance suffers in situations where the coherent integration time (CIT) of the radar is short. To overcome this limitation, this paper presents a matched-field estimation approach which uses multiple consecutive dwells on the target. The technique exploits the altitude dependence of dwell-to-dwell shape changes in the complex delay-Doppler multipath return. Monte Carlo simulations results indicate that using short CIT’s, moderate signal bandwidth, and a 30 second revisit rate, multi-dwell matched-field estimation can achieve better than 5,000 ft. accuracy after as few as four radar dwells. The results of processing actual radar data for a high flying commercial aircraft of known altitude are presented which serve to validate the technique.
ICASSP98 Paper Abstract

Direction Finding for Unstructured Emitters in the Presence of Structured Interferers

Authors:
Matthew C. Bromberg, Radix Technologies, (U.S.A.)
Brian G. Agee, Radix Technologies, (U.S.A.)

Volume 4, page 2537, paper no. 2415

Abstract:
This paper addresses the problem of direction finding for unstructured emitters in environments that contain interference signals with exploitable properties. The CA-JML algorithm, presented here, offers dramatic angle estimation improvement in environments that contain exploitable interference. The performance improvement is equivalent to the removal of the interferers from the environment. The complexity of the algorithm, however, is comparable to MUSIC. Even when structured interferers are not present, the CA-JML algorithm can reduce the angle error bias exhibited in MUSIC in challenging environments. The CA-JML algorithm also admits a simple relaxation technique as each emitter is localized, which reduces the probability of angle estimation error due to the presence of ambiguous peaks in the angle objective function.
ICASSP98 Paper Abstract
A Least-Squares Approach to Joint Schur Decomposition

Authors:
Karim Abed-Meraiam, University of Melbourne, (Australia)
Yingbo Hua, University of Melbourne, (Australia)

Volume 4, page 2541, paper no. 1327

Abstract:
We address the problem of joint Schur decomposition (JSD) of several matrices. This problem is of great importance for many signal processing applications such as sonar, biomedicine, and mobile communications. We first present a least-squares (LS) approach for computing the JSD. The LS approach is shown to coincide with that proposed intuitively by Haardt et al, thus establishing the optimality of their criterion in the least-squares sense. Following the LS criterion, we then propose new Jacobi-like algorithms that extend and improve the existing JSD algorithms. An application of the new JSD algorithms to multidimensional harmonic retrieval is also presented.
ICASSP98 Paper Abstract

A Lossless Image Coder with Context Classification, Adaptive Prediction and Adaptive Entropy Coding

Authors:
Farshid Golchin, Griffith University, (Australia)
Kuldip K. Paliwal, Griffith University, (Australia)

Volume 5, page 2545, paper no. 1279

Abstract:
In this paper, we combine a context classification scheme with adaptive prediction and entropy coding to produce an adaptive lossless image coder. In this coder, we maximize the benefits of adaptivity using both adaptive prediction and entropy coding. The adaptive prediction is closely tied with the classification of contexts within the image. Contexts are defined with respect to the local edge, texture or gradient characteristics as well as local activity within small blocks of the image. For each context an optimal predictor is found which is used for the prediction of all pixels belonging to that particular context. A Clustering algorithm is used to design an optimal entropy coding scheme for the prediction residual. The combination of these two adaptive techniques produces some of the best lossless coding results reported so far.
ICASSP98 Paper Abstract

A New Approach for Reducing Blockiness in DCT Image Coders

Authors:
Stephen A Martucci, Scitex Digital Video, (U.S.A.)

Abstract:
This paper presents a new approach for reducing the blockiness that occurs when using DCT image coders at high compression ratios. The method is simply the replacement of the inverse DCT-2 in the decoder by a larger inverse DCT-1 followed by overlapping and averaging of the enlarged blocks to reconstruct the image. The modified decoder can decode any bitstream generated by a standard encoder. Blockiness is reduced but there is no noticeable distortion or loss of sharpness in the image. There is also no significant increase in complexity when using this method.
ICASSP98 Paper Abstract

Stack-Run-End Compression for Low Bit Rate Color Image Communication

Authors:
Min-Jen Tsai, National Chiao-Tung University, (Taiwan)

Abstract:
A new wavelet image coding algorithm was designed for color image compression in this paper. This algorithm utilizes multi-ary symbol set to represent the meaningful coefficients in the wavelet transform domain which are necessary for the image reconstruction in the respective color channel. The scheme works first by color space conversion, followed by raster scanning the individual subband for data conversion to symbol representation. Adaptive arithmetic coder is then used to compress the symbols with high efficiency. Unlike zerotree coding or its variations which are essentially the intersubband coding approach with the complexity in addressing the location relationship across the subbands, this work is a low complexity intrasubband based coding method with context specification within the subband, and termination symbol across subbands. Compared with the zerotree refined schemes, this algorithm results in competitive PSNR values and perceptually high quality images at the same compression ratio for color image compression.
ICASSP98 Paper Abstract

DC Coefficient Restoration Using MAP Estimation Technique

Authors:
Fu-wing Tse, The Chinese University of Hong Kong, (Hong Kong)
Wai-Kuen Cham, The Chinese University of Hong Kong, (Hong Kong)

Abstract:
DC coefficient restoration scheme is a technique which can be used to increase the compression ability of transform image coding by not transmitting the DC coefficients but estimating them from the transmitted AC component. In the last decade, the minimum edge difference criterion is used in the scheme. However the criterion fails at the locations where the discontinuities are along the block boundaries and therefore results in observable blocking effect around these locations or higher bit rate. In this paper, we propose a new criterion using the maximum a posterior (MAP) estimation technique which preserves the discontinuities during the DC coefficients restoration and solves the blocking effects in the restored images.
ICASSP98 Paper Abstract

Model-Based Edge Reconstruction for Low Bit-Rate Wavelet-Based Image Coding

Authors:
G.L. Fan, The Chinese University of Hong Kong, (Hong Kong)
Wai-Kuen Cham, The Chinese University of Hong Kong, (Hong Kong)
J.Z. Liu, The Chinese University of Hong Kong, (Hong Kong)

Abstract:
Low bit-rate image coding brings about obvious degradation to the compressed images, among which distortions at edges are particular objectionable. In this paper, a model-based edge reconstruction algorithm is proposed for wavelet-based image coding at low bit-rate. Our approach applies a general model to represent varieties of edges existing in an image. Based on this model, the problem of edge reconstruction is formulated as finding original edge model parameters from the lossy image. The proposed method is able to improve the subjective visual quality and fidelity (PSNR) of images coded by wavelet-based coding using zerotree quantization. Our algorithm can also be adapted to other wavelet-based coding methods which have the same quantization results as the zerotree quantization. Experimental results show that it performs well for most images with notable structures. Our approach is promising in stretching the performance of wavelet-based coding at low bit-rate. More demonstrations of edge reconstruction results can be found at http://www.ee.cuhk.edu.hk/glfan/icassp98.html
ICASSP98 Paper Abstract

Compression Algorithms for Classification of Remotely Sensed Images

Authors:
Frank Tintrup, University of Cagliari, (Italy)
Francesco G.B. De Natale, University of Cagliari, (Italy)
Daniele D. Giusto, University of Cagliari, (Italy)

Volume 5, page 2565, paper no. 2484

Abstract:
The paper presents a comparison of the principal lossy compression algorithms, Vector Quantization (VQ), JPEG and Wavelets (WV) posterior KLT applied to multispectral remotely sensed images and evaluated by the classification algorithm K-NN. The main goal of the compression of remotely sensed images is a reduction of the huge requirements for downlink and storage. The Karhunen Loeve Transform first removes the interband correlation to produce the principal components of the image which are then compressed by the principal algorithms. The quality evaluation was done by a supervised classification with the well known algorithm K-NN for remote sensing applications and the MSE for visual aspects. The obtained results of these accurate and particular analysis of the current compression techniques are quite surprisingly compared to other recent works.
ICASSP98 Paper Abstract

Locally-adaptive Image Coding based on a Perceptual Target Distortion

Authors:
Ingo Hontsch, Arizona State University, (U.S.A.)
Lina J Karam, Arizona State University, (U.S.A.)

Volume 5, page 2569, paper no. 2532

Abstract:
This paper presents a perceptual-based image coder, which discriminates between image components based on their perceptual relevance for achieving increased performance in terms of quality and bit-rate. The new coder uses a locally-adaptive perceptual quantization scheme based on a tractable perceptual distortion metric. Our strategy is to exploit human visual masking properties by deriving visual masking thresholds in a locally-adaptive fashion. The derived masking thresholds are used in controlling the quantization stage by adapting the quantizer reconstruction levels in order to meet a desired target perceptual distortion. The proposed coding scheme is flexible in that it works with any subband-based decomposition and with block-based transform methods. Compared to the existing perceptual transform-based and block-based methods, the proposed perceptual coding method exhibits superior performance in terms of bit rate and distortion control. Coding results are presented to illustrate the performance of the presented coding scheme.
A Non Uniform Segmentation Optimal Hybrid Fractal/DCT Image Compression Algorithm

Abstract:
In this paper a hybrid fractal and Discrete Cosine Transform (DCT) coder is developed. Drawing on the ability of DCT to remove inter-pixel redundancies and on the ability of fractal transforms to capitalize on long-range correlations in the image, the hybrid coder performs an optimal, in the Rate-Distortion sense, bit allocation among coding parameters. An orthogonal basis framework is used within which an image segmentation and a hybrid block-based transform are selected jointly. A Lagrangian multiplier approach is used to optimize the hybrid parameters and the segmentation. Differential encoding of the DC coefficient is employed, with the scanning path based on a 3rd-order Hilbert curve. Simulation results show a significant improvement in quality with respect to the JPEG standard.
ICASSP98 Paper Abstract
Orthogonal Subspace Projection Filtering for Stereo Image Compression

Authors:
Sang-Hoon Seo, Colorado State University, (U.S.A.)
Mahmood R. Azimi-Sadjadi, Colorado State University, (U.S.A.)

Volume 5, page 2577, paper no. 2354

Abstract:
This paper presents a 2-D filtering scheme for stereo image compression using orthogonal subspace projection. To provide more candidate blocks for input data, the support region for input data is extended in the reference image. In addition, edge blocks are added to the candidate input blocks in order to provide better compensation ability for edges and boundaries of objects. The best blocks for input data are selected one by one in order of importance to reconstruct the desired block using Gram-Schmidt orthogonalization algorithm. Simulation results exhibit excellent performance of the proposed scheme when compared to those of the standard block-matching and least-squares (LS)-based 2-D filtering schemes.
ICASSP98 Paper Abstract

Finding a Suitable Wavelet for Image Compression Applications

Authors:
Shahid Masud, The Queens University of Belfast, (Northern Ireland)
John V. McCanny, The Queens University of Belfast, (Northern Ireland)

Volume 5, page 2581, paper no. 1056

Abstract:
In this paper we assess the relative merits of various types of wavelet functions for use in a wide range of image compression scenarios. We have delineated different algorithmic criteria that can be used for wavelet evaluation. The assessment undertaken includes both algorithmic aspects (fidelity, perceptual quality) as well as suitability for real time implementation in hardware. The results obtained indicate that of the wavelets studied the biorthogonal 9&7 taps wavelet is most suitable from a compression perspective and that the Daubechies 8 taps gives best performance when assessed solely in terms of statistical measures.
ICASSP98 Paper Abstract

Representation and Estimation of Motion Using a Dictionary of Models

Authors:
Daniel Lauzon, INRS-Telecommunications, (Canada)
Eric Dubois, INRS-Telecommunications, (Canada)

Volume 5, page 2585, paper no. 1848

Abstract:
This paper presents a novel method for representing motion information based on a Dictionary of Motion Models and a Tag Image which indicates which motion model is used at any given image position. Each model is composed of low-order polynomial-based motion fields. The motion in most sequences can be adequately represented by a very small number of such motion models. We further present an efficient way of estimating and coding this representation. Comparative results are presented which indicate a performance superior to that of motion representations found in classical block-based codecs.
ICASSP98 Paper Abstract

Global Motion Estimation and Robust Regression for Video Coding

Authors:
Kui Zhang, CVSSP, University of Surrey, (U.K.)
Josef Kittler, CVSSP, University of Surrey, (U.K.)

Volume 5, page 2589, paper no. 2367

Abstract:
In the H.263 Version 2 (H.263+) coding standard, the global motion compensation can be introduced by using Reference Picture Resampling (Annex P) syntax. Such an application requires that the global motion parameters be estimated automatically. In this paper, we propose a global motion estimation algorithm based on the Taylor Expansion Equation and robust regression technique using probabilistic thresholding. The experimental results confirm that the proposed algorithm can improve both coding efficiency and the quality of motion compensation on sequences involving camera movement.
ICASSP98 Paper Abstract

Parametric Motion Modeling Based on Trilinear Constraints for Object-Based Video Compression

Authors:
Zhaohui Sun, University of Rochester, (U.S.A.)
A. Murat Tekalp, University of Rochester, (U.S.A.)

Volume 5, page 2593, paper no. 1317

Abstract:
We propose a new parametric motion model based on the so-called “trifocal tensor” representation, which captures rigid 3D motion of static scenes with a depth of field. The proposed parametric representation, called the trilinear model, is superior to other forms such as translational, affine, perspective, and bilinear models, because it can implicitly encode the depth of the scene and 3D motion of the scene/camera under perspective projection unlike others. A video object can thus be represented by its first VOP, a set of trifocal sensors and the corresponding prediction residues. Motion estimation and compensation based on the new parametric model are incorporated into the MPEG-4 Video Verification Model to compare its efficacy for object-based video compression with the state-of-the-art motion compensation methods. Experimental results are provided to demonstrate the performance of the trilinear model for object-based video compression.
ICASSP98 Paper Abstract

Adaptive Thresholding for Detection of Nonsignificant Vectors in Noisy Image Sequences

Authors:
Luc Martel, Laval University, (Canada)
Andre Zaccarin, Laval University, (Canada)

Volume 5, page 2597, paper no. 2498

Abstract:
In noisy image sequences, block matching motion estimation generates erroneous motion vectors since the algorithm tries to correlate noise. We present an adaptive threshold test to detect blocks for which only nonsignificant motion vectors can be estimated. Vectors of these blocks are then assigned the zero vector before any block motion estimation is performed. By nonsignificant, we refer to motion vectors of non moving areas as well as vectors of moving areas for which the noise level is too high to allow a good estimation of the motion. The detection of these vectors reduces the computational complexity of the BMA and the entropy of the motion field. The algorithm is embedded in a hierarchical BMA and takes advantage of their different spectral characteristics to discriminate between the frame difference energy due to noise and due to motion. The algorithm is also efficient for low noise sequences where it can be used to initialize a segmentation of moving objects from the background.
Computing Optical Flow for Motion Images Sequences by Applying Multi-constraints to Multi-locations

Authors:
Chunke S Yang, University of Tokushima, (Japan)
Shunitiro Oe, University of Tokushima, (Japan)

Abstract:
Computing optical flow is one of the most fundamental problem to the motion image analysis. Many methods have been proposed for computing optical flow, among them gradient-based methods are the most well-known and most used. In the paper, a new gradient-based method for the computation of optical flow was proposed. In this method, optical flow was computed by minimizing a weighted least-squares error estimator for a constant motion vector model in a local spatial neighborhood, where the weight of each image location in the neighborhood was determined by its multiple constraints. Several experiments on real and synthetic image sequences have been carried out to verify the efficacy and the reliability of the new method.
ICASSP98 Paper Abstract
Active Mesh Reconstruction of Block-Based Motion Information

Authors:
Xavier Marichal, Lab de Telecommunication & Teledetection, (Belgium)
Benoît Macq, Lab de Telecommunication & Teledetection, (Belgium)

Volume 5, page 2605, paper no. 1976

Abstract:
This paper proposes an asymmetric scheme for motion estimation/compensation. While the estimation is performed with a classical Block Matching Algorithm, the motion information is decoded by using an active mesh in order to implement the compensation stage. A mesh is positioned by taking into account the relevant spatial information of the image to be compensated and is used afterwards to reconstruct the motion information. Two main issues have to be addressed for conducting such a motion compensation technique: i) how to optimally design an active mesh, ii) how to reverse and interpolate a backward motion field estimated on an a priori grid of fixed-size blocks so as to determine a forward motion field on variable size triangular patches. While proposing a solution to these two problems, particular attention is paid to the computational burden. Such a scheme opens the possibility for added manipulation functionalities because of mesh capabilities.
ICASSP98 Paper Abstract

Automatic Fitting and Tracking of Facial Features in Head-And-Shoulders Sequences

Authors:
Paul M Antoszczyszyn, The University of Edinburgh, Scotland, (U.K.)
John M. Hannah, The University of Edinburgh, Scotland, (U.K.)
Peter M. Grant, The University of Edinburgh, Scotland, (U.K.)

Volume 5, page 2609, paper no. 1027

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'). A method of automatic semantic wire-frame fitting and tracking based on principal component analysis using an independent reference data-base of facial images is presented. The method has been tested on widely used 'head-and-shoulders' video sequences with very good results. It was possible to accurately retrieve the position of the desired facial features in all cases. The position of the facial features in initial frames was subsequently used in automatic tracking. Experimental results are presented as a part of this contribution. Compressed movies illustrating these results can be viewed from our Internet site http://www.ee.ed.ac.uk/plma/.
ICASSP98 Paper Abstract

Fast Rate-Constrained N-Step Search Algorithm for Motion Estimation

Authors:
Muhammed Z Coban, Georgia Institute of Technology, (U.S.A.)
Russell M. Mersereau, Georgia Institute of Technology, (U.S.A.)

Volume 5, page 2613, paper no. 1306

Abstract:
A fast N-step search algorithm for rate-constrained motion estimation is presented. The motion vectors are selected from a search window based on a rate-distortion criterion by successively eliminating the search positions at each step. The performance of the proposed algorithm is identical to the performance of the conventional rate-constrained N-step search algorithm, with considerable reduction in computation. Computational savings increase in parallel with the increases in the rate constraint and the number of steps.
ICASSP98 Paper Abstract
An Embedded DCT-Based Still Image Coding Algorithm

Authors:
David Nister, *Ericsson Telecom AB, (Sweden)*
Charilaos Christopoulos, *Ericsson Telecom AB, (Sweden)*

Volume 5, page 2617, paper no. 1048

Abstract:
In this paper, an embedded DCT-based image coding algorithm is described. The decoder can cut the bitstream at any point and therefore reconstruct an image at lower rate. The quality of the reconstructed image at this lower rate would be the same as if the image was coded directly at that rate. The algorithm outperforms any other DCT-based coders published in the literature, including the JPEG algorithm. Moreover, our DCT-based embedded image coder gives results close to the best wavelet-based coders. The algorithm is very useful in various applications, like WWW, fast browsing of databases, etc.
ICASSP98 Paper Abstract

On the Trade-Off between Contour-Adaptive Texture Coding and Lossy Shape Coding

Authors:
Karsten Schroeder, University of Dortmund, (Germany)

Volume 5, page 2621, paper no. 5231

Abstract:
Shape-adaptive texture coding is often being perceived as a mean to achieve increased coding efficiency, e.g. due to a better presentation of correlation properties along the boundary between two objects which should lead to a higher transform gain. However, if data rates demand a lossy encoding of contours, an improvement in transform gain becomes questionable. This paper investigates the potential coding gain which may result from shape-adaptive texture coding using DCT basis functions under the constraint of both lossless and lossy shape coding. For this purpose, a 1D texture model is derived with reflects synthetical as well as "natural" borders between objects.
ICASSP98 Paper Abstract

A Fast Encoding Method without Search for Fractal Image Compression

Authors:
In Kwon Kim, Sogang University, (Korea)
Rae Hong Park, Sogang University, (Korea)

Volume 5, page 2625, paper no. 1273

Abstract:
A fast coding algorithm for images using vector quantization (VQ) and pixelwise fractal approximation is proposed. The low frequency component of an input image is approximated and its residual is used to calculate the scaling factor of fractal transform. The scaling factor is compressed by transform VQ (TVQ). In the proposed method, to encode a digital image by an iterated function system (IFS), we use the pixel-based IFS (PIFS) rather than the block-based IFS: the scaling factor is computed for each pixel. In the proposed method, the scaling factor of each pixel is calculated with the constraint of contraction mapping and it is transformed by wavelet and quantized by VQ. For approximation of an original image, the variable block-size segmentation using quadtree is employed. Because the proposed method calculates the scaling factor using the PIFS, the encoding time is faster than the conventional algorithm using block-based IFS with search.
ICASSP98 Paper Abstract

Adaptive-Rate Image Compression for Wireless Digital Data Transmission Systems

Authors:
John E Kleider, Motorola SSTG, (U.S.A.)
Glen P Abousleman, Motorola SSTG, (U.S.A.)

Volume 5, page 2629, paper no. 1765

Abstract:
The vast amount of data needed to represent digital imagery motivates the use of advanced compression systems to reduce the bandwidth required to transmit high-resolution source imagery. We propose two methods to provide optimal image quality at a fixed image delivery rate. The first method, channel-controlled variable-rate (CCVR) image coding, operates within the constraint that the modulation symbol rate is fixed. The second method, adaptive-rate coding-modulation (ARCM), utilizes adaptive modulation, and is less complex, while providing increased performance. Both methods use a variable-compression-ratio image coder and variable-rate channel coding. The objective is to maximize the quality of the reconstructed image at the receiver for Rayleigh fading and AWGN. The ARCM system achieves up to a 17 dB improvement over the peak signal-to-noise ratio performance of a system designed assuming a fixed-compression-ratio image coder and fixed-rate channel coding.
ICASSP98 Paper Abstract
JPEG Compliant Efficient Progressive Image Coding

Authors:
Jaehan In, University of British Columbia, (Canada)
Shahram Shirani, University of British Columbia, (Canada)
Faouzi Kossentini, University of British Columbia, (Canada)

Volume 5, page 2633, paper no. 2414

Abstract:
Among the different modes of operations allowed in the current JPEG standard, the sequential and progressive modes are the most widely used. While the sequential JPEG mode yields essentially the same level of compression performance for most encoder implementations, the performance of progressive JPEG depends highly upon the designed encoder structure. This is due to the flexibility the standard leaves open in designing progressive JPEG encoders. In this paper, a rate-distortion optimized JPEG compliant encoder is presented that produces a sequence of bit scans, ordered in terms of decreasing importance. Our encoder outperforms a baseline JPEG encoder in terms of compression, significantly at medium and high bit rates, and substantially at low bit rates. Moreover, unlike baseline JPEG encoders, ours can achieve precise rate/distortion control. Good rate-distortion performance at low bit rates and precise rate control, provided by our progressive JPEG compliant encoder, are two highly desired features currently sought for JPEG-2000.
A Low Bit Rate Segmented Video Codec with Hybrid Motion Estimation and Inherent Bit Rate Control Capability

Authors:
Vassilios A Christopoulos, Vrije Universiteit Brussel, (Belgium)
Jan Cornelis, Vrije Universiteit Brussel, (Belgium)

Abstract:
In this paper a segmented video codec with hybrid motion estimation and inherent bit rate control capability is presented. The first frame in the data is always encoded in intraframe mode, while the rest of the frames are encoded in interframe mode. The interframe encoding is based on (1) hybrid conventional/perspective block motion vector estimation and coding, and (2) coding of the prediction error (Displaced Frame Difference, “DFD”) using segmented image coding techniques. We present a way to partition the DFD in moving and static regions and we explain how this classification strategy can be used as a means to control the bit rate. The simulation results show that the hybrid motion estimation technique outperforms the conventional full search block-matching method by improving the overall PSNR for reduced bit rate, and that the bit rate control strategy, although simple, is very efficient for monitoring both the bit rate and the quality degradation.
ICASSP98 Paper Abstract

Rank Order Polynomial Decomposition for Image Compression

Authors:
Olivier Egger, Swiss Federal Institute of Technology, (Switzerland)
Reto Grueter, Swiss Federal Institute of Technology, (Switzerland)
Jean-Marc Vesin, Swiss Federal Institute of Technology, (Switzerland)
Murat Kunt, Swiss Federal Institute of Technology, (Switzerland)

Volume 5, page 2641, paper no. 2008

Abstract:
In this paper, a novel decomposition scheme for image compression is presented. It is capable to apply any nonlinear model to compress images in a lossless way. Here, a very efficient polynomial model that considers spatial information as well as order statistic information is introduced. This new rank order polynomial decomposition (ROPD) that allows also for a progressive bitstream is applied to various images of different nature and compared to the morphological subband decomposition (MSD) and to the best prediction mode for lossless compression of the international standard JPEG. For all compressed images, ROPD provides better compression results than MSD and clearly outperforms the lossless mode of JPEG.
ICASSP98 Paper Abstract

Bit Error Prediction for Digital Image Data

Authors:
J. Q Trelewicz, Arizona State University, (U.S.A.)
Douglas Cochran, Arizona State University, (U.S.A.)

Volume 5, page 2645, paper no. 2344

Abstract:
A nonideal two-dimensional optical system, as encountered in digital holographic data storage applications, can modify the intensity of transmitted digital data through beam shaping, focal surface distortion, and moire patterns. Such changes in intensity can have significant adverse effects on digital data recovery at the receiver (e.g., a CCD camera). Current research seeks to detect and correct classes of such distortion so that recovery methods can be applied to the received data. This paper discusses methods used to predict the locations of bit errors in the recovered data. Prediction information may be used as weighting information in the recovery algorithm and in the design of channel codes. Furthermore, the higher the level of distortion that can be tolerated in the system, the lower the cost of the corresponding lenses, making the system more tractable for commercialization.
ICASSP98 Paper Abstract
Joint Compression and Restoration of Images Using Wavelets and Non-Linear Interpolative Vector Quantization

Authors:
Kannan Panchapakesan, University of Arizona, (U.S.A.)
Ali Bilgin, University of Arizona, (U.S.A.)
Michael W Marcellin, University of Arizona, (U.S.A.)
Bobby R Hunt, University of Arizona, (U.S.A.)

Volume 5, page 2649, paper no. 1807

Abstract:
In this paper, we present a wavelet based non-linear interpolative vector quantization scheme for joint compression and restoration of images; two tasks which are traditionally regarded as having conflicting goals. Vector quantizer codebook training is done using a training set consisting of pairs of the original image and its diffraction-limited counterpart. The designed VQ is then used to compress and simultaneously restore diffraction-limited images. Results from simulations indicate that the image produced at the output of the decoder is quantitatively and visually superior to the diffraction-limited image at the input to the encoder. We also compare the performance of several wavelet filters in our algorithm.
ICASSP98 Paper Abstract
Comparative Investigation of A Non-Linear Predictive Codec versus JPEG Lossless Compression

Authors:
Jianmin Jiang, Loughborough University, (U.K.)
Meiying Lo, Loughborough University, (U.K.)

Volume 5, page 2653, paper no. 1070

Abstract:
A non-linear predictive coding based algorithm is proposed in the paper for lossless image compression. The algorithm uses two neighbouring pixels, one left and the other top, as a pioneering block to search for the best matched blocks inside a pre-defined window. The corresponding pixels associated with the best matched blocks are then taken to produce the predictive value, together with the two pioneering pixels. Comparative investigation is carried out by experiments which show clearly that the proposed algorithm constantly outperform JPEG lossless compression mode.
ICASSP98 Paper Abstract

A Generalized Interpolative VQ Method for Jointly Optimal Quantization and Interpolation of Images

Authors:
Faramarz Fekri, Georgia Institute of Technology, (U.S.A.)
Russell M. Mersereau, Georgia Institute of Technology, (U.S.A.)
Ronald W Schafer, Georgia Institute of Technology, (U.S.A.)

Volume 5, page 2657, paper no. 2124

Abstract:
In this paper we discuss the problem of reconstruction of a high resolution image from a lower resolution image by a jointly optimum interpolative vector quantization method. The interpolative vector quantizer maps quantized low dimensional 2x2 image blocks to higher dimensional 4x4 blocks by a table lookup method. As a special case of generalized vector quantization (GVQ), a jointly optimal quantizer and interpolator (GIVQ) is introduced to find the corresponding codebooks for the low and high resolution image. In order to incorporate the nearest neighborhood constraint on the quantizer and also to obtain the desired distortion in the interpolated image, a deterministic annealing based optimization technique has been applied. With a small interpolation example, we will demonstrate the superior performance of this method over nonlinear interpolative vector quantization (NLIVQ), in which the interpolator is optimized for a given input quantizer.
ICASSP98 Paper Abstract

Novel Codebook Generation Algorithms for Vector Quantization Image Compression

Authors:
Konstantinos Masselos, University of Patras, (Greece)
Thanos Stouraitis, University of Patras, (Greece)
Costas E Goutis, University of Patras, (Greece)

Volume 5, page 2661, paper no. 1820

Abstract:
Novel algorithms for vector quantization codebook design are presented in this paper. Two basic techniques are proposed. The first technique takes into consideration specific characteristics of the blocks of the training sequence during the generation of the initial codebook. In this way a representative initial codebook is generated. Starting from a high quality initial codebook the iterative optimization procedure converges fast to a representative final codebook which in turn leads to high output image quality. The second proposed technique extends small codebooks computationally. The main idea is the application of simple transformations on the codewords. This technique reduces the memory requirements of the traditional vector quantization making it useful for applications requiring low-power consumption.
ICASSP98 Paper Abstract

Entropy-Constrained Gradient-Match Vector Quantization for Image Coding

Authors:
Shin-Chou Juan, National Chiao-Tung University, (Taiwan)
Chen-Yi Lee, National Chiao-Tung University, (Taiwan)

Volume 5, page 2665, paper no. 1275

Abstract:
Side-match VQ (SMVQ) is a well-known class of FSVQ used for low-bit rate image/video coding. It exploits the spatial correlation between the neighboring blocks to select several codewords that are very close to the encoding block from the master codebook. But if the block boundary is in the region edge area, the spatial correlations are not high and the SMVQ can't select proper codewords to encode blocks. In this paper, an Entropy-Constrained Gradient-Match VQ (ECGMVQ) is proposed. Instead of exploiting the spatial correlation, the ECGMVQ uses the gradient contiguity property to select the codewords. State function of ECGMVQ can select better codevectors than the SMVQ. In addition, the entropy-constrained rule is applied to the encoding process to reduce bit rate. Simulation results show that the improvement of ECGMVQ over the SMVQ is up to 4.5 dB at nearly the same bit rate. Further, the perceptual image quality is better than that of SMVQ, especially in the region edge area.
ICASSP98 Paper Abstract

Three-Sided Side Match Finite State Vector Quantization

Authors:
Hsien-Chung Wei, National Tsing Hua University, (Taiwan)
Pao-Chin Tsai, National Tsing Hua University, (Taiwan)
Jia-Shung Wang, National Tsing Hua University, (Taiwan)

Volume 5, page 2669, paper no. 1907

Abstract:
Several effective low bit rate still image compression methods have been presented in these two years, such as SPHIT [9], Hybrid VQ [7], Wu and Chen method [10]. These methods exercise the analysis techniques (wavelet or subband) before distributing the bit rate to each piece of image, thus the tradeoff between bit rate and distortion can be resolved. In this paper, we try to propose a simple but comparable method that adopts the technique of side match VQ only. Side match vector quantization (SMVQ) is an efficient VQ coding scheme for low bit rate coding. Conventional side match (two-sided) utilizes the codeword information of two neighboring blocks to predict the state codebook of an input vector. In this paper, we propose a hierarchical three-sided side match finite-state vector quantization (HTSMVQ) method that can (1) make the state codebook size as small as possible, it can be reduced to 1 if the prediction is performed perfectly; (2) improve the prediction quality for edge blocks; (3) regularly refresh the codewords to alleviate the error propagation of side match. In the simulation results, the image "Lena" can be coded with PSNR 34.682 dB at 0.25 bpp. It is better than SPIHT, EZW, FSSQ and hybrid VQ with 34.1, 33.17, 33.1 and 33.7 dB, respectively. At the bit rate lower than 0.15 bpp, only the enhanced versions of EZW perform better than our method about 0.14 dB.
ICASSP98 Paper Abstract

Real Time Low Bit-Rate Video Coding Algorithm Using Multi-Stage Hierarchical Vector Quantization

Authors:
Kazuhiko Terada, Kyoto University, (Japan)
Masahiro Takeuchi, Kyoto University, (Japan)
Kazutoshi Kobayashi, Kyoto University, (Japan)
Keikichi Tamaru, Kyoto University, (Japan)

Volume 5, page 2673, paper no. 2278

Abstract:
In this paper, we propose a low bit-rate coding algorithm for wireless communication based on multi-stage hierarchical vector quantization, motion compensation and differential pulse code modulation. Our method adapts bit allocation to spatial and temporal correlation. Conventional schemes based on discrete cosine transform (DCT) need a large amount of computation on both encoding and decoding. On the other hand, our proposed method consists of addition, subtraction and shift operation. It does not use multiplication. It can decode in real time on a conventional serial processor. Encoding by vector quantization (VQ), however, consumes a large amount of computation. We developed a new LSI to accelerate VQ. Our scheme can send 10 frames of QCIF video sequences through a 29.2kbps line. The quality of reconstructed image is over 30dB.
ICASSP98 Paper Abstract

A Novel Subtree Partitioning Algorithm for Wavelet-Based Fractal Image Coding

Authors:
Lai-Man Po, City University of Hong Kong, (Hong Kong)
Ying Zhang, Guangdong Posts & Telecom, (China)
Kwok-Wai Cheung, City University of Hong Kong, (Hong Kong)
Chun-Ho Cheung, City University of Hong Kong, (Hong Kong)

Volume 5, page 2677, paper no. 1161

Abstract:
In this paper, a novel wavelet subtree partitioning algorithm is proposed, which divides a subtree into scalar quantized wavelet coefficients and fractal coded sub-subtree. Based on this new technique, a variable size wavelet subtree fractal coding scheme for still image compression is developed. Experimental results show that the new scheme can achieve nearly optimal partition of wavelet subtree with substantially computational reduction as compared with Davis’ scheme.
ICASSP98 Paper Abstract
Biorthogonal Modified Coiflet Filters for Image Compression

Authors:
Lowell L Winger, University of Toronto, (Canada)
Anastasios N Venetsanopoulos, University of Toronto, (Canada)

Volume 5, page 2681, paper no. 1265

Abstract:
The selection of filter bank in wavelet compression is crucial, affecting image quality and system design. Recently, the biorthogonal coiflet (coolet) family of wavelet filters has been constructed [2][4], and explicit frequency domain formulae have been developed [2] in the Bernstein polynomial basis. In this paper we use the Bernstein basis for frequency domain design and construction of biorthogonal nearly coiflet wavelet bases. In particular, we construct a previously unpublished nearly coiflet 17/11 biorthogonal wavelet filter pair. Key filter quality evaluation metrics due to Villasenor demonstrate this filter pair to be well suited for image compression. Comparison is made to the 17/11 biorthogonal coiflet (coolet), Villasenor 10/18, Odegard 9/7, and classical CDF 9/7 wavelet bases. Simulation results with the SPIHT algorithm due to Said and Pearlman [3], and with our SRSFQ [7][5], confirm that the new 17/11 wavelet basis outperforms the others for still image compression.
ICASSP98 Paper Abstract
A Flexible Zerotree Coding with Low Entropy

Authors:
Sang-hyun Joo, Niigata University, (Japan)
Hisakazu Kikuchi, Niigata University, (Japan)
Shigenobu Sasaki, Niigata University, (Japan)
Jaeho Shin, Dongguk University, (Korea)

Volume 5, page 2685, paper no. 1719

Abstract:
We introduce a new zerotree scheme that effectively exploits the inter-scale self-similarities found in the octave decomposition by a wavelet transform. A zerotree is useful to code wavelet coefficients and its effectiveness was proved by Shapiro’s EZW. In this paper, we analyze symbols produced from the EZW and discuss the entropy per symbol. Since the entropy depends on the produced symbols, we modify the procedure of symbol generation. First, we extend the relation between a parent and children used in the EZW to increase the probability such that a significant parent has significant children. The proposed relation is flexibly extended according to the fact that a significant coefficient is likely to have significant coefficients in its neighborhood. Our coding results are compared with the published results. Our proposed coder owes its improved performance to the use of lower entropy per symbol. The comparison of the number of produced symbols is also given.
ICASSP98 Paper Abstract

Image Coding Based on Edges and Textures via Wavelet Transform

Authors:
Sergio Rodrigues Neves, Instituto de Pesquisas da Marinha, (Brazil)
Gelson Vieira Mendonça, COPPE/EE/UFRJ, (Brazil)

Volume 5, page 2689, paper no. 1808

Abstract:
The increasing interest on image compression is due to the increasing necessity of transmission velocity and memory to keep digital images, on many kind of midia. In order to reduce the quantity of bits needed to store images and, consequently, increase their transmission velocity, many coding methods have been studied, especially those which utilize transforms. This paper presents an image compression technique that codifies edges and textures separately using the wavelet transform. Several test images have been coded with the proposed method and the results show that the reconstructed images have good performance in relation to their PSNR and, especially, in terms of visual perception.
ICASSP98 Paper Abstract

Joint Optimal Bit Allocation and Best-Basis Selection for Wavelet Packet Trees

Authors:
Jill R. Goldschneider, University of Washington, (U.S.A.)
Eve A. Riskin, University of Washington, (U.S.A.)

Volume 5, page 2693, paper no. 1036

Abstract:
In this paper, an algorithm for wavelet packet trees that can systematically identify all bit allocations/best-basis selections on the lower convex hull of the rate-distortion curve is presented. The algorithm is applied to tree-structured vector quantizers used to code image subbands that result from the wavelet packet decomposition. This method is compared to optimal bit allocation for the discrete wavelet transform.
ICASSP98 Paper Abstract

An Operational Approach to Monitor Vegetation Using Remote Sensing

Authors:
Sonia Bouzidi, INRIA, (France)
Jean-Paul Berroir, INRIA, (France)
Isabelle L Herlin, INRIA, (France)

Abstract:
This paper addresses vegetation monitoring in European agricultural areas using Earth Observation satellites. Due to the small size of typical European fields, two complementary sensors are used, SPOT and NOAA-AVHRR, bringing the spatial and the temporal information respectively. A subpixel analysis of NOAA data using one SPOT image is performed to characterize fields with high spatial and temporal resolutions. To be used in an operational context, the method must have realistic data requirements. We define an operational scenario making use of only one SPOT image per site and a one year NOAA sequence, covering a large part of Europe. We first proceed to an unsupervised segmentation of the SPOT image; the NOAA data analysis on test sites provides the temporal evolution of vegetation; then, identification of fields is performed by minimizing a cost function measuring the similarity between the global reflectance observed on NOAA pixels and the reflectance computed from corresponding regions at SPOT resolution.
ICASSP98 Paper Abstract

Face Extraction from Non-Uniform Background and Recognition in Compressed Domain

Authors:
Nicolas A Tsapatsoulis, National Technical University of Athens, (Greece)
Nikolaos D Doulamis, National Technical University of Athens, (Greece)
Anastasios D Doulamis, National Technical University of Athens, (Greece)
Stefanos D Kollias, National Technical University of Athens, (Greece)

Abstract:
A complete face recognition system is proposed in this paper by introducing the concepts of foreground objects, which are currently used during the MPEG-4 standardization phase, to human identification. The system automatically detects and extracts the human face from the background, even if it is not uniform, based on a combination of a retrainable neural network structure and the morphological sizedistribution technique. In order to combine face images of high quality and low computational complexity, the recognition stage is performed in compressed domain. Thus, in contrast to existing recognition schemes, the face images are available in their original quality and not only in their transformed representation.
ICASSP98 Paper Abstract

An Integral Stochastic Approach to Image Sequence Segmentation and Classification

Authors:
Peter Morguet, Munich University of Technology, (Germany)
Manfred Lang, Munich University of Technology, (Germany)

Volume 5, page 2705, paper no. 1185

Abstract:
Finding and identifying characteristic or meaningful image sequences in a continuous video stream is a challenging task with many applications. This paper presents a new and efficient approach to these temporal segmentation and classification problems based on Hidden Markov Models (HMMs). The basic principle consists in continuously observing the output scores of the HMMs at every time step. Peaks, which appear in the individual HMM output scores, allow to determine in an integral way which image sequence occurred at what time. The application of our method to the spotting of connected dynamic hand gestures provided excellent recognition results and a high temporal accuracy.
ICASSP98 Paper Abstract
Characterization of One-Dimensional Texture – A Point Process Approach

Authors:
Miriam Zacksenhouse, Technion - Institute of Technology, (Israel)
Gil Abramovich, Technion - Institute of Technology, (Israel)
Gad Hetsroni, Technion - Institute of Technology, (Israel)

Volume 5, page 2709, paper no. 1502

Abstract:
The distance between texture primitives is of major interest in characterizing texture images. This is especially natural when the texture primitives are elongated structures aligned in parallel to a common main axis, and the distance is measured along the perpendicular axis. Such images arise, for example, in flow visualization studies, where the elongated structures are low-speed streaks. A point process based texture generation model is developed for the one-dimensional texture along lines perpendicular to the streaks. The point process models the location of the edges of the streaks, and using edge detection techniques, its probability density function (pdf) can be estimated by the histogram of the distances between the edges. It is shown that for the studied images the resulting histogram is wide (coefficient of variation greater than half), and demonstrated that in this case, previously suggested auto-correlation based methods are not adequate.
ICASSP98 Paper Abstract

Entropy-Based Detection of Microcalcifications in Wavelet Space

Authors:
Giuseppe Boccignone, Università di Salerno, (Italy)
Angelo Chianese, Università di Napoli Federico II, (Italy)
Antonio Picariello, Università di Napoli Federico II, (Italy)

Volume 5, page 2713, paper no. 1388

Abstract:
In this paper we present a method for the detection of microcalcifications in digital mammographic images. Our approach is based on the wavelet transform, but differently from other techniques proposed in the literature, the detection is directly accomplished into the wavelet domain and no inverse transform is required. After a preliminary de-noising pass, microcalcifications are separated from background tissue by exploiting information gained through evaluation of Renyi's entropy at the different decomposition levels of the wavelet space. Experimental results achieved on the standard Nijmegen data set are shown and discussed.
ICASSP98 Paper Abstract

Improved Automatic Target Recognition Using Singular Value Decomposition

Authors:
Vijay Bhatnagar, Wright State University, (U.S.A.)
Arnab K. Shaw, Wright State University, (U.S.A.)
Rob W Williams, Wright Patterson AFB, (U.S.A.)

Volume 5, page 2717, paper no. 2570

Abstract:
A new algorithm is presented for Automatic Target Recognition (ATR) where the templates are obtained via Singular Value Decomposition (SVD) of High Range Resolution (HRR) profiles. SVD analysis of a large class of HRR data reveals that the Range-space eigenvectors corresponding to the largest singular value accounts for more than 90% of target energy. Hence, it is proposed that the Range-space eigenvectors be used as templates for classification. The effectiveness of data normalization and Gaussianization of profile data for improved classification performance is also studied. With extensive simulation studies it is shown that the proposed Eigen-template based ATR approach provides consistent superior performance with recognition rate reaching 99.5% for the four class XPATCH database.
ICASSP98 Paper Abstract
Hidden Markov Models for Face Recognition

Authors:
Ara V Nefian, Georgia Institute of Technology, (U.S.A.)
Monson H Hayes III, Georgia Institute of Technology, (U.S.A.)

Volume 5, page 2721, paper no. 1789

Abstract:
The work presented in this paper focuses on the use of Hidden Markov Models for face recognition. A new method based on the extraction of 2D-DCT feature vectors is described, and the recognition results are compared with other face recognition approaches. The method introduced in this paper reduces significantly the computational complexity of previous HMM-based face recognition system, while preserving the same recognition rate.
ICASSP98 Paper Abstract

Texture Characterization Using 2D Cumulant-Based Lattice Adaptive Filtering

Authors:
Mounir Sayadi, ESSTT, (France)
Veronique Buzenac-Settineri, LESTER-UBS, (France)
Mohamed Najim, ESI-ENSERB, (France)

Volume 5, page 2725, paper no. 1994

Abstract:
In this work, we take into account the non gaussian properties of textures and we propose a new approach for their characterization based on bidimensionnal adaptive modelization using high order statistics. The 2D-OLRIV (Bidimensionnal Overdetermined Lattice Recursive Instrumental Variable) algorithm allows accurate texture estimation. Sets of 2D-AR coefficients obtained from the reflection coefficient of the lattice model are used to characterize the texture. This algorithm has the advantage of yielding non biased estimates of the 2D-AR model even when the textured image is disturbed by gaussian noise. A multilayer neural network deals with these coefficients in order to classify different textures. In order to evaluate the performances of this approach, classification sensitivity is evaluated on a set of eight different textures. This characterization approach gives very promising results.
ICASSP98 Paper Abstract

Land Use Classification of SAR Images Using a Type II Local Discriminant Basis for Preprocessing

Authors:
Laura S Rogers, Vexcel Corporation, (U.S.A.)
Carolyn Johnston, Vexcel Corporation, (U.S.A.)

Volume 5, page 2729, paper no. 2087

Abstract:
In this paper, we present the application of the Type II Local Discriminant Basis (LDB) technique to feature extraction for land use classification in Synthetic Aperature Radar (SAR) images. Our classification algorithm incorporates spatial information into the decision process by classifying small image blocks, instead of single pixels. A feature vector composed of all the values in the image blocks is large for even small image blocks and, therefore, degrades the performance of many classifiers. The LDB technique greatly compresses the dimensionality of the feature vector by indicating the most discriminant coordinates within the wavelet packet decomposition of an image block.
ICASSP98 Paper Abstract

Cropland Detection with SAR Interferometry: A Segmentation Model

Authors:
Etienne G Huot, INRIA, (France)
Isabelle L Herlin, INRIA, (France)

Volume 5, page 2733, paper no. 2450

Abstract:
Repeat-pass SAR interferometric data are multitemporal and display changes occurring between two acquisitions. As a consequence, phase and correlation images contain meaningful information usable for cropland monitoring. This paper proposes a statistical model to segment high phasimetric structures. It is expressed in a Markov random field framework by using cooperatively phase and correlation information.
ICASSP98 Paper Abstract

Steerable Filters and Invariant Recognition in Space-time

Authors:
Reiner Lenz, Linkoping University, (Sweden)

Volume 5, page 2737, paper no. 1486

Abstract:
The groups which have received most attention in signal processing research are the affine groups and the Heisenberg-Weyl group related to wavelets and time-frequency methods. In low-level image processing the rotation-groups SO(2) and SO(3) were studied in detail. In this paper we argue that the Lorentz group SO(1; 2) provides a natural framework in the study of dynamic processes like the analysis of image sequences. We summarize the connection between the group SO(1; 2) and the groups SU(1; 1) and SL(2; R) and give an overview over their representations. We show that their representation theory is in parts similar to the corresponding theory for the three-dimensional rotation group. The main differences between the compact groups (like SO(2) and SO(3)) is however that the Fourier transforms for these groups involves infinite-dimensional representations and that the finite-dimensional representations are no longer unitary. In the signal processing context this means that the filter vectors computed by finite-dimensional steerable filter systems no longer transform as unitary vector transformations under the symmetry operations in SO(1; 2).
ICASSP98 Paper Abstract

Statistical Model and Genetic Optimization: Application to Pattern Detection in Sonar Images

Authors:
Max Mignotte, Ecole Navale, (France)
Christophe Collet, Ecole Navale, (France)
Patrick Perez, IRISA/INRIA, (France)
Patrick Bouthemy, IRISA/INRIA, (France)

Volume 5, page 2741, paper no. 1099

Abstract:
We present a new classification method using deformable template model to separate natural objects from man made objects in an image given by a high resolution sonar. A prior knowledge of the manufactured object shadow shape is described by a prototype template and a set of admissible linear transformations to take into account the shape variability. Then, the classification problem is defined as a two step process; firstly the detection problem of a region of interest in the input image is stated in a Bayesian framework and is posed as an equivalent energy minimization problem of an objective function: in this paper, this energy minimization problem is solved by using a hybrid Genetic Algorithm. Secondly, the value of this function at convergence allows to determine the presence of the desired object in the sonar image. This method has been successfully tested on real and synthetic sonar images, yielding promising results.
ICASSP98 Paper Abstract

Target Aspect Estimation from Single and Multi-Pass SAR Images

Authors:
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Rama Chellappa, University of Maryland, (U.S.A.)
Shyam Kuttikkad, Etak Inc., (U.S.A.)

Volume 5, page 2745, paper no. 2106

Abstract:
A technique is presented for estimating the aspect of targets in SAR imagery for use in indexing, feature extraction and recognition. Aspect estimation is enhanced by combining multiple images of the same target. In order to properly combine the estimation of multiple passes, it is necessary to accurately register the images to a common coordinate frame. An algorithm for registering multiple high resolution SAR images, is presented. A global affine transformation derived from the sensor acquisition parameters is used to automatically register the images, followed by a refinement to correct for translational errors. The registered SAR images are used for improving the estimates of target orientation angles, detecting the presence of occlusion and indicating poor target segmentation.
ICASSP98 Paper Abstract
An Edge Detection by Using Self-Organization

Authors:
Hironori Nagai, Hokkaido University, (Japan)
Yoshikazu Miyanaga, Hokkaido University, (Japan)
Koji Tochinai, Hokkaido University, (Japan)

Volume 5, page 2749, paper no. 1676

Abstract:
This paper proposes a self-organized edge detection. In this method, several clusters are yielded and self-organized according to a gray scale level and the location of pixels. In addition, the comparison among these clusters results in estimated edge. However, after self-organization, clusters are classified into some group according to their properties. In this report, the method which represents the detail distribution of each cluster is introduced. In addition, by using this method, it is shown that the proposed detection of edges can improve the accuracy in some experiments.
ICASSP98 Paper Abstract

A New Multiresolution Algorithm for Image Segmentation

Authors:
Mohammed Saeed, MIT, (U.S.A.)
W. C. Karl, Boston University, (U.S.A.)
T. Q. Nguyen, University of Wisconsin, (U.S.A.)
Hamid R. Rabiee, Intel Corporation, (U.S.A.)

Volume 5, page 2753, paper no. 5228

Abstract:
We present here a novel multiresolution-based image segmentation algorithm. The proposed method extends and improves the Gaussian mixture model (GMM) paradigm by incorporating a multiscale correlation model of pixel dependence into the standard approach. In particular, the standard GMM is modified by introducing a multiscale neighborhood clique that incorporates the correlation between pixels in space and scale. We modify the log likelihood function of the image field by a penalization term that is derived from a multiscale neighborhood clique. Maximum Likelihood (ML) estimation via the Expectation Maximization (EM) algorithm is used to estimate the parameters of the new model. Then, utilizing the parameter estimates, the image field is segmented with a MAP classifier. It is demonstrated that the proposed algorithm provides superior segmentations of synthetic images, yet is computationally efficient.
Unsupervised Image Segmentation

Authors:
Simon A Barker, Cambridge University, (U.K.)
Peter J.W. Rayner, Cambridge University, (U.K.)

Abstract:
We present an unsupervised segmentation algorithm comprising an annealing process to select the maximum a posteriori (MAP) realization of a Hierarchical Markov Random Field (MRF) Model. The algorithm consists of a sampling framework which unifies the processes of model selection, parameter estimation and image segmentation, in a single Markov Chain. To achieve this, Reversible Jumps are incorporated into the Markov Chain to allow movement between model spaces. By using partial decoupling to segment the MRF it is possible to generate jump proposals efficiently while providing a mechanism for the use of deterministic methods, such as Gabor filtering, to speed up convergence.
ICASSP98 Paper Abstract
Unsupervised Multidimensional Hierarchical Clustering

Authors:
Rakesh C Dugad, *University of Illinois, Urbana-Champaign, (U.S.A.)*
Narendra Ahuja, *University of Illinois, Urbana-Champaign, (U.S.A.)*

Volume 5, page 2761, paper no. 2318

Abstract:
A method for multidimensional hierarchical clustering that is invariant to monotonic transformations of the distance metric is presented. The method derives a tree of clusters organized according to the homogeneity of intracluster and interpoint distances. Higher levels correspond to coarser clusters. At any level the method can detect clusters of different densities, shapes and sizes. The number of clusters and the parameters for clustering are determined automatically and adaptively for a given data set which makes it unsupervised and non-parametric. The method is simple, noniterative and requires low computation. Results on various sample data sets are presented.
ICASSP98 Paper Abstract

Fast and Robust Level-set Segmentation of Deformable Structures

Authors:
Hussein M Yahia, INRIA, (France)
Jean-Paul Berroir, INRIA, (France)
Gilles Mazars, INRIA, (France)

Volume 5, page 2765, paper no. 2438

Abstract:
Level-sets provide powerful methods for the segmentation of deformable structures. They are able to handle protrusions and specific topological effects. In this work a particle system formulation of level-sets is introduced. It keeps all the advantages of the level-set approach for the segmentation of deformable structures, while it overcomes some of its drawbacks. In this approach the level-sets are controlled by particles, which is of particular interest for interactive control. The particle system records the internal energy of the level-set, while the external force field comes from image data. The energy minimization process is fast, stable and robust. The use of skeleton techniques provide a reliable initialization of the particles, and it is coherent with simple affine motion. The paper is illustrated by examples coming from real image sequences.
ICASSP98 Paper Abstract

A New Stereo Matching Algorithm Based on Bayesian Model

Authors:
Sang-Hwa Lee, Seoul National University, (Korea)
Jong-Il Park, MIC3, ATR, (Japan)
Choong-Woong Lee, Seoul National University, (Korea)

Abstract:
In this paper we derive the general formula of Bayesian model for stereo matching algorithm and implement it with simplified probabilistic models. The probabilistic models are independence property and similarity between the neighborhood disparities in the configuration. The formula is the generalization of Bayesian model for stereo matching, and can be implemented into some different forms corresponding to the probabilistic models in the configuration. We propose a new probabilistic model in order to simplify the joint probability distribution of disparities in the configuration. According to the experimental results, we can conclude that the derived formula generalizes the Bayesian model for stereo matching, and the simplified probabilistic model reason the pure joint probability distribution very well. Compared with the conventional method of Bayesian model and sum of squared difference (SSD) algorithm, the proposed algorithm outperforms the other ones.
ICASSP98 Paper Abstract
Errors-in-Variables Modeling in Optical Flow Problems

Authors:
Lydia L Ng, Macquarie University, (Australia)
Victor Solo, Macquarie University, (Australia)

Volume 5, page 2773, paper no. 1098

Abstract:
Although still in practice, the use of total least squares (TLS) in optical flow estimation is unreliable. TLS implicitly assumes that the error terms affecting the partial derivatives of the image intensities are independent. The usual methods for estimating the partial derivatives ensures that the errors are strongly correlated. Due to this correlation, an alternative method is required to treat the resulting errors-in-variables (EIV) problem. In this paper we propose a new method for estimating optical flow based on Sprent's procedure. This method incorporates a general EIV model and provides a far simpler computational procedure than found in previous solutions.
ICASSP98 Paper Abstract

Wavelet Based Analysis of Rotational Motion in Digital Image Sequences

Authors:
Mingqi Kong, Washington University in Saint Louis, (U.S.A.)
Jean-Pierre Leduc, Washington University in Saint Louis, (U.S.A.)
Bijoy K Ghosh, Washington University in Saint Louis, (U.S.A.)
Jon Corbett, Washington University in Saint Louis, (U.S.A.)
Victor M Wickerhauser, Washington University in Saint Louis, (U.S.A.)

Volume 5, page 2777, paper no. 1406

Abstract:
This paper addresses the problem of estimating, analyzing and tracking objects moving with spatio-temporal rotational motion (spin or orbit). It is assumed that the digital signals of interest are acquired from a camera and structured as digital image sequences. The trajectories in the signal are two-dimensional spatial projection in time of motion taking place in a three-dimensional space. The purpose of this work is to focus on the rotational motion i.e. estimate the angular velocity. In natural scenes, the rotational motion usually composes with translational or accelerated motion on a trajectory. In this paper, we show that the trajectory can be estimated and tracked either simultaneously or separately from the rotational motion and that the analysis of the trajectory and the rotational motion can be done efficiently. The final goal of this work is to provide selective reconstructions of moving objects of interest. This paper constructs new continuous wavelet transforms that can be tuned to both translational and rotational motion. The link between rotational motion, symmetry and critical sampling is also presented. Applications are presented with tracking and estimation. The parameters of analysis that are taken into account in these rotational wavelet transforms are translation (space and time), velocity, spatial scale, angular position and angular velocity. The continuous wavelet functions are finally discretized for signal processing.
ICASSP98 Paper Abstract

Accelerated Spatio-Temporal Wavelet Transforms: An Iterative Trajectory Estimation

Authors:
Jean-Pierre Leduc, Washington University in Saint Louis, (U.S.A.)
Jon Corbett, Washington University in Saint Louis, (U.S.A.)
Mingqi Kong, Washington University in Saint Louis, (U.S.A.)
Victor M Wickerhauser, Washington University in Saint Louis, (U.S.A.)
Bijoy K Ghosh, Washington University in Saint Louis, (U.S.A.)

Volume 5, page 2781, paper no. 1417

Abstract:
This paper addresses the problem of estimating and analyzing accelerated motion in spatio-temporal discrete signals. It is assumed that the digital signals of interest are acquired from imaging sensors and structured as digital image sequences. The motion trajectories in the signal are two-dimensional spatial projections in time of three-dimensional motions. Consequently, they contain all the orders of acceleration. The purpose of this work is to estimate the trajectory and the motion parameters of selected moving objects in the scene. The final goal is to provide selective reconstructions of accelerated objects of interest. This paper presents the construction of new continuous wavelet transforms that can be tuned to any order of accelerations, we demonstrate their existence and provide the related admissibility conditions. The parameters for analysis that are taken into account in these accelerated wavelet transforms are spatial and temporal translations, velocity, acceleration (second or nth order), spatial scale and angular orientation. The continuous wavelet functions are finally discretized for signal processing.
ICASSP98 Paper Abstract

Estimation of Object Location from Short Pulse Scatter Data

Authors:
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Anthony J Devaney, Northeastern University, (U.S.A.)
E. Heyman, Tel-Aviv University, (Israel)

Volume 5, page 2785, paper no. 2297

Abstract:
We derive an efficient algorithm for the computation of the maximum likelihood estimate of the location of a known target from short pulse wave scatter data. The algorithm constitutes a three step procedure: (i) convolutional data filtering, (ii) time-domain backpropagation, and (iii) summation and consists of a number of projection and backprojection operations integrated in a tomographic scheme. A computer simulation is included for illustration purposes and relevant applications in radar target identification and buried object detection are discussed.
Evaluation of Image Stabilization Algorithms

Authors:
Carlos H Morimoto, IBM Almaden Research Center, (U.S.A.)
Rama Chellappa, University of Maryland, (U.S.A.)

Abstract:
Several techniques for electronic image stabilization have recently been proposed, but very little research has been done to compare and evaluate such techniques. In this paper we propose a set of measures to evaluate image stabilization algorithms based on their fidelity, displacement range, and performance. These measures do not require calibration or ground truth, making the evaluation procedure very simple and flexible, i.e., it provides the means to compare techniques based on different motion models. We have used this procedure to compare several image stabilization algorithms and also evaluate the sensitivity of these algorithms to some of its parameters. These same procedures could also be used for the comparison and evaluation of motion estimation and image registration techniques.
High Precision Image Matching Using 4D Optimal Minimisation

Authors:
Marius C Vasiliu, Paris-Sud University, (France)
Bertrand Zavidovique, Paris-Sud University, (France)

Abstract:
We propose a global method to match pair of images using the similarity information. Using a generic similarity distance between pixel pairs, this method can match any kind of images (gray levels, RGB, IR) or more generally any pair of 2D matrix (like spectrogram or wavelet transformations). Our algorithms search the best matching map in a 4-dimensional space defined by the Cartesian product of two input images. Several parameters like topological cost functions or global minimum search method can be adapted, function of specific applications. One of the proposed search method is an original extension of dynamic programming in 4D space. Other methods like iterated global searching or simulated annealing are proposed and their performances are compared. Typical applications are 3D stereo reconstruction, optical flow and velocity field computing or (sub-)pixel texture stretching measurement.
ICASSP98 Paper Abstract

A New Fast Motion Estimation Method Based on Total Least Squares for Video Encoding

Authors:
Sachin G Deshpande, University of Washington, (U.S.A.)
Jenq-Neng Hwang, University of Washington, (U.S.A.)

Volume 5, page 2797, paper no. 2206

Abstract:
We present a new fast motion estimation method useful for high speed video encoding. Most of the motion estimation methods for video coding can be classified as Block Matching (BM) methods or Pel Recursive (PR) methods. Majority of the current fast motion estimation methods belong to block matching category. These methods try to reduce the number of search locations. Our proposed method is based on the pel recursive formulation. However, in order to achieve fast estimation, we operate on a block of pixels using a Total Least Squares (TLS) based estimation scheme which tries to estimate the true motion vector for each block. The major advantages of the proposed method include very fast estimation, almost constant time for motion estimation for all the video sequences, fractional pel accuracy, and better performance for noisy sequences. We present extensive simulation results to illustrate the performance of the proposed method.
Video Coding Based on Motion Estimation in the Wavelet Detail Images

Authors:
Geert Van der Auwera, VUB, ETRO/IRIS Research Group, (Belgium)
Adrian Munteanu, VUB, ETRO/IRIS Research Group, (Belgium)
Gauthier Lafruit, IMEC - Leuven, (Belgium)
Jan Cornelis, VUB, ETRO/IRIS Research Group, (Belgium)

Volume 5, page 2801, paper no. 2423

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 uses two matching criteria, namely the absolute difference and the absolute sum. For a wavelet decomposed one-dimensional step function, it is shown that for odd translations of the step, the absolute sum reaches a smaller minimum than the absolute difference. We also derive in this case a constraint on the highpass filter coefficients so that a zero prediction error can be reached by using the absolute sum. Although this cannot be easily generalized for an arbitrary signal profile, experimental results obtained with photorealistic image sequences indicate that the prediction error can be reduced with respect to techniques that only use the absolute difference as matching criterion.
ICASSP98 Paper Abstract

Motion Segmentation on the TMS320C80 Multimedia Video Processor

Authors:
Robert J. Fergusson, University of Strathclyde, Scotland, (U.K.)
John J. Soraghan, University of Strathclyde, Scotland, (U.K.)

Volume 5, page 2805, paper no. 1592

Abstract:
In MPEG-4 that uses Video Object Planes (VOP’s), systems to efficiently extract objects are required. Based on research into the Human Visual System (HVS), this paper presents an Object Extraction System that uses Motion Segmentation and Image Segmentation in parallel, and then combines them to efficiently extract objects using Fuzzy Reasoning. The most computationally demanding part of this system, the Motion Segmentation (MS), is implemented on the TMS320C80 DSP MVP. The Motion Estimation part of the MS algorithm and its fast implementation on C80 architecture are also described.
ICASSP98 Paper Abstract

Vector Set Partitioning with Classified Successive Refinement VQ for Embedded Wavelet Image and Video Coding

Authors:
Debargha Mukherjee, University of California, Santa Barbara, (U.S.A.)
Sanjit K. Mitra, University of California, Santa Barbara, (U.S.A.)

Volume 5, page 2809, paper no. 2521

Abstract:
The Set Partitioning in Hierarchical Trees (SPIHT) approach for still image compression proposed by Said and Pearlman, is one of the most efficient embedded gray image compression schemes till date. The algorithm relies on a very efficient scanning cum bit-allocation scheme for quantizing the coefficients obtained by a wavelet decomposition of an image. In this paper, we adopt this scheme to scan vectors of wavelet coefficients, and use successive refinement VQ techniques with staggered bit-allocation to quantize several wavelet coefficients at once. The new scheme is named VSPIHT (Vector SPIHT). We present some coding results comparing VSPIHT to the scalar counterpart in the mean-squared-error sense. The method readily generalizes to color images and video where the vector-based approach makes more sense. We present the coding results on INTRA frames of QCIF sequences as compared against H.263.
ICASSP98 Paper Abstract
Transcoder Architectures for Video Coding

Authors:
Niklas Björk, Ericsson Telecom AB, (Sweden)
Charilaos Christopoulos, Ericsson Telecom AB, (Sweden)

Volume 5, page 2813, paper no. 1035

Abstract:
Two different models for transcoding of H.263-based video streams are examined: rate reduction and resolution reduction. Results show that the computational complexity of the basic transcoding model can be reduced for each model by an average of 39% and 23% with less than 1 dB loss in quality for sequences with high motion. Comparisons with scalable video coding model are also presented.
ICASSP98 Paper Abstract

Efficient Coding of DCT Coefficients by Joint Position-Dependent Encoding

Authors:
Eric C Reed, MIT, (U.S.A.)
Jae S. Lim, MIT, (U.S.A.)

Volume 5, page 2817, paper no. 2254

Abstract:
In a typical MC-DCT encoding scheme, a large portion of the bit rate is used to encode the location and amplitude information of the nonzero quantized DCT coefficients. Therefore efficient encoding of the DCT coefficients is extremely important. In this paper we describe the Joint Position-Dependent Encoding (PDE) approach to encode the DCT coefficients. Joint PDE exploits the variations in statistical properties of the runlengths and amplitudes as a function of position by introducing a set of 2-D codebooks in which each DCT coefficient is assigned to one codebook in the set based on its location. Utilizing an MPEG-2 codec, we compare the bit rates using the joint PDE variable length codes (VLC’s) with the bit rates produced by the MPEG-2 VLC’s. We also examine how performance is affected by the number of codebooks.
ICASSP98 Paper Abstract
Efficient SNR-Scalability in Predictive Video Coding

Authors:
Kenneth Rose, University of California, Santa Barbara, (U.S.A.)
Peng Wu, University of California, Santa Barbara, (U.S.A.)
Shankar L. Regunathan, University of California, Santa Barbara, (U.S.A.)

Volume 5, page 2821, paper no. 2303

Abstract:
A new method is proposed for efficient SNR scalability in predictive video coding. It is of low complexity, and it is applicable to standard DCT-based video compression with motion compensation. Information that is only available to the enhancement layers is exploited to improve the quality of their frame prediction without compromising the usefulness of the compressed data provided by the base layer(s). More specifically, the next frame prediction for use by an enhancement-layer decoder is obtained by combining, or switching between transform coefficients from: i) the reconstructed base-layer frame; and ii) the predicted enhancement-layer frame. The combining rule depends on the compressed residual of the base layer, and on the parameters used for this compression. The method is applied to standard DCT-based predictive video coding, and preliminary simulation shows consistent, substantial improvement in the performance of enhancement layers. The proposed method may be easily combined with known temporal scalability methods to provide further improvement of the performance of enhancement layers over a wide range of bit rates.
ICASSP98 Paper Abstract
MPEG-2 Video Coding with Image Partitioning

Authors:
Ekaterina G Barzykina, University of British Columbia, (Canada)
Panos Nasiopoulos, University of British Columbia, (Canada)
Rabab K Ward, University of British Columbia, (Canada)

Volume 5, page 2825, paper no. 2511

Abstract:
We present an original motion compensation strategy based on frame partitioning. The proposed method uses different temporal resolutions within a single frame to improve compression. We present a new bit allocation and rate control algorithm complementing our motion compensation technique. Our approach to bit allocation ensures the consistency of quality throughout a frame and a GOP. For the same picture quality, frame partitioning alone yields an additional increase of up to 20 percent or more in the encoding efficiency, while our bit allocation algorithm eliminates fluctuation in visual quality.
ICASSP98 Paper Abstract
Significance-Linked Wavelet Video Coder

Authors:
Jozsef Vass, University of Missouri-Columbia, (U.S.A.)
Bing-Bing Chai, Sarnoff Corporation, (U.S.A.)
Xinhua Zhuang, University of Missouri-Columbia, (U.S.A.)

Volume 5, page 2829, paper no. 2023

Abstract:
Perhaps, Sarnoff Corporation's zerotree entropy (ZTE) coder is the most successful wavelet video coder published so far which exploits the statistical properties of wavelet-transformed images by utilizing novel data representation and organization strategies. In this paper, a high performance hybrid video coding algorithm termed video significance-linked connected component analysis (VSLCCA) is developed. It is quite encouraging that, at least empirically convinced, the wavelet transform with aids of those recently published innovative data representation and organization methods can be an invaluable asset in video coding if motion-compensated error frames are coherent. In VSLCCA, time domain motion estimation followed by exhaustive overlapped block motion compensation is utilized to ensure coherency, and then wavelet transform is applied to each error frame with significant wavelet coefficients being encoded by highly efficient SLCCA technique. Experimental results on standard MPEG-4 test sequences show that VSLCCA is superior to H.263 and ZTE by 0.48 dB and 0.77 dB on average, respectively.
Novel Error Concealment Techniques for Images in ATM Environments

Authors:
Moh’d A Hasan, King’s College London, (U.K.)
Atif I Sharaf, King’s College London, (U.K.)
Farokh A Marvasti, King’s College London, (U.K.)

Abstract:
Images transmitted via ATM networks suffer from quality degradation due to buffer overflow or cell header errors which cause ATM cells to be lost. This paper presents a new approach to conceal the errors in the received images by the application of novel error recovery techniques to the decomposed DCT-coefficient subimages of the corrupted image. These techniques were developed to recover images corrupted by impulsive noise. Since decomposing the corrupted image into the DCT-coefficient subimages generates low resolution images corrupted by impulsive noise, all the techniques used to recover images corrupted by impulsive noise can be used to recover the subimages and hence the corrupted image. In this paper, we study the performance of new techniques to recover the corrupted subimages. The quality of the recovered image using these techniques is better than the quality obtained by many classical error concealment techniques.
ICASSP98 Paper Abstract

Detail Selection Incorporating Subjective Factors for Very Low Bit-Rate Image Coding

Authors:
Yongqin Zeng, Imperial College, (U.K.)

Abstract:
This paper is concerned with the subjective selection of image details for very low bit-rate coding scheme. An approach is proposed for extracting and ranking the details in the order of perceptual significance to the Human Visual System (HVS). A new perceptual ranking model has been established based on multivariate regression analysis. This ranking model provides better results in detail selection than that by an empirical ranking formula proposed in a previous study in terms of the correlation between the objective ranking and subjective ranking of perceptual significance. Segmented strong contours and selected details are combined and coded to illustrate the efficiency of coding meaningful details. Compared with the pure segmented images, the addition of the selected details shows better subjective image quality at lower bit rates.
ICASSP98 Paper Abstract

An Adaptive Quantization Algorithm for MPEG-2 Video Coding

Authors:
Lijun Luo, *Southeast University*, (China)
Cairong Zou, *Southeast University*, (China)
Zhen-Ya He, *Southeast University*, (China)
Isao Shirakawa, *Osaka University*, (Japan)

Abstract:
An adaptive quantization algorithm for MPEG-2 video coding using neural network is presented in this paper. The proposed algorithm uses a BP neural network to divide the macroblock activity into one of four categories: flat, edge, texture, fine-texture, and thus the macroblock can be quantized adaptively according to the human vision system (HVS) sensitivity. Experiment results show that this method can reduce blocky artifacts of flat area and distortion at edge effectively. Meanwhile, the picture subjective quality and objective quality of each frame are improved.
ICASSP98 Paper Abstract

A Novel Similarity Measure for Compression and Classification

Authors:
Yusuf Ozturk, San Diego State University, (U.S.A.)
Huseyin Abut, San Diego State University, (U.S.A.)

Volume 5, page 2845, paper no. 1890

Abstract:
In this study we propose a new architecture for texture classification based on pair-wise pixel associations as an extension of the recently developed Multivalued Recursive Network (MAREN) architecture. Maybe more critically we propose a novel similarity measure and classification algorithm to be used with this network. The proposed fidelity criterion has been observed to be tightly coupled with the ubiquitous mean-square error (MSE) distance measure. Both SOAR and MAREN structures can be considered as extensions of the associative memory concept frequently used in neural networks. Our proposed similarity measure is based on the principle of directional divergence of interpixel relationships in a given texture and promises a number of advantages over the MSE measure. In this paper, SOAR will be discussed within the framework of a texture classification problem, but we believe it would be very easy to extend to other applications where interpixel relationship is the primary focus.
ICASSP98 Paper Abstract
A Generalized Weighted Median Filter Structure Admitting Real-Valued Weights

Authors:
Gonzalo R Arce, University of Delaware, (U.S.A.)

Volume 5, page 2849, paper no. 1261

Abstract:
Weighted median filters (smoothers) have been shown to be analogous to normalize FIR linear filters constrained to have only positive weights. In this paper, it is shown that much like the mean is generalized to the rich class of linear FIR filters, the median can be generalized to a richer class of weighted median (WM) filters admitting positive and negative weights. The generalization follows naturally and is surprisingly simple. In order to analyze and design this class of WM filters, a new threshold decomposition theory admitting real-valued input signals is developed which, in turn, is used to develop fast adaptive algorithms to optimally design the real-valued filter coefficients. The new WM filter formulation leads to significantly more powerful estimators capable of effectively addressing a number of fundamental problems in signal processing which could not adequately be addressed by prior WM filter (smoother) structures.
ICASSP98 Paper Abstract

Resolution Enhancement of Colored Images by Inverse Diffusion Processes

Authors:
Nir Sochen, Technion - Institute of Technology, (Israel)
Yehoshua Y Zeevi, Technion - Institute of Technology, (Israel)

Volume 5, page 2853, paper no. 1504

Abstract:
Algorithms for resolution enhancement are needed in various applications of image processing and communication such as compression, and HDTV, in which the enhancement of low resolution images acquired by CCD-based electronic color cameras is required. We develop a geometrical algorithm, based on diffusion processes, which are used both for smoothing of the enlarged image, when "time" is flowing forward, and for enhancement. The latter is accomplished by allowing the "time" to flow backwards, i.e. "solving" an inverse diffusion problem which is ill posed. In order to stabilize the flow as well as to enhance important features (e.g. edges) on the expense of less important image domains, we use a modified Beltrami diffusion equation.
Iterative Multiframe Super-Resolution Algorithms for Atmospheric Turbulence-Degraded Imagery

Authors:
David G Sheppard, University of Arizona, (U.S.A.)
Bobby R Hunt, University of Arizona, (U.S.A.)
Michael W Marcellin, University of Arizona, (U.S.A.)

Abstract:
Algorithms for image recovery with super-resolution from sequences of short-exposure images are presented in this paper. Both deconvolution from wavefront sensing (DWFS) and blind deconvolution are explored. A multiframe algorithm is presented for DWFS which is based on maximum a posteriori (MAP) formulation. A multiframe blind deconvolution algorithm is presented based on a maximum likelihood formulation with strict constraints incorporated using nonlinear reparameterizations. Quantitative simulation of imaging through atmospheric turbulence and wavefront sensing are used to demonstrate the super-resolution performance of the algorithms.
ICASSP98 Paper Abstract
Generically Sufficient Conditions for Exact Multichannel Blind Image Restoration

Authors:
Hung-Ta Pai, University of Texas, Austin, (U.S.A.)
John W Havlicek, University of Texas, Austin, (U.S.A.)
Alan C Bovik, University of Texas, Austin, (U.S.A.)

Volume 5, page 2861, paper no. 2141

Abstract:
We have previously developed an algorithm and sufficient conditions for exact blind image restoration. In this paper, we use the resultant matrix theorem and techniques of algebraic geometry to prove that the sufficient conditions hold generically given three blurred versions of the same image and some restrictions on the size of the original image. Moreover, the extension to multichannel blind n-dimensional signal restoration is described.
ICASSP98 Paper Abstract

Penalized Maximum Likelihood Image Reconstruction with Min-Max Incorporation of Noisy Side Information

Authors:
Robinson Piramuthu, *University of Michigan, (U.S.A.)*
Alfred O. Hero III, *University of Michigan, (U.S.A.)*

Volume 5, page 2865, paper no. 2211

Abstract:
A method for incorporating anatomical MRI boundary side information into penalized maximum likelihood (PML) Emission Computed Tomography (ECT) image reconstructions using a set of averaged Gibbs weights was proposed in an earlier paper. A quadratic penalty based on Gibbs weights was used to enforce smoothness constraints everywhere in the image except across the estimated boundary of ROI. In this methodology, a limiting form of the posterior distribution of the MRI boundary parameters was used to average the Gibbs weights obtained using other techniques. We show an improvement in performance when the variance of boundary estimates from the MRI data becomes significant. In this paper, we present the empirical performance analysis of the proposed method of averaged Gibbs weights.
ICASSP98 Paper Abstract

Wavelet-Vaguelette Restoration in Photon-Limited Imaging

Authors:
Robert D. Nowak, Michigan State University, (U.S.A.)
Michael J Thul, University of Kaiserslautern, (Germany)

Abstract:
This paper studies linear shift-invariant inverse problems arising in photon-limited imaging. The problem we consider is the recovery of an intensity image from a distorted version degraded with Poisson noise. This problem arises in medical and astronomical imaging. It is shown that the wavelet-vaguelette decomposition (WVD) can provide much better estimates of the underlying intensity compared to classical frequency domain methods. The paper combines recently developed wavelet-based filtering techniques for photon imaging with new results in WVD methods for inverse problems. Furthermore, we show that the WVD can be interpreted as a prefiltered wavelet transform, and that it can be very efficiently computed. The new method is applied to nuclear medicine imaging.
ICASSP98 Paper Abstract

Perceptually Optimal Restoration of Images with Stack Filters

Authors:
Jr-Jen Huang, Purdue University, (U.S.A.)
Edward J. Coyle, Purdue University, (U.S.A.)

Volume 5, page 2873, paper no. 2526

Abstract:
The present approach to the MAE-based design of stack filters for image restoration does not always produce the desired visual result. Thus, in this paper, a new stack filter design algorithm is developed. It is based upon a Weighted Mean Absolute Error (WMAE) criterion instead of the traditional MAE criterion, which assigns the same weights to all errors. The weights in this WMAE criterion are designed with the aid of the Visible Differences Predictor (VDP), which can estimate the sensitivity of the human visual system to changes in images. Experiments with this WMAE approach show that the stack filters it produces perform significantly better in image processing applications than those designed with the MAE approach.
ICASSP98 Paper Abstract
A Directional Image Decomposition for Ultra-Wideband SAR

Authors:
Richard Rau, Georgia Institute of Technology, (U.S.A.)
James H. McClellan, Georgia Institute of Technology, (U.S.A.)

Volume 5, page 2877, paper no. 1809

Abstract:
This paper presents a theoretical analysis of the structure of wide angle, ultra-wideband SAR images formed by a constant integration angle backprojection image former. It is shown that the effects of the image former can be modeled as a filtering operation on the original data. Furthermore, SAR images for different squint angles can be obtained from the original images by directional filtering. As a result, it will be shown that perfect reconstructing directional filterbanks can be used as a unitary transform between SAR images and a 3-D representation containing additional aspect-angle information. It will be demonstrated how this new representation can be used to enhance targets.
ICASSP98 Paper Abstract

Affine Equivariance in Multichannel OS-filtering

Authors:
Visa V Koivunen, Tampere University of Technology, (Finland)
Simo Luukkonen, University of Oulu, (Finland)
Hannu Oja, University of Oulu, (Finland)

Volume 5, page 2881, paper no. 1664

Abstract:
Nonlinear multichannel filters have successfully been applied to biomedical signals, multichannel images as well as processing of vector fields. In multichannel signals, component variances and correlations among components may be unequal and time-varying. Such changes can be expressed as an affine transformation of the input signal. In this paper, we investigate how the performance and statistical properties of multichannel filters stemming from order statistics (OS) change under affine transformations. An affine equivariant multichannel filter is introduced and the use of affine equivariant performance metric replacing the Mean Square Error is proposed. Advantages of affine equivariance are demonstrated in simulation, and filtering examples using real data are given.
ICASSP98 Paper Abstract

Grid Filters for Local Nonlinear Image Restoration

Authors:
Todd L Veldhuizen, University of Waterloo, (Canada)
M. Ed Jernigan, University of Waterloo, (Canada)

Volume 5, page 2885, paper no. 2066

Abstract:
We describe a new approach to local nonlinear image restoration, based on approximating functions using a regular grid of points in a many-dimensional space. Symmetry reductions and compression of the sparse grid make it feasible to work with eight-dimensional grids as large as $14^8$. Unlike polynomials and neural networks whose filtering complexity per pixel is linear in the number of filter coefficients, grid filters have $O(1)$ complexity per pixel. Grid filters require only a single presentation of the training samples, are numerically stable, leave unusual image features unchanged, and are a superset of order statistic filters. Results are presented for blurring and additive noise.
ICASSP98 Paper Abstract
Denoising by Extracting Fractional Order Singularities

Authors:
Hassan Shekarforoush, University of Maryland, (U.S.A.)
Josiane B. Zerubia, INRIA, (France)
Marc Berthod, INRIA, (France)

Volume 5, page 2889, paper no. 2467

Abstract:
In this paper we will introduce a method of isolating and extracting certain class of local singular behaviours of signals/images which in turn will lead to a method of point-wise noise estimation and suppression. The underlying motivation is to decompose functions directly in terms of components which would naturally represent different orders of regular or singular behaviours defined by the local Holder exponents. We have shown that such a decomposition can lead to a factorization of the spectrum of the singular portion of the signal in terms of the spectrum of the original signal and that of a denoising filter.
Perception Based Adaptive Image Restoration

Authors:
Stuart W. Perry, *University of Sydney, (Australia)*
Ling Guan, *University of Sydney, (Australia)*

Volume 5, page 2893, paper no. 1421

Abstract:
This paper presents an image restoration technique which uses a cost function based on a novel image error measure. The cost function presented here takes into account local statistical information of the image when performing restoration. It is shown that the technique compares favourably with other techniques, especially when applied to color images.
ICASSP98 Paper Abstract
On Estimating the Quality of Noisy Images

Authors:
Zhong Zhang, Lehigh University, (U.S.A.)
Rick S Blum, Lehigh University, (U.S.A.)

Abstract:
Some new techniques are proposed for estimating the quality of a noisy image of a natural scene. Analytical justifications are given which explain why these techniques work. Experimental results are provided which indicate that the techniques work well in practice. These techniques need only the images to be evaluated and do not use detailed information about the formation of the image. The focus is on the case where the image is only corrupted by additive Gaussian noise, which is independent from pixel to pixel, but some cases with blurring are also considered. These results should be useful in the process of fusing several images to obtain a higher quality image. Quality measures of this type are needed for fusion, but they have not received much attention to date. In this research, a mixture model is used in conjugation with the expectation-maximization (EM) algorithm to model edge images. This approach yields an accurate representation which should also be useful in other image processing research.
ICASSP98 Paper Abstract
Adaptive Fuzzy Morphological Filtering of Images

Authors:
Jinsung Oh, University of Pittsburgh, (U.S.A.)
Luis F. Chaparro, University of Pittsburgh, (U.S.A.)

Abstract:
In this paper we introduce a neural network implementation of fuzzy mathematical morphology operators and apply it to image denoising. Using a supervised training method and differentiable equivalent representations for the fuzzy morphological operators, we derive efficient adaptation algorithms to optimize the structuring elements. We can then design fuzzy morphological filters for processing multi-level or binary images. The convergence behavior of basic structuring elements for the opening filter and different signals, and its significance for other structuring elements of different shape is discussed. To illustrate the performance of the fuzzy opening filter we consider the removal of impulse noise in multi-level and binary images.
ICASSP98 Paper Abstract

Hierarchical Bayesian Image Restoration from Partially Known Blurs

Authors:
Vladimir Z Mesarovic, Crystal Semiconductor Corporation, (U.S.A.)
Nikolas P Galatsanos, Illinois Institute. of Technology, (U.S.A.)
Rafael Molina, University of Granada, (Spain)
Aggelos K Katsaggelos, Northwestern University, (U.S.A.)

Volume 5, page 2905, paper no. 2543

Abstract:
A number of restoration filters have been proposed for the restoration problem from partially-known blurs. We derived the regularized constrained total least-squares (RCTLS) filter for this problem and we showed that it has a number of advantages over previous filters for this problem. However, the problem of estimating the parameters that define this filter has not been addressed yet. In this paper we propose a two-step algorithm based on the hierarchical Bayesian approach to simultaneously restore the image and estimate the parameters of the RCTLS restoration filter. The algorithm is derived in the DFT domain; thus, it is very efficient even for very large images. The algorithm is demonstrated in numerical experiments.
ICASSP98 Paper Abstract

Bayesian Estimation in an Image Restoration Problem in X-Ray Fiber Diffraction.

Authors:
Shyamsunder Baskaran, Purdue University, (U.S.A.)
Rick P Millane, Purdue University, (U.S.A.)

Volume 5, page 2909, paper no. 2507

Abstract:
The restoration of an incomplete image from a known part and experimental data in the form of the Fourier amplitude squared sums is formulated as a Bayesian estimation problem. This problem is motivated by the structure completion problem in x-ray fiber diffraction analysis. An appropriate prior of uniformly distributed impulses is used. The Bayesian MMSE and MAP estimates are obtained. Simulations are used to compare the performance of the estimates as a function of image and data reduction. The results show that the MMSE estimate significantly outperforms the other estimates. The restored images exhibit some bias towards the known part of the image. This can be partly reduced by an unbiasing procedure.
ICASSP98 Paper Abstract

A Natural Pixel Decomposition for Tomographic Imaging of the Ionosphere

Authors:
Joshua I. Semeter, Max Planck Institute, (Germany)
Farzad Kamalabadi, Boston University, (U.S.A.)

Volume 5, page 2913, paper no. 2515

Abstract:
We apply a natural pixel (NP) decomposition to the problem of computerized ionospheric tomography (CIT). For tomography from very few angles, the NP approach provides some distinct advantages over standard basis expansions. The NP solution requires no prior assumption and as such embeds into the solution the natural spatial resolution supported by the data. For the uniquely constrained CIT geometry, however, NP estimated fields will contain significant negative values. We propose a method of improving the NP estimates by enforcing positivity through an entropy regularization algorithm. These techniques are demonstrated in an example.
ICASSP98 Paper Abstract
A Block-Iterative Quadratic Signal Recovery Algorithm

Authors:
Patrick L. Combettes, City University of New York, (U.S.A.)

Volume 5, page 2917, paper no. 1499

Abstract:
We propose a block-iterative parallel decomposition method to solve quadratic signal recovery problems under convex constraints. The idea of the method is to disintegrate the original multi-constraint problem into a sequence of simple quadratic minimizations over the intersection of two half-spaces constructed by linearizing blocks of constraints. The implementation of the algorithm is quite flexible thanks to its block-parallel structure. In addition, a wide range of complex constraints can be incorporated since the algorithm does not require exact constraint enforcement at each step but merely approximate enforcement via linearization. An application to deconvolution is demonstrated.
ICASSP98 Paper Abstract
Resolution Enhancement by Polyphase Back-Projection Filtering

Authors:
Boaz Cohen, Ben-Gurion University of the Negev, (Israel)
Its'hak Dinstein, Ben-Gurion University of the Negev, (Israel)

Volume 5, page 2921, paper no. 1156

Abstract:
The method for reconstruction and restoration of super resolution images from low resolution sequences presented here is an extension of Irani and Peleg's algorithm ("Improving Resolution by Image Registration", CVGIP: Graphical Models and Image Processing, Vol. 53, No. 3, pp. 231-239, 1991). The input is a set of low resolution images that have been registered to a pixel translation accuracy. A high resolution image is initialized and iteratively improved by back-projecting the errors between the low resolution images and the respective images obtained by simulating the imaging system. The sub-pixel translations between the low resolution images are quantized. The imaging system's PSF and back projection function are estimated with a resolution higher than that of the super resolution image and decimated so that two banks of polyphase filters are obtained. The use of the polyphase filters allows exploitation of all the input images without any smoothing or interpolation operations.
ICASSP98 Paper Abstract

Wavelet Filtering of SAR Images Based on Non Gaussian Assumptions

Authors:
Samuel Foucher, Universite de Sherbrooke, (Canada)
Goze B Bénié, Universite de Sherbrooke, (Canada)
Jean-Marc Boucher, ENSTB, (France)

Volume 5, page 2925, paper no. 1582

Abstract:
Radar images are affected by a multiplicative noise depending on the underlying signal (the ground reflectivity) due to the coherence of the radar wavelength. Images present a strong pixel to pixel variability considerably reducing the efficiency of target detection and classification algorithms. We propose in this study filtering this noise using image multiresolution analysis. The value of the wavelet coefficients of the radar reflectance is estimated by a Bayesian model by maximizing the a posteriori density and by modeling the different densities using the Pearson distributions system. The resulting filter combines a classical adaptive approach and wavelet decomposition using the local variance of the wavelet coefficients for segmenting and weighting the latter taking into account the multiplicative nature of the noise.
ICASSP98 Paper Abstract

Blind Image Restoration Using Local Bound Constraints

Authors:
Kaaren L. May, Imperial College, (U.K.)
Tania Stathaki, Imperial College, (U.K.)
Aggelos K. Katsaggelos, Northwestern University, (U.S.A.)

Volume 5, page 2929, paper no. 1493

Abstract:
A new method of incorporating local image characteristics into blind image restoration is proposed. The local variance of the degraded image is used as a measure of spatial activity, from which individual pixel bounds are determined. A parameter defined by the user controls the degree of smoothing. The local bounds define the solution more precisely than smoothness constraints on the image (including those that are spatially-adaptive), reducing the number of possible solutions and leading to a faster rate of convergence. Experimental results demonstrate the potential of this method as an alternative/supplement to smoothing constraints in blind image restoration.
ICASSP98 Paper Abstract

Symmetry-Constrained 3D Interpolation for Virus X-Ray Crystallography

Authors:
Yibin Zheng, GE Corporate R&D, (U.S.A.)
Peter C Doerschuk, Purdue University, (U.S.A.)
John E Johnson, The Scripps Research Institute, (U.S.A.)

Volume 5, page 2933, paper no. 1494

Abstract:
An interpolation problem that is important in viral x-ray crystallography is considered. The problem requires new methods because (1) the function is known to have icosahedral symmetry, (2) the data is corrupted by experimental errors and therefore lacks the symmetry, (3) the problem is 3D, (4) the measurements are irregularly spaced, and (5) the number of measurements is large (10^4). A least-squares approach is taken using two sets of basis functions: the functions implied by a minimum-energy band-limited exact interpolation problem and a complete orthonormal set of band-limited functions. A numerical example on Cowpea Mosaic Virus is described.
An Iterative Method for Image Enhancement Based on Fuzzy Logic

Authors:
Farzam Farbiz, Amirkabir University of Technology, (Iran)
Seyed Ahmad Motamedi, Amirkabir University of Technology, (Iran)
Mohammad Bagher Menhaj, Amirkabir University of Technology, (Iran)

Volume 5, page 2937, paper no. 2259

Abstract:
This paper presents a new filtering approach based on fuzzy-logic which has high performance in mixed-noise environments. This filter is mainly based on the idea that each pixel is not allowed to be uniformly fired by each of the fuzzy rules. We perform several test experiments in order to highlight the merit of the proposed method. The results are very promising and indicating the high performance of the proposed filter in image restoration in compared with the filters which have been recently cited in image processing literature.
ICASSP98 Paper Abstract
Direct Gray Scale Ridge Reconstruction in Fingerprint Images

Authors:
Carlotta Domeniconi, University of California, Riverside, (U.S.A.)
Sibel Tari, University of California, Riverside, (U.S.A.)
Ping Liang, University of California, Riverside, (U.S.A.)

Volume 5, page 2941, paper no. 2346

Abstract:
An original technique, based on ridge point detection directly from gray scale fingerprint images, is proposed. Our method avoids serious problems that algorithms which perform binarization of fingerprint images have. Each step can be easily hardware implemented, allowing a relevant speed up of the whole process.
ICASSP98 Paper Abstract

Diffusion of the Attractor of Fractal Coding for Edge Restoration

Authors:
Nikki M Bruner, Oklahoma State University, (U.S.A.)
Rao Yarlagadda, Oklahoma State University, (U.S.A.)

Volume 5, page 2945, paper no. 2296

Abstract:
Diffusion of the attractor or reconstructed image of the fractal code provides us a technique to restore edge information. Because of coding error associated with the fractal mappings, edges are degraded at high compression ratios. Partitioning compensates for the degradation, but lowers the compression ratios significantly and does not insure the retention of significant edges. The diffusion technique uses the image gradient to control the rate and direction of diffusion. This allows for smoothing in flat (low intensity transitions) regions and sharpening in edge (high intensity transitions) regions. The usage of the image gradient in this method insures the retention of significant edges. The diffusion technique presented in this paper lessens the degree of degradation of edges from fractal coding at a lower bit rate cost than partitioning at small blocks sizes.
ICASSP98 Paper Abstract

Printer Models and the Direct Binary Search Algorithm

Authors:
Farhan A Baqai, Purdue University, (U.S.A.)
Jan P Allebach, Purdue University, (U.S.A.)

Volume 5, page 2949, paper no. 2327

Abstract:
We incorporate a higher order measurement-based model for printer dot interactions within the iterative direct binary search (DBS) halftoning algorithm. We also present an efficient strategy for evaluating the change in computational cost as the search progresses. Experimental results are shown which demonstrate the efficacy of the approach.
ICASSP98 Paper Abstract
Wreath Products for Edge Detection

Authors:
Valerie Chickanosky, University of Vermont, (U.S.A.)
Gagan Mirchandani, University of Vermont, (U.S.A.)

Volume 5, page 2953, paper no. 2384

Abstract:
Wreath product group based spectral analysis has led to the development of the wreath product transform, a new multiresolution transform closely related to the wavelet transform. In this work, we derive the filter bank implementation of a simple wreath product transform and show that it is in fact, a multiresolution Roberts edge detector. We also derive the relationship between this transform and the two-dimensional Haar wavelet transform. We prove that using a nontraditional metric for measuring edge amplitude with the wreath product transform yields a rotation and translational invariant edge detector. We introduce a novel method for measuring the orientation of an edge and show that it is without error in the noise-free case. Wreath product edge detection performance is shown to be superior to many standard edge detectors.
ICASSP98 Paper Abstract

Film Grain Noise Removal and Generation for Color Images

Authors:
Jacky Chun Kit Yan, University of Toronto, (Canada)
Patrizio Campisi, Universita degli Studi di Roma Tre, (Italy)
Dimitrios Hatzinakos, University of Toronto, (Canada)

Volume 5, page 2957, paper no. 1593

Abstract:
In this paper, we propose a noise filtering scheme, which is based on a multichannel homomorphic transformation, for color photographic images corrupted by signal-dependent film grain noise. The proposed method performs the estimation of the noise parameter using the higher-order statistics (skewness or kurtosis) of the corrupted image and the filtered image statistics. This parameter estimation technique can be used to generate color film grain noise that has applications in motion picture productions. After a theoretical description of the method employed, experimental results are provided.
ICASSP98 Paper Abstract

Video Scene Change Detection Using the Generalized Sequence Trace

Authors:
Cuneyt Taskiran, Purdue University, (U.S.A.)
Edward J. Delp, Purdue University, (U.S.A.)

Volume 5, page 2961, paper no. 2291

Abstract:
We propose an algorithm to detect scene changes in a video sequence in the compressed domain. We define a feature vector extracted from each frame that we call the generalized trace. We examine various ways of processing the generalized trace to determine the temporal location of scene changes in a video stream.
ICASSP98 Paper Abstract

A Multi-Resolution Video Segmentation Scheme for Wipe Transition Identification

Authors:
Hong H Yu, Princeton University, (U.S.A.)
Wayne Wolf, Princeton University, (U.S.A.)

Volume 5, page 2965, paper no. 2540

Abstract:
This paper presents a new methodology for wipe transition identification. Shot transition detection is an important technique for making videos easier to handle. Due to the wide variety, wipe transition appears to be the most difficult one to be detected among all types of shot transitions. We propose an approach that takes advantage of the production aspect of video. Each video frame is first decomposed into low-resolution and high-resolution components which are analyzed respectively and further recom-bined together to form a wipe transition detector. In our system, wavelet transformation is used for multi-resolution decomposition.
ICASSP98 Paper Abstract
Digital Watermarking Using Multiresolution Wavelet Decomposition

Authors:
Deepa Kundur, University of Toronto, (Canada)
Dimitrios Hatzinakos, University of Toronto, (Canada)

Volume 5, page 2969, paper no. 1310

Abstract:
We present a novel technique for the digital watermarking of still images based on the concept of multiresolution wavelet fusion. The algorithm is robust to a variety of signal distortions. The original unmarked image is not required for watermark extraction. We provide analysis to describe the behaviour of the method for varying system parameter values. We compare our approach with another transform domain watermarking method. Simulation results show the superior performance of the technique and demonstrate its potential for the robust watermarking of photographic imagery.
ICASSP98 Paper Abstract

The Impact of Channel Coding on the Performance of Spatial Watermarking for Copyright Protection

Authors:
Juan R Hernandez, Universidad de Vigo, (Spain)
Fernando Perez-Gonzalez, Universidad de Vigo, (Spain)
Jose M Rodriguez, Universidad de Vigo, (Spain)

Volume 5, page 2973, paper no. 1721

Abstract:
In this paper we analyze the effect that the application of channel coding produces on the performance of the watermark detection and decoding tests for copyright protection of images. Detector structures are derived for both tests and analytical bounds and approximations are obtained for the bit error rate (BER) and the receiver operating characteristic (ROC) associated with the watermark decoding and detection tests when block codes are employed. The extension to other families of codes is discussed. Finally, the analytical expressions are contrasted with experimental results in several cases of interest.
ICASSP98 Paper Abstract
The Two-Dimensional Wold Decomposition for Segmentation and Indexing in Image Libraries

Authors:
Radu S Stoica, INRIA, (France)
Josiane B. Zerubia, INRIA, (France)
Joseph M Francos, Ben-Gurion University, (Israel)

Volume 5, page 2977, paper no. 1093

Abstract:
This paper presents a method for indexing and retrieval of multimedia data through texture segmentation, using the Wold decomposition. The texture field is assumed to be a realisation of a regular homogeneous random field. On the basis of a 2-D Wold-like decomposition, the field is represented as the sum of a purely indeterministic component, a harmonic component and a countable number of evanescent fields. A new rigorous distance measure between textures is derived, using Wold parameters. Adopting the MRF framework, we construct a segmentation procedure using the Wold parameters.
ICASSP98 Paper Abstract
Sprite-Based Video Coding Using On-Line Segmentation

Authors:
Regis Crinon, Sharp Laboratories of America, (U.S.A.)
Ibrahim Sezan, Sharp Laboratories of America, (U.S.A.)

Volume 5, page 2981, paper no. 1304

Abstract:
We address the problem of on-line sprite-based video coding in cases where scene segmentation is not available a priori or explicit transmission of such segmentation cannot be afforded due to low bit rate requirements. We propose an on-line segmentation method that can be integrated into an MPEG4 on-line sprite-based video codec. The proposed method uses macroblock coding types as well as motion compensated residuals to perform the on-line segmentation. It produces a background mosaic without requiring a priori foreground-background segmentation information. Our results demonstrate the coding efficiency and functionality benefits of the proposed approach.
ICASSP98 Paper Abstract

Segmentation and Compression of Video for Delay-flow Multimedia Networks

Authors:
Yuan-Chi Chang, University of California, Berkeley, (U.S.A.)
David G Messerschmitt, University of California, Berkeley, (U.S.A.)

Volume 5, page 2985, paper no. 2483

Abstract:
Digital video coding has traditionally used frame-by-frame synchronous reconstruction. The transport must then be delay-jitter-free, forcing the modern integrated service packet network such as the Internet to operate in an inefficient "circuit emulation" mode. This mode results in a jitter-free delay representative of the worst-case network delay, which is problematic for delay-sensitive interactive applications. In response, we have proposed and demonstrated a "delay cognizant" model of video coding (DCVC) that operates in an asynchronous reconstruction mode. DCVC minimizes the perceptual delay observed by the user, and still achieves good quality and high compression. Furthermore, the feasibility of asynchronous reconstruction is evidenced by vision science studies of spatiotemporal masking in human visual systems at temporal edges of video.
ICASSP98 Paper Abstract

A Lagrangian Optimization Approach to Rate Control for Delay-Constrained Video Transmission over Burst-Error Channels

Authors:
Chi-Yuan Hsu, University of Southern California, (U.S.A.)
Antonio Ortega, University of Southern California, (U.S.A.)

Volume 5, page 2989, paper no. 2544

Abstract:
We propose a rate control algorithm based on Lagrangian optimization for video transmission over burst-error channels. In our rate control approach, the delay and channel capacity constraints in the video transmission are translated into rate constraints at the encoder. Given that a feedback channel is available, the rate control mechanism can dynamically adjust the video encoding rate to meet the changing rate constraints as the channel conditions vary. Lagrangian optimization is used to find the optimal bit-allocation for the input video frames under the rate constraints, with the objective of minimizing the overall distortion at the decoder. We show how the performance of the transmission system, as measured in terms of the received video quality or the data loss rate, can be improved when information about the channel state is available and the encoder has an a priori probabilistic model of the channel behavior.
ICASSP98 Paper Abstract

A 35uW 1.1V Gate Array 8x8 IDCT Processor for Video-Telephony

Authors:
Roberto Rambaldi, Universita di Bologna, (Italy)
Alessandro Uguzzoni, Universita di Bologna, (Italy)
Roberto Guerrieri, Universita di Bologna, (Italy)

Volume 5, page 2993, paper no. 1503

Abstract:
We have designed and fabricated a low power IC to perform the Inverse 8X8 DCT transform according to the CCITT precision specifications, suitable for portable video communication devices. Several design techniques have been used to reduce the power, such as a fast algorithm, an architecture that can exploit input signal correlation, and large amount of parallelism. The chip is fabricated in a triple metal 0.5um Gate Array CMOS technology. The maximum throughput is 400 Kpix/s at 1.1V, and 27 Mpix/s at 3.3V and the measured power consumption is 35 uW for typical image sequences in color QCIF format at 10 frames/sec with a 1.1 V power supply.
ICASSP98 Paper Abstract

Discrete Cosine Transform Generator for VLSI Synthesis

Authors:
Jill K Hunter, The Queens University of Belfast, (Northern Ireland)
John V. McCanny, The Queens University of Belfast, (Northern Ireland)

Volume 5, page 2997, paper no. 1053

Abstract:
A generator for the automated design of Discrete Cosine Transform (DCT) cores is presented. This can be used to rapidly create silicon circuits from a high level specification. These compare very favourably with existing designs. The DCT cores produced are scaleable in terms of point size as well as input/output and coefficient wordlengths. This provides a high degree of flexibility. An example, 8-point 1D DCT design, produced occupies less than 0.92 mm$^2$ when implemented in a 0.35m double level metal CMOS technology. This can be clocked at a rate of 100MHz.
ICASSP98 Paper Abstract
A New Architecture for In-Memory Image Convolution

Authors:
Vasily G. Moshnyaga, Kyoto University, (Japan)
Kazuhiro Suzuki, Kyoto University, (Japan)
Keikichi Tamaru, Kyoto University, (Japan)

Volume 5, page 3001, paper no. 2443

Abstract:
A new memory-based architecture for real-time image convolution with variable kernels is proposed. The architecture exploits the highest possible bandwidth inherent in memory and achieves the fine-grain parallelism of computations inside the memory. Unlike existing approaches, the architecture ensures convolution with very large kernels under the real time constraints of video applications. It does not require external memory banks or large I/O count and features single chip VLSI implementation.
Abstract:
We present the architecture of a programmable FIR filter for use in DSP and communication applications. A filter with this architecture is capable of running a wide variety of single-rate and multirate filtering algorithms with low latency. Flexibility is achieved by distributed register files that store input data and filter coefficients. The functionality of the filter is programmed by a set of pipelined control signals that are independent of the filter length. We demonstrate how to generate these control signals for a variety of configurations. In addition to its flexibility, the architecture is scalable, modular, and has no broadcast signals, making it ideally suited for VLSI implementations.
ICASSP98 Paper Abstract

Low Power FIR Filter Realization with Differential Coefficients and Input

Authors:
Tian-Sheuan Chang, National Chiao-Tung University, (Taiwan)
Chein-Wei Jen, National Chiao-Tung University, (Taiwan)

Abstract:
Most FIR filter realizations use the inputs and coefficients directly to compute the convolution. In this paper, we present a low power and high speed FIR filter designs by using first order difference between inputs and various orders of differences between coefficients. This design first reformulates the FIR operations with the differences in algorithm level. Then, in architecture level, we adopt the DA architecture to exploit the probability distribution such that power consumption can be reduced further. The design is applied to an example FIR filter to quantify the energy savings and speedup. It shows lower power consumption than the previous design with the comparable performance.
ICASSP98 Paper Abstract

A New Approach to Data Conversion: Direct Analog-to-Residue Converter

Authors:
Damu Radhakrishnan, Nanyang Technological University, (Singapore)
Adimathara P Preethy, Nanyang Technological University, (Singapore)

Volume 5, page 3013, paper no. 1893

Abstract:
A novel design of a direct analog-to-residue converter is presented in this paper. The design makes use of two successive approximation analog-to-digital (A/D) converters, a few modulo adders and a small look-up table. One of the digital-to-analog converters is modified to generate outputs which are weighted by a constant factor, and one of the comparators is replaced by a difference amplifier. The look-up table needed is a very small percentage of the entire chip area and is shown to be only 840 bytes for a 36 bit residue number system converter.
ICASSP98 Paper Abstract

Low Power Signal Processing Architectures Using Residue Arithmetic

Authors:
Manish Bhardwaj, Siemens Components Pte Ltd, (Singapore)
Arjun Balaram, Siemens Components Pte Ltd, (Singapore)

Abstract:
Recent trends like increasing frequencies, larger die sizes and demand for greater portability make power reduction a hard taskmaster. It is acknowledged that the greatest returns come from optimisations at the architectural and technology level. In this paper, we present, for the first time, residue architectures that reduce power by more than 70% without changes in technology. This reduction is achieved without sacrificing performance and with minimal sacrifice in area (less than 60%). The key to such low power solutions is trading-off the speed gained by parallelism for lower power. Existing proposals that achieve similar trade-offs demand an area increase of more than a factor of two and also increase control complexity. Other benefits of using residue arithmetic for low power is the significant reduction in peak current and increased design locality. The role of the number of computations per forward (or reverse) conversion in determining the power characteristics of the system are also analysed and explained. The effectiveness of the methodology is illustrated using a system that extracts a 256-point FFT of the input signal.
Designing Efficient Residue Arithmetic Based VLSI Correlators

Authors:
Aniruddha A Deodhar, Siemens Components Pte Ltd, (Singapore)
Manish Bhardwaj, Siemens Components Pte Ltd, (Singapore)
C. T Clarke, Submetrics, (U.K.)
T Srikanthan, Nanyang Technological University, (Singapore)

Volume 5, page 3021, paper no. 2408

Abstract:
The most important reason for the lack of commercial residue arithmetic (RA) based systems is not the "slow" and area consuming reverse conversion, but the absence of research that explores the system-level trade-offs of such arithmetic in actual VLSI implementations. Such system-level issues are - choice of the moduli set, effect of moduli imbalance on resulting VLSI implementation, choice of the reverse and forward converters, use of lookup versus computation for modular operations, system characteristics that indicate RA suitability and finally, typical VLSI area and performance figures. This paper explains these concerns by presenting novel RA architectures for VLSI correlators employed in radio-astronomy and ultrasonic blood flow measurement. A state-of-the-art, high-performance (80-100 MHz), RA-based correlator ASIC was successfully fabricated as a result of this research.
ICASSP98 Paper Abstract

Pipelined Cordic Based QRD-MVDR Adaptive Beamforming

Authors:
Jun Ma, University of Minnesota, (U.S.A.)
Keshab K. Parhi, University of Minnesota, (U.S.A.)
Ed F. Deprettere, Delft University of Technology, (The Netherlands)

Volume 5, page 3025, paper no. 1324

Abstract:
Cordic based QRD-MVDR adaptive beamforming algorithms possess desirable properties for VLSI implementation such as regularity and good finite-word length behavior. But this algorithm suffers from speed limitation constraint due to the presence of recursive operations in the algorithm. In this paper, a fine-grain pipelined Cordic based QRD-MVDR adaptive beamforming algorithm is developed using the matrix lookahead technique. The proposed architecture can operate at arbitrarily high sample rates, and consists of only Givens rotations which can be mapped onto a Jacobi specific dataflow processor. It requires a complexity of $O(M(p^2 + Kp))$ Givens rotations per sample time, where $p$ is the number of antenna elements, $K$ is the number of look direction constrains, and $M$ is the pipelining level.
ICASSP98 Paper Abstract

A Systolic VLSI Implementation of Kalman-Filter-Based Algorithms for Signal Reconstruction

Authors:
Daniel Massicotte, Universite du Quebec a Trois-Rivieres, (Canada)

Abstract:
The problem of improving the performance of the implementation in VLSI technology of Kalman-based algorithms for signal reconstruction in real time is discussed. A systolic approach is proposed to develop architecture expressly for this specific application. Implemented algorithms are based on the steady-state version of the Kalman filter, which performs for a broad field of specific applications, but the use of a co-processor for the Kalman gain is allowed. We show that the autoregressive model of Kalman filtering is particularly adapted to parallel processing and is well suited for implementation. Although intended to improve signal reconstruction, other applications where a similar autoregressive model of Kalman filtering is required are allowed. The performance of the systolic architecture is validated by comparison with Motorola’s general-purpose DSP56002 digital signal for real-world spectrometric signal reconstruction.
ICASSP98 Paper Abstract

A Class of Vector-Tracing Motion Estimation Architectures for MPEG2 Type Coding For TV and HDTV

Authors:
Martin Gumm, C3I/EPFL, (Switzerland)
Friederich Mombers, C3I/EPFL, (Switzerland)
Stephanie Dogimont, C3I/EPFL, (Switzerland)
Daniel Mlynek, C3I/EPFL, (Switzerland)

Volume 5, page 3033, paper no. 1678

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 RSIC/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.
ICASSP98 Paper Abstract
Reconfiguration for Power Saving in Real-Time Motion Estimation

Authors:
S.R. Park, University of Massachusetts, (U.S.A.)
Wayne Burleson, University of Massachusetts, (U.S.A.)

Volume 5, page 3037, paper no. 2310

Abstract:
In this paper we propose a reconfigurable approach to motion estimation. The statistics of motion vectors can be monitored on a frame by frame basis to choose appropriate hardware configurations. A novel aspect of this work is that we use power savings as a motivation for the reconfiguration. Although FPGAs are not a very power efficient technology, careful design of array architectures can allow power to be saved by avoiding unnecessary computation. This is done by adjusting the search area according to the changing characteristics of an input video signal. Unlike some proposed applications of dynamic reconfiguration, this rate can easily be supported by existing FPGA technology. A more general result is that further power saving can be achieved by utilizing free FPGA resources as local memory to avoid power-hungry off-chip communication. Practical implementation issues using Xilinx 6200 series FPGAs are also discussed.
ICASSP98 Paper Abstract

H.263 Mobile Video Codec Based on a Low Power Consumption Digital Signal Processor

Authors:
Yukihiro Naito, NEC Corporation, (Japan)
Ichiro Kuroda, NEC Corporation, (Japan)

Volume 5, page 3041, paper no. 1313

Abstract:
This paper describes an H.263 video codec implementation based on a low power consumption general purpose DSP. Fast algorithms, such as a fast motion estimation algorithms and a low complexity noise reduction filter, are proposed to implement the video codec on a single DSP chip maintaining sufficient picture quality. By using a 50MIPS, 100mW DSP, the developed codec encodes and decodes 7.5 QCIF frames per second, which is sufficient performance for low bit-rate video compression, typically below 64kbps.
ICASSP98 Paper Abstract

Hardware Software Tri-Design of Encryption for Mobile Communication Units

Authors:
Oskar Mencer, Stanford University, (U.S.A.)
Martin Morf, Stanford University, (U.S.A.)
Michael J Flynn, Stanford University, (U.S.A.)

Volume 5, page 3045, paper no. 2332

Abstract:
We explore the design space of Field Programmable Gate Arrays (FPGAs), Processors and ASICs – Hardware-Software Tri-design – in the framework of encryption for hand-held communication units. IDEA (International Data Encryption Algorithm) is used to show the tradeoffs for the suggested technologies. The measures for comparing different options are: Performance, Programmability and Power ($P^3$). More specifically we use the Performance to Power, or Operations to Energy ratio MOPS/Watt and Mbits/s/Watt to compare processors, FPGAs and ASICs. We compare the latest Digital Signal Processor (DSP) from Texas Instruments to Xilinx XC4000 series FPGAs. Many DSP-like applications perform very well on FPGAs. We show the benefits and limitations of FPGA technology for IDEA.
ICASSP98 Paper Abstract

Low-Energy Heterogeneous Digit-Serial Reed-Solomon Codecs

Authors:
Leilei Song, University of Minnesota, (U.S.A.)
Keshab K. Parhi, University of Minnesota, (U.S.A.)
Ichiro Kuroda, NEC Corporation, (Japan)
Takao Nishitani, NEC Corporation, (Japan)

Volume 5, page 3049, paper no. 1344

Abstract:
Reed-Solomon (RS) codecs are used for error control coding in many applications such as digital audio, digital TV, software radio, CD players, and wireless and satellite communications. This paper considers software-based implementation of RS codecs where special instructions are assumed to be used to program finite field multiplication datapaths inside a domain-specific programmable digital-signal processor (DS-PDSP). A heterogeneous digit-serial approach is presented, where the heterogeneity corresponds to the use of different digit-sizes in the multiply-accumulate (MAC for polynomial multiplication) and degree reduction (DEGRED for polynomial modulo operation) subarrays. The salient feature of this digit-serial approach is that only the digit-cells are implemented in hardware, the finite field multiplications are performed digit-serially in software by dynamically scheduling the internal digit-level operations in RS encoders and decoders. A hardware-software co-design approach is used to select the best digit-size parameters and minimize the energy consumption of both RS encoders and decoders. Low-energy Reed-Solomon codecs are designed in software based on various finite field datapath architectures. It is concluded that, for 2-error-correcting RS(n,k) codec implementations over finite field GF(2^8), a parallel MAC unit (of digit-size 8) and a DEGRED unit with digit-size 2 is the best datapath, with respect to least energy consumption and energy-delay products; with this datapath architecture and appropriate digit-serial scheduling strategies, more than 60% energy reduction and more than 1/3 energy-delay reduction can be achieved compared with the parallel multiplication datapath based approach.
ICASSP98 Paper Abstract

Software Implementation of ADSL Application with a Convolution Co-Processor

Authors:
Eric Dujardin, Philips, (France)
Olivier Gay-Bellile, Philips, (France)

Volume 5, page 3053, paper no. 1958

Abstract:
More and more applications have software implementations in order to cope with the cohabitation of several emerging standards and the fast evolution of consumer products. We show in this paper how to implement efficiently digital communications applications into DSPs with the help of the Convolution Co-Processors, which is described in this paper, and how co-processors are useful to empower DSP performances at a small cost. ADSL application is taken as an example for broadband communications.
ICASSP98 Paper Abstract

Implementation of a Car Handsfree Speech Enhancement Application on a TI TMS320C54x DSP

Authors:
Jamil Chaoui, Texas Instruments, (France)
Sebastien de Gregorio, Texas Instruments, (France)
Daravith Kho, Texas Instruments, (France)
Stephane Sintes, Texas Instruments, (France)
Yves Masse, Texas Instruments, (France)

Abstract:
This paper describes the implementation of a car handsfree speech enhancement application on a TI TMS320C54x DSP. This fixed-point DSP family is especially suited for wireless applications and it is shown how, by taking full advantage of the DSP architecture and instruction set, advanced wireless speech processing applications can be efficiently implemented on these devices. The performances of the complete speech enhancement application in a car environment are presented, showing that even with a fixed-point arithmetic implementation, high performances close to a floating point implementation can be achieved.
ICASSP98 Paper Abstract

A Low-Power VLSI Feature Extractor for Speech Recognition

Authors:
Marco Felici, University of Bologna, (Italy)
Michele Borgatti, University of Bologna, (Italy)
Alberto Ferrari, University of Bologna, (Italy)
Roberto Guerrieri, University of Bologna, (Italy)

Volume 5, page 3061, paper no. 1464

Abstract:
A low-power feature extraction chip computing cepstral coefficients from linear predictive analysis on one-bit quantized speech signal is presented and its VLSI implementation is evaluated. An isolated-word small-vocabulary speech recognizer based on these features has been developed. Its recognition accuracy is within 2% below a system based on standard linear predictive cepstral features. The power consumption of the feature extractor chip is 30uW at 0.9V.
ICASSP98 Paper Abstract
Software Pipelining of Nested Loops for Real-Time DSP Applications

Authors:
Jian Wang, Speech Recognition Software, (Canada)
Bogong Su, William Paterson University of New Jersey, (U.S.A.)
Volume 5, page 3065, paper no. 1219

Abstract:
Modern DSP Processors have been integrated with Instruction-Level Parallelism (ILP), which presents a challenge to exploit ILP within DSP applications. Software Pipelining is an efficient technique to expose ILP for loop programs and has been used widely for current microprocessors. Recently it has been used in DSP compilers but only for the innermost loops. This paper proposes a new approach to extend software pipelining from innermost loops to the whole nested loops in DSP applications. For a perfect loop, after applying any existing software pipelining approach for the innermost loop, we use the so-called pipelining-dovetailing transformation to extend software pipelining to the outer loops. We also present a transformation to convert a non-perfect nested loop into a perfect one. The above transformations have been verified with some nested loops selected from DSP compiler challenge C code. The preliminary results are also presented.
ICASSP98 Paper Abstract

Improving the Throughput of Flexible-Precision DSPs via Algorithm Transformation

Authors:
Manoj Aggarwal, *University of Illinois, (U.S.A.)*
Naresh R. Shanbhag, *University of Illinois, (U.S.A.)*
Narendra Ahuja, *University of Illinois, (U.S.A.)*

Volume 5, page 3069, paper no. 2313

Abstract:
In this paper, we have presented a systematic technique to improve throughput of signal/image processing algorithms when implemented on flexible precision hardware. Many image/signal processing algorithms need 8-16 bit precision while the DSPs available are of much higher precision (32-bit). Significant performance gain can be obtained if multiple low precision computations can be performed in one cycle of a high precision DSP. We have proposed a framework based on algorithm transformation techniques of unfolding and retiming to systematically map low precision algorithms onto high precision DSPs. The improvement in throughput obtained by this framework is linearly related to the ratio of precision used by the processor and that required by the algorithm. The efficacy of this technique has been demonstrated on a IIR filter. We have also established some theoretical bounds on the maximum throughput that can be achieved using the proposed methodology.
Loop Scheduling Algorithms for Power Reduction

Authors:
Zhihong Yu, University of Notre Dame, (U.S.A.)
Fei Chen, University of Notre Dame, (U.S.A.)
Edwin H.M. Sha, University of Notre Dame, (U.S.A.)

Volume 5, page 3073, paper no. 1819

Abstract:
The increasing demand for portable computing has elevated power consumption to be one of the most critical parameters for execution of loops which constitute most of the computation of scientific applications. The reduction of a schedule length is usually considered to be opposite to the reduction of power. This paper presents a novel loop pipelining approach to reduce power consumption while reducing the schedule length. Power consumption is measured by transition activity between operands of successive operations. Both initial scheduling and loop scheduling across iterations try to reduce the transition activity at the inputs to the functional units. A series of experiments show that our method achieves considerable power dissipation and schedule length reduction.
ICASSP98 Paper Abstract

Performance Evaluation of Register Allocator for the Advanced DSP of TMS320C80

Authors:
Jihong Kim, Seoul National University, (Korea)
Graham Short, Texas Instruments, (U.K.)

Abstract:
PPCA is an assembly language-level register allocator and instruction compactor for the Advanced DSPs (ADSPs) of the TMS320C80 digital signal processor. It was developed to help the implementation of time-critical ADSP assembly programs which heavily utilize powerful ADSP features optimized for multimedia and image computing applications for maximum efficiency. PPCA takes as an input ADSP assembly operations with symbolic variables. It then allocates the ADSP’s physical registers to the symbolic variables and rearranges the operations into a highly-parallelized compact format. In this paper, we have evaluated the performance of a register allocation capability of PPCA using an extensive image computing library for the TMS320C80. We present the basic algorithm of the PPCA’s register allocation module and describe the performance evaluation approach used. The result shows that PPCA essentially achieves optimal register allocation for the test cases based on the image computing library functions.
ICASSP98 Paper Abstract

Low-Power Reconfigurable Signal Processing via Dynamic
Algorithm Transformations (DAT)

Authors:
Manish Goel, University of Illinois, (U.S.A.)
Naresh R. Shanbhag, University of Illinois, (U.S.A.)

Volume 5, page 3081, paper no. 2504

Abstract:
Presented in this paper are dynamic algorithm transformation (DAT) for systematic design of reconfigurable computing engines. These techniques allow dynamic alteration of algorithm properties in response to input non-stationarities. The input is modeled as a set of states with an underlying probability distribution, \( P_S \). For each input state \( s \), a signal monitoring algorithm SMA computes a power-optimal configuration for the signal processing algorithm SPA block. A fraction \( \alpha \) of the SPA block is hardwired and the remaining \( 1 - \alpha \) is reconfigurable. Similarly, the SMA block computation is partitioned into a fraction \( \beta \) for the memory and the remaining \( 1 - \beta \) for the datapath. For the given input state distribution, the optimal values of \( \alpha (\alpha_{opt}) \) and \( \beta (\beta_{opt}) \) are determined. It is shown that for frequency selective filtering, the power savings of 35% - 45% can be achieved by DAT-based reconfigurable system as compared to the traditional design based on the worst-cased scenario.
ICASSP98 Paper Abstract

Pipelined Hogenauer CIC Filters Using Field-Programmable Logic and Residue Number System

Authors:
Antonio Garcia, University of Granada, (Spain)
Uwe Meyer-Baese, University of Florida, (U.S.A.)
Fred Taylor, University of Florida, (U.S.A.)

Volume 5, page 3085, paper no. 2052

Abstract:
Field-Programmable Logic (FPL) is on the verge of revolutionizing digital signal processing (DSP) in the manner that programmable DSP microprocessors did nearly two decades ago. While FPL densities and performance have steadily improved to the point where some DSP solutions can be integrated into a single FPL chip, they still have limited use in high-precision high-bandwidth applications. In this paper it is shown that in such cases, the residue number system (RNS) can be an enabling technology. The design of a high-decimation rate digital filter is presented which demonstrates the RNS-FPL synergy.
ICASSP98 Paper Abstract

Synthesis of Folded, Pipelined Architectures for Multi-Dimensional Multirate Systems

Authors:
Vijay Sundararajan, University of Minnesota, (U.S.A.)
Keshab K. Parhi, University of Minnesota, (U.S.A.)

Volume 5, page 3089, paper no. 1224

Abstract:
Motivated by the need for designing efficient architectures for two-dimensional discrete wavelet transforms (DWTs), this paper presents a novel multi-dimensional (MD) folding transformation technique which can be used to synthesize control circuits for pipelined architectures for a specific class of multirate MD digital signal processing (DSP) algorithms. Although a multirate MD DSP algorithm contains decimators and expanders which change the effective sample rate of a MD discrete time signal, MD folding time-multiplexes the algorithm to hardware in such a manner that the resulting synchronous architecture requires only a single clock signal for the clocking of the datapath. Feasibility constraints are derived for folding a 2-D data-flow graph (DFG) onto a given set of hardware functional units according to a specified schedule. Area/power efficient architectures are derived for 1-4 level 2-D discrete wavelet transforms (DWT) with 18.5%-23.3% savings in storage area.
ICASSP98 Paper Abstract

Minimization of Data Address Computation Overhead in DSP Programs

Authors:
Bernhard R Wess, University of Technology, Vienna, (Austria)
Martin Gotschlich, University of Technology, Vienna, (Austria)

Volume 5, page 3093, paper no. 1690

Abstract:
Digital signal processors (DSPs) provide dedicated data address generation units (AGUs) with multiple register files. These units allow data memory access by indirect addressing with automatic address modification. Typically, both linear and modulo addressing are supported. There is no address computation overhead if the next address is within the auto-modify range. Often, this range can be adapted to the application by assigning static values to modify registers. In this paper, we discuss optimized data memory address generation in DSP programs. Here the goal is to minimize data address computation and register initialization costs by optimizing data memory layout, address register assignment, and auto-modify range. The investigated combinatorial optimization problems can have an extremely large solution space. However, experimental results indicate that random neighbourhood sampling by simulated annealing allows to produce highly optimized solutions.
ICASSP98 Paper Abstract

A RISC Architecture with Uncompromised Digital Signal Processing and Microcontroller Operation

Authors:
Daniel Martin, Siemens Microelectronics Inc., (U.S.A.)
Robert E. Owen, Data/Time International, (U.S.A.)

Volume 5, page 3097, paper no. 1166

Abstract:
Digital signal processors are paired with microcontrollers in many applications. Various attempts have been made to combine the two processor functions in one architecture, but there have remained two unresolved conflicts. These are different data and program memory hierarchy choices in speed and size, and different real-time control needs. This paper reviews the basic processing requirements for digital signal processing (DSP) and controllers and shows how a new 32-bit RISC architecture has resolved these conflicts and successfully integrated the two functions seamlessly into one processor core. This is confirmed with a detailed FIR filter example. Major innovations in this Tricore architecture are a novel memory organization used along with variable instruction word sizes and multiple issuing of instructions.
ICASSP98 Paper Abstract

Real Time Execution of Optimal Edge Detectors on RISC and DSP Processors

Authors:
Lionel Lacassagne, LIS/Electronique-Informatique-Applications (EIA), (France)
Frantz Lohier, LIS/Electronique-Informatique-Applications (EIA), (France)
Patrick Garda, Laboratoire des Instruments et Systemes, (France)

Volume 5, page 3101, paper no. 1735

Abstract:
This paper presents the real time software implementation of the Canny-Deriche’s optimal edge detectors on RISC and high performance DSP processors. For each type of architecture, the most leading algorithmic and programming optimization techniques are described. We have shown that real time is achieved for 256x256 images on RISC processors and for 512x512 on state of the art DSPs. Those results outperform the best software and FPGA implementations of optimal edge detectors.
ICASSP98 Paper Abstract

**Fast 2D IDCT Implementation with Multimedia Instructions for a Software MPEG2 Decoder**

**Authors:**
Eri Murata, NEC Corporation, (Japan)
Masao Ikekawa, NEC Corporation, (Japan)
Ichiro Kuroda, NEC Corporation, (Japan)

**Abstract:**
This paper presents an implementation of a fast two-dimensional Inverse Discrete Cosine Transform (IDCT) with multimedia instructions for a software MPEG-2 decoder. IDCT algorithms for sparse blocks which eliminate the calculation for zero coefficients are realized by using multimedia instructions. To reduce the cycle count for IDCT, an adaptive control method for these IDCT algorithms, based on the bit rate and picture type, is proposed and its performance is described. In the implementation of a software MPEG-2 decoder, the execution time for IDCT is reduced to 10% by using MMX instructions from original C program. Moreover, using proposed adaptive control, it can be further be reduced to 76%.
ICASSP98 Paper Abstract

Associative Architecture for Fast DCT

Authors:
Yossi Shain, Associative Computing Ltd., (Israel)
Avidan Akerib, Associative Computing Ltd., (Israel)
Rutie Adar, Associative Computing Ltd., (Israel)

Volume 5, page 3109, paper no. 1572

Abstract:
This paper discusses an associative processor architecture designed to meet the demands of real-time image processing applications. In a single chip, this architecture provides thousands of processors - one for each pixel, in the form of associative memory. This paper focuses on a generic, proprietary associative processor architecture and discusses implementing the discrete cosine transform (DCT) using processors based on this architecture. Associative Computing Ltd. has developed a commercial associative chip based on this architecture, and while the DCT implementation discussed refers to future generations based on this architecture, reference is made throughout to the Company’s present processor. Processors based on our associative architecture can process the large amounts of data typically required in real-time imaging applications at a lower cost-performance ratio than conventional processors. The scalable nature of memory-based processor architecture allows developers to rapidly increase processing power without altering the fundamental processor, or system architecture. The underlying technologies used in the Company’s present processor can significantly facilitate the development of associative processing as an alternative to conventional processing for video applications including compression and video editing.
ICASSP98 Paper Abstract

Behavioral Synthesis Optimization Using Multiple Precision Arithmetic

Authors:
Milos Ercegovac, University of California, Los Angeles, (U.S.A.)
Darko Kirovski, University of California, Los Angeles, (U.S.A.)
George Mustafa, University of California, Los Angeles, (U.S.A.)
Miodrag Potkonjak, University of California, Los Angeles, (U.S.A.)

Volume 5, page 3113, paper no. 2548

Abstract:
Modern image and video processing applications are characterized by a unique combination of arithmetic and computational features: fixed point arithmetic, a variety of short data types, high degree of instruction-level parallelism, strict timing constraints, high computational requirements, and high cost sensitivity. The current generation of behavioral synthesis tools does not address well this type of application. In this paper we explore the potential of using multiple precision arithmetic units to effectively support implementation of image and video processing applications as application specific integrated circuits. A new architectural scheme for collaborative addition of sets of variable precision data is proposed as well as an allocation and assignment methodology for multiple precision arithmetic units. Experimental results indicate strong advantages of the proposed approach.
ICASSP98 Paper Abstract

4-Way Superscalar DSP Processor for Audio Codec Applications

Authors:
Joon Seok Kim, Yonsei University, (Korea)
Sun Kook Yoo, Yonsei University, (Korea)
Sung Wook Park, Yonsei University, (Korea)
Nam Hoon Jung, Yonsei University, (Korea)
Woo Suk Ko, Yonsei University, (Korea)
Keun Sup Lee, Yonsei University, (Korea)
Dae Hee Youn, Yonsei University, (Korea)

Volume 5, page 3117, paper no. 1516

Abstract:
The recent audio CODEC (Coding/Decoding) algorithms are complex of several coding techniques, and can be divided into DSP tasks, controller tasks and mixed tasks. The traditional DSP processor has been designed for fast processing of DSP tasks only, but not for controller and mixed tasks. This paper presents a new architecture that achieves high throughput on both controller and mixed tasks of such algorithms while maintaining high performance for DSP tasks. The proposed processor, YSP-3, operates four functional units (Multiplier, two ALUs, Load/Store Unit) in parallel via 4-issue super-scalar instruction structure. The performance evaluation of YSP-3 has been done through the implementation of the common DSP algorithms and AC-3 decoder.
ICASSP98 Paper Abstract

A Low-Power DSP Core Architecture for Low Bitrate Speech Codec

Authors:
Hiroyuki Okuhata, Osaka University, (Japan)
Morgan H Miki, Osaka University, (Japan)
Takao Onoye, Osaka University, (Japan)
Isao Shirakawa, Osaka University, (Japan)

Volume 5, page 3121, paper no. 1905

Abstract:
A VLSI implementation of a low-power DSP is described, which is dedicated to the G.723.1 low bitrate speech codec. A number of sophisticated DSP microarchitectures are devised mainly on dual multiply accumulators, rounding and saturation mechanisms, and two-banked on-chip memory. The proposed DSP architecture has been integrated in the total area of $7.75 \text{mm}^2$ by using a 0.35um CMOS technology, which can operate at 10MHz with the dissipation of 45mW from a single 3V supply.
ICASSP98 Paper Abstract

Number Representations for Reducing Switched Capacitance in Subband Coding

Authors:
John R Sacha, The Pennsylvania State University, (U.S.A.)
Mary Jane Irwin, The Pennsylvania State University, (U.S.A.)

Volume 5, page 3125, paper no. 2180

Abstract:
In low power VLSI design, fixed point number representations are standard. For some signal processing applications, however, achieving sufficient dynamic range with fixed point may lead to computations utilizing more precision than necessary. In such cases, trading precision for dynamic range through the use of floating point and logarithmic number system representations can potentially provide power savings. This is demonstrated for a subband speech coding application using architectural-level capacitance modeling.
ICASSP98 Paper Abstract

An Architectural Study of a Digital Signal Processor for Block Codes

Authors:
Wolfram K Drescher, Dresden University of Technology, (Germany)
Menno Mennenga, Dresden University of Technology, (Germany)
Gerhard P Fettweis, Dresden University of Technology, (Germany)

Volume 5, page 3129, paper no. 2424

Abstract:
This paper examines architectural issues for a domain specific digital signal processor (DS-DSP) which is capable of fast decoding of block codes. In real time systems it was not possible before to employ common processors for this task because of a lack of architectural and arithmetical support. We proposed solutions for the arithmetical problem in previous work. In this paper we focus on architectures for implementation of different block decoding algorithms on a new DS-DSP architecture. The paper also contains benchmarks for our architecture for some selected codes and compares our DS-DSP to common digital signal processors (DSP) and dedicated logic solutions.
Techniques for Intellectual Property Protection of DSP Designs

Authors:
Inki Hong, University of California, Los Angeles, (U.S.A.)
Miodrag Potkonjak, University of California, Los Angeles, (U.S.A.)

Abstract:
Recently, numerous watermarking-based techniques for intellectual property protection of DSP artifacts, such as images, compressed and uncompressed audio and video data, and text documents have been proposed. However, the applicability of all techniques proposed until now are limited to digital data and they either implicitly or explicitly exploit the imperfection of human perception to audio and video. We propose the first watermarking technique for protecting the intellectual property of DSP designs. The essence of the technique is the use of additional synthesis constraints to encode the authorship signature. The constraints are selected in such a way that they result in minimal hardware overhead while embedding the signature which is unique and difficult to detect and remove. The technique is applicable to all levels of design process, from the algorithm, system and behavioral synthesis to logic synthesis and physical design levels. The technique is illustrated on a set of DSP design examples on all levels of design process.
ICASSP98 Paper Abstract

Design Space Exploration for Future TriMedia CPUs

Authors:
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Bram Riemens, Philips Research, (The Netherlands)
Kees Vissers, Philips Research, (The Netherlands)
Selliah Rathnam, Philips Semiconductors, (U.S.A.)
Gert Slavenburg, Philips Semiconductors, (U.S.A.)

Abstract:
It is widely recognized that fine-grain parallelism can greatly enhance a processor's performance for signal processing applications. For this reason, future generation TriMedias will combine VLIW and subword parallelism in a single CPU. In this article, we present a snapshot of the new CPU's design process: the outlines are clear but fine tuning is still ongoing. We present the design flow and 'work bench' that the designers use for further tuning.
ICASSP98 Paper Abstract

Software MPEG-2 Video Decoder on a 200-Mhz, Low-Power Multimedia Microprocessor

Authors:
Kouhei Nadehara, NEC Corporation, (Japan)
Hanno Lieske, University of Hannover, (Germany)
Ichiro Kuroda, NEC Corporation, (Japan)

Volume 5, page 3141, paper no. 1911

Abstract:
This paper presents a low-power, 32-bit RISC microprocessor with a 64-bit “single-instruction multiple-data” multimedia coprocessor, V830R/AV, and its MPEG-2 video decoding performance. This coprocessor basically performs multimedia-oriented four 16-bit operations every clock, such as multiply-accumulate with symmetric rounding and saturation, and accelerates computationally intensive procedures of the video decoding; an 8x8 IDCT is performed in 201 clocks. The processor employs the Concurrent Rambus DRAM interface, and facilities for controlling cache behaviors explicitly by software to speed up enormous memory accesses necessary to motion compensation. The 200-MHz V830R/AV processor with the 600-Mbyte/sec. Concurrent Rambus DRAMs decodes MPEG-2 MP@ML video in real-time (30 frames/sec.).
ICASSP98 Paper Abstract

Codevelopment of the TMS320C6x VelociTI Architecture and Compiler

Authors:
Ray Simar Jr, Texas Instruments, (U.S.A.)

Volume 5, page 3145, paper no. 1762

Abstract:
Continuing dramatic improvements in semiconductor manufacturing processes are enabling radical new signal-processing architectures at the chip level. The development of these new architectures must be coupled with clearly defined target applications, a thorough analysis of applicable signal processing algorithms, and significant advancements in code-generation technology. The TMS320C6x development program involved the codevelopment of the VelociTI architecture, a new code-generation capability, and a large set of representative benchmarks.
ICASSP98 Paper Abstract

TI DSP Implementation of A Medium Speed DSL (MDSL) for Multimedia Applications

Authors:
Song Wu, Texas Instruments, (U.S.A.)
Xiaolin Lu, Texas Instruments, (U.S.A.)
Walter Chen, Texas Instruments, (U.S.A.)

Volume 5, page 3149, paper no. 2150

Abstract:
High performance general purpose Digital Signal Processor (DSP) provides a cost efficient solution for broadband Digital Subscriber Line (DSL) transceiver. A DSL modem with a transmission throughput between 400 kbps and 2 Mbps operating over most of existing telephone subscriber loops has been implemented on a single TI TMS320c548 DSP for consumer multimedia applications such as internet access. Except the analog front end (AFE), all the Discrete Multitone Modem (DMT) algorithms are implemented with DSP software. DSP based software DSL modem also provides a convenient interface to Microsoft point-to-point protocol (PPP) for network access.
ICASSP98 Paper Abstract
A Flexible Processor Architecture for MPEG-4 Image Compositing

Authors:
Mladen Berekovic, Laboratorium für Informationstechnologie, (Germany)
Rainer Frase, Laboratorium für Informationstechnologie, (Germany)
Peter Pirsch, Laboratorium für Informationstechnologie, (Germany)

Abstract:
This paper proposes a new array architecture for MPEG-4 image compositing. The emerging MPEG-4 standard for multimedia applications allows script-based compositing of audiovisual scenes from multiple audio and visual objects at the decoder side. A coprocessor architecture is presented that works in parallel to an MPEG-4 video- and audio-decoder, and performs computation and bandwidth intensive low-level tasks for image compositing. The processor consists of an SIMD array of 16 DSPs to reach the required processing power for real-time image warping, alpha blending and 3D rendering tasks. A programmable architecture allows to adapt processing resources to the specific needs of different tasks and applications. The processor has an object-oriented cache architecture with 2D virtual address space (e.g. textures), that allows concurrent and conflict-free access to shared data objects for all 16 DSPs. Especially I/O intensive tasks like texture mapping, alpha blending, image warping, z-buffer and shading algorithms benefit from shared memory caches and the possibility to preload data before it is accessed.
ICASSP98 Paper Abstract

Multimedia Applications of Microprocessor with Embedded DRAM

Authors:
Yasuhiro Nunomura, Mitsubishi Electric Corporation, (Japan)
Toru Shimizu, Mitsubishi Electric Corporation, (Japan)
Kazunori Saitoh, Mitsubishi Electric Corporation, (Japan)
Koji Tsuchihashi, Mitsubishi Electric Corporation, (Japan)

Volume 5, page 3157, paper no. 1553

Abstract:
The M32R/D is a 32-bit microprocessor with large-capacity on-chip DRAM. It consists of a 32-bit RISC CPU, a 32-bit x 16-bit multiply and accumulator (MAC), either 1-Mbyte or 2-Mbyte DRAM, 4-Kbyte cache memory, and a memory controller. The CPU, DRAM, and cache memory are connected via a 128-bit 66.6 MHz internal bus yielding high performance and low power dissipation. The chip is capable of coping with a wide range of applications and thus provides system designer with great flexibility. For instance, a portable multimedia system can be realized by only three chips: an M32R/D chip, an I/O ASIC chip, and programming ROM. This means that a total system solution can be achieved at a lower cost with higher performance. Personal digital assistants (PDAs) and digital still cameras are such examples.
ICASSP98 Paper Abstract
Blind Multi-User MMSE Detection of CDMA Signals

Authors:
David J. Gesbert, Stanford University, (U.S.A.)
Joakim Sorelius, Uppsala University, (Sweden)
Arogyaswami J. Paulraj, Stanford University, (U.S.A.)

Volume 6, page 3161, paper no. 2336

Abstract:
The recovery of code-division multiple access (CDMA) information signals in a frequency selective fading channel is a problem of great theoretical and practical interest. This paper addresses the estimation of an optimal (within the class of linear detectors) multi-user CDMA receiver. A novel approach is introduced that enables the estimation of the minimum mean-square error (MMSE) detector in a blind setting. The MMSE detector is obtained through a double subspace projection that exploits the subspace structure associated with both the code of the desired user and the estimated signal subspace of the covariance matrix for the observed signals. The technique allows for interference rejection without requiring the knowledge of the codes for the interferers.
ICASSP98 Paper Abstract
Closed-Form Blind Identification of MIMO Channels

Authors:
João MM. F. Xavier, Instituto Superior Técnico, (Portugal)
Victor A.N. Barroso, Instituto Superior Técnico, (Portugal)
José M.F. Moura, CMU, (U.S.A.)

Abstract:
We present a closed-form algorithm for blind identification of multiple-input/multiple-output (MIMO) finite-impulse response (FIR) systems driven by digital sources. The algorithm is based on second order statistics and yields an asymptotically exact estimate of the MIMO channel. We assign distinct spectral signatures to each user through transmitter correlative filters, and exploits this spectral asymmetry to derive the closed-form solution. Simulation results illustrate the good performance of the proposed approach. We compare the mean square error (MSE) of the MIMO channel estimate against the Cramer-Rao bound, and assess the algorithm capability in rejecting inter-user crosstalk interference.
ICASSP98 Paper Abstract
Blind Adaptive Interference Suppression in DS-CDMA Communications with Impulsive Noise

Authors:
Xiaodong Wang, Princeton University, (U.S.A.)
H. Vincent Poor, Princeton University, (U.S.A.)

Volume 6, page 3169, paper no. 1325

Abstract:
In many wireless systems where multiuser detection techniques may be applied, the ambient channel noise is known through experimental measurements to be decidedly non-Gaussian, due largely to impulsive phenomena. The performance of many multiuser detectors can degrade substantially in the presence of such impulsive ambient noise. In this paper, a blind adaptive robust multiuser detection technique is developed for combating both multiple-access interference and impulsive noise in CDMA communication systems. This technique is nonlinear in nature and it is based on the signal subspace tracking method and the $M$-estimation method for robust regression. It is seen that the proposed technique offers significant performance gain over linear adaptive multiuser detectors in impulsive noise, with little attendant increase in computational complexity.
ICASSP98 Paper Abstract

Blind 2-D Rake Receivers Based on RLS-Type Space-Time Adaptive Filtering For DS-CDMA System

Authors:
Yung-Fang Chen, Purdue University, (U.S.A.)
Michael D. Zoltowski, Purdue University, (U.S.A.)

Abstract:
We previously presented a blind 2D RAKE receiver for CDMA that cancels strong multi-user access interference and optimally combines multipath. The weight vector yielding the optimum signal to interference plus noise ratio for bit decisions is the “largest” generalized eigenvector of the spatio-frequency (spatio-temporal) correlation matrix pencil. However, the eigen-analysis based algorithm is on the order of $O(N^3)$ computational complexity and the resulting spatio-frequency (spatio-temporal) correlation matrix pencil is of large dimension. This detracts from the real-time applicability of that scheme. A blind 2-D RAKE receiver is thus presented based on an RLS-type space-time adaptive filtering scheme which offers $O(N^2)$ computational complexity and competitive performance. The applicability of the scheme to the IS-95 uplink is also addressed as in a decision directed fashion.
A New Blind Zeroforcing Equalizer for Multichannel Systems

Abstract:
Blind channel equalization has recently been a very active research topic due to its potential application in mobile communications and digital TV systems. In this paper, we present a new blind zero-forcing equalizer that utilizes second order statistics from the multi-channel configuration. The algorithm is simple and relies only on nullspace decomposition. It can actively select the desired delay of the equalizer output signal. The performance of this new algorithm is demonstrated through simulation examples.
ICASSP98 Paper Abstract

Convergence Properties of Blind Algorithms for Base Station CDMA Receivers

Authors:
Alex Stéphenne, INRS-Telecommunications, (Canada)
Benoit Champagne, INRS-Telecommunications, (Canada)

Volume 6, page 3181, paper no. 1814

Abstract:
In this paper, the blind estimation of wireless CDMA receiver coefficients from the second order statistics of the signals is considered. Although many algorithms have been proposed so far, their performance analysis has always been carried out assuming perfect receiver coefficients estimation and/or under time-invariant conditions. In this article, we present some decision-directed blind algorithms and use a time-varying vector channel simulator to compare their performance with those of many recently proposed algorithms. It is shown that decision-directed chip-level algorithms can operate without the use of training sequences to avoid catastrophic error propagation, and that one should not expect an increase in performance from using least squares instead of least-mean-square. Furthermore the unpracticability of the bit-level algorithm and of the least significant algorithm under time varying environment is outlined. The performance of the principal component (Stanford) algorithm is also studied.
ICASSP98 Paper Abstract

Blind and Semi-Blind Maximum Likelihood Methods for FIR Multichannel Identification

Authors:
Jaouhar Ayadi, Institut EURECOM, (France)
Elisabeth De Carvalho, Institut EURECOM, (France)
Dirk T.M. Slock, Institut EURECOM, (France)

Abstract:
We investigate Maximum Likelihood (ML) methods for blind and semi-blind estimation of multiple FIR channels. Two blind Deterministic ML (DML) strategies are presented. In the first one, we propose to modify the Iterative Quadratic ML (IQML) algorithm in order to "denoise" it and hence obtain consistent channel estimates. The second strategy, called Pseudo-Quadratic ML (PQML), is naturally asymptotically denoised. Links between these two approaches are established and their global convergence is proved. Furthermore, we propose semi-blind ML techniques combining PQML with two different training sequence estimation methods and compare their performance. These semi-blind techniques, exploiting the presence of known symbols, outperform their blind version. They also allow channel estimation in situations where blind and training sequence methods fail separately. Simulations are presented to demonstrate the performance of all the proposed algorithms, and comparisons between them are discussed in a blind and/or semi-blind context.
ICASSP98 Paper Abstract
The Blind QRD-DMS Beamformer and its VLSI Systolic Designs for DS/CDMA Systems

Authors:
Shang-Chieh Liu, University of Maryland, (U.S.A.)
Evaggelos Geraniotis, University of Maryland, (U.S.A.)

Volume 6, page 3189, paper no. 1505

Abstract:
In this paper, we present a blindly adaptive beamforming algorithm which is based on the second order interference estimation to maximize the received SINR. Using the desired signature code and the orthogonal to it code, a new code filter is introduced to decompose the receiving signals into two parts: desired information and interference. The above method motivates us to develop the QR-decomposition based dominant eigen mode search (QRD-DMS) algorithm which is more numerically stable than the DMS one. The corresponding wavefront systolic architecture is also proposed for the VLSI implementations. Compare to the families of minimum mean square error (MMSE) algorithms which need training sequences, we have completed the maximum SINR families, as shown in Table 1, QRD-DMS method which not only blindly updates the beamforming weights and converges as fast as the QRD-RLS method.
ICASSP98 Paper Abstract

Coding Enhanced Joint Detection for Multiple Access Communications

Authors:
Rachel E Learned, Sanders, A Lockheed Martin Company, (U.S.A.)
Andrew C Singer, Sanders, A Lockheed Martin Company, (U.S.A.)

Volume 6, page 3193, paper no. 1259

Abstract:
A low complexity approach to coding-enhanced multi-user detection is developed to mitigate the problems associated with near-far effect and to permit a more efficient assignment of channel resources relative to current multiple access (MA) communications systems. Through prudent integration of error correction decoding, multi-user and inter-symbol interference equalization and stripping, a low complexity near-far resistant multi-user joint detector/decoder has been developed which exhibits significant performance gains relative to the best known low complexity joint detection/decoding procedures reported in recent literature. Empirical analysis of the coding-enhanced multi-user detector (CMD) for a case of heavy inter-symbol interference and multi-user interference shows a 3 dB improvement over these best known methods.
ICASSP98 Paper Abstract

An Analysis of a Modulated Orthogonal Sequence

Authors:
Seong Ill Park, Korea Advanced Institute of Science and Technology, (Korea)
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Kwang Soon Kim, Korea Advanced Institute of Science and Technology, (Korea)
Naoki Suehiro, University of Tsukuba, (Japan)
Jooshik Lee, Korea Advanced Institute of Science and Technology, (Korea)

Volume 6, page 3197, paper no. 1311

Abstract:
In this paper, a new generation and an information construction methods for a modulated orthogonal sequence are suggested: these sequence is generated by only integer sums and modular techniques. The autocorrelation and cross-correlation characteristics of the sequence are investigated via a new procedure. A modified sequence also having the orthogonality and satisfying the mathematical lower bound of the cross-correlation is proposed, and the symbol error probability of the sequence is investigated.
ICASSP98 Paper Abstract
Subspace-based Detection for CDMA Communications

Authors:
Yong-Suk Moon, University of Minnesota, (U.S.A.)
Mostafa Kaveh, University of Minnesota, (U.S.A.)
Laurie B Nelson, University of Minnesota, (U.S.A.)

Volume 6, page 3201, paper no. 1584

Abstract:
The oblique projection supports the framework to resolve a signal space into desired signal and interference subspaces. This paper presents subspace-based detection methods using the oblique projection for the CDMA channel. For the synchronous case, it is shown that this detector represents the geometrical form of the decorrelating detector, and performs a complete rejection of interfering signals. This paper also suggests the approach of combining the subspace-based detection with the MUSIC algorithm for an asynchronous CDMA channel. It is shown that the BER performance of this detection approach, which depends on the accuracy of the code timing acquisition, is better than that of the blind adaptive demodulation technique.
ICASSP98 Paper Abstract
Adaptive Reception of Wireless CDMA Signals Using Empirical Detection

Authors:
Lin Yue, Rice University, (U.S.A.)
Don H. Johnson, Rice University, (U.S.A.)

Volume 6, page 3205, paper no. 1260

Abstract:
Channel characteristics of practical code division multiple access (CDMA) systems are usually unknown and difficult to model accurately. Type-based receivers, without assuming any a priori model, extract signals successfully from background noise. In this paper, we develop type-based receivers that address two major issues in CDMA signal reception: multiple access interference and multipath fading. We first present the type-based receiver with interference suppression capability, assuming the knowledge of the code and timing of the intended user only. We then show that equal-gain combining (EGC) of type-based statistics is the asymptotically optimal technique for diversity empirical detection. Compared with maximal ratio combining of matched filter outputs, the diversity receiver with EGC of type-based statistics assumes less channel knowledge and yields competitive detection performance in Gaussian noise and better performance in Laplacian noise.
ICASSP98 Paper Abstract
Reduced Complexity M-ary Hypotheses Testing in Wireless Communications

Authors:
Mohammed H Nafie, University of Minnesota, (U.S.A.)
Ahmed H. Tewfik, University of Minnesota, (U.S.A.)

Volume 6, page 3209, paper no. 2364

Abstract:
We present a progressive refinement approach to M-ary detection problems. The approach leads on average to a logarithmic reduction in the complexity of the detector. It relies on designing binary decision trees that trade complexity with probability of error. We also discuss simplified solutions that can be used in several cases of interest in wireless communications such as CDMA multiuser detection and blind equalization.
ICASSP98 Paper Abstract

A Recent Subspace-Based Approach to Joint Detection and Time-delay Estimation in DS-CDMA Systems

Authors:
Anders Ranheim, Chalmers University of Technology, (Sweden)

Abstract:
This paper presents a novel method for jointly estimating the time-delay parameters and detecting the transmitted symbols in a DS-CDMA system. A short training sequence is used to get an initial estimate of the time-delay, which is consequently used to detect the symbols. The method then iterates while exploiting signal structure, to improve the performance. Simulation results are presented to compare the algorithm with the decorrelating criterion and the matched-filter receiver in terms of bit-error rate, and with the MUSIC algorithm and the sliding correlator in terms of the variance of the time-delay estimates.
ICASSP98 Paper Abstract

Adaptive H-infinity Delay Tracking in Asynchronous DS-CDMA Systems

Authors:
A. Manikas, Imperial College, (U.K.)
S. S. Lim, Imperial College, (U.K.)
P. Wilkinson, Imperial College, (U.K.)

Volume 6, page 3217, paper no. 5205

Abstract:
In this paper a new H-infinity-type delay acquisition/tracking approach is proposed which is based on partitioning of the PN-code matrix of the CDMA users into submatrices which are then used to form a 'state-space' H-infinity-type linear combiner. The proposed novel formulation and approach is robust to modelling errors such as over- and underestimation of the number of signals present and provides a powerful near-far resistant delay-tracking solution in a multiuser DS-CDMA signal environment.
ICASSP98 Paper Abstract
Efficient Implementation of Linear Multiuser Detectors

Authors:
Xiao-feng Wang, University of Victoria, (Canada)
Wu-sheng Lu, University of Victoria, (Canada)
Andreas Antoniou, University of Victoria, (Canada)

Volume 6, page 3221, paper no. 1603

Abstract:
A recursive algorithm for updating linear multiuser detectors in code-division multiple-access systems is proposed. Based on this algorithm, a window-based implementation with a signal-based criterion for determining the window length is developed. Performance analysis and numerical experiments are conducted that show the merits of the proposed implementation method.
ICASSP98 Paper Abstract

Implementation of Linear Multiuser Detectors for Asynchronous CDMA Systems by Linear Multi-stage Interference Cancellation

Authors:
Harald Elders-Boll, Aachen University of Technology, (Germany)
Hans Dieter Schotten, Aachen University of Technology, (Germany)
Axel Busboom, Aachen University of Technology, (Germany)

Abstract:
The decorrelating and the linear, minimum mean-squared error (MMSE) detector for asynchronous code-division multiple-access communications ideally are infinite memory-length detectors. Finite memory approximations of these detectors require the inversion of a correlation matrix whose dimension is given by the product of the number of active users and the length of the processing window. With increasing number of active users or increasing length of the processing window, the calculation of the inverse may soon become numerically very expensive. In this paper, we prove that the decorrelating and the linear MMSE detector can both be realized by linear multi-stage interference cancellation algorithms with ideally an infinite number of stages. It will be shown that for serial multi-stage interference cancellation, depending on the signal-to-noise ratio and the number of active users, only a few stages are necessary to obtain the same BER performance as the ideal detectors. Thus, the complexity can be reduced considerably.
ICASSP98 Paper Abstract

Pipelined Implementation of Adaptive Multiple-Antenna CDMA Mobile Receivers

Authors:
Ramin Baghaie, Helsinki University of Technology, (Finland)
Stefan Werner, Helsinki University of Technology, (Finland)
Timo I. Laakso, Helsinki University of Technology, (Finland)

Volume 6, page 3229, paper no. 1660

Abstract:
Pipelined implementation of adaptive Direct-Sequence Code Division Multiple Access (DS-CDMA) receiver is proposed when multiple antennas are utilized for mobile communications. Adaptive multiple-antenna receivers can provide insensitivity to the interfering powers and room for more users or require smaller number of antennas than the matched filter solution. In this paper, a number of approximation techniques are utilized to pipeline the adaptive algorithm used for the proposed multiple-antenna receiver. The resulting pipelined receiver requires minimal hardware increase and achieves a higher throughput or requires lower power as compared to the receiver using the serial algorithm. Simulation results illustrate the signal-to-interference (SIR) versus the relative interfering power for different number of antennas and different levels of pipelining.
ICASSP98 Paper Abstract

Branch-Hopped Wavelet Packet Division Multiplexing

Authors:
Timothy N Davidson, McMaster University, (Canada)
Anne-Jeanne Schott, McMaster University, (Canada)
K M Wong, McMaster University, (Canada)

Volume 6, page 3233, paper no. 2295

Abstract:
Wavelet Packet Division Multiplexing (WPDM) is a high-capacity, flexible and robust orthogonal multiplexing technique in which wavelet packet basis functions are chosen as the coding waveforms. By analogy with frequency-hopped communication schemes, incorporation of time variation into the WPDM scheme offers the potential for further performance improvements, especially in frequency-selective fading channels. We consider a 'Branch-Hopped' WPDM scheme which employs an efficient modular switched transmultiplexer structure to induce the time variation. We determine classes of 'slow' and 'fast' hopping schemes analogous to their frequency-hopped counterparts, and evaluate several switching strategies for the transmultiplexer. For a given switching strategy we then design the filters within the transmultiplexer modules to provide further robustness to frequency-selective fading channels.
ICASSP98 Paper Abstract

Joint Multipath-Doppler Diversity in Fast Fading Channels

Authors:
Akbar M Sayeed, University of Wisconsin-Madison, (U.S.A.)
Behnaam Aazhang, Rice University, (U.S.A.)

Volume 6, page 3237, paper no. 2102

Abstract:
We present a new framework for code-division multiple access (CDMA) communication over fast fading mobile wireless channels. The performance of the RAKE receiver, which is at the heart of existing CDMA systems, degrades substantially under fast fading encountered in many mobile scenarios. Due to the time-varying nature of the fast fading channel, we employ joint time-frequency processing, which is a powerful approach to time-varying signal processing. Whereas the RAKE receiver exploits multipath diversity to combat fading, our framework is based on joint multipath-Doppler diversity facilitated by a fundamental time-frequency decomposition of the channel into independent flat fading channels. Diversity processing is achieved by a time-frequency generalization of the RAKE receiver which can be leveraged into several important aspects of system design. Performance analysis shows that CDMA systems based on the time-frequency RAKE receiver, due to their inherently higher level of diversity, can potentially deliver significant performance gains over existing systems.
ICASSP98 Paper Abstract

A Blind Joint Estimator for Multipath Diversity and PN Timing Error in Direct-Sequence Spread-Spectrum Receivers

Authors:
Jia-Chin Lin, National Taiwan University, (Taiwan)

Volume 6, page 3241, paper no. 1525

Abstract:
A blind joint estimator for multipath diversity and PN code timing error is proposed in this paper for direct-sequence spread-spectrum signaling on a frequency-selective fading channel. In the multipath diversity combiner, a modified known modulus adaptive (KMA) algorithm is used to cope with time-varying multipath effects and to perform multipath diversity combining in the blind mode. In the code timing recovery, the timing error signal is extracted from each propagation path independently and also combined in the same fashion as the multipath diversity combining process. By taking advantage of the inherent diversity based on a known modulus adaptive (KMA) algorithm, this modified code timing recovery can avoid the problem due to the drift or flutter effects of the error signals, and provide better performance on frequency selective fading channels. Extensive computer simulation results have verified the analysis and indicated very attractive behavior of the proposed joint estimator for multipath diversity and PN code timing error.
ICASSP98 Paper Abstract

A Receiver Diversity Based Code-Timing Estimator for Asynchronous DS-CDMA Systems

Authors:
Zheng-She Liu, University of Florida, (U.S.A.)
Jian Li, University of Florida, (U.S.A.)
Scott L Miller, University of Florida, (U.S.A.)

Volume 6, page 3245, paper no. 1009

Abstract:
We propose a receiver diversity based code-timing estimator for DS-CDMA systems. The systems are assumed to work in a flat fading and near-far environment, where an arbitrary antenna array is used at the receiver of the system to achieve the spatial diversity. We show that by utilizing the information collected via multiple antenna sensors, the length of the training sequences can be greatly reduced. We also show that the algorithm is an asymptotic maximum likelihood estimator. As a result, the mean-squared error of the code-timing estimates obtained by the algorithm approaches the Cramer-Rao lower bound as the length of the training sequence increases. Moreover, the algorithm does not require the search over a parameter space and the code-timing is obtained by rooting a second-order polynomial, which is computationally very efficient.
ICASSP98 Paper Abstract

Fixed Point Error Analysis of Multiuser Detection and Synchronization Algorithms for CDMA Communication Systems

Authors:
Chaitali Sengupta, Rice University, (U.S.A.)
Suman Das, Rice University, (U.S.A.)
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Behnaam Aazhang, Rice University, (U.S.A.)

Volume 6, page 3249, paper no. 2205

Abstract:
Conventional correlation based single-user techniques for Direct Sequence Code Division Multiple Access (DS-CDMA) wireless communication systems is susceptible to performance degradation due to interference from other users. Recent research has focused on development of several multiuser techniques where information about multiple users is used to improve performance for each individual user. Due to performance benefits of these methods, they are attractive candidates for implementation in future cellular systems. In this paper we present an error analysis of fixed point implementation of some of these techniques.
ICASSP98 Paper Abstract

Real Time Implementation of a Symbol Timing Recovery Algorithm for a Narrowband Wireless Modem

Authors:
Jimm Grimm, Purdue University, (U.S.A.)
Ramesh N Kumar, Purdue University, (U.S.A.)
Julius Kusuma, Purdue University, (U.S.A.)
James V Krogmeier, Purdue University, (U.S.A.)

Volume 6, page 3253, paper no. 2266

Abstract:
This paper examines some of the complexity issues arising in the real time implementation of the symbol timing subsystem of a narrowband wireless modem. The modem prototype has been designed for 4 kHz channels in the 220-222 MHz land mobile band. In an effort to achieve high bandwidth efficiency, the modem architecture employs transmitter diversity, pilot symbol assisted modulation, and trellis coded modulations. An experimental, non-real time system has been implemented and extensively field tested demonstrating bandwidth efficiencies in excess of 3 bits per second per Hz. The work on this project is currently focused on the real-time implementation of the baseband receiver functions using the Texas Instruments C54x fixed point digital signal processor. Here we report on some of the performance/complexity tradeoffs present in the design of a DSP implementation of the symbol timing recovery. Symbol timing is the first function performed in the baseband receiver.
ICASSP98 Paper Abstract

Broadband Nonstationary Interference Excision for Spread
Spectrum Communications Using Time-Frequency Syn-
thesis

Authors:
Stephen R. Lach, Villanova University, (U.S.A.)
Moeness G. Amin, Villanova University, (U.S.A.)
Alan R. Lindsey, Rome Laboratory, (U.S.A.)

Abstract:
A new method is introduced for interference excision in spread spectrum communications. Time-
frequency synthesis techniques are used to synthesize the nonstationary jammer from the time-frequency
domain using least-squares methods. The synthesized jammer is then subtracted from the incoming
data in the time domain, leading to increased signal to interference ratio at the input of the correlator.
The paper focuses on jammers with constant modulus where the jamming signal is a polynomial phase.
With this apriori knowledge, the jammer signal amplitude is restored by projecting each sample of the
synthesized signal to a circle representing its constant modulus. With the phase matching provided by
the least-squares synthesis method and amplitude matching underlying the projection operation, the
paper shows a significant improvement in receiver performance/bit error rates over the case where no
projection is performed.
ICASSP98 Paper Abstract

A New Non-Linear System for Estimating and Suppressing Narrowband Interference in PN Spread Spectrum Modulation

Authors:
Ana I. Perez-Neira, Universitat Politecnica de Catalunya, (Spain)
Joan Roca, Universitat Politecnica de Catalunya, (Spain)
Miguel Angel Lagunas, Universitat Politecnica de Catalunya, (Spain)

Volume 6, page 3261, paper no. 1942

Abstract:
This work develops a novel dynamic fuzzy logic system that, based on a fuzzy basis function expansion, successfully solves the non-linear problem of narrowband interference prediction and rejection in DS-SS. A fuzzy basis function representation provides a natural framework for combining both numerical and linguistic information in a uniform fashion. The result is a low complexity non-linear adaptive line enhancer, which offers a faster convergence rate and an overall better performance over other well-known non-linear line enhancers.
ICASSP98 Paper Abstract

A Novel Time-Frequency Exciser in Spread Spectrum Communications for Chirp-Like Interference

Authors:
Aykut Bultan, New Jersey Institute of Technology, (U.S.A.)
Ali N. Akansu, New Jersey Institute of Technology, (U.S.A.)

Volume 6, page 3265, paper no. 5236

Abstract:
A novel time-frequency exciser is developed for the removal of chirp-like interferences in direct sequence spread spectrum (DSSS) communications. The chirplet decomposition iteratively expands signals in terms of time-frequency localized waveforms. The interference signal components are highly correlated with chirplets, and are represented by a few of the highest energy components of the decomposition. These components are excised from the received signal, and an excised signal goes through a detector for a decision. The proposed time-frequency exciser outperforms the conventional Fourier transform based excisers for chirp-like interference classes.
ICASSP98 Paper Abstract

A Comparative Performance Evaluation of DMT (OFDM) and DWMT (DSBMT) Based DSL Communications Systems for Single and Multitone Interference

Authors:
Ali N. Akansu, New Jersey Institute of Technology, (U.S.A.)
Xueming Lin, New Jersey Institute of Technology, (U.S.A.)

Volume 6, page 3269, paper no. 2536

Abstract:
Multicarrier modulation techniques have been proposed in the digital subscriber line (DSL) applications. In this paper, the performance of DMT (OFDM) and DWMT (DSBMT) techniques for single and multitone interference are investigated. It is shown that a DMT system is sensitive to the location of narrow band interference. DMT technique needs additional narrow band interference canceller before forward FFT transform for performance improvements. In DSBMT technique, due to a limited spectral overlap between its subcarriers, single (multi)-tone interference could effect only a few subchannels which correspond to these interferences. DSBMT has a superior performance than DMT.
ICASSP98 Paper Abstract
Low-Complexity Digital Encoding Strategies for Wireless Sensor Networks

Authors:
Haralabos C. Papadopoulos, MIT, (U.S.A.)
Gregory W. Wornell, MIT, (U.S.A.)
Alan V. Oppenheim, MIT, (U.S.A.)

Volume 6, page 3273, paper no. 1642

Abstract:
Low-complexity schemes for digital encoding of a noise-corrupted signal and associated signal estimators are presented. This problem arises in wireless distributed sensor networks where an environmental signal of interest is to be estimated at a central site from low-bandwidth digitized information received from collections of remote sensors. We show that the use of a properly designed and often easily implemented additive control input before signal quantization can significantly enhance overall system performance. In particular, efficient estimators can be constructed and used with optimized pseudo-noise, deterministic, and feedback-based control inputs, resulting in a hierarchy of practical systems with very attractive performance-complexity characteristics.
Applications of Blind Equalization in Wireless ATM Network

Authors:
Jeffrey Q Bao, University of Connecticut, (U.S.A.)
Lang Tong, University of Connecticut, (U.S.A.)

Volume 6, page 3277, paper no. 2165

Abstract:
We investigated the feasibility of applying blind equalization to wireless ATM networks. Making use of the information exploited from the wireless ATM cell structure and Medium Access Control (MAC), blind channel estimation together with a Non-linear Data Directed Estimator achieve good equalization performance without transmitting extra preamble. Simulation results are presented for ATM CBR and ABR traffic.
ICASSP98 Paper Abstract
Demodulation of CPM Signals Using Piecewise Polynomial-Phase Modeling

Authors:
Sergio Barbarossa, University of Rome, La Sapienza, (Italy)
Anna Scaglione, University of Rome, La Sapienza, (Italy)

Volume 6, page 3281, paper no. 1608

Abstract:
In this work we propose a novel approach for demodulating Continuous Phase Modulation (CPM) signals based on the modeling of the instantaneous phase as a piecewise polynomial-phase function. The crucial step in the demodulation process is then the estimation of the polynomial coefficients, which is carried out using the so called product high order ambiguity function (PHAF). The proposed approach is suboptimal with respect to the optimal maximum likelihood sequence estimation (MLSE) method, but is much simpler to implement and offers important advantages such as independence of initial phase, tolerance to Doppler shift and time-offset, blind channel identification. We show theoretical results concerning the minimum distance together with some simulation results.
ICASSP98 Paper Abstract

Digital Modulation Classification Using Power Moment Matrices

Authors:
Alfred O. Hero III, University of Michigan, (U.S.A.)
Hafez Hadinejad-Mahram, University of Michigan, (U.S.A.)

Volume 6, page 3285, paper no. 2459

Abstract:
With the rising number of modulation types used in multi-user and multi-service digital communication systems, the need to find efficient methods for their discrimination in the presence of noise has become increasingly important. Here, we present a new approach based on a recently developed pattern recognition method previously applied to word spotting problems in binary images. In this approach, a large number of spatial moments are arranged in a symmetric positive definite matrix for which eigen-decomposition and noise subspace processing methods can be applied. The resultant denoised moment matrix has entries which are used in place of the raw moments for improved pattern classification. In this paper, we generalize the moment matrix technique to grey scale images and apply the technique to discrimination between M-ary PSK and QAM constellations in signal space. Invariance to unknown phase angle and signal amplitude is achieved by representing the in-phase and quadrature components of the signal in the complex plane, and computing joint moments of normalized magnitude and phase components.
ICASSP98 Paper Abstract
A Simple and Robust Modulation Classification Method via Counting

Authors:
Xiaoming Huo, Stanford University; (U.S.A.)
David Donoho, Stanford University; (U.S.A.)

Volume 6, page 3289, paper no. 2399

Abstract:
Automatic modulation classification (or recognition) is an intrinsically interesting problem with a variety of regulatory and military applications. We developed a method which is simple, fast, efficient and robust. The feature being used is the counts of signals falling into different parts of the signal plane. Compared with the likelihood method and the High Order Correlation method, it is much easier to be implemented, and the execution is much faster. When the channel model is correct, our method is efficient, in the sense that it will achieve the "optimal" classification rate. When unknown contamination is present, our method can automatically overcome to certain degree. At SNR being 10 and 15 dB, examples of classifying two modulation types—QAM4 and PSK6—are given. Simulations demonstrate its ability to deal with unknown noises.
ICASSP98 Paper Abstract

Co-channel FM Voice Separation via Cross Coupled Phase Locked Loops

Authors:
Edgar H Satorius, Jet Propulsion Laboratory, (U.S.A.)

Abstract:
This paper presents the results of simulation experiments that successfully demonstrate FM co-channel voice separation via cross-coupled phase locked loops (CCPLL). Unlike previous CCPLL studies which are typically restricted to the situation where the FM modulation waveforms are steady state sinusoidal, triangular, etc., we have empirically determined CCPLL loop parameters that provide for stable separation of co-channel FM voice signals with comparable bandwidth (100% spectral overlap) and comparable modulation indices. The resulting CCPLL parameters differ somewhat from existing CCPLL design rules; however, the differences can yield a significant improvement in CCPLL performance.
A Joint Viterbi Algorithm to Separate Cochannel FM Signals

Authors:
Jon Hamkins, Jet Propulsion Laboratory, (U.S.A.)

Volume 6, page 3297, paper no. 1416

Abstract:
This paper presents a method for separating cochannel FM signals. We show that the Viterbi algorithm, traditionally limited to estimation of digital sequences, can jointly track analog FM signals by separately quantizing the derivatives of their instantaneous frequencies. We employ per-survivor processing in the trellis to estimate unknown channel effects. The approach works well when the signal to interference ratio (SIR) is less than or equal to zero, in contrast to conventional interference suppression algorithms that degrade as SIR approaches zero and fail catastrophically when SIR < 0. Comparisons of mean squared error (MSE) between the estimates and the true signals are given for varying SIR, SNR, Doppler offsets, and frequency deviations. The same approach can also be used for any other continuous phase modulation scheme, such as continuous-phase frequency-shift keying (CPFSK).
ICASSP98 Paper Abstract

COPERM: Transform-Domain Energy Compaction by Optimal Permutation

Authors:
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Volume 6, page 3301, paper no. 1399

Abstract:
COPERM is a novel paradigm for energy compaction and signal compression, whose foundation is a simple but powerful idea: any signal can be transformed to resemble a more desirable signal from a class of “target” signals, by means of a suitable permutation of its samples. The approach is well-suited for transform domain energy compaction prior to transform-domain compression of persistent broadband signals. The associated optimal permutation precoders are surprisingly simple, and the permutation precoding overhead can be made modest - resulting in improved overall rate-distortion performance.
ICASSP98 Paper Abstract
A Fast Blind Source Separation for Digital Wireless Applications

Authors:
Murat Torlak, University of Texas, Austin, (U.S.A.)
Lars K Hansen, University of Texas, Austin, (U.S.A.)
Guanghan Xu, University of Texas, Austin, (U.S.A.)

Volume 6, page 3305, paper no. 2320

Abstract:
The problem of blindly estimating multiple digital co-channel communication signals using an antenna array is studied in the presence of multipath fading. We develop a fast sequential-estimation algorithm for separating multi-user signals based on the geometric observation made by Hansen and Xu. When the signals are constrained to a finite alphabet, it is possible to visualize the geometric properties of the problem, which can be exploited to sequentially extract the digital co-channel communication signals. We will present simulation results comparing speed and BER performance with different methods.
ICASSP98 Paper Abstract

Two Stage Code Reference Beamformer in Mobile Communications

Authors:
Xavier Mestre, UPC, (Spain)
Montse Nájar, UPC, (Spain)
Miguel Angel Lagunas, UPC, (Spain)

Volume 6, page 3309, paper no. 2422

Abstract:
This paper addresses a new architecture for blind adaptive beamforming when dealing with Frequency Hopping (FH) modulation in cellular mobile communications systems. The proposed Code Reference Beamformer (CRB) takes advantage of the inherent frequency diversity to estimate beforehand the noise plus interference correlation matrix, which is employed as the first part of the framework. Then, a second stage is adaptively obtained without any a priori knowledge of either the direction of arrival or the array manifold. Using this information, the first stage is in turn readjusted and, as a result, the scheme is able to track non-stationary scenarios following the channel variations with no previous references.
ICASSP98 Paper Abstract

Downlink Beamforming Avoiding DOA Estimation for Cellular Mobile Communications

Authors:
Thierry Asté, Laboratoire d'Electronique CNAM, (France)
Philippe Forster, Laboratoire d'Electronique CNAM, (France)
Luc Féty, Laboratoire d'Electronique CNAM, (France)
Sylvie Mayrargue, CNET, (France)

Volume 6, page 3313, paper no. 1290

Abstract:
A new technique to overcome induced difficulties of FDD for the design of a forward link beamformer for cellular mobile communications is presented. It uses the array topology at the basestation in order to transpose second order statistics of the propagation channel from uplink frequency to downlink frequency, thus enabling to optimize directly any beamforming criterion based on these statistics at downlink frequency, without feedback nor DOA (Direction Of Arrival) estimation. It can be applied whatever criterion is used to design the beamformer is. Effectiveness is verified by the mean of simulation results.
ICASSP98 Paper Abstract

Maximum Likelihood Multichannel Estimation under Reduced Rank Constraint

Authors:
Philippe Forster, CNAM, (France)
Thierry Asté, CNAM, (France)

Volume 6, page 3317, paper no. 1288

Abstract:
This paper deals with the maximum likelihood estimation of the multichannel impulse response in a mobile communication system whose base stations are equipped with antennas arrays. The following problem is solved: using the training sequence, find the maximum likelihood multichannel impulse response from one mobile to the base station under a reduced rank constraint in the presence of gaussian noise and jammers with unknown covariance matrix. Our results find applications in equalization (the reduced rank channel estimate can be used in a Viterbi algorithm), and in the estimation of the directions of arrival (DOA) of the paths from the mobile to the base station. In this last application, a MUSIC like algorithm is developed using the estimated channel subspace.
ICASSP98 Paper Abstract

Performance Enhancement of the Decorrelating Detector Using Antenna Arrays

Authors:
Kwang Soon Kim, Advanced Institute of Science and Technology, Korea, (Korea)
Seong Ill Park, Advanced Institute of Science and Technology, Korea, (Korea)
Hong Gil Kim, Advanced Institute of Science and Technology, Korea, (Korea)
Yun Hee Kim, Advanced Institute of Science and Technology, Korea, (Korea)
Iickho Song, Advanced Institute of Science and Technology, Korea, (Korea)

Volume 6, page 3321, paper no. 1268

Abstract:
In this paper, a vector channel model is proposed and some statistical properties of the asymptotic efficiency of the decorrelating detector with base-station antenna arrays are investigated. It is shown that we can get gain over the conventional decorrelator by employing antenna arrays and the gain increases as the number of antennas increases and the angle dispersion decreases. It is also shown that we can increase the asymptotic efficiency of the decorrelating detector with base-station antenna arrays up to unity if we use infinite number of antennas when the channel is angle-nondispersive: we cannot, however, increase the asymptotic efficiency of the decorrelating detector by employing base-station antenna arrays if the dispersion is infinite.
Vector-Sensor Array Processing for Estimating Angles and Times of Arrival of Multipath Communication Signals

Authors:
Peng-Huat Chua, DSO National Laboratories, (Singapore)
Chong-Meng See, DSO National Laboratories, (Singapore)
Arye Nehorai, University of Illinois, Chicago, (U.S.A.)

Volume 6, page 3325, paper no. 1539

Abstract:
We develop vector-sensor array processing to estimate the angles-of-arrival (AOAs) and time delays of multipath channels in the space-time-polarization domain. A MUSIC-type algorithm for joint angle and delay estimation with a vector-sensor array is derived. Potential applications include multipath channel estimation and mobile localization. Simulation results show that the space-time-polarization parameterization of the multipath channels results in improved accuracy and resolution performance.
ICASSP98 Paper Abstract
Decoupled Direction Finding: Detection

Authors:
Per Pelin, Chalmers University of Technology, (Sweden)

Volume 6, page 3329, paper no. 1565

Abstract:
Antenna arrays are likely to be an important feature of future mobile communication systems. With an antenna array, mobile users can be separated by a spatial filtering procedure allowing several users on the same carrier frequency. The uplink part (mobile to base) not only can, but is better solved without using any spatial knowledge in terms of direction of arrival (DOA). However, DOA estimation remains an important issue in the overall system, both for downlink beamforming, as well as channel allocation. Previous results have shown that DOA-estimation is best performed in a post-detection manner, i.e., using the estimated symbol sequence for DOA estimation. In this way, the estimation problem can be decoupled to treat individual users separately. To estimate the number of propagation paths from a specific user, a detection scheme is derived based on the DOA estimation criterion function.
ICASSP98 Paper Abstract

The Effects of Local Scattering on Direction of Arrival Estimation with MUSIC and ESPRIT

Authors:
David Asztely, Royal Institute of Technology, (Sweden)
Björn Ottersten, Royal Institute of Technology, (Sweden)

Volume 6, page 3333, paper no. 1538

Abstract:
In wireless communication scenarios, multipath propagation from local scatterers in the vicinity of mobile sources may cause angular spreading as seen from a base station antenna array. This paper studies the effects of such local scattering on direction of arrival (DOA) estimation with the MUSIC and ESPRIT algorithms. Previous work has considered rapidly time-varying scenarios, and concluded that local scattering has a minor effect on DOA estimation in such scenarios. This work considers the case in which the channel is time-invariant during the observation period. The distribution of the DOA estimates is derived, and the results show that local scattering has significant impact on DOA estimation in the time-invariant case. In addition, numerical examples are included to illustrate the analysis, and to demonstrate that the results may be used to formulate simple estimators of angular spread.
ICASSP98 Paper Abstract

Optimal Downlink Power Assignment for Smart Antenna Systems

Authors:
Weidong Yang, The University of Texas, Austin, (U.S.A.)
Guanghan Xu, The University of Texas, Austin, (U.S.A.)

Volume 6, page 3337, paper no. 2525

Abstract:
Smart antenna systems have the potential to substantially increase the range of base stations and boost the SINRs of signals. In this paper, we study several criteria in downlink weighting vector design which is key to exploit the full potential of smart antenna systems, and give the optimal power assignment when the orientations of the weighting vectors are known. Simulation results have shown significant improvement offered by the proposed optimal power assignment method. In particular, we can equalize each user’s downlink performance by significantly reducing the output power. Since the power amplifiers at the base station are the most expensive subsystems, this approach can lead to significant cost reduction for a base station.
ICASSP98 Paper Abstract
Optimal Array Combiner for Sequence Detectors

Authors:
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Ana I. Perez-Neira, UPC, Barcelona, (Spain)
Josep Vidal, UPC, Barcelona, (Spain)

Volume 6, page 3341, paper no. 1695

Abstract:
The use of spatial diversity at the receiver front-end together with a sequence detector implies a joint
design problem of the spatial combiner and the sequence detector impulse response. This joint design
is usually faced under the constraint that the impulse response of the sequence detector is matched to
the channel plus combiner response. This procedure maximizes the signal to noise ratio at the input
of the detector but does not guarantee that the so-called effective signal to noise ratio is maximized.
This work presents a procedure that, starting from the matched criteria, faces directly the maximization
of the effective signal to noise ratio, yet preserving all the features of the spatial processor in terms of
co-channel and high order intersymbol interference rejection.
ICASSP98 Paper Abstract

The Local Minima of Fractionally-Space CMA Blind Equalizer Cost Function in the Presence of Channel Noise

Authors:
Wonzoo Chung, Cornell University, (U.S.A.)
James P LeBlanc, New Mexico State University, (U.S.A.)

Volume 6, page 3345, paper no. 2298

Abstract:
We study the local minima relocation of the fractionally spaced Constant Modulus Algorithm (FSE-CMA) cost function in the presence of noise. Local minima move in a particular direction as the noise power increases and their number may be eventually reduced. In such cases the performance of FSE-CMA may fail to adequately reduce inter symbol interference (ISI), but achieve an approximated MMSE by reducing its equalizer noise gain under certain constraints. We analyze the mechanism of relocation of FSE-CMA cost function local minima in terms of the auto-correlation matrix of sub-channel convolution matrix and its eigenvectors.
ICASSP98 Paper Abstract

Fractionally - Spaced Equalization of Time-Varying Mobile Communications

Authors:
Marie-Line Alberi, ETIS, (France)
Inbar Fijalkow, ETIS, (France)
Thomas J Endres, Sarnoff Digital Comm, (U.S.A.)

Volume 6, page 3349, paper no. 2458

Abstract:
The improved convergence speed and tracking properties of fractionally-spaced equalizers are analyzed. We consider in particular the effect of a frequency offset between the transmitter baud rate and the receiver sampling clock that induces important time-variations. We show that a fractionally-spaced equalizer can handle the intersymbol interferences (ISI) induced when the propagation channel doesn't introduce too much ISI.
ICASSP98 Paper Abstract
The Dithered Signed-Error Constant Modulus Algorithm

Authors:
Philip Schniter, Cornell University, (U.S.A.)
C. Richard Johnson Jr, Cornell University, (U.S.A.)

Volume 6, page 3353, paper no. 1509

Abstract:
Adaptive blind equalization has gained widespread use in communication systems that operate without training signals. In particular, the Constant Modulus Algorithm (CMA) has become a favorite of practitioners due to its LMS-like complexity and desirable robustness properties. The desire for further reduction in computational complexity has motivated signed-error versions of CMA, which have been found to lack the robustness properties of CMA. This paper presents a simple modification of signed error CMA, based on the judicious use of dither, that results in an algorithm with robustness properties closely resembling those of CMA. An approximation to the steady-state mean-squared error performance of the new algorithm is derived for comparison to that of CMA.
ICASSP98 Paper Abstract

A Hybrid Equalizer Merging the Advantages of Baud Spaced and Fractionally Spaced Equalizers

Authors:
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Hans-Martin Bluethgen, University of Technology RWTH Aachen, (Germany)
Tobias G Noll, University of Technology RWTH Aachen, (Germany)

Volume 6, page 3357, paper no. 1966

Abstract:
A transversal equalizer with half Baud spaced taps in the center and extended with Baud spaced taps on both sides is presented. This hybrid equalizer combines the benefits of Baud spaced equalizers - like superior equalization of notches in the middle of the transmission band - and fractionally spaced equalizers, which have a superior performance when equalizing asymmetric notches in the slope of the transmission band, when the same number of coefficients are used. The hybrid equalizer offers the reduced sensitivity to sampling time changes and the ability to model the matched filter in the receiver as the fractionally spaced equalizer. The problem of tap-wandering, which is present in fractionally spaced equalizers, is reduced due to the reduced degree of freedom in the coefficient adjustment.
ICASSP98 Paper Abstract
Optimum Delay and Mean Square Error Using CMA

Authors:
Duncan J Brooks, Imperial College, (U.K.)
Sangarapillai Lambotharan, Imperial College, (U.K.)
Jonathon A Chambers, Imperial College, (U.K.)

Volume 6, page 3361, paper no. 1774

Abstract:
The performance of the Constant Modulus Algorithm can suffer because of the existence of local minima with large Mean Square Error (MSE). This paper presents a new way of obtaining the optimum MSE over all delays using a second equalizer under a mixed Constant Modulus and Cross Correlation Algorithm (CM-CCA). Proof of convergence is obtained for the noiseless case. Simulations demonstrate the potential of the method.
ICASSP98 Paper Abstract
A Sign-Error Algorithm for Blind Equalization of Real Signals

Authors:
Monisha Ghosh, Philips Research, ( U.S.A.)

Volume 6, page 3365, paper no. 1500

Abstract:
The two criteria most commonly used in blind equalization are Sato’s cost function and Godard’s cost function. In this paper we analyze a sign-error cost function for real signals which gives an error term that can be viewed as the sign of either the Sato or the Godard error. We show that the conventional definition of equalizer convergence is not suitable for analyzing this cost function. A more realistic definition of convergence for low to medium SNR situations is presented and used to analyze this sign-error cost function. The performance of this cost function is evaluated via simulations and shown to have excellent performance as compared to the Godard cost function, with substantially less complexity.
ICASSP98 Paper Abstract

A Forward Backward Approach to Adaptive Space-Time Identification and Equalization of Time-Varying Channels

Authors:
Chong-Meng See, DSO National Laboratories, (Singapore)
Colin F.N. Cowan, The Queens University of Belfast, (Northern Ireland)

Volume 6, page 3369, paper no. 1547

Abstract:
In this paper, we present an adaptive algorithm for space-time channel identification and equalization. The proposed algorithm performs joint channel estimation and sequence detection by optimizing a least squares cost function iteratively in a forward and backward manner. Simulation results demonstrate the proposed algorithm to be data efficient and fast converging. In addition, good BER performance is achieved in time-varying channels at relatively low SNR and with an extremely short start-up sequence. These attributes render it suitable for wireless mobile communications using short burst data format.
ICASSP98 Paper Abstract

Non-Linear Channel Equalisation Using Minimal Radial Basis Function Neural Networks

Authors:
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Volume 6, page 3373, paper no. 1151

Abstract:
This paper presents the study results of non-linear channel equalisation problems in data communications using a recently developed minimal radial basis function neural network structure, referred to as MRAN (Minimal Resource Allocation Network). MRAN algorithm uses on-line learning and has the capability to grow and prune the RBF network's hidden neurons ensuring a parsimonious network structure. Compared to earlier methods, the proposed scheme does not have to estimate the channel order first, and fix the model parameters. Results showing the superior performance of the MRAN algorithm for two different non-linear channel equalisation problems, along with a linear non-minimum phase problem, are presented.
ICASSP98 Paper Abstract

Equalization of Satellite Mobile Communication Channels Using Combined Self-Organizing Maps and RBF Networks

Authors:
Steven Bouchired, ENSEEIHT-SIC, (France)
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Volume 6, page 3377, paper no. 2013

Abstract:
The paper proposes a neural network approach to equalize time varying nonlinear channels. The approach is applied to a satellite UMTS channel composed of time invariant linear filters, a non-linear memoryless amplifier and a time varying multipath propagation channel. The neural network equalizer has a Radial Basis Function structure. The usual k-mean clustering algorithm is replaced by a Kohonen learning rule. This results in an RBF-SOM equalizer which outperforms the LMS equalizer, and which has better recovering abilities (after passing through a high fading area) than the former RBF equalizer.
ICASSP98 Paper Abstract
A Truly Recursive Blind Equalization Algorithm

Authors:
Arnaud Bouttier, Thomson-CSF, (France)

Volume 6, page 3381, paper no. 2557

Abstract:
This paper describes a new adaptive blind equalization algorithm based on a truly IIR structure that enables the correction of ISI over severely distorted channels. The recursive feedback filter is in lattice form to allow an easy monitoring of the filter stability. During blind training, the adaptation of the equalizer is carried out via the usual stochastic gradient algorithm by minimizing the Shtrom-Fan cost function, a CMA like functional robust to ill-convergence. Once in steady state, the algorithm switches automatically into a classical DFE structure adapted via the DD-MMSE criterion. Simulation results show that this new equalizer outperforms most of the traditional blind FIR equalizers.
ICASSP98 Paper Abstract

A Semi-Blind Approach to Structured Channel Equalization

Authors:
Boon C Ng, Stanford University, (U.S.A.)
David J. Gesbert, Stanford University, (U.S.A.)
Arogyaswami J. Paulraj, Stanford University, (U.S.A.)

Abstract:
This paper describes a direct equalization approach for channels with some underlying structure. A semi-blind approach is taken here where a small amount of training symbols is available. A family of MMSE equalizers is obtained that includes some prior information about the channel structure. The channel structure assumed in this paper is that the channel vector lies approximately in the subspace of a matrix associated with the samples of the transmit pulse shape. Blind identifiability issues of the structured equalizer are also addressed. Numerical results using experimental indoor channel data indicate that these structured equalizers can achieve bit error rates that are significantly lower than traditional non-blind MMSE equalizers.
Semi-Blind Channel Identification Method For GSM Receivers

Authors:
Ge Li, Auburn University, (U.S.A.)
Zhi Ding, Auburn University, (U.S.A.)

Abstract:
In this paper, we present a semi-blind channel identification scheme for GSM system. Even though the GMSK signal has almost zero excess bandwidth (oversample will give no more information), two diversity channels for each GMSK signal can be generated using a de-rotation scheme without additional antennas. Based on this single input and two output system, the semi-blind algorithm is applied to GSM signals successfully. Simulation results are presented.
ICASSP98 Paper Abstract
On the Use of Kernel Structure for Blind Equalization

Authors:
Jacob H Gunther, Brigham Young University, (U.S.A.)
Arnold Lee Swindlehurst, Brigham Young University, (U.S.A.)

Volume 6, page 3393, paper no. 2131

Abstract:
The mathematical theory of kernel (null space) structure of Hankel and Hankel-like matrices is applied to the problem of blind equalization of co-channel signals. This work builds on recently introduced ideas in blind equalization where the symbols are treated as deterministic parameters and estimated directly without estimating the channel first. The main contribution of the new approach is that it allows the simultaneous exploitation of shift structure in the data model and the finite alphabet property of the signals.
ICASSP98 Paper Abstract

Performance Analysis of Blind Channel Estimators Based on Non-Redundant Periodic Modulation Precoders

Authors:
Antoine Chevreuil, Telecom Paris, (France)
Erchin Serpedin, University of Virginia, (U.S.A.)
Philippe Loubaton, Universite de Marne-la-Valle, (France)
Georgios B. Giannakis, University of Virginia, (U.S.A.)

Volume 6, page 3397, paper no. 2250

Abstract:
Periodic modulation precoders allow blind identifiability of SISO channels from the output second-order cyclic statistics, irrespective of the location of channel zeros, color of additive stationary noise, or channel order overestimation errors. In the present paper the performance of blind channel estimators is investigated. Some criteria for optimally designing the periodic modulation precoders are also presented.
ICASSP98 Paper Abstract
Second Order Blind Equalization: The Band Limited Case

Authors:
Philippe Ciblat, University of Marne-la-Vallee, (France)
Philippe Loubaton, University of Marne-la-Vallee, (France)

Volume 6, page 3401, paper no. 1397

Abstract:
Most of the second order based fractionally sampled blind equalizers are known to perform poorly in the context of band limited signals. In this paper, we analyse the behaviour of the subspace method in the particular context of band limited signals. As it is well known, the subspace channel estimate is obtained as the eigenvector associated to the eigenvalue 0 of a certain positive quadratic form Q. We show that apart 0, Q has quite small eigenvalues, and that this induces poor statistical performance. More importantly, we characterize the numerical kernel of Q, and show that it contains vectors constructed from certain spheroidal wave sequences. From this, we deduce that the subspace method does not allow to estimate accurately the transfer function of the channel on a certain frequency interval.
ICASSP98 Paper Abstract

Fast Blind Identification of FIR Communications Channels

Authors:
Claudio Becchetti, University Roma "La Sapienza", (Italy)
Gaetano Scarano, University Roma "La Sapienza", (Italy)
Giovanni Jacovitti, University Roma "La Sapienza", (Italy)

Volume 6, page 3405, paper no. 2272

Abstract:
This contribution describes a fast frequency domain approach for blind channel identification which does not rely on the statistic of the symbols. The proposed approach is based on the so-called "intraspectral relations" of DFT's of PAM fractionally sampled signals. The use of DFT's is allowed under certain conditions commonly encountered in data communication systems. From the intraspectral relations, asymptotically efficient solutions are derived which turn out to be either more accurate or less expensive in term of complexity w.r.t. the time domain counterparts. Simulation results are provided to assess the validity of the proposed approach in comparison with the Rao Cramer bound and with other approaches from the literature.
ICASSP98 Paper Abstract


Authors:
Bin Huang, Auburn University, (U.S.A.)
Jitendra K. Tugnait, Auburn University, (U.S.A.)

Volume 6, page 3409, paper no. 1106

Abstract:
The problem of blind equalization of SIMO (single-input multiple-output) communications channels is considered using only the second-order statistics of the data. Such models arise when a single receiver data is fractionally sampled (assuming that there is excess bandwidth), or when an antenna array is used with or without fractional sampling. We focus on direct design of finite-length MMSE (minimum mean-square error) blind equalizers. Unlike the past work on this problem, we allow infinite impulse response (IIR) channels. Our approaches also work when the “subchannel” transfer functions have common zeros so long as the common zeros are minimum-phase zeros. Illustrative simulation examples are provided.
ICASSP98 Paper Abstract

Blind Linearization, and Identification of Nonlinear Systems - A Least Squares P-th Order Inverse Approach

Authors:
Gil M Raz, University of Wisconsin-Madison, (U.S.A.)
Barry D Van Veen, University of Wisconsin-Madison, (U.S.A.)

Volume 6, page 3413, paper no. 1780

Abstract:
A deterministic approach to blind nonlinear channel equalization and identification is presented. This approach applies to nonlinear channels that can be approximately linearized by finite memory, finite order Volterra filters. Both the Volterra equalizers and the linearized channels are identified. This method also applies to blind identification of linear IIR channels. General conditions for existence and uniqueness are discussed and numerical examples are given.
ICASSP98 Paper Abstract
Pilot Symbol Assisted Diversity Reception for a Fading Channel

Authors:
Selaka B Bulumulla, University of Pennsylvania, (U.S.A.)
Saleem A Kassam, University of Pennsylvania, (U.S.A.)
Santosh S Venkatesh, University of Pennsylvania, (U.S.A.)

Volume 6, page 3417, paper no. 1803

Abstract:
Pilot symbol assisted modulation is a promising scheme to mitigate the effect of fading in a wireless channel. Analytical results for the performance of this scheme are available. Although the use of diversity is known to improve the performance of receivers used in fading channels, pilot symbol assisted diversity reception has not been studied. In this paper, we derive an exact probability of error expression for such a receiver as a function of the channel estimation error variance and the number of diversity channels. An upper bound for the probability of error, which illustrates the advantage of using diversity, is also obtained. A numerical example is provided.
ICASSP98 Paper Abstract

Estimation of FM Modulation of Multi-Component Signals from the Fourier Phase

Authors:

Volume 6, page 3421, paper no. 5059

Abstract:
Spectral phase is a quantity which is normally discarded in analyzing signals. In this paper, the concept of a complex time-frequency representation is presented. In this representation, the rows are narrow bandpass filtered representations of the original signal, and the columns are broadband Fourier spectra. Methods are developed which exploit the spectral phase of the surface to recover the FM modulating functions of an FM modulated tone and an FM modulated multi-component harmonic signal.
ICASSP98 Paper Abstract

Conditional Maximum Likelihood Frequency Estimation for Staggered Modulations

Authors:
Jaume Riba, UPC, (Spain)
Gregori Vázquez, UPC, (Spain)
Sergio Calvo, UPC, (Spain)

Volume 6, page 3425, paper no. 2167

Abstract:
The use of spectrally efficient continuous phase modulations for mobile communications may lead to a serious performance degradation of the classical frequency error detectors (FEDs) due to the presence of self-noise. This contribution presents a new statistically efficient frequency estimation algorithm for staggered modulations. The cancellation of the self-noise is accomplished by the use of the Conditional ML principle, well known in the context of array processing, as an alternative to the Unconditional ML, typically applied in the communications field. The paper also provides a new Cramer Rao Bound (CRB) which is more accurate than the so-called Modified CRB (MCRB) extensively applied to synchronization problems.
ICASSP98 Paper Abstract

A Signal Processing Approach for Effective Reduction of Timing Jitter due to the Acoustic Effect

Authors:
Tulay Adali, University of Maryland, Baltimore, (U.S.A.)
Bo Wang, University of Maryland, Baltimore, (U.S.A.)
Alexei N. Pilipetski, University of Maryland, Baltimore, (U.S.A.)
Curtis R. Menyuk, University of Maryland, Baltimore, (U.S.A.)

Abstract:
We introduce a signal processing approach to compensate for the timing jitter produced by the acoustic effect in soliton communications. The other main sources of timing jitter, the Gordon-Haus effect and the polarization effect, are inherently stochastic. By contrast, the acoustic effect is deterministic and becomes the dominant source of bit error rates in standard soliton systems when the bit rates are more than 10 Gbits/s and the transmission distance is more than several thousand kilometers. We exploit the deterministic nature of the acoustic effect to introduce a scheme that predicts the amount of timing jitter as a function of the previous transmitted bits and uses the information to adjust the sampling period of the received soliton pulses. We demonstrate successful application of the scheme by simulations and discuss implementation issues.
ICASSP98 Paper Abstract

Cost-Efficient Approximation of Linear Systems with Repeated and Multi-channel Filtering Configurations

Authors:
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Fatih M Erden, Tampere University of Technology, (Finland)
Haldun M. Ozaktas, Bilkent University, (Turkey)
Orhan Arikani, Bilkent University, (Turkey)
Cagatay Candan, Bilkent University, (Turkey)
Ozgur Guleryuz, Bilkent University, (Turkey)

Volume 6, page 3433, paper no. 2027

Abstract:
It is possible to obtain either exact realizations or useful approximations of linear systems or matrix-vector products arising in many different applications, by synthesizing them in the form of repeated or multi-channel filtering operations in fractional Fourier domains, resulting in much more efficient implementations with acceptable decreases in accuracy. By varying the number and configuration of filter blocks, which may take the form of arbitrary flow graphs, it is possible to trade off between accuracy and efficiency in the desired manner. The proposed scheme constitutes a systematic way of exploiting the information inherent in the regularity or structure of a given linear system or matrix, even when that structure is not readily apparent.
ICASSP98 Paper Abstract

QoS Considerations for DMT-based ADSL and VDSL Systems

Authors:
Marie Colin, IEMN/OAE, (France)
Cory Modlin, Amati Communications Corporation, (U.S.A.)
Mohamed Gharbi, IEMN/OAE, (France)
Marc Gazalet, IEMN/OAE, (France)

Volume 6, page 3437, paper no. 2080

Abstract:
Thanks to their high bandwidth ability, Asymmetric Digital Subscriber Lines (ADSL) and Very high speed DSL (VDSL) are access technologies that permit the transmission of several applications simultaneously on telephonic subscriber lines. Considering that these applications may require a different Quality of Service (QoS), and particularly different Bit Error Rates (BER), for transmission, this paper addresses the problem of providing simultaneously two BERs for transmission over a DMT-based ADSL or VDSL link. Both coded and uncoded systems are considered.
ICASSP98 Paper Abstract

Computational Reduction During Idle Transmission in DSL Modems

Authors:
Adam M Chellali, Texas Instruments, (U.S.A.)
Mike Polley, Texas Instruments, (U.S.A.)
Alan Gatherer, Texas Instruments, (U.S.A.)

Abstract:
This document describes two methods for reducing computational requirements during idle transmission in remote access systems incorporating digital subscriber line (DSL) modems, including asymmetrical DSL (ADSL) systems. These methods save processing power during idle transmission by generating an idle signal using low complexity techniques. The generated idle signal is made spectrally compatible with xDSL systems and a non-disruptive signaling scheme is used to indicate to the far end receiver the transition between idle to active or active to idle status. A technique is presented that modulates the phase of the pilot tone to signal status transitions to the remote receiver. The computational complexity at the receiver is reduced because full demodulation and decoding is not required to determine that an idle signal is being transmitted.
ICASSP98 Paper Abstract

On the Implementation of Adaptive Equalizers for Wireless ATM with an Extended QR-Decomposition-Based RLS-Algorithm

Authors:
Christian Drewes, Technische Universitaet Muenchen, (Germany)
Ralph Hasholzner, Technische Universitaet Muenchen, (Germany)
Joachim S Hammerschmidt, Technische Universitaet Muenchen, (Germany)

Abstract:
An extended QR-decomposition (QRD)-based RLS-algorithm is introduced, applicable for equalization, allowing an estimate of the transmitted symbol, when it is a-priori unknown. To achieve low-cost implementations strength reduction and square-root free Givens-rotations are applied. Several QRD-based RLS-algorithms are compared in terms of the number of mathematical operations. The QR-RLS-algorithm indicates lowest complexity and is always a good candidate, when a huge trainig overhead has to be avoided, resulting in an equalizer of less than 15 taps. Finally, the hardware complexity of a 13 Mbaud-DFE for wireless ATM is estimated.
ICASSP98 Paper Abstract
A New Test of Stationarity and its Application to Tele-traffic Data

Authors:
Sandrine Vaton, ENST Paris, (France)

Abstract:
In this contribution we generalize the test of sphericity as a test of stationarity for time-series. The sphericity statistics is in our case a measure of distance between the empirical correlations calculated on two contiguous segments of the same process. We prove that under the hypothesis of stationarity the logarithm of the sphericity converges in distribution to a quadratic form in a multidimensional gaussian random variable with a convergence rate that is equal to the length of the observation window. We then derive a test of proportionality of the correlations of the process on the two segments. This new test is applied to test if the traffic measured on today’s broadband networks is stationary. The results that we obtain are connected to many previous works according to which the traffic generated by modern high-speed networks is a stationary and long-range dependent process.
Probability of Detection of Residual Echo Based on Magnitude-Squared Coherence Estimate

Authors:
Cliff Sui, Lucent Technologies, (U.S.A.)

Abstract:
Residual echo is a distorted, partially-canceled, and transient echo of the near-end speech signal returned from the remote network. Previous work has shown that residual echo can be detected and reduced by using a nonlinear processing technique based on frequency coherence. In this paper a mathematical approach to evaluate this nonlinear processor and the algorithm for computing the probability of detection using the magnitude-squared coherence estimate are presented.
ICASSP98 Paper Abstract

Joint Source Channel Coding over Channels with Intersymbol Interference Using Vector Channels and Discrete Multitone

Authors:
Venceslav Kafedziski, Arizona State University, (U.S.A.)
Darryl Morrell, Arizona State University, (U.S.A.)

Volume 6, page 3457, paper no. 2389

Abstract:
We investigate joint source channel coding for channels with intersymbol interference (ISI) where the source coder is a vector quantizer. In our previous work we used a block MAP equalizer, which takes into account the residual correlation between the VQ outputs and also provides for soft decisions to improve the performance. In this paper we propose the use of a vector channel approach and discrete multitone modulation for joint source channel coding on channels with ISI. Using these modulation procedures, intersymbol interference can be eliminated and the problem of joint source channel coding for ISI channels is reduced to the problem of coding for vector Gaussian channels. Optimization of the signal set is performed through optimal power allocation to the subchannels. Simulation results are presented for both vector and discrete multitone channels and compared to the results obtained by using the block MAP equalizer and to the OPTA (optimum performance theoretically attainable) bound.
ICASSP98 Paper Abstract

A Joint Source/Channel Coder with Block Constraints

Authors:
Hasan H Otu, University of Nebraska, (U.S.A.)
Khalid Sayood, University of Nebraska, (U.S.A.)

Volume 6, page 3461, paper no. 2342

Abstract:
Joint source/channel coders obtained using MAP decoders tend to fail at low probability of error. In this paper we propose a modification of the standard approach which provides protection at low error rates as well.
ICASSP98 Paper Abstract

Design of Channel Optimized Vector Quantizers in the Presence of Channel Mismatch

Authors:
Hamid Jafarkhani, AT&T Labs, (U.S.A.)
Nariman Farvardin, University of Maryland, (U.S.A.)

Volume 6, page 3465, paper no. 2473

Abstract:
We propose algorithms to design channel-optimized vector quantizers in the presence of channel mismatch. We consider two cases: (i) no information about the statistics of the channel bit error rate is available and (ii) the probability density function of the channel bit error rate is known. We also consider the use of an estimate of the channel signal-to-noise ratio to improve performance. Simulation results demonstrate the advantages of new design algorithms.
ICASSP98 Paper Abstract

Design of Neural Network Quantizers for a Distributed Estimation System with Communication Constraints

Authors:
Vasileios Megalooikonomou, University of Maryland Baltimore County, (U.S.A.)
Yaacov Yesha, University of Maryland Baltimore County, (U.S.A.)

Volume 6, page 3469, paper no. 1409

Abstract:
We consider the problem of quantizer design in a distributed estimation system with communication constraints at the channels in the case where the observation statistics are unknown and one must rely on a training set. The method that we propose applies a variation of the Cyclic Generalized Lloyd Algorithm (CGLA) on every point of the training set and then uses a neural network for each quantizer to represent the training points along with their associated codewords. The codeword of every training point is initialized using a regression tree approach. Simulations show that the combined approach i.e. building the regression tree system and using its quantizers to initialize the neural networks provides an improvement over the regression tree approach except in the case of high noise variance.
ICASSP98 Paper Abstract

Performance of Predictive Coders over Noisy Channels with Feedback

Authors:
Masoud Khansari, Hewlett-Packard Labs, (U.S.A.)

Volume 6, page 3473, paper no. 2576

Abstract:
Predictive coding methods such as DPCM used for lossless coding of images or motion compensated hybrid video coders MPEG family are shown to compress the input signals well with a reasonable complexity. The performance of these coders, however, degrades considerably when the transmission channel is not error-free. This is due to the error propagation at the decoder where a single error can have catastrophic consequences. A low-rate feedback channel is shown to improve the overall performance. In this paper, we consider two such methods and provide the analysis and investigate different trade-offs.
ICASSP98 Paper Abstract

Finite-state Differential Coding for Wireless Communications with Multipath Channels

Authors:
Hui Liu, University of Virginia, (U.S.A.)
Kemin Li, University of Virginia, (U.S.A.)

Volume 6, page 3477, paper no. 1657

Abstract:
Differential coding allows signal demodulation without carrier phase estimation, and thus is commonly used to cope with phase ambiguity and residual carriers. In a wireless scenario where the system transfer function is FIR due to multipath reflections, channel estimation and equalization is usually required. Inspired by the recent work by Tong on blind sequence estimation, we propose a vector differential coding scheme that allows instantaneous signal detection at the receiver without knowledge of the channel. The new technique can be regarded as a generalization of the standard differential coding method for removing convolutional ambiguities.
ICASSP98 Paper Abstract
Constrained Equalizers and Precoding for Magnetic Storage Channels

Authors:
German S Feyh, Cirrus Logic, (U.S.A.)
Volume 6, page 3481, paper no. 1900

Abstract:
The nonlinear write process of magnetic recording allows to write the symbols +1/-1 only. The magnetic channel is a differentiating channel. The locations of the transitions from +1 to -1 and vice versa in the input signal to the magnetic channel are important for the received waveform. This paper defines a noise enhancement constrained, finite dimensional equalizer. This equalizer trades some misequalization of the data signal for less noise enhancement after the equalizer. Additionally the misequalization is decreased by precoding. Precoding shapes the signal before entering the channel. Since precoding in magnetic channels is limited to shifting the positions of the transitions around, precoding does not allow for full equalization at the receiver. Therefore the equalizer in the receiver and the precoder are optimized. In order to find the optimal transition positions a linearized representation of the transition shift is produced. This representation leads to a constraint optimization problem.
ICASSP98 Paper Abstract

Optimum Open Eye Equalizer Design for Non-Minimum Phase Channels

Authors:
Mark E Halpern, University of Melbourne, (Australia)
Murk Bottema, Flinders University, (Australia)
William Moran, Flinders University, (Australia)
Soura Dasgupta, University of Iowa, (U.S.A.)

Volume 6, page 3485, paper no. 2299

Abstract:
This paper contains results on the design of optimum equalizers to eliminate intersymbol interference in linear non-minimum phase channels conveying binary signals. The optimization is with respect to an open eye condition with a given delay. For causal stable channels with non-minimum phase zeros, we argue that this problem requires only the consideration of the FIR modified channel that has all the non-minimum phase zeros of the original channel. We show that if this modified channel can be equalized to yield an equalized system that is open eye with a specified delay, then the optimizing equalizer is, in fact FIR with all zeros outside the unit circle, and the impulse response of the equalised channel does not extend beyond the delay. We also give a simple necessary and sufficient condition to determine if for a particular delay, a given channel can be equalized to achieve an equalized response that is open eye.
ICASSP98 Paper Abstract
H-Infinity Equalization of Communication Channels

Authors:
Alper T Erdogan, Stanford University, (U.S.A.)
Babak Hassibi, Stanford University, (U.S.A.)
Thomas Kailath, Stanford University, (U.S.A.)

Volume 6, page 3489, paper no. 2417

Abstract:
As an alternative to existing techniques and algorithms, we investigate the merit of the H-infinity approach to the equalization of communication channels. We first look at causal H-infinity equalization problem and then look at the improvement due to finite delay. By introducing the risk sensitive property, we compare the average performance of the central H-infinity Equalizer with the MMSE equalizer in equalizing minimum phase channels.
ICASSP98 Paper Abstract

Fast Computation of Efficient Decision Feedback Equalizers for High Speed Wireless Communications

Authors:
Ian J Fevrier, Purdue University, (U.S.A.)
Saul B Gelfand, Purdue University, (U.S.A.)
Michael P Fitz, Ohio State University, (U.S.A.)

Volume 6, page 3493, paper no. 2391

Abstract:
Decision feedback equalization (DFE) structures have recently been proposed for the efficient equalization of wireless channels with long postcursor response, which is a bottleneck problem for high speed communications over multipath channels with large delay spreads. These structures are equivalent to the conventional DFE, but remove postcursor intersymbol interference (ISI) prior to feedforward filtering. We investigate the relationship between these structures and fast equalizer coefficient computation. Based on this relationship, we obtain a fast algorithm for computing optimal DFE settings which has significantly lower complexity than other known approaches for these high speed wireless channels. An example is given for data rates and channel profiles of the type considered for the proposed North American high definition television (HDTV) terrestrial broadcast mode.
ICASSP98 Paper Abstract

Efficient Filterbank Channelizers for Software Radio Receivers

Authors:
Kambiz C Zangi, Ericsson Inc., (U.S.A.)
David R Koilpillai, Ericsson Inc., (U.S.A.)

Volume 6, page 3497, paper no. 2329

Abstract:
For cellular software radio receivers, this paper presents a computationally efficient algorithm for extracting individual radio channels from the output of the wideband A/D converter. In a software radio, the extraction of individual channels from the output of the wideband A/D converter is by far the most computationally demanding task; hence, it is very important to devise computationally efficient algorithms for this task. Our algorithm is obtained by modifying the DFT filter bank structure that is well known in the multi-rate signal processing literature. We show that the complexity of the proposed algorithm is significantly less (2X-50X) than the complexity of the conventional channelizers.
ICASSP98 Paper Abstract
Self-Recovering Multirate Equalizers Using Redundant Filterbank Precoders

Authors:
Anna Scaglione, University of Rome, La Sapienza, (Italy)
Georgios B. Giannakis, University of Virginia, (U.S.A.)
Sergio Barbarossa, University of Rome, La Sapienza, (Italy)

Volume 6, page 3501, paper no. 1475

Abstract:
Transmitter redundancy introduced using FIR filterbank precoders offers a unifying framework for single-and multi-user transmissions. With minimal rate reduction, FIR filterbank transmitters with trailing zeros allow for perfect (in the absence of noise) equalization of FIR channels with FIR zero-forcing equalizer filterbanks, irrespective of the input color and the channel zero locations. Exploiting input diversity, blind channel estimators, block synchronizers, and direct self-recovering equalizing filterbanks are derived in this paper. The resulting algorithms are computationally simple, require small data sizes, can be implemented online, and remain consistent (after appropriate modifications) even at low SNR colored noise. Simulations illustrate applications to multi-carrier modulations through channels with deep fades, and superior performance relative to CMA and existing output diversity techniques relying on multiple antennas and fractional sampling.
ICASSP98 Paper Abstract

Nonmaximally Decimated Filterbank Based Precoder / Post-equalizer for Blind Channel Identification and Optimal MMSE Equalization

Authors:
Xueming Lin, New Jersey Institute of Technology, (U.S.A.)
Ali N. Akansu, New Jersey Institute of Technology, (U.S.A.)

Volume 6, page 3505, paper no. 2539

Abstract:
A novel nonmaximally decimated multirate filterbank structure is proposed for blind identification of communication channels. This structure is shown to be very similar to a form proposed earlier in the literature. It is presented that the proposed blind channel identification algorithm is not sensitive to the characteristics of unknown channel, including mixed phase and zeros on the unit circle. An optimal minimum mean square error based linear equalizer using the blind channel identification scheme is investigated. It is shown that the proposed system outperforms the existing zero-forcing blind equalization algorithms in literature. It can simultaneously cancel the intersymbol interference (ISI) and suppress the noise enhancement. The reconstructed signal to noise ratio is maximized by the proposed algorithm. Simulation results show the superior performance and robustness of the proposed blind identification and equalization scheme.
Blind Carrier Synchronization and Channel Identification for OFDM Communications

Authors:
Ufuk Tureli, University of Virginia, (U.S.A.)
Hui Liu, University of Virginia, (U.S.A.)

Abstract:
In OFDM communications, the loss of orthogonality due to carrier offset must be compensated before DFT-based demodulation can be performed. In this paper, we present a high accuracy blind carrier offset estimation algorithm and a blind channel equalizer which exploit the intrinsic structure information of OFDM signals. The latter method allows the receiver to perform coherent demodulation in changing environments without the overhead required for additional pilots.
Robust Equalization for Spread-Response Precoding Systems

Authors:
J. Nicholas Laneman, *MIT, (U.S.A.)*
Gregory W. Wornell, *MIT, (U.S.A.)*

Volume 6, page 3513, paper no. 1677

Abstract:
The problem of equalization for spread-response precoding systems based on minimum mean-square error (MMSE) estimates of the fading channel coefficients is considered. These systems are attractive, low complexity alternatives to the combination of interleaving and error-control coding for achieving time diversity in fading environments. To make the performance of these systems robust to channel estimation errors, we derive the linear equalizer at the receiver that maximizes the effective signal-to-noise-and-interference ratio (SNIR) subject to uncertainty in the channel measurements. We examine the bit-error rate performance and develop fixed and dynamic solutions to the associated problem of optimal power allocation between the data transmissions and channel measurements. The effectiveness of these algorithms is demonstrated through measurements obtained from an indoor wireless setting.
Cost-efficient Parallel Lattice VLSI Architecture for the IFFT/FFT in DMT Transceiver Technology

Authors:
An-Yeu Wu, National Central University, (Taiwan)
Tsun-Shan Chan, National Central University, (Taiwan)

Volume 6, page 3517, paper no. 1595

Abstract:
The discrete multitone (DMT) modulation/demodulation scheme is the standard transmission technique in the application of asymmetric digital subscriber lines (ADSL). Although the DMT can achieve higher data rate compared with other modulation/demodulation schemes, its computational complexity is too high for cost-efficient implementations. For example, it requires 512-point IFFT/FFT as the modulation/demodulation kernel. The large block size results in heavy computational load in running programmable DSP processors. It also makes the VLSI implementation not feasible. In this paper, we derive the parallel lattice structure for the IFFT/FFT based on the time-recursive approach. The resulting architectures are regular, modular, and without global communications so that they are very suitable for VLSI implementation. Also, the proposed structure requires only 11% number of multipliers and 9% number of adders compared with the direct implementation approach.
ICASSP98 Paper Abstract

Nonlinear Channel Identification and Equalization for OFDM Systems

Authors:
Arthur J Redfern, Georgia Institute of Technology, (U.S.A.)
G. Tong Zhou, Georgia Institute of Technology, (U.S.A.)

Volume 6, page 3521, paper no. 1810

Abstract:
Orthogonal Frequency Division Multiplexing (OFDM) has become increasingly popular due to its potential applications in digital audio broadcasting, digital terrestrial TV broadcasting, and satellite communication. A notable drawback of OFDM systems is their sensitivity to nonlinear distortion. For maximum power efficiency, amplifiers and transmitters of modern communication systems often operate near their saturation regions which leads to nonlinear distortion. In this paper, we use the special property that the transmitted OFDM symbols are asymptotically white Gaussian to derive an algorithm that identifies the nonlinear channel. A nonlinear equalizer is built to compensate for the undesired nonlinearities. Simulation results show that the nonlinear equalizer outperforms its linear counterpart when nonlinear distortion is present.
ICASSP98 Paper Abstract

Design and Implementation of a DVB On-Board Multi-Carrier Demodulator

Authors:
Josep Sala-Alvarez, UPC, (Spain)
Alba Pages-Zamora, UPC, (Spain)
Sergio Calvo, UPC, (Spain)
Josep Prat, UPC, (Spain)

Volume 6, page 3525, paper no. 1975

Abstract:
A description of the signal processing stage of an on-board integrated VLSI multi-carrier demodulator at the demultiplexer level is presented in this paper, along with a description of the optimization procedure that has been developed for the signal processing functions. The varying adjacent carrier interference and channel noise distribution are modeled to provide the best performing demultiplexing scheme under the given carrier distribution with minimum complexity.
ICASSP98 Paper Abstract

Head-Related Transfer Function Modeling in 3-D Sound Systems with Genetic Algorithms

Authors:
Ngai-Man Cheung, Texas Instruments Tsukuba R&D Center, (Japan)
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Andrew Horner, Hong Kong University of Science & Technology, (Hong Kong)

Volume 6, page 3529, paper no. 1244

Abstract:
Head-related transfer functions (HRTFs) describe the spectral filtering that occurs between a source sound and the listener's eardrum. Since HRTFs vary as a function of relative source location and subject, practical implementation of 3D audio must take into account a large set of HRTFs for different azimuths and elevations. Previous work has proposed several HRTF models for data reduction. This paper describes our work in applying genetic algorithms to find a set of HRTF basis spectra, and then normal equation method to compute the optimal combination of linear weights to represent the individual HRTFs at different azimuths and elevations. The genetic algorithm selects the basis spectra from the set of original HRTF amplitude responses, using an average relative spectral error as the fitness function. Encouraging results from the experiments suggest that genetic algorithms provide an effective approach to this data reduction problem.
ICASSP98 Paper Abstract
A Bottle Model for Head-Related Transfer Functions

Authors:
Bradley S Ferguson, University of Adelaide, (Australia)
Robert E. Bogner, University of Adelaide, (Australia)
Steve Wawryk, University of Adelaide, (Australia)

Volume 6, page 3533, paper no. 1650

Abstract:
We describe a parsimonious model for the direction-dependent transfer function of the pinna. The model describes the transfer function with reference to resonators located in particular physical positions relative to the ear canal. The purpose of the work is to provide a parametric model that permits identification with moderate data-gathering, and filter specification for any direction without the need for interpolation of responses.
ICASSP98 Paper Abstract
Simulation of Three-Dimensional Sound Propagation with Multidimensional Wave Digital Filters

Authors:
Thomas Schetelig, University Erlangen, (Germany)
Rudolf Rabenstein, University Erlangen, (Germany)

Volume 6, page 3537, paper no. 1827

Abstract:
The propagation of sound waves is described by partial differential equations for the acoustic pressure and the acoustic fluid velocity. The solution depends on the shape of the enclosure and on the boundary conditions. Among various methods for the discretization of partial differential equations, the multidimensional wave digital filter approach is known to yield robust algorithms for the discrete simulation of continuous problems. This paper describes the derivation of a discrete model for three-dimensional sound propagation according to multidimensional wave digital filtering principles. The correct treatment of boundary conditions for various wall impedances is shown. A numerical example for the sound propagation in three interconnected rooms of a building demonstrates the capabilities of the method.
ICASSP98 Paper Abstract

Optimum Loudspeaker Spacing for Robust Crosstalk Cancellation

Authors:
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Gary W Elko, Bell Labs, Lucent Technologies, (U.S.A.)

Volume 6, page 3541, paper no. 1600

Abstract:
Crosstalk cancellation is a signal processing technique whereby two (or more) loudspeakers are used to deliver desired signals exactly at the listener's ears. Such a system is useful for 3D audio applications, and removes the requirement for the listener to wear headphones. However, crosstalk cancelers are notoriously non-robust to slight movements in head position, and there currently exists no clear method for determining the best loudspeaker placement in a given situation. In this paper we propose a robustness measure to evaluate the performance of crosstalk cancelers as a function of loudspeaker spacing. Based on this analysis we conclude that certain loudspeaker spacings give far better robustness performance, and provide a simple empirically-derived equation for determining the optimum loudspeaker spacing in a given situation.
ICASSP98 Paper Abstract
Mean Weight Behavior of the Filtered-X LMS Algorithm

Authors:
Orlando J. Tobias, Universidade Federal de Santa Catarina, (Brazil)
Jose C.M. Bermudez, Universidade Federal de Santa Catarina, (Brazil)
Neil J. Bershad, University of California, Irvine, (U.S.A.)
Rui Seara, Universidade Federal de Santa Catarina, (Brazil)

Volume 6, page 3545, paper no. 1117

Abstract:
This paper presents a stochastic analysis of the Filtered-X LMS algorithm. The mean weight vector recursion is derived for slow adaptation and for a white reference signal without use of independence theory. The Wiener solution is determined explicitly as a function of the input statistics and the impulse responses of the primary and secondary signal paths. It is shown that the steady-state mean weights for the Filtered-X LMS algorithm converge to the Wiener solution only if the estimate of the secondary path is without error. Monte Carlo simulations show excellent agreement with the behavior predicted by the theoretical model.
ICASSP98 Paper Abstract

A Method of Optimizing Source Configuration in Active Control Systems Using Gram-Schmidt Orthogonalization

Authors:
Futoshi Asano, Electrotechnical Laboratory, (Japan)
Yoiti Suzuki, R.I.E.C Tohoku University, (Japan)
David C Swanson, A.R.L Pennsylvania State University, (U.S.A.)

Abstract:
In this paper, a method for optimizing the number and the configuration of control sources in an active control system is proposed. In the optimization process, sources are selected one by one so that the corresponding transfer impedance vector is the most linearly independent. From the results of the simulation, it is shown that the optimized configuration yields not only small average control error but also small condition number in the transfer impedance matrix, which contributes to the robustness of the system against the environmental change.
GMDF-alpha with Adaptive Reconstruction Filters and Zero Throughput Delay

Authors:
Jeff P Lariviere, Carleton University, (Canada)
Rafik A Goubran, Carleton University, (Canada)

Abstract:
With reduction of the block size (increasing the number of subfilters) regular gmdf can achieve low throughput delay at the expense of system performance. In situations where zero delay is desirable, we propose a new method which is not dependent on the block size. In addition, by using an adaptive reconstruction filter, further performance gains can be achieved with minimal additional computation complexity. Results from experiments performed in a conference room show an increase in the average Echo Return Loss Enhancement (ERLE) of > 2.5 dB for acoustic echo cancelation over the traditional moving average reconstruction filter.
ICASSP98 Paper Abstract

Analysis and Design of Narrowband Active Noise Control Systems

Authors:
Sen M. Kuo, Northern Illinois University, (U.S.A.)
Xuan Kong, Northern Illinois University, (U.S.A.)
Shaojie Chen, Northern Illinois University, (U.S.A.)
Wenge Hao, Northern Illinois University, (U.S.A.)

Volume 6, page 3557, paper no. 5195

Abstract:
This paper presents an analysis and optimization of narrowband active noise control (ANC) systems using the filtered-X least mean-square (LMS) algorithm. First, we derive an upper bound for the eigenvalue spread of the filtered reference signal's covariance matrix, which provides insights into algorithm convergence speed. Amplitude of internally generated sinusoidal reference signal is optimized as the inverse of the secondary path's magnitude response at the corresponding frequency to improve the convergence speed. Second, we analyze the characteristic of asymmetric out-of-band overshoot. Based on the analysis result, the phase of sinusoidal reference signal is optimized to compensate for the phase shift of the secondary path. This phase optimization leads to the minimization of the out-of-band overshoot.
ICASSP98 Paper Abstract

Amplitude Modulated Sinusoidal Modeling Using Least-square Infinite Series Approximation with Applications to Timbre Analysis

Authors:
Wooi-Boon Goh, Nanyang Technological University, (Singapore)
Kai-Yun Chan, Nanyang Technological University, (Singapore)

Volume 6, page 3561, paper no. 1287

Abstract:
A least-square infinite series approximation (L-SISA) technique is proposed for modeling amplitude modulated (AM) sinusoidal components of naturally occurring signals, such as those produced by traditional musical instruments. Each AM sinusoid is iteratively extracted using an analysis-by-synthesis technique and the problem of parameter estimation is linearised for least-square approximation through a systematic search in the frequency vector space. Some timbre analysis results obtained using the AM sinusoidal model are presented.
ICASSP98 Paper Abstract
Multi-Pitch Estimation for Polyphonic Musical Signals

Authors:
Pablo Fernandez-Cid, GAPS-SSR-ETSIT-UPM, (Spain)
Francisco Javier Casajus-Quiros, GAPS-SSR-ETSIT-UPM, (Spain)

Volume 6, page 3565, paper no. 1402

Abstract:
Automatic Score Transcription goal is to achieve an score-like (notes pitches through time) representation from musical signals. Reliable pitch extraction methods for monophonic signals exist, but polyphonic signals are much more difficult, often ambiguous, to analyze. We propose a computationally efficient technique for automatic recognition of notes from a polyphonic signal. It looks for correctly shaped (magnitude and phase wise) peaks in a, time and frequency oversampled, multiscale decomposition of the signal. Peaks (partial candidates) get accepted/discarded by their match to the window spectrum shape and continuity-across-scale constraints. The final partial list builds a resharpened and equalized spectrum. Note candidates are found searching for harmonic patterns. Perceptual and source based rejection criteria help discard false notes, frame-by-frame. Slightly non-causal postprocessing uses continuity (across a <150 ms. observation time) to kill too short notes, fill in the gaps, and correct (sub)octave jumps.
ICASSP98 Paper Abstract

Suppression of Transients in Time-Varying Recursive Filters for Audio Signals

Authors:
Vesa Valimaki, Helsinki University of Technology, (Finland)
Timo I. Laakso, Helsinki University of Technology, (Finland)

Volume 6, page 3569, paper no. 1497

Abstract:
A new method for suppressing transients in time-varying recursive filters is proposed. The technique is based on modifying the state variables when the filter coefficients are changed so that the filter enters a new state smoothly without transient attacks, as originally proposed by Zetterberg and Zhang. In this contribution we modify the Zetterberg-Zhang algorithm to render it feasible for efficient implementation. We explain how to determine an optimal transient suppresser to cancel the transients down to a desired level at the minimum complexity of implementation. The application of the method to time-varying all-pole and direct-form II filter structures is studied. The algorithm may be generalized for any recursive filter structure. The transient suppression technique finds applications in audio signal processing where the characteristics of a recursive filter needs to be changed in real time, such as in music synthesis, auralization, and equalization.
ICASSP98 Paper Abstract

An Analysis/Synthesis Tool for Transient Signals That Allows a Flexible Sines+Transients+Noise Model for Audio

Authors:
Tony S. Verma, Stanford University, (U.S.A.)
Teresa H.Y. Meng, Stanford University, (U.S.A.)

Volume 6, page 3573, paper no. 1631

Abstract:
We present a flexible analysis/synthesis tool for transient signals that extends current sinusoidal and sines-noise models for audio to sines-transients-noise. The explicit handling of transients provides a more realistic and robust signal model. Because the transient model presented is the frequency domain dual to sinusoidal modeling, it has similar flexibility and allows for a wide range of transformations on the parameterized signal. In addition, due to this duality, a major portion of the transient model is sinusoidal modeling performed in a frequency domain. In order to make the transient and sinusoidal models work more effectively together, we present a formulation of sinusoidal modeling (and therefore transient modeling) in terms of matching pursuits and overlap-add synthesis. This formulation provides a tight coupling between the sines-transients-noisemodel because it allows a simple heuristic, based on tonality, as to when an audio signal should be modeled as sines and/or transients and/or noise.
ICASSP98 Paper Abstract

A New Frequency Domain Approach to Time-Scale Expansion of Audio Signals

Authors:
Anibal J.S. Ferreira, The University of Porto, (Portugal)

Abstract:
We present a new algorithm for time-scale expansion of audio signals that comprises: time interpolation, frequency-scale expansion and modification of a spectral representation of the signal. The algorithm relies on an accurate model of signal analysis and synthesis, and was constrained to a non-iterative modification of the magnitudes and the wrapped phases of the relevant sinusoidal components of the signal. The structure of the algorithm is described and its performance is illustrated. A few examples of time-expanded wideband speech can be found on the Internet.
ICASSP98 Paper Abstract

Robust Exponential Modeling of Audio Signals

Authors:
Joost Nieuwenhuijse, Delft University of Technology, (The Netherlands)
Richard Heusdens, Delft University of Technology, (The Netherlands)
Ed F. Deprettere, Delft University of Technology, (The Netherlands)

Volume 6, page 3581, paper no. 1997

Abstract:
In this paper we present a numerically robust method for modeling audio signals which is based on an exponential data model. This model is a generalization of the classical sinusoidal model in the sense that it allows the amplitude of the sinusoids to evolve exponentially. We show that, using this model, so-called attacks can be represented very efficiently and we propose an algorithm for finding the exponentials in a robust way. Moreover, we show that by using a proper segmentation of the input data into variable length segments the signal-to-noise ratio can be drastically improved as compared to a fixed-length analysis.
ICASSP98 Paper Abstract
Multiresolution Sinusoidal Modeling for Wideband Audio with Modifications

Authors:
Scott N Levine, Stanford University, (U.S.A.)
Tony S. Verma, Stanford University, (U.S.A.)
Julius O Smith III, Stanford University, (U.S.A.)

Volume 6, page 3585, paper no. 2104

Abstract:
In this paper, we describe an computationally efficient method of generating more accurate sinusoidal parameters amplitude, frequency, phase from a wideband polyphonic audio source in a multiresolution, non-aliased fashion. This significantly improves upon previous work of sinusoidal modeling that assumes a single-pitched monophonic source, such as speech or an individual musical instrument, while using approximately the same number of sinusoids. In addition to a more general analysis, we can now perform high-quality modifications such as time-stretching and pitch-shifting on polyphonic audio with ease.
ICASSP98 Paper Abstract

Efficient Analysis/Synthesis of Percussion Musical Instrument Sounds Using an All-Pole Model

Authors:
Michael W Macon, Oregon Graduate Institute, (U.S.A.)
Alan V. McCree, Texas Instruments, (U.S.A.)
Wai-Ming Lai, Texas Instruments, (U.S.A.)
Vishu Viswanathan, Texas Instruments, (U.S.A.)

Volume 6, page 3589, paper no. 2207

Abstract:
It is well-known that an impulse-excited, all-pole filter is capable of representing many physical phenomena, including the oscillatory modes of percussion musical instruments like woodblocks, xylophones, or chimes. In contrast to the more common application of all-pole models to speech, however, practical problems arise in music synthesis due to the location of poles very close to the unit circle. The objective of this work was to develop algorithms to find excitation and filter parameters for synthesis of percussion instrument sounds using only an inexpensive all-pole filter chip (TI TSP50C1x). The paper describes analysis methods for dealing with pole locations near the unit circle, as well as a general method for modeling the transient attack characteristics of a particular sound while independently controlling the amplitudes of each oscillatory mode.
ICASSP98 Paper Abstract

Music Recognition Using Note Transition Context

Authors:
Kunio Kashino, NTT Basic Research Laboratories, (Japan)
Hiroshi Murase, NTT Basic Research Laboratories, (Japan)

Volume 6, page 3593, paper no. 2234

Abstract:
As a typical example of sound-mixture recognition, the recognition of ensemble music is addressed. Here music recognition is defined as recognizing the pitch and the name of an instrument for each musical note in monaural or stereo recordings of real music performances. The first key part of the proposed method is adaptive template matching that can cope with variability in musical sounds. This is employed in the hypothesis-generation stage. The second key part of the proposed method is musical context integration based on the probabilistic networks. This is employed in the hypothesis-verification stage. The evaluation results clearly show the advantages of these two processes.
ICASSP98 Paper Abstract
A System for Machine Recognition of Music Patterns

Authors:
Edward J. Coyle, Purdue University, (U.S.A.)
Ilya Shmulevich, University of Nijmegen, (The Netherlands)

Volume 6, page 3597, paper no. 2541

Abstract:
We introduce a system for machine recognition of music patterns. The problem is put into a pattern recognition framework in the sense that an error between a target pattern and scanned pattern is minimized. The error takes into account pitch and rhythm information. The pitch error measure consists of an absolute error and a perceptual error. The latter depends on an algorithm for establishing the tonal context which is based on Krumhansl's key-finding algorithm. The sequence of maximum correlations that it outputs is smoothed with a cubic spline and is used to determine weights for perceptual and absolute pitch errors. Maximum correlations are used to create the assigned key sequence, which is then filtered by a recursive median filter to improve the structure of the output of the key finding algorithm. A procedure for choosing the weights given to pitch and rhythm errors is discussed.
ICASSP98 Paper Abstract

Dichotic Presentation of Speech Signal with Critical Band Filtering for Improving Speech Perception

Authors:
Devendra S Chaudhari, Institute of Technology, Bombay, (India)
Prem C Pandey, Institute of Technology, Bombay, (India)

Volume 6, page 3601, paper no. 1159

Abstract:
Spread of spectral masking along the cochlear partion is one of the major factors contributing to the relatively poor speech reception in cases of hearing impairment of sensorineural origin. We have carried out experimental evaluation of splitting speech into two signals on the basis of frequency and presenting it dichotically over two ears for increasing the speech intelligibility. In this scheme, input speech signal is filtered into two signals by using a bank of critical bank filters where odd numbered critical bands are presented to one ear and even numbered ones to the other. Thus, the effect of spectral masking on speech information in the cochlea is reduced, and the dichotically presented signals are perceptually integrated in the auditory cortex. The processing of speech signal quantized with 12-bit resolution with centre frequencies ranging from 150 Hz to 4.80 kHz. The scheme was evaluated using normal hearing subjects, with sensorineural loss being simulated by adding white noise to the speech signal as a masker at different SNRs. Listening tests were carried out to record stimulus-response confusion matrices. Test stimuli consisted of twelve English consonants in vowel-consonant -vowel and consonant-vowel context with vowel /a/. The relative improvements in recognition scores was about 15%. Improvement in speech reception was contributed by the features of voicing, place, and manner.
ICASSP98 Paper Abstract

A Realtime Robust Adaptive Microphone Array Controlled by an SNR Estimate

Authors:
Osamu Hoshuyama, NEC Corporation, (Japan)
Brigitte Begasse, INSA, (France)
Akihiko Sugiyama, NEC Corporation, (Japan)
Akihiro Hirano, NEC Corporation, (Japan)

Volume 6, page 3605, paper no. 1312

Abstract:
A robust adaptive microphone array (RAMA) using a new adaptation-mode control method (AMC) and its evaluation by hardware are presented. The adaptation of the RAMA is controlled based on an SNR (signal-to-noise) estimate using the output powers of the fixed beamformer and the adaptive blocking matrix. The RAMA is implemented on a multi-DSP realtime signal-processing system with a C-compiler. Simulation results with real acoustic data show that the AMC based on the SNR estimate causes less breathing noise than the conventional AMC and that it obtains 1.0-point higher score on a 5-point mean opinion score scale. Evaluation through a the realtime signal-processing system demonstrates that noise reduction achieved by the RAMA is over 12 dB even in reverberant environments.
ICASSP98 Paper Abstract

Automatic Classification of Environmental Noise Events by Hidden Markov Models

Authors:
Paul Gaunard, Faculte Polytechnique de Mons, (Belgium)
Corine Ginette Mubikangiey, Faculte Polytechnique de Mons, (Belgium)
Christophe Couvreur, Faculte Polytechnique de Mons, (Belgium)
Vincent Fontaine, Faculte Polytechnique de Mons, (Belgium)

Volume 6, page 3609, paper no. 1684

Abstract:
The automatic classification of environmental noise sources from their acoustic signatures recorded at the microphone of a noise monitoring system (NMS) is an active subject of research nowadays. This paper shows how hidden Markov models (HMM's) can be used to build an environmental noise recognition system based on a time-frequency analysis of the noise signal. The performance of the proposed HMM-based approach is evaluated experimentally for the classification of five types of noise events (car, truck, moped, aircraft, train). The HMM-based approach is found to outperform previously proposed classifiers based on the average spectrum of noise event with more than 95% of correct classifications. For comparison, a classification test is performed with human listeners for the same data which shows that the best HMM-based classifier outperforms the “average” human listener who achieves only 91.8% of correct classification for the same task.
ICASSP98 Paper Abstract

On the Use of Explicit Speech Modeling in Microphone Array Applications

Authors:
Michael S Brandstein, Harvard University, (U.S.A.)

Abstract:
This paper addresses the limitations of current approaches to distant-talker speech acquisition and advocates the development of techniques which explicitly incorporate the nature of the speech signal (e.g., statistical non-stationarity, method of production, pitch, voicing, formant structure, and source radiator model) into a multi-channel context. The goal is to combine the advantages of spatial filtering achieved through beamforming with knowledge of the desired time-series attributes. The potential utility of such an approach is demonstrated through the application of a multi-channel version of the Dual Excitation speech model.
ICASSP98 Paper Abstract

Construction of a Joint Peak-Interval Histogram Using Higher-Order Cumulant-Based Inverse Filtering

Authors:
Sheau-Fang Lei, Plattsburgh State University, (U.S.A.)
Roger P Hamernik, Plattsburgh State University, (U.S.A.)

Volume 6, page 3617, paper no. 2115

Abstract:
Conventional metrics used to quantify signals in noise/hearing research rely primarily on time-averaged energy and spectral analyses. Such metrics, while appropriate for Gaussian-distributed waveforms, are of limited value in the more complex sound environments encountered in industrial/military settings that have non-Gaussian and nonstationary-distributed waveforms. Recent research has shown that metrics incorporating the temporal characteristics of a waveform are needed to evaluate hazardous acoustic environments for purposes of hearing conservation. The joint peak-interval histogram is a prospective candidate for use in such an application. This paper shows that the joint peak-interval histogram can be obtained from an estimation of the temporal pattern of a complex noise waveform by using higher-order cumulant-based inverse filtering.
ICASSP98 Paper Abstract

Classification of Audio Signals Using Statistical Features on Time and Wavelet Transform Domains

Authors:
Tryphon Lambrou, University College London, (U.K.)
Panos Kudumakis, King's College London, (U.K.)
Robert Speller, University College London, (U.K.)
Mark Sandler, King's College London, (U.K.)
Alfred Linney, University College London, (U.K.)

Volume 6, page 3621, paper no. 2120

Abstract:
This paper presents a study on musical signal classification, using wavelet transform analysis in conjunction with statistical pattern recognition techniques. A comparative evaluation between different wavelet analysis architectures in terms of their classification ability, as well as between different classifiers is carried out. We seek to establish which statistical measurements clearly distinguish between the three different musical styles of rock, piano, and jazz. Our preliminary results suggest that the features collected by the adaptive splitting wavelet transform technique performed better compared to the other wavelet based techniques, achieving overall classifications accuracy of 91.67%, using either the Minimum Distance Classifier or the Least Squares Minimum Distance Classifier. Such a system can play a useful part in multimedia applications which require content based search, classification, and retrieval of audio signals, as defined in MPEG-7.
ICASSP98 Paper Abstract

Personal Computer Software Vowel Training Aid for the Hearing Impaired

Authors:
Andrew Matthew Zimmer, Old Dominion University, (U.S.A.)
Bingjun Dai, Old Dominion University, (U.S.A.)
Stephen A Zahorian, Old Dominion University, (U.S.A.)

Volume 6, page 3625, paper no. 2300

Abstract:
A vowel training aid system for hearing impaired persons which uses a Windows-based multimedia computer has been developed. The system provides two main displays which give visual feedback for vowels spoken in isolation and short word contexts. Feature extraction methods and neural network processing techniques provide a high degree of accuracy for speaker independent vowel training. The system typically provides correct classification of over 85% of steady state vowels spoken by adult male, adult female and child (both genders combined) speakers. Similar classification accuracy is also observed for vowels spoken in short words. Low cost and good performance make this system potentially useful for speech training at home.
Optimal Truncation Time for Matched Filter Array Processing

Authors:
Daniel V Rabinkin, CAIP Center, Rutgers University, (U.S.A.)
Dwight F Macomber, SEAS, University of Pennsylvania, (U.S.A.)
Richard J Renomeron, CAIP Center, Rutgers University, (U.S.A.)
James L Flanagan, CAIP Center, Rutgers University, (U.S.A.)

Abstract:
Matched filter array processing (MFA) has been shown to improve signal-to-noise (SNR) quality for array speech capture in reverberant environments. However, under non-optimum conditions, MFA processing is computationally costly, and may produce little improvement or even subjective quality degradation as compared with simple time delay compensation (TDC). Appropriate truncation of the MFA filter bank is shown to reduce the computational burden without significantly reducing the capture SNR. This work attempts to find an optimal truncation time with respect to room size, wall absorption and the number of microphones used for the system. Simulations were conducted to evaluate MFA performance as a function of truncation length as these parameters were varied in situations typical of teleconferencing applications. It was demonstrated that judicious MFA truncation allows a reduction in computation load without sacrificing capture SNR.
ICASSP98 Paper Abstract

Multi-Microphone Noise Cancellation For Improvement of Hearing Aid Performance

Authors:
Paul W Shields, University of Paisley, Scotland, (U.K.)
Douglas R Campbell, University of Paisley, Scotland, (U.K.)

Volume 6, page 3633, paper no. 2476

Abstract:
A scheme for binaural pre-processing of speech signals for input to a standard linear hearing aid has been investigated. The system is based on that of Toner & Campbell** who applied the Least Mean Squares (LMS) algorithm in sub-bands to speech signals from various acoustic environments and signal to noise ratios (SNR). The processing scheme attempts to take advantage of the multiple inputs to perform noise cancellation. The use of sub-bands enables a diverse processing mechanism to be employed, where the wide-band signal is split into smaller frequency limited sub-bands, which can subsequently be processed according to their signal characteristics. The results of a large scale series of intelligibility tests are presented from experiments in which acoustic speech and noise data, generated using simulated and real-room acoustics was tested on hearing impaired volunteers. ** Toner, E., Campbell, D.R., (1993), 'Speech Enhancement using Sub-Band intermittent adaptation', Speech Communication, 12, 253-259.
Novel Brick-Wall Filters Based on the Auditory System

Authors:
Amitava Biswas, Speech & Hearing Sciences, Indiana University, (U.S.A.)

Volume 6, page 3637, paper no. 2566

Abstract:
A novel class of narrow band filters is presented that offers great rejection of out of band noise and a flat top at the peak. The filter is unconventional, as the output of such a filter is a series of spikes, like the action potential in the auditory nerve. The product of its time and frequency window is less than unity even using 100% output cutoff points. A bank of such filters has been used to compute a spectrogram like display that has no streaks as seen in conventional spectrograms. This model differs from Lyon’s cochlea in several areas particularly energy management and damping factor requirements. Its Q factor depends on the signal intensity, and it produces acoustic cubic distortion products similar to the auditory system, however, it is basically linear with a very wide dynamic range and its dynamics can be analyzed using linear filter theory.
ICASSP98 Paper Abstract

Vector Quantization of Scale Factors in Advanced Audio Coder (AAC)

Authors:
Thippur V Sreenivas, Fraunhofer Institute for Integrated Circuits, (Germany)
Martin Dietz, Fraunhofer Institute for Integrated Circuits, (Germany)

Abstract:
This paper describes some experiments to reduce the load of side information in the MPEG AAC scheme using vector quantization (VQ) methods. The VQ replaces the existing differential and entropy coding of the scale factors. Various types of VQ are considered, such as sub-vector/product VQ, multi-stage VQ and tree-structured VQ which provide some advantages in the context of AAC applications such as scalability, etc. However, the VQ being a lossy compression scheme, psycho-acoustic sensitivity of the losses is very important. These are studied using objective measures such as NMR and listening tests to make proper choices for the VQ design.
ICASSP98 Paper Abstract

Low Delay Coding of Wideband Audio (20 Hz - 15 kHz) at 64 kbps

Authors:
Arouggou Jbira, Ecole Nationale Superieure des Telecommunications, ( France)
Nicolas Moreau, Ecole Nationale Superieure des Telecommunications, ( France)
Przemyslaw Dymarski, Technical University of Warsaw, ( Poland)

Volume 6, page 3645, paper no. 1395

Abstract:
A 64 kbps coder of wideband (15 kHz) monophonic audio signals is described. Its structure is based on the transform coded excitation scheme, adopted to 7 kHz band signals. Significant modifications are proposed, that yield reduction of delay while keeping an almost transparent quality of speech and music, equivalent to that provided by the MPEG1, layer II audio standard at the same bit rate. Algorithmic delay has been reduced to 17 ms - approximately 1/3 the delay of the MPEG coder.
ICASSP98 Paper Abstract

Influence of Audio Coding on Stereophonic Acoustic Echo Cancellation

Authors:
Tomas Gängler, Lund University, (Sweden)
Peter Eneroth, Lund University, (Sweden)

Volume 6, page 3649, paper no. 1557

Abstract:
Sterephononic acoustic echo cancellation has been found more difficult than echo cancellation in mono due to a high correlation between the two audio channels. Different methods to decorrelate the channels have been proposed so that the stereophonic echo canceller identifies the true echo paths and its convergence rate increases. In this paper it is shown that the use of a perceptual audio coder effectively reduces the correlation between the channels and thus convergence to the true echo paths is insured. Furthermore, in those frequency regions where the encoder introduced quantization noise is below the global perceptual masking threshold, an extra amount of inaudible noise can be added to the channels. Thereby the channel correlation is further decreased and the solution is stabilized. In subband audio coders with high frequency resolution only minor modifications are needed in the decoder.
ICASSP98 Paper Abstract

A Time-Varying, Analysis/Synthesis Auditory Filterbank Using the Gammachirp

Authors:
Toshio Irino, ATR Human Information Processing Research Labs., (Japan)
Masashi Unoki, Japan Advanced Institute of Science and Technology, (Japan)

Volume 6, page 3653, paper no. 1645

Abstract:
A time-varying, analysis/synthesis auditory filterbank has been developed using a new implementation of the "gammachirp", which has been shown to be an excellent function for the asymmetric, level-dependent auditory filter. The gammachirp filter is shown to be implemented through a combination of a gammatone filter and an IIR asymmetric compensation filter; which largely reduces the computational cost for time-varying filtering. The gammachirp filterbank is designed using a linear gammatone filterbank and a bank of time-varying asymmetric compensation filters controlled by the sound pressure level estimated at the output of the filterbank. Since the inverse filter of the asymmetric compensation filter is always stable, it is possible to resynthesize signals from time-varying, level-dependent auditory representations. The resynthesis error is only determined by the linear analysis/synthesis gammatone filterbank. The proposed filterbank is applicable to various types of signal processing required to model human auditory filtering. URL: http://www.hip.atr.co.jp/irino/.
ICASSP98 Paper Abstract

Hybrid LPC and Discrete Wavelet Transform Audio Coding with a Novel Bit Allocation Algorithm

Authors:
Simon D Boland, Queensland University of Technology, (Australia)
Mohamed Deriche, Queensland University of Technology, (Australia)

Volume 6, page 3657, paper no. 1868

Abstract:
This paper examines a new method for coding high quality digital audio signals based on a combination of Linear Predictive Coding (LPC) and the Discrete Wavelet Transform (DWT). In this method, a linear predictor is first used to model each audio frame. Then, the prediction error is analyzed using the DWT. The LPC coefficients and DWT coefficients are quantized using a novel bit allocation scheme which minimizes the overall quantization error with respect to the masking threshold. The proposed coder is capable of delivering near-transparent audio signal quality at encoding bitrates of around 90-96 kb/s. Objective and subjective results suggest that the proposed coder operating at 90-96 kb/s has a performance comparable to that of the MPEG layer II codec operating at 128 kb/s.
ICASSP98 Paper Abstract
A New Subband Perceptual Audio Coder Using CELP

Authors:
Olivier Van Der Vrecken, BaBel Technologies sa, (Belgium)
Laurent Hubaut, TCTS - Faculté Polytechnique de Mons, (Belgium)
Florence Coulon, TCTS - Faculté Polytechnique de Mons, (Belgium)

Volume 6, page 3661, paper no. 1915

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.
ICASSP98 Paper Abstract
Perceptually Hidden Data Transmission over Audio Signals

Authors:
Paolo Prandoni, EPFL, Lausanne, (Switzerland)
Martin Vetterli, EPFL, Lausanne, (Switzerland)

Abstract:
A data transmission framework is proposed to embed digital data into an audio signal in a perceptually undetectable or almost undetectable way. The resulting signal can be reproduced as is with no loss of acoustic quality; the embedded data can be exactly retrieved at the decoder. The transmission process exploits the perceptual redundancy of the audio signal to conceal the acoustic impact of the embedded data; encoding of side information is used to inform the receiver of the time-varying structure of the masking properties of the audio signal. A sample implementation is described with a throughput of the order of 30 kbit/sec over CD-quality audio.
ICASSP98 Paper Abstract

MPEG Audio Bit Rate Scaling on Coded Data Domain

Authors:
Yasuyuki Nakajima, KDD Co. Ltd., (Japan)
Hiromasa Yanagihara, KDD Co. Ltd., (Japan)
Akio Yoneyama, KDD Co. Ltd., (Japan)
Masaru Sugano, KDD Co. Ltd., (Japan)

Volume 6, page 3669, paper no. 5237

Abstract:
Formerly, once the audio data is compressed, transcoding is used to scale the bit rate, where decoding and re-encoding are taken place. Therefore, data manipulation of coded data has been very complex and time consuming work. In this paper, we describe three algorithms for bit rate scaling on coded MPEG data domain. One is bandwidth limitation method cutting higher frequency components until target data rate is satisfied. The other two use re-quantization process where a quantization step in each subband is modified. One of them reflects psychoacoustic model from bit allocation information obtained in the bitstream in order to improve bit rate scaling efficiency. The simulation results show that re-quantization process provides very high conversion efficiency and nearly equal sound quality to direct coding one can be obtained by reflecting psychoacoustic model. It is also shown that very fast scaling (factor of six) have been achieved when compared with transcoding method.
ICASSP98 Paper Abstract

Stereophonic Acoustic Echo Cancellation Using Nonlinear Transformations and Comb Filtering

Authors:
Jacob Benesty, Bell Labs, (U.S.A.)
Dennis R Morgan, Bell Labs, (U.S.A.)
Joseph L Hall, Bell Labs, (U.S.A.)
Man M Sondhi, Bell Labs, (U.S.A.)

Volume 6, page 3673, paper no. 1059

Abstract:
Stereophonic sound becomes more and more important in a growing number of applications (such as teleconferencing, multimedia workstations, televideo gaming, etc.) where spatial realism is demanded. Such hands-free systems need stereophonic acoustic echo cancelers (AECs) to reduce echos that result from coupling between loudspeakers and microphones in full-duplex communication. In this paper we propose a new stereo AEC based on two experimental observations: (a) the stereo effect is due mostly to sound energy below about 1 kHz and (b) comb filtering above 1 kHz does not degrade auditory localization. The principle of the proposed structure is to use one stereo AEC at low frequencies (e.g. below 1 kHz) with nonlinear transformations on the input signals and another stereo AEC at higher frequencies (e.g. above 1 kHz) with complementary comb filters on the input signals.
ICASSP98 Paper Abstract

A Stereo Echo Canceler with Pre-Processing for Correct Echo-Path Identification

Authors:
Yann Joncour, NEC Corporation, (Japan)
Akihiko Sugiyama, NEC Corporation, (Japan)

Volume 6, page 3677, paper no. 1727

Abstract:
A new stereo echo canceler with pre-processing for correct echo-path identification is proposed. The pre-processing is accomplished by a two-tap time-varying filter which delays the input signal periodically by one sample in one of the two channels. Aliasing components and audible clicks by pre-processing are made inaudible by selecting appropriate parameters for the filter. Simulations with the NLMS algorithm and a white Gaussian signal confirm the correct echo-path identification. For speech signals, the convergence speed of the proposed echo canceler is more than three times faster than that of an echo canceler with nonlinear transformations. Results of a subjective listening test demonstrate that quality of the pre-processed signals is 4.38 using the CCIR five-grade impairment scale. This is acceptable for general teleconference applications.
ICASSP98 Paper Abstract

Using Auditory Properties to Improve the Behavior of Stereophonic Acoustic Echo Cancellers

Authors:
Andre Gilloire, Telecom / CNET, (France)
Valerie Turbin, Telecom / CNET, (France)

Volume 6, page 3681, paper no. 1772

Abstract:
We focus on the problem of stereophonic acoustic echo cancellation for teleconference applications. To limit the well-known detrimental effect of the correlation between the loudspeaker input signals, we propose a new method which consists in adding to these signals random noises controlled by auditory properties. We describe this method in some details and we show that its complexity can be fairly low. We demonstrate experimentally that the improvement yielded by this method is higher than the one provided by a former method based on the use of a non-linearity.
New Configuration for a Stereo Echo Canceller with Non-linear Pre-Processing

Authors:
Suehiro Shimauchi, NTT Human Interface Laboratories, (Japan)
Yoichi Haneda, NTT Human Interface Laboratories, (Japan)
Shoji Makino, NTT Human Interface Laboratories, (Japan)
Yutaka Kaneda, NTT Human Interface Laboratories, (Japan)

Abstract:
A new configuration for a stereo echo canceller with nonlinear pre-processing is proposed. The pre-processor which adds uncorrelated components to the original received stereo signals improves the adaptive filter convergence even in the conventional configuration. However, because of the inaudibility restriction, the pre-processed signals still include a large amount of the original stereo signals which are often highly cross-correlated. Therefore, the improvement is limited. To overcome this, our new stereo echo canceller includes exclusive adaptive filters whose inputs are the uncorrelated signals generated in the pre-processor. These exclusive adaptive filters converge to true solutions without suffering from cross-correlation between the original stereo signals. This is demonstrated through computer simulation results.
ICASSP98 Paper Abstract

Stereophonic Acoustic Echo Cancellation System Using Time-Varying All-Pass Filtering for Signal Decorrelation

Authors:
Murtaza Ali, Texas Instruments, (U.S.A.)

Abstract:
This paper describes a novel technique for decorrelating the stereo signals in stereophonic acoustic echo cancellation (AEC) systems. At present, most teleconferencing systems use a single full-duplex audio channel for voice communications. However, in order to introduce spatial realism, future teleconferencing systems are expected to have more than one channel (at least stereo with two channels). However, in stereophonic AEC systems, the correlation between the stereo signals does not allow correct identification of the echo path responses. In this paper, we develop a signal decorrelation technique based on time-varying all-pass filtering of the individual stereo signals. Experiments show that this technique does not effect the perception of the stereo signals, but identifies the echo path responses correctly.
ICASSP98 Paper Abstract

A Syntactic Approach to Automatic Lip Feature Extraction for Speaker Identification

Authors:
Timothy J Wark, Queensland University of Technology, (Australia)
Sridha Sridharan, Queensland University of Technology, (Australia)

Volume 6, page 3693, paper no. 1227

Abstract:
This paper presents a novel technique for the tracking and extraction of features from lips for the purpose of speaker identification. In noisy or other adverse conditions, identification performance via the speech signal can significantly reduce, hence additional information which can complement the speech signal is of particular interest. In our system, syntactic information is derived from chromatic information in the lip region. A model of the lip contour is formed directly from the syntactic information, with no minimization procedure required to refine estimates. Colour features are then extracted from the lips via profiles taken around the lip contour. Further improvement in lip features is obtained via linear discriminant analysis (LDA). Speaker models are built from the lip features based on the Gaussian Mixture Model (GMM). Identification experiments are performed on the M2VTS database, with encouraging results.
ICASSP98 Paper Abstract
Video Sequence Matching

Authors:
Rakesh Mohan, IBM, (U.S.A.)

Volume 6, page 3697, paper no. 1801

Abstract:
We present a novel scheme to match a video clip against a large database of videos. Unlike previous schemes that match videos based on image similarity, this scheme matches videos based on similarity of temporal activity, i.e., it finds similar 'actions.' Furthermore, it provides precise temporal localization of the actions in the matched videos. Video sequences are represented as a sequence of feature vectors called fingerprints. The fingerprint of the query video is matched against the fingerprints of videos in a database using sequential matching. The fingerprints are computed directly from compressed MPEG videos. The matching is much faster than real-time. We have used this scheme to find similar actions in sporting events, such as diving and baseball. Keywords: video matching, video search, video databases.
Progressive Resolution Motion Indexing of Video Object

Authors:
Jeho Nam, University of Minnesota, (U.S.A.)
Ahmed H. Tewfik, University of Minnesota, (U.S.A.)

Volume 6, page 3701, paper no. 2563

Abstract:
We present a novel motion-based video indexing scheme for fast content-based browsing and retrieval in a video database. The proposed technique constructs a dictionary of prototype objects to support query by motion. The first step in our approach extracts moving objects by analyzing layered images constructed from the coarse data in a 3-D wavelet decomposition of the video sequence. These images capture motion information only. Moving objects are modeled as collections of interconnected rigid polygonal shapes in the motion sequences that we derive from the wavelet representation. The motion signatures of the object are computed from the rotational and translational motions associated to the elemental polygons that form the objects. These signatures are finally stored as potential query terms.
ICASSP98 Paper Abstract

A Multiresolution Color Clustering Approach to Image Indexing and Retrieval

Authors:
Xia Wan, University of Southern California, (U.S.A.)
C. C. Jay Kuo, University of Southern California, (U.S.A.)

Volume 6, page 3705, paper no. 5226

Abstract:
We propose a multiresolution color feature extraction scheme based on octree data structure to achieve efficient and robust image retrieval. With the proposed method, multiple color features, including the dominant color, the number of distinctive colors and the color histogram, can be naturally integrated into one framework. A selective filtering strategy is also described to speed up the retrieval process. Retrieval examples are given to illustrate the performance of the proposed approach.
ICASSP98 Paper Abstract
Multiresolutional Encoding and Decoding in Embedded Image and Video Coders

Authors:
Zixiang Xiong, University of Hawaii, (U.S.A.)
Beong-Jo Kim, Rensselaer Polytechnic Institute, (U.S.A.)
William A Pearlman, Rensselaer Polytechnic Institute, (U.S.A.)

Volume 6, page 3709, paper no. 2133

Abstract:
We address multiresolutional encoding and decoding within the embedded zerotree wavelet (EZW) framework for both images and video. By varying a resolution parameter, one can obtain decoded images at different resolutions from one single encoded bitstream, which is already rate scalable for EZW coders. Similarly one can decode video sequences at different rates and different spatial and temporal resolutions from one bitstream. Furthermore, a layered bitstream can be generated with multiresolutional encoding, from which the higher resolution layers can be used to increase the spatial/temporal resolution of the images/video obtained from the low resolution layer. In other words, we have achieved full scalability in rate and partial scalability in space and time. This added spatial/temporal scalability is significant for emerging multimedia applications such as fast decoding, image/video database browsing, telemedicine, multipoint video conferencing, and distance learning.
ICASSP98 Paper Abstract
Extraction of Detailed Image Regions for Content-Based Image Retrieval

Authors:
Dimitrios Androutsos, University of Toronto, (Canada)
Kostas N Plataniotis, Ryerson Polytechnic University, (Canada)
Anastasios N Venetsanopoulos, University of Toronto, (Canada)

Volume 6, page 3713, paper no. 2481

Abstract:
We present a technique for coarsely extracting the regions of natural color images which contain directional detail, e.g., edges, texture, etc., 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. Database indices are then created from the busy regions using the directional detail histogram technique and retrieval is performed using these.
ICASSP98 Paper Abstract
Optimal Object Allocation for Multimedia Broadcast

Authors:
Edwin A Heredia, Thomson Consumer Electronics, (U.S.A.)
Volume 6, page 3717, paper no. 1864

Abstract:
With the arrival of terrestrial digital TV, a distribution network able to deliver up to 19 Mbits/s in each of the physical transmission channels will become available. Using the adopted data broadcast protocols, simultaneous transmission of multimedia documents to large population segments can be achieved. While these protocols describe methods for recognizing files in data streams, no method is known yet on how to distribute large collections of files in one or more data streams. This paper addresses this problem. The method proposed in the paper allocates objects in multiple streams according to their sizes and access probabilities, in such a way that average access latency is minimized. We show that the minimization problem can be described as a particular form of the NP hard quadratic allocation model for which an algorithmic solution for finding local minima exists.
An Integrated Progressive Image Coding and Watermark System

Authors:
Houngjyh Wang, University of Southern California, (U.S.A.)
C. C. Jay Kuo, University of Southern California, (U.S.A.)

Volume 6, page 3721, paper no. 5225

Abstract:
The design of an integrated image coding and watermark system with the wavelet transform is examined in this work. First, the multi-threshold wavelet codec (MTWC) is used to achieve the image compression purpose. Unlike other embedded wavelet coders which use a single initial threshold in their successive approximate quantization (SAQ), MTWC adopts different initial thresholds in different subbands. A superior rate-distortion tradeoff is achieved by MTWC with a low computational complexity. Then, a non-invertible progressive watermark scheme is incorporated in MTWC for copyright protection. This watermark scheme uses the user input data to produce a Gaussian distribution pseudorandom watermark in the wavelet domain. The performance of the proposed watermark technology is supported by experimental results.
ICASSP98 Paper Abstract
On Combining Watermarking with Perceptual Coding

Authors:
Jack B Lacy, AT&T Labs, (U.S.A.)
Schuyler R. Quackenbush, AT&T Labs, (U.S.A.)
Amy R Reibman, AT&T Labs, (U.S.A.)
David H Shur, AT&T Labs, (U.S.A.)
James H Snyder, AT&T Labs, (U.S.A.)

Volume 6, page 3725, paper no. 2530

Abstract:
A watermark is a data stream inserted into multimedia content. It contains information relevant to the ownership or authorized use of the content. A watermark which could be recovered without a priori knowledge of the identify of the content could be used by web search mechanisms to flag unauthorized distribution of the content. Since media will be compressed on these sites, a mark detection algorithm that operates in the compressed domain would be useful. We describe in this paper a watermark algorithm which operates in the compressed domain and does not require a reference.
ICASSP98 Paper Abstract

Face Authentication Using Variants of Elastic Graph Matching Based on Mathematical Morphology that Incorporate Local Discriminant Coefficients

Authors:
Constantine Kotropoulos, Aristotle University of Thessaloniki, (Greece)
Anastasios Tefas, Aristotle University of Thessaloniki, (Greece)
Ioannis Pitas, Aristotle University of Thessaloniki, (Greece)

Volume 6, page 3729, paper no. 5224

Abstract:
Two novel variants of Dynamic Link Architecture that are based on mathematical morphology and incorporate local coefficients which weigh the contribution of each node according to its discriminatory power in elastic graph matching are proposed, namely, the Morphological Dynamic Link Architecture and the Morphological Signal Decomposition-Dynamic Link Architecture. They are tested for face authentication in a cooperative scenario where the candidates claim an identity to be checked. Their performance is evaluated in terms of their receiver operating characteristic and the Equal Error Rate achieved in M2VTS database. An Equal Error Rate of 6.6% - 6.8% is reported.
Discriminative Training of HMM Stream Exponents for Audio-Visual Speech Recognition

Authors:
Gerasimos Potamianos, AT&T Labs, (U.S.A.)
Hans Peter Graf, AT&T Labs, (U.S.A.)

Volume 6, page 3733, paper no. 1802

Abstract:
We propose the use of discriminative training by means of the generalized probabilistic descent (GPD) algorithm to estimate hidden Markov model (HMM) stream exponents for audio-visual speech recognition. Synchronized audio and visual features are used to respectively train audio-only and visual-only single-stream HMMs of identical topology by maximum likelihood. A two-stream HMM is then obtained by combining the two single-stream HMMs and introducing exponents that weigh the log-likelihood of each stream. We present the GPD algorithm for stream exponent estimation, consider a possible initialization, and apply it to the single speaker connected letters task of the AT&T bimodal database. We demonstrate the superior performance of the resulting multi-stream HMM to the audio-only, visual-only, and audio-visual single-stream HMMs.
ICASSP98 Paper Abstract
A Hybrid Real-Time Face Tracking System

Authors:
Ce Wang, *Harvard University*, (U.S.A.)
Michael S Brandstein, *Harvard University*, (U.S.A.)

Volume 6, page 3737, paper no. 1917

Abstract:
A hybrid real-time face tracker based on both sound and visual cues is presented. Initial talker locations are estimated acoustically from microphone array data while precise localization and tracking are derived from image information. A computationally efficient algorithm for face detection via motion analysis is employed to track individual faces at rates up to 30 frames per second. The system is robust to nonlinear source motions, complex backgrounds, varying lighting conditions, and a variety of source-camera depths. While the direct focus of this work is automated video conferencing, the face tracking capability has utility to many multimedia and virtual reality applications.
ICASSP98 Paper Abstract

A Hidden Markov Model Framework for Video Segmentation Using Audio and Image Features

Authors:
John S Boreczky, FX Palo Alto Laboratory, (U.S.A.)
Lynn D Wilcox, FX Palo Alto Laboratory, (U.S.A.)

Abstract:
This paper describes a technique for segmenting video using hidden Markov models (HMM). Video is segmented into regions defined by shots, shot boundaries, and camera movement within shots. Features for segmentation include an image-based distance between adjacent video frames, an audio distance based on the acoustic difference in intervals just before and after the frames, and an estimate of motion between the two frames. Typical video segmentation algorithms classify shot boundaries by computing an image-based distance between adjacent frames and comparing this distance to fixed, manually determined thresholds. Motion and audio information is used separately. In contrast, our segmentation technique allows features to be combined within the HMM framework. Further, thresholds are not required since automatically trained HMMs take their place. This algorithm has been tested on a video data base, and has been shown to improve the accuracy of video segmentation over standard threshold-based systems.
ICASSP98 Paper Abstract

Text-to-Visual Speech Synthesis Based on Parameter Generation from HMM

Authors:
Takashi Masuko, Tokyo Institute of Technology, (Japan)
Takao Kobayashi, Tokyo Institute of Technology, (Japan)
Masatsune Tamura, Tokyo Institute of Technology, (Japan)
Jun Masubuchi, Tokyo Institute of Technology, (Japan)
Keiichi Tokuda, Nagoya Institute of Technology, (Japan)

Abstract:
This paper presents a new technique for synthesizing visual speech from arbitrarily given text. The technique is based on an algorithm for parameter generation from HMM with dynamic features, which has been successfully applied to text-to-speech synthesis. In the training phase, syllable HMMs are trained with visual speech parameter sequences that represent lip movements. In the synthesis phase, a sentence HMM is constructed by concatenating syllable HMMs corresponding to the phonetic transcription for the input text. Then an optimum visual speech parameter sequence is generated from the sentence HMM in ML sense. The proposed technique can generate synchronized lip movements with speech in a unified framework. Furthermore, coartication is implicitly incorporated into generated mouth shapes. As a result, synthetic lip motion becomes smooth and realistic.
ICASSP98 Paper Abstract
Digital Processing of Affective Signals

Authors:
Jennifer A Healey, MIT Media Lab, (U.S.A.)
Rosalind W Picard, MIT Media Lab, (U.S.A.)

Volume 6, page 3749, paper no. 2285

Abstract:
Affective signal processing algorithms were developed to allow a digital computer to recognize the affective state of a user who is intentionally expressing that state. This paper describes the method used for collecting the training data, the feature extraction algorithms used and the results of pattern recognition using a Fisher linear discriminant and the leave one out test method. Four physiological signals, skin conductivity, blood volume pressure, respiration and an electromyogram (EMG) on the masseter muscle were analyzed. It was found that anger was well differentiated from peaceful emotions (90%-100%), that high and low arousal states were distinguished (80%-88%), but positive and negative valence states were difficult to distinguish (50%-82%). Subsets of three emotion states could be well separated (75%-87%) and characteristic patterns for single emotions were found.
ICASSP98 Paper Abstract

Immersive Audio for the Desktop

Authors:
Chris Kyriakakis, University of Southern California, (U.S.A.)
Tomlinson Holman, University of Southern California, (U.S.A.)

Volume 6, page 3753, paper no. 2337

Abstract:
Integrated media workstations are increasingly being used for creating, editing, and monitoring sound that is associated with video or computer-generated images. While the requirements for high quality reproduction in large-scale systems are well understood, these have not yet been adequately translated to the workstation environment. In this paper we discuss several factors that pertain to high quality sound reproduction at the desktop including acoustical considerations, signal processing requirements, and listener location issues. We also present a novel desktop system design with integrated listener-tracking capability that circumvents several of the problems faced by current digital audio and video workstations.
ICASSP98 Paper Abstract

Speech Interaction in Virtual Reality

Authors:
Johannes Mueller, Munich University of Technology, (Germany)
Christian Krapichler, GSF, Neuherberg, (Germany)
Lam Son Nguyen, Munich University of Technology, (Germany)
Karl-Hans Englmeier, GSF, Neuherberg, (Germany)
Manfred Lang, Munich University of Technology, (Germany)

Volume 6, page 3757, paper no. 1076

Abstract:
A system for the visualization of three-dimensional anatomical data, derived from Magnetic Resonance Imaging (MRI) or Computed Tomography (CT), enables the physician to navigate through and interact with the patient’s 3D scans in a virtual environment. This paper presents the multimodal human-machine interaction focusing the speech input. For the concerning task, a speech understanding front-end using a special kind of semantic decoder was successfully adopted. Now, the navigation as well as certain parameters and functions can be directly accessed by spoken commands. Using the implemented interaction modalities, speed and efficiency of the diagnosis could be considerably improved.
ICASSP98 Paper Abstract

Word Learning in a Multimodal Environment

Authors:
Deb K Roy, MIT Media Lab, (U.S.A.)
Alex Pentland, MIT Media Lab, (U.S.A.)

Volume 6, page 3761, paper no. 2153

Abstract:
We are creating human machine interfaces which let people communicate with machines using natural modalities including speech and gesture. A problem with current multimodal interfaces is that users are forced to learn the set of words and gestures which the interface understands. We report on a trainable interface which lets the user teach the system words of their choice through natural multimodal interactions.
ICASSP98 Paper Abstract

Sign Language Communication between Japanese-Korean and Japanese-Portuguese using CG Animation

Authors:
Yoshinao Aoki, Hokkaido University, (Japan)
Ricardo Mitsumori, Hokkaido University, (Japan)
Jincan Li, Hokkaido University, (Japan)
Alexander Burger, Langweid, (Germany)

Volume 6, page 3765, paper no. 1334

Abstract:
In this paper we propose a sign language communication between different languages such as Japanese-Korean and Japanese-Portuguese using CG animation of sign language based on the intelligent image communication method. For this purpose sign language animation is produced using data of gesture or text data expressing sign language. In the reduction process of CG animation of sign language, MATLAB and LIFO language are used, where MATLAB is useful for three-dimensional signal processing of gestures and for displaying animation of sign language. On the other hand LIFO language, which is a descendant of the LISP and FORTH language families, is developed and used to produce live CG animations, resulting in a high-speed interactive system of designing and displaying sign language animations. A simple experiment was conducted to translate Japanese sign language into Korean and Portuguese sign languages using the developed CG animation system.
ICASSP98 Paper Abstract

Probing the Relationship between Qualitative and Quantitative Performance Measures for Voice-Enabled Telecommunication Services

Authors:
Shrikanth Narayanan, AT&T Labs, (U.S.A.)
Mani Subramaniam, AT&T Labs, (U.S.A.)
Benjamin Stern, AT&T Labs, (U.S.A.)
Barbara Hollister, AT&T Labs, (U.S.A.)
Chih-mei Lin, AT&T Labs, (U.S.A.)

Volume 6, page 3769, paper no. 2142

Abstract:
The relationship between objective speech recognition performance measures and perceived performance is analyzed and modeled using data obtained from a voice-dialing trial with 798 AT&T customers. The ability of these models for predicting user perception and overall demand for such voice-enabled services is discussed.
ICASSP98 Paper Abstract

Signal Processing for Recognition of Human Frustration

Authors:
Raul Fernandez, MIT Media Lab, (U.S.A.)
Rosalind W. Picard, MIT Media Lab, (U.S.A.)

Volume 6, page 3773, paper no. 2468

Abstract:
In this work, inspired by the application of human-machine interaction and the potential use that human-computer interfaces can make of knowledge regarding the affective state of a user, we investigate the problem of sensing and recognizing typical affective experiences that arise when people communicate with computers. In particular, we address the problem of detecting "frustration" in human computer interfaces. By first sensing human biophysiological correlates of internal affective states, we proceed to stochastically model the biological time series with Hidden Markov Models to obtain user-dependent recognition systems that learn affective patterns from a set of training data. Labeling criteria to classify the data are discussed, and generalization of the results to a set of unobserved data is evaluated. Significant recognition results (greater than random) are reported for 21 of 24 subjects.
ICASSP98 Paper Abstract
Quick Audio Retrieval Using Active Search

Authors:
Gavin A Smith, NTT Basic Research Laboratories, (Japan)
Hiroshi Murase, NTT Basic Research Laboratories, (Japan)
Kunio Kashino, NTT Basic Research Laboratories, (Japan)

Volume 6, page 3777, paper no. 1987

Abstract:
This paper discusses a method to search quickly through broadcast audio data to detect and locate known sounds using reference templates, based on the active search algorithm and histogram modeling of zero-crossing features. Active search reduces the number of candidate matches between reference and test template by up to 36 times compared to exhaustive search, while still remaining optimal. Computation is further reduced by using computationally inexpensive zero-crossing features. The method is robust against white noise addition down to 20dB signal-to-noise ratios and digitization noise.
ICASSP98 Paper Abstract

Retrieval of Broadcast News Documents with the THISL System

Authors:
David C Abberley, Sheffield University, (U.K.)
Steve J Renals, Sheffield University, (U.K.)
Gary D Cook, Cambridge University, (U.K.)

Volume 6, page 3781, paper no. 2015

Abstract:
This paper describes a spoken document retrieval system, combining the Abbot large vocabulary continuous speech recognition (LVCSR) system developed by Cambridge University, Sheffield University and SoftSound, and the PRISE information retrieval engine developed by NIST. The system was constructed to enable us to participate in the TREC 6 Spoken Document Retrieval experimental evaluation. Our key aims in this work were to produce a complete system for the SDR task, to investigate the effect of a word error rate of 30-50% on retrieval performance and to investigate the integration of LVCSR and word spotting in a retrieval task.
ICASSP98 Paper Abstract

Digital Image/Video Library and MPEG-7: Standardization and Research Issues

Authors:
Yong Rui, Beckman Institute, University of Illinois-Urbana/Champaign, (U.S.A.)
Thomas S. Huang, Beckman Institute, University of Illinois-Urbana/Champaign, (U.S.A.)
Shih-Fu Chang, Columbia University, (U.S.A.)

Volume 6, page 3785, paper no. 5250

Abstract:
Much research activity and interest has emerged recently in two closely related areas: Digital Image/Video Library (DIVL) and MPEG-7. In this paper, we review the critical research issues in DIVL from a signal processing viewpoint, the objectives and scope of MPEG-7, and the relationships between these two.
ICASSP98 Paper Abstract

Multimedia Content Description in the InfoPyramid

Authors:
Chung-Sheng Li, IBM T.J. Watson Research Center, (U.S.A.)
Rakesh Mohan, IBM T.J. Watson Research Center, (U.S.A.)
John R. Smith, IBM T.J. Watson Research Center, (U.S.A.)

Abstract:
There is a growing need for developing a content description language for multimedia that improves searching, indexing and managing of the multimedia content. The MPEG group recently established the MPEG-7 effort to standardize the multimedia content interface. The proposed interface will bridge the gap between various types of content meta-data, such as content features, annotations, relationships, and the search engines. In this paper, we develop a method of handling multimedia content description in a new multi-abstraction, multi-modal content representation framework called InfoPyramid. The InfoPyramid facilitates the search, retrieval, manipulation, and transmission of multimedia data by providing a hierarchy for content descriptors. We illustrate the suitability of the InfoPyramid multimedia content description to MPEG-7 by examining four multimedia retrieval applications: a Web-image search engine, a satellite image retrieval system, an Internet content delivery system, and a TV news storage and retrieval system.
ICASSP98 Paper Abstract

H.263+: The New ITU-T Recommendation for Video Coding at Low Bit Rates

Authors:
Thomas R. Gardos, Intel Corporation, (U.S.A.)

Volume 6, page 3793, paper no. 5254

Abstract:
H.263+ is a revision to the 1996 version of ITU-T Recommendation H.263 that brings incremental improvements to compression performance, better support for packet-based networks, expanded support for video formats as well as other new functionality. All the new capabilities can be negotiated individually or disabled for backwards compatibility with H.263. In this paper, we review all the major new features of H.263+. 
ICASSP98 Paper Abstract
Coding of Natural Audio in MPEG-4

Authors:
Schuyler R. Quackenbush, AT&T Labs, (U.S.A.)

Volume 6, page 3797, paper no. 5252

Abstract:
MPEG-4 standardizes natural audio coding at bitrates ranging from 2 kbit/s, suitable for intelligible speech coding, to 64 kbit/s per channel, suitable for high-quality audio coding. Within this range, three categories of coding are defined: parametric coding, Code Excited Linear Predictive coding (CELP) and time/frequency (T/F) coding. The unique contribution of MPEG-4 audio is that not only does it scale across a wide range of bitrates, but it also scales across a broad set of other parameters, such as sampling rate, bandwidth, voice pitch and complexity. This paper presents an overview of the MPEG-4 natural audio coding framework and each of its component coding techniques.
ICASSP98 Paper Abstract
The MPEG-4 Structured Audio Standard

Authors:
Eric D. Scheirer, MIT Media Lab, (U.S.A.)

Volume 6, page 3801, paper no. 5253

Abstract:
The MPEG-4 standard defines numerous tools that represent the state-of-the-art in representation, transmission, and decoding of multimedia data. Among these is a new type of audio standard, termed "Structured Audio". The MPEG-4 standard for structured audio allows for the efficient, flexible description of synthetic sound in synchronization with natural sound in interactive multimedia scenes. A discussion of the capabilities, technological underpinnings, and application of MPEG-4 Structured Audio is presented.
ICASSP98 Paper Abstract
Natural and Synthetic Video in MPEG-4

Authors:
Joern Ostermann, AT&T Labs - Research, (U.S.A.)
Atul Puri, AT&T Labs - Research, (U.S.A.)

Volume 6, page 3805, paper no. 5248

Abstract:
The ISO MPEG committee, after successful completion of the MPEG-1 and MPEG-2 standards, has recently completed the Committee Draft for MPEG-4, its third standard. MPEG-4 is designed to be an object-based standard for multimedia coding. The visual part of the standard specifies coding of both natural and synthetic video. The MPEG-4 visual standard supports coding of natural video not only in a conventional manner (using frames) but also as a collection of arbitrary shape objects (using video object planes). Further, it supports functionalities such as spatial and temporal scalability, both conventional and with arbitrary shape objects. It also supports error resilient coding for delivery of coded video on error prone channels. MPEG-4 visual standard also supports coding of synthetic video which includes still texture maps used in 3D graphics models, mesh geometry for object animation, and parameters for facial animation.
ICASSP98 Paper Abstract

Protocols for Real-Time Multimedia Data Transmission over the Internet

Authors:
M. Reha Civanlar, AT&T Labs - Research, (U.S.A.)

Volume 6, page 3809, paper no. 5255

Abstract:
The explosive growth of the Internet and the intranets have attracted a great deal of attention to the implementation and performance of networked multimedia services, which involve the transport of real-time multimedia data streams over non-guaranteed quality of service (QoS) networks based on the Internet Protocol (IP). In this paper, I present an overview of the existing architectural elements supporting real-time data transmission over the Internet. Effective implementations of such systems require a thorough understanding of both the network protocols and the coding systems used for compressing the signals to be transmitted in real-time. This paper includes a section discussing the issues to be considered in designing signal compression applications suitable for network use.
MEMIS - MHEG Environment for Multimedia Information and Simulation

Authors:
Alan Gauton, University of Strathclyde, Scotland, (U.K.)
Tariq S Durrani, University of Strathclyde, Scotland, (U.K.)

Abstract:
MHEG represents a new multimedia and hypermedia standard proposed by ISO/IEC. This paper presents a new software authoring environment based around MHEG-5 that offers users a vehicle for creating multimedia applications that can interact with external programs which involve intense computational tasks. MEMIS provides a linkage between a multimedia front-end and externally available computational processes. The paper provides a background to the development of the environment, by identifying the facilities offered by MHEG, discusses the efficacy offered by MHEG Vs JAVA for multimedia development; and then covers the development process and includes specific exemplars of the environment for managing multimedia applications which include (a) real-time signal processing embedded within a LabVIEW kernel, (b) ATM, (c) Set Top Box, and (d) Kiosk for Internet commerce that utilises MATLAB type calls. The results includes system level architecture for multimedia implementation, and the timing requirements for such applications.
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