AOARD-TR-94-39

AOARD REPORT

1994 IEEE International Workshop on Intelligent Signal Processing and Communication Systems (ISPACS94)

October 5-7, 1994
T. Davis
AOARD

DTIC
ELECTE
APR 1 4 1995
S D
G

A summary of the 1994 IEEE International Workshop on Intelligent Signal Processing and Communication Systems (ISPACS94), conducted October 5-7, 1994 at Yonsei University, Seoul, Korea is presented. Abstracts of all presented papers are included. This report is based upon information collected via workshop attendance, review of the proceedings, and conversations with other attendees.

19950321 077

DISTRIBUTION STATEMENT A:
APPROVED FOR PUBLIC RELEASE; DISTRIBUTION IS UNLIMITED.

AIR FORCE OFFICE OF SCIENTIFIC RESEARCH

ASIAN OFFICE OF AEROSPACE RESEARCH AND DEVELOPMENT

TOKYO, JAPAN
UNIT 45002
APO AP 96337-0007
DSN: (315)229-3212
Comm: 81-3-5410-4409

DTIC QUALITY INSPECTED 1

1. Event Organization and Background.

The 1994 IEEE International Workshop on Intelligent Signal Processing and Communication Systems (ISPACS94), the 1994 installment in a series of annual advanced signal processing workshops, was conducted October 5-7, 1994 at Yonsei University, Seoul, Korea. ISPACS94 was co-sponsored by the IEEE Communications Society and the IEEE Korea Council. The Korean members of the Workshop General Chair and the Technical Program Committee Chair were respectively Professor Byeong Gi Lee of Seoul National University and Professor Yeong Ho Ha of Kyungpook National University. Their contact information, along with that of Professor Dae Hee Youn from Yonsei University, who handled local arrangements, appears below.

Professor Byeong Gi Lee
Seoul National University
Shinlim-dong
Kwanak-gu
Seoul 151-742 Korea
Tel: +82 2 880-8426
Fax: +82 2 885-4459

Professor Yeong Ho Ha
Dept. of Electronic Engineering
Kyungpook National University
1370 Sankyuk-Dong
Taegu, 702-701 Korea
Tel: +82 53-950-5535
Fax: +82 53-950-5505
e-mail: yhha@ee.kyungpook.ac.kr

Professor Dae Hee Youn
Dept. of Electronic Engineering
Yonsei University
Shinchong-Dong 134
Sudaemoon-Ku
Seoul, Korea
Tel: +82 2 962-8801
Fax: +82 2 969-3754
e-mail: dolhana@bubble.yonsei.ac.kr

Accesion For
NTIS CRA&I ☒
DTIC TAB ☐
Unannounced ☐
Justification
By
Distribution /
Availability Codes
Dist | Avail and / or Special
A-1

The ISPACS94 technical program is summarized in Section 2 below. As noted there, it was composed of a keynote speech followed by 9 serial paper presentation sessions over the three day workshop schedule. Additionally, 4 off-line poster sessions were conducted in parallel with some of the paper sessions. According to the conference organizers, a total of 109 papers were submitted to the program committee, of which 87 were selected for presentation and publication in the workshop proceedings. Of the 87 selected presentations, 78 were completed and published in the proceedings. Abstracts of all workshop papers, for both presentation and poster sessions, are included in Section 3.

The technical presentations at ISPACS94 encompassed a representative array of advanced signal processing topics, including sessions on equalization, coding, visual signal processing, neural networks and fuzzy logic based processing. The single most recurrent application theme of the program was video compression. Other themes included speech processing and both printed and handwritten character recognition.

ISPACS94 attracted about eighty people representing nine different nationalities, with the largest non-Korean contingent being from Japan. The U. S. based contingent consisted of three people, all of whom are Korean nationals either studying or working in the U. S. Although the workshop had a very extensive organizing committee array, including some forty people serving on international steering, advisory and technical program committees, very few of the named representatives were in attendance.

2. Technical Program Synopsis.

The ISPACS94 technical program was opened on Wednesday morning, 5 Oct. Following a few introductory remarks by the technical co-chairs, Dr. Sakae Okubo, the conference keynote speaker, was introduced. Dr. Koubo is from the Graphics Communication Laboratories in Tokyo, Japan. His contact information is provided below.

Dr. Sakae Okubo
Graphics Communication Laboratories
4-36-19 Yoyogi
Shibuya-ku
Tokyo 151, Japan
Tel: +81 3 5351-0181
Fax: +81 3 5351-0185
e-mail: okubo@gctech.co.jp

The topic of Dr. Okubo's address was "Digital Audio-Visual Standards and Their Applications". He traced the history of the H.26X and MPEG standards from the early 1970s through 1994, and offered the observation that while hybrid coding (DPCM, motion compensation and DCT) has proven versatile, it is probably inadequate for future compression standard requirements (i.e. MPEG-4).

The remainder of the conference technical program was divided into nine serial presentation sessions and four poster sessions. The poster sessions were conducted in parallel with some of the presentation sessions. Abstracts of all papers published in the proceedings (both presentation and poster session) are provided in Section 3. They are collected by session title and appear in the order presented.

The first presentation session of the workshop was devoted to equalization. Five papers were accepted for the session but two were withdrawn prior to the workshop. The three presented papers all concerned adaptive channel equalization techniques in one form or another.

Two subsequent presentation sessions were devoted to visual signal processing. Ten papers were accepted for the sessions but one was withdrawn before the workshop. The remaining nine papers appear in the proceedings but several of the corresponding presentations were cancelled on site. Topics include image restoration, pyramid coding, camera parameter estimation and Korean print character recognition.

Two successive presentation sessions were devoted to neural networks. Ten papers were accepted for the sessions and eight were presented. One was withdrawn before the workshop and one was cancelled on site, though the corresponding paper appears in the proceedings. Topics include pattern recognition (speech and handwritten Japanese character recognition), image compression, a specialized channel equalizer (unshielded twisted pair) and an adaptive time delay recurrent NN for studying spatio-temporal correlations. None of the session papers directly concern NN theoretical convergence or learning topics.

A substantial portion of the workshop technical program, three successive sessions, was devoted to coding. Fifteen papers were accepted for the sessions and fourteen of them were presented. The papers cover a wide variety of coding topics, however they almost universally share very low bit rate image coding as a central theme. It is also noteworthy that almost the entire complement of contributors to the coding sessions represent either Korean or Japanese institutions.

The topic of the final presentation session was fuzzy logic based processing. Five papers were accepted for the session but two were withdrawn. The remaining three were presented.

The four poster sessions were conducted two per day on Thursday and Friday afternoon, 6 and 7 Oct. They were conducted in parallel with the scheduled presentation sessions. A total of forty two papers were accepted for the poster sessions, of which two were withdrawn and three cancelled on site. The poster sessions do not appear to have been organized by topic area, but seem rather to have been the repository for papers which for one reason or another did not fit into the presentation sessions.

3. Abstracts of Presented Papers.

Following is a complete set of abstracts for all papers presented at ISPACS94. They are collected by session title and in the order presented at the conference.

Session 1: Equalization

Equalization of Data Transmission through Nonlinear Channels

Ludwik Kurz and Yeong-Sam Kang Department of Electrical Engineering
Polytechnic University
Six Metrotech Center, Brooklyn, New York 11201
e-mail:ykang~photon.poly. edu

Abstract

The purpose of this paper is to design robust equalizers which have an ability to suppress dependent impulse noise and nongaussian noise based on training information. The algorithm proposed here is based on piecewise approximations to a regression function involving only quantiles and partition moments which can be estimated by Robins-Monro Stochastic Approximation (RMSA) algorithm. For the purpose of testing the robustness, we adopted mixture distribution models, e.g. e-contaminated Gaussian distribution introduced by Tukey and Huber as noise models. The performance of the robust MMSE equalizers in satellite channels is considered for a wide range of SNR and various forms of non.gaussian noise. The results are compared to a single sample detector and conventional equalizers, in terms of error rate and MMSE. The superiority of the proposed robust equalizers is clearly demonstrated. Due to extreme difficulty in calculating PE and MMSE, Monte-Carlo simulations were selected as means of comparison.

EQUALIZATION OF NONLINEAR DIGITAL SATELLITE CHANNELS USING A FREQUENCY-DOMAIN ADAPTIVE VOLTERRA FILTER

Sungbin Im and Edward J. Powers
Department of Electrical and Computer Engineering, and Electronics Research Center
The University of Texas at Austin
Austin, Texas 78712-1084, USA
FAX: (512) 471-1856, TEL: (512) 471-1430

ABSTRACT

The objective of this paper is to present a new structure for an adaptive nonlinear equalizer for digital transmission over a nonlinear satellite chamiel with PSK modulation. The adaptive nonlinear equalizer is based on a discrete frequency-domain third-order Volterra filter with the multidimensional overlap-save filtering technique, and on the block least mean square (BLMS) algorithm for updating the filter coefficients. Since the proposed equalizer performs fast convolution by blocks, in decision-directed mode, it provides larger savings in computational complexity compared to the conventional time-domain Volterra equalizers. The feasibility and practicality of the proposed equalizer structure is demonstrated by utilizing it to equalize a nonlinear satellite channel.

This work is supported in part by the Joint Services Electronics Program AFOSR Contract F-49620-92-C-0027.

Gradient-Type Instrumental Variable Method and Its Application to Channel Equalization

Tetsuya Shimamura, Tomoyuki Takada and Jouji Suzuki

Department of Information and Computer Sciences,
Faculty of Engineering, Saitama University
255 Shimo-okubo, Urawa, Saitama 338 Japan

Abstract

In this paper, we present an adaptive IIR equaiizer using system identification techniques. Gradient-type instrumental vaxiable (GIV) algorithm is used for the purpose of channel estimation. The proposed

equalizer has a cascade structure of an ARMA prefilter and an adaptive FIR equalizer. The ARMA prefilter is designed based on the transfer function estimated *by* the GIV algorithm. The ARMA prefiltering leads to reduction in a degree of correlation of the channel output. The LMS algorithm of the cascaded FIR adaptive equalizer provides faster convergence and smaller residual errors than the commonly used adaptive FIR equalizer. Simulation results are shown to confirm the performance of the proposed adaptive IIR equalizer.

Session 2: Visual Signal Processing (I)

The Dynamics of the Anisotropic Diffusion

Fu Liyan Yu Song Yu
Institute of Image Processing and Pattern Recognition Shanghai Jiao Tong University
Shanghai 200030, P.R. China

Abstract

This paper analyzes the anisotropic diffusion from the point of view of numerical analysis and dynamical system. The iteration functions defined by the numerical solution of the anisotropic diffusion define a dynamical system. It is shown that the immediate localization, the piecewise smooth and the edge enhancement property of the anisotropic diffusion results from the underlying characteristics of the iteration functions. This leads to constraints for choosing the diffusion coefficient and a criterion for defecting edge.

Improved Design of the 2-D Quadrature Polar Separable Filter for Texture Processing

Soobong Park, Iksu Choy, Jongan Park
Dept. of Electronic Engineering, Cho-Sun University
Kwang-Ju City, South Korea 501-759

Abstract

An improved Quadrature Polar Separable (QPS) filter, which uses an exponential attenuation function and the shift paramater to the central frequency, is proposed in this paper.

It is easier to control the frequency characteristics of the improved filter compared with Knutsson's. The central frequency change is simpler as it can be shifted algebraically. The filter symmetry is similar to the Knutsson filter.

In order to estimate the orientation and the frequency component of local texture in the frequency domain, a series of experiments have been carried out. The results show that it can be used as an efficient tool in texture processing.

Wavelet Construction Using Beta Functions

Seyed M. Hamidi, Reza R. Adhami, Lori M. Bruce
Electrical and Computer Engineering Department
University of Alabama in Huntsville

1. Introduction

In recent years wavelet theory has been studied extensively for application in many areas of communications and signal processing, such as speech processing, image processing, data compression, and subband coding. Most of these applications involve orthogonal dyadic wavelets. The construction of this type of wavelets is an active area in the study of wavelet theory.

We present a simple method of constructing orthogonal dyadic wavelets based on maximally flat finite impulse response (FIR) filters generated by normalized incomplete Beta functions. We also show that these hailband Beta filters can be appropriately factored to yield conjugate quadrature filters (CQF) for use in exact reconstruction analysis/synthesis subband coders. Finally, we show that Daubechies' orthogonal compactly supported bases, as well as wavelets constructed from Lagrange halfband filters, are precisely the solutions derived from normalized incomplete Beta functions.

A New Pyramid Structure for Progressive Transmission of Color Palettized Images

Young-Woo Cho, Dong-Ho Lee, Young-Mo Kim, Dept. of
Elec. Eng., Kyungpook Nat'l Univ., Taegu, Korea

Abstract

The conventional pyramid structure schemes are not adequate for progressive transmission of color palettized images that have lower spatial continuity. In this paper, a new pyramid structure for progressive transmission of color palettized images is proposed. in the new pyramid structure, the color of a node is represented by a configuration type code and the color which appears the most in the nodes at lower level. The experimental results show that the proposed pyramid structure allows intermediate images to express the original color distribution of color palettized images very well. Furthermore, the proposed pyramid structure gives lower total lossless transmission rate than conventional pyramid structures.

Pyramid-structured Progressive Image Transmission Using the Quantization Error Delivery in Transform Domains

Ji-Honh Kim Woo-Jin Song
Department of Electronic and Electrical Engineering
. Pohang University of Science and echnology
Pohang, Republic of Korea

Abstract

In this paper, we propose a pyramid-structured progressive image transmission scheme in which the quantization error delivery is applied to the transform coefficients. To make it possible to deliver and correct the quantization errors in the transform domain, the proposed method is made up of two stages as follows: 1) the appropriate decomposition of the difference images into transform input subimages and 2) the quantization error delivery to the lower level. Through these procedures, the quantization errors at level k-1 can be completely removed at level k. The simulation results show that the proposed method achieves the lossless reproduction of the original image and increases the coding performance even at the intermediate levels.

Session 3: Visual Signal Processing (II)

Image Coding and Processing in Personal Computer

R.W.L.Cheung, P.C.K.Liu, C.K.Leung

Hong Kong Polytechnic
Hunghom, Hong Kong

Abstract

A simple technique based on PC image coding is developed to simulate a printer output of a picture on the monitor screen in a PC. During the process of simulation of the printed page produced by the selected matrix printer, the image processbig techniques are applied. Most importantly, a palette consists of 20 pre-defined gray levels is read through by a scanner and the resulted file is sent to the PC for subsequent processing and analysis.

ROBUST ESTIMATION OF CAMERA PARAMETERS FROM IMAGE SEQUENCE FOR VIDEO COMPOSITION

Jong-Il Park and Choong Woong Lee

Dep. of Electronics Eng., Seoul National University
56-1, Shinlim-dong, Kwanak-gu, Seoul 151-742, Korea
Tel: +82-2-880-7281 e-mail: pji@tbhp4.snu.ac.kr

ABSTRACT

In video composition systems, viewing points and image planes of multiple images to be composed have to be properly integrated for natural composed images, especially when some of the images undergo camera operations. Camera parameters are needed for the integration.

In this paper, we propose a robust method for the estimation of camera parameters from image sequence. We first establish correspondence of feature points between consecutive image fields. After the establishment, we formulate a nonlinear least-square data fitting problem. When the image sequence contains moving objects, and/or when the correspondence establishment is not successful for some feature points, we get bad observations, outliers. They should be eliminated properly for a good estimation. Thus, we propose an iterative algorithm for rejecting the outliers and fitting the camera parameters alternatively. We attempt to decrease both the number of outliers and the energy of residual errors, given finite observations. Thus, we formulate a budget problem, outlier rejection formula. Then we construct a simple scheme for solving the problem. Finally, we show the validity of the proposed method using computer generated data sets and real image sequences.

Multichannel Iterative Image Restoration

Woon Na and Joon Ki Paik
Department of Electronic Engineering
Chung-Ang University
Seoul 156-756, Korea

Abstract

Various single channel or monochrome image restoration techniques have been proposed, and they have been applied in broad areas of image processing. However multichannel or color image restoration technique is strongly required in current image processing systems, because most of them adopt color imaging structure. Color images that are used in this paper are assumed to have red, green, and blue color components. In this paper, a multichannel image restoration algorithm using an iterative method is proposed, and experimental results obtained by the proposed algorithm are presented.

Shape Invariant Printed Korean Character Recognition Using Holographic Associative Memory Implemented with MMACE and CMF

Jeong Woo Kim*, Cheol Soo Kim*, Yang Hoi Doh**, and Soo Joong Kim*

*Department of Electronic Engineering, Kyungpook National University, Taegu 702-701, Korea
**Department of Electronic engineering, Cheju National University, Cheju 690-756, Korea

Abstract

A concept of the printed Korean character recognition and multiplexing method of MACE filter into only one filter plane using carrier frequencies is presented. The structure of the recognition system is a single-layer neural network employing feedback. The 14 consonant MACE filters and 10 vowel MACE filters are reduced to four consonant MMACE filters and three vowel MMACE filters using the multiplexed technique, respectively. In order to recognize the full printed Korean character, the system must have four consonant recognition loops and three vowel ones, and it is carried out in parallel.

Computer simulations are in good agreement and demonstrate that the proposed technique is a feasible approach for printed Korean character recognition.

Session 4: Neural Networks (I)

Neural Whitened Matched Filter

Uwe Latsch
University of Siegen, Germany
latsch@nue.et-inf.umi-siegen.de

Abstract

Recently, artificial neural networks (ANN's) have been applied to many tasks in the fields of communications and signal processing. Most frequently, additive white gaussian noise (AWGN) is assumed as the only transmission impairment. In this case neural networks lead to almost the same performance as classical approaches do. Decoding of AWGN-corrupted block and convolutional codes by means of neural networks and the equalization of unknown channels are well studied [1], [2], [3]. The idea of this paper is to apply neural networks to communication systems with statistically correlated noise, so-called colored noise. The classical solution to this problem uses a 'whitening filter' which transforms the colored noise into white noise. Then the standard algorithms, such as matched filtering and maximum likelihood decision, may be employed to perform further processing of the signal.

For different system configurations, the bit error rate (BER) of neural based receivers is compared with the BER of classical approaches.

Word Recognition by Neural Networks Using Adaptive Mel-Cepstral Analysis

Tetsuya Hoya, Hitoshi Ihara, Takamitsu Asano and Yoshihisa Ishida
Department of Electronics and Communication, Meiji University
1-1-1, Higashi-Mita, Tama-Ku, Kawasaki, 214 Japan

Abstract

Neural networks have played an important role in many fields of signal processing. One of the fruitful application using neural networks is the development of speech recognition system. In this paper, a practical word recognition system using the technique of adaptive mel-cepstral analysis is presented. The experimental results show the proposed method is suitable for the practical word recognition system.

Image Compression Using 3-Layer Perceptton with Variable Block Size

Jun Geun Jeon, Sung Man Kim, and Kyu Tae Park
Department of Electronic Engineering,
Yonsei University, Seoul Korea

Abstract

In image compression using neural networks, for the purpose of increasing the generalization capability, the entire image is divided into sub-images (typically 8x8 block). In system using fixed block size, the image compression ratio is fixed. This means that bits are over allocated for the plain block as under allocated for the complex block. In the proposed method, an image is divided into sub-images with block size 32x32, 16x16, 8x8 and 4x4. As the block size decreases, the neural network of the lower compression ratio is used. As the property of the sub-image, the neural network used for the compression is adaptively chosen. The computer simulation results show that the PSNR of the reconstructed images is increased by 1-2 dB by the proposed method.

An Efficient Parallel BP for Large Scale Training-sets and its Implementation on the Multicomputer System

Seon-Jong Kim and Heung-Moon Choi
Department of Electronics, Kyungpook National University,
1370 Sangyuk-dong Pukgu Taegu, 702-701, Korea
pdp@ee.kyungpook.ac.kr

Abstract

This paper proposes an efficient parallel BP (backpropagation) for large scale training-sets, and its implementation on the multicomputer system. The proposed model maps the PTS (partial training-set) into the interdependent one by introducing a modified Hopfield neural network onto the output side of BP. From the information of dependent set of patterns, we can obtain the complete energy function for the CTS (complete training-set). Also, due to the complete algorithm of the proposed scheme, the convergence problem usually met in parallel implementation can be solved regardless of the number of processors in the multicomputer. Thus, it is efficient for the larger number of pattems to be learned rapidly into the multilayer neural network. We implement the proposed model on the Transputer system connected in ring, and the results show that we can obtain the shorter convergence time, and thereby the higher speedup than the conventional BP.

Session 5: Neural Networks (II)

Neural network based equalizers for unshielded twisted pair channel

Alexander M.C. YU, Peter C.K. LIU and K.C. LI
Department of Electronic Engineering, Hong Kong Polytechnic,
Hung Hom, Kowloon, Hong Kong

Abstract

This paper is to investigate the problems and possible solutions for the equalizer in unshielded twisted pair (UTP) transmission based on neural network approach. The objectives are 1) to develop a zeroforcing equalizer using the approximation function of multilayer feedforward neural network (MFNN) with back propagation weights adjustment algorithm over the UTP channel; 2) to compare results of BER and sum square error criterion of those using the conventional approach and other neural network based equalizer.

ADAPTIVE RECOGNITION OF HAND-WRITTEN KANJI CHARACTERS USING SELF-ORGANIZED NEURAL NETWORK

Yoshikazu Miyanaga*, IEEE member, Masanori Kondo*, Masato Hayashi** and Koji Tochinai*

*Department of Electronic Engineering, Faculty of Engineering
Hokkaido University, Sapporo-shi 060, Japan
E-mail: miyanaga@hudk.hokudai.ac.jp

**Information Technology Res. Labs, NEC, Kawasaki-shi 216, Japan

ABSTRACT

This report introduces an image recognition system for hand-written Kanji characters. The method is based on a self-organized neural network and a single layer perceptron network. The self-organized neural network is used for adaptive clustering. The single perceptron network is used for recognition. It is well known that a large amount of time is required in the training by multi-layered perceptron when some cluster distributions have complicated structures. However, since only the simplest perceptron is applied in this

proposed system, a quite short time is enough to learn training data, The reason why multi-layered perceptron is not required to recognize data in this system is based on the use of a self-organized network. The self-organized network can change a complicated structure of cluster distribution to a simple structure without the loss of information. Thus, it can be shown that the simple perceptron is enough to recognize even nonlinear characteristic distribution.

Dynamics Learning Network with Structured Recurrent Modules

Yisheng Li, Yoshikazu Miyanaga. and Koji Tochinai
Faculty of Engineering, Hokkaido University
Sapporo 060, Japan

Abstract

In this report, a local connected recurrent neural network, and a. new learning algorithm are proposed.

The network which has the ability to memorize and regenerate complex dynamics is constructed by adaptive oscillating modules. This module consists of two simple neuron nodes with recurrent connections.

In the new learning algorithm, each module can be trained independently with suitable speed for given input data. The network's size is also adaptively determined in learning process.

Finally, some simulation results are demonstrated to verify the effectiveness of the proposed network structnre and the learning algorithm.

An Adaptive Time-Delay Recurrent Neural Network for Learning Spatiotemporal Correlations

Sung Suk Kim*, Sul Gi Yim**, Chong Koo An** and Tai Ilo Lee**

*Dept. of Computer Science, Sangji Junior College
Andong 760-070, South Korea
and
**Dept. of Electronic Engineering, Ulsan University
Ulsan 680-749, South Korea

Abstract

Spatiotemporal pattern recognition and temporal sequence learning are challenging tasks in neural network researches. This paper presents an Adpative Time-Delay Recurrent Neural Network (ATRN) for learning and recognition of spatiotemporal correlations of temporal patterns. The ATRN employs adpative time-delays and feedback connections, which is inspired from neurobiology: in biological systems, neurons have a great variety of different time-delays and there is a great deal of feedback between the cortical strata. In the ATRN, the adaptive time-delays make the ATRN choose the optimal values of time-delays for the temporal location of the important information in the input patterns, and the feedback connections enable the network to encode and integrate temporal information of sequences which have arbitrary interval time and arbitrary length of temporal context. The ATRN described in this paper, ATNN proposed by Lin, and TDNN introduced by Waibel were simulated and applied to chaotic time series preditcion of Mackey-Glass delay-differential equation. The simulation results show that the normalized mean square error (NMSE) of ATRN is 0.0036, while the NMSE values of ATNN and TDNN are 0.0114, 0.0117, respectively, and the best performance is attained by the ATRN. The ATRN will be well applicable for temporally continuous domains, such as speech recognition, motor control, prediction, nonlinear system identification, and trajectory recognition.

Classification of Document Image Blocks Based on Textural Features and BP

Joong-Soo Kim*, Jae-Chang Shim*, Jeong-Hwan Lee*,
and Heung-Moon Choi**
*Dept. of Computer Science, Andong National University, Kyungpook,
760-380 Korea
**Dept. of Electronics, Kyungpook National University, Taegu, 702-701 Korea

Abstract

This paper introduces an efficient document block classification method based on statistical textural features and using BP (backpropagation) algorithm. The image is first processed by Sobel operator and then segmentation is done by smearing the block. Seven features are extracted from SGLDM (spatial grey level dependence matrix) which is the textural features of grey level document image. Block classification is accomplished using a neural network of BP algorithm. This method classifies the image block into nine types in detail. The experiments show excellent results of very few errors and the fine classification of document blocks.

Session 6: Coding (I)

A Fast Coding Algorithm for Iterated Transformation Theory-Based Coding by Multi-Resolution Tree Search

Masayuki Kawamata, Masaki Nagahisa and Tatsuo Higuchi

Graduate School of Information Sciences Tohoku University
Aoba, Aramaki, Aoba-ku, Sendai 980-77, Japan

Email: kawamataAhiguchi.ecei.tohoku.ac.jp

Abstract

Iterated Transformation Theory-Based Coding (ITTBC) suffers from very high computational complexity in encodiug phase. This is due to its exhaustive search. This paper proposes Multiresolution Tree Search ITTBC (MTS-ITTBC) to reduce the encoding complexity. MTS-ITTBC searches ITT-code using a multiresolution searchtree. The computational load of MTS-ITTBC is O(NIogN), while that of full search ITTBC is $O(N^2)$, where N is the number of range blocks or domain blocks in an image. Numerical examples show that the encoding time of MTS-ITTBC can be reduced to 12% of that of the full search ITTBC, while the loss in quality relative to the full search ITTBC is about 0.5 dB in PSNR, which is visually negligible.

Fast Minimum Distortion Codevector Search Algorithm for VQ

Jong Kil Chae and Chan Sik Hwang
Dept. of Electronics, Kyungpook National University
Taegu 702-701, Korea

Abstract

A critical problem associated with the use of vector quantization (VQ) is the computational complexity incurred in the encoding, where the complexity increases exponentially according to codebook size and limits practical application. In this paper, the initialization of fast searching algorithm using prior comprison is proposed to decrease the coding time. This initialization of fast searching algorithm is adopted to reduce the chance of selecting the inappropriate reference codevector which seems to happen in the fast searching algorithm using prior comparison. The proposed algorithm needs only a small

fraction of mathematical operations of exhaustive full search VQ and the reduction of the encoding time is shown in the fast searching algorithm with the proposed initialization, compared with the conventional fast searching algorithm.

A Hybrid Fractal-Based Image Coding

Sung-Hyun Han and Jong-Soo Choi

Department of Electronic Engineering, Chung-Ang University
221 Huksuk-dong Dongjak-ku Seoul, 156-756, KOREA

ABSTRACT

The complexity of the fractal encoding is growing very quickly with the image size. In this paper, we present a novel algorithm which reduces the encoding time of fractal-based image coding. To begin with, the simplest type of image pyramid is constructed. In searching a optimal domain block for each range block, this structure allows to initially search lowresolution version of an image. After decoding in the next higher resolution, distortion measures are computed between the decoded range block and the block of the same position in the same resolution of the original image. For every range block such that the distortion is above the threshold, searching is performed in that resolution again. In the original solution, we consider two-level square partition. Simulation results show good visual quality of decoded images with a lower computational cost.

Improved IFS Coding Scheme Using Fractional Differencing Model

Yong-Goo Kim , Yoonsik Choe
Dept. of Electrical Engineering, Yonsei University, Seoul, Korea

Abstract

In this paper, we propose an improved fractal coding scheme modeled by the Fractional Differencing Model(FDM) to save the computational time for encoding.

Since the Fractional Differencing Model can be considered as the discrete version of Fractional Brownian Motion Process and its fractal parameter can be estimated by simple least mean square scheme, this estimated value which represents the scale property in Fractional Brownian Motion Process is used for encoding and to restrict the searching area for block searching, thus, the processing time can be dramatically reduced.

To demonstrate the capabilities of this model, the existing algorithms are compared with our proposed scheme.

Session 7: Coding (II)

Image Coding Based on Restoration Method for a Very Low Bit Rate Image Transmission

Byung Ha Hwang, Oh Kyun Jeong, Jae Ho Choi and Hoon Sung Kwak
Dept. of Computer Engineering, Chonbuk Nat'l Univ, Korea

Abstract

In this paper we present a block-based predictive coding in the transform domain using the energy correlation and directional correlation properties. The discrete cosine transform (DCT) spreads the texture component into the low frequency and high frequency edge component with relatively high human visual sensitivity. After band split of the directional component in the high frequency region. Our coding method employs a predictive inter-block scheme depending on the stationary region and nonstationar

region of the directional components. Simulation shows the better performance in data compression and processing time compared to the previous predictire coding using the band-split in the spatial domain.

A Fast Huffman Decoder via Pattern Matching.

Seung Bae Choi, Moon Ho Lee
Dept. of Information & Communication Engineering, Chon Buk National Univ.

Abstract

The fast Huffman decoding algorithm has been used in JPEG, MPEG and image data compression standards, etc. And code compression is a key element in high speed digital data transport. A major compression is performed by converting the fixed-length codes to variable-length codes through a entropy coding scheme. Huffman coding combined with run-length coding is shown to be a very efficient coding shceme. To speed up the process of search for a symbol in a Huffman tree and to reduce the memory size we have proposed a tree clustering and a pattern matching algorithm to avoid high sparsity of the tree. The method is shown to be very efficient in memory size and fast searching for the symbol. For an experimental video data with Huffman codes extended up to 16 bits in length, i.e. it is used for the standard JPEG, the result of experiments show that the proposed algorithm has a very high speed and performance. The design of the decoder is carried out using silicon-gate CMOS process.

Selective ECVQ of Images on the Wavelet Transform Domain

Kyeong Ho Yang, Chul-Woo Kim and Choong Woong Lee

Dept. of Electronics Eng., Seoul National University
San 56-1, Shillim-dong Kwanak-gu, Seoul 151-742, Korea
TEL:+82(2)880-8426 FAX:+82(2)880-8222

Abstract

This paper proposes the selective entropy-constrained vector quantization (ECVQ) scheme for wavelet decomposed images. Each of high frequency subimages is segmented into basic blocks, and then the blocks are selectively encoded by ECVQ according to the energy of the samples. We introduce an efficient method to encode the map representing which blocks are encoded, based on inter-band prediction followed by a quadtree encoding. And the coefficients in the lowest frequency subimage are encoded with an adaptive prediction followed by entropyconstrained scalar quantization (ECSQ) of the resultant errors. The proposed scheme optimafly allocates the distortion among all the subimages and introduces a preprocessing of signals which normalizes the input vectors of ECVQ in order to reduce the imagedependency of ECVQ codebooks. Simulation results show that our encoding scheme provides good PSNR(peak-topeak signal-to-noise ratio) performance.

A Video Coder Using Motion Compensation, DCT, and Sequential Vector Quantization

Jun Seok Song, Dong Wook Kang, and Choong Woong Lee
Dept. of Electronics Eng., Seoul National University
Shinlim-dong, Kwanak-gu, Seoul 151-7,42 Korea
e-mail: sjs@tbhp4.snu.ac.kr

Abstract

For still image encoding, there has been recently proposed an efficient transform-domain vector quantization technique which employs the DCT, directional decomposition of the transform coefficients, and sequential vector quantization (DCTDD-SVQ). In this paper, we apply the DCT-DDSVQ for encoding the motion compensated residual signals, of which energy is assumed to be concentrated mostly along the

edges of the original image. The adaptive prediction scheme is applied for motion estimation/compensation with variable block size based on the full-search block matching algorithm. For the purpose of regulating the transmission bit rate, the stopping criterion of the SVQ is adjusted according to the status of the transmission buffer. The simulation results show that the proposed scheme provides high quality of moving pictures both objectively and subjectively in case of encoding of rodeo signals for 4Mb/s channel.

Image Compression Using Projective Vector Quantization with Variable Block Size Segmentation

Kyeong-Hoon Jung and Choong Woong Lee

Dept. of Electronics Engineering, Seoul National University
Shinlim-dong, Kwanak-gu, Seoul 151-742, KOREA
Fax. :+82-2-885-4459, e-mail:timothy@tbhp4.snu.ac.kr

Abstract

A new image compression algorithm that uses quadtree decomposition and projective vector quantization is described in this paper. It segments an image into rectangular blocks of variable size so that each block has homogeneous information content, and estimates projection data that produce the best approximation for the original block, and finally performs vector quantization on the projection data. The advantages of the proposed algorithm are that 1) it can easily deal with larger blocks than the conventional VQ scheme and 2) it can allocate the bits according to the details of the image. The proposed algorithm shows improved performance both in subjective and objective quality, especially at low bit rate. Therefore, it is expected to be applicable to low bit rate image coding.

Session 8: Coding (III)

Entropy Coded TCVQ Using O-spline Based Wavelet Transform

Choong Il Ryu, In Gyeom Kim and Kyu Tae Park
Yonsei University, Seoul, Korea

Abstract

In this paper, we incorporate TCVQ into the encoding of the wavelet transformed(WT) image followed by a variable length coding (VLC) or an entropy coding (EC). We used an orthogonal spline wavelet transform. We metioned a modification to the set partitioning algorithm presented by Wang et al. The PSNR result of the 256x256 LENA image excluded from the training images is 36.38 dB at 1.078 bpp.

Block Motion Vector Recovery by Neighborhood Matching

Kook-Yeol Yoo student member and Jae-Kyoon Kim member
Visual Communications Lab., KAIST, Taejon, Korea

Abstract

The digital video services in the B-ISDN network require the transmission of a large amount of information within a highly restricted bandwidth. Transmitting coded bitstream over networks, channel errors are inevitable due to transmission constraints. Currently most of video coders adopt the MCDCT techniques. In such coders, the loss of motion information result in fatal quality degradation, and furthermore, errors are propagated into the next reconstruction images. A novel recovery method of lost motion information, based on motion estimation at the decoder side, is proposed in this paper. Luminanee intensities enclosing block with motion information loss are used in the estimation. The emperical results show that the proposed algorithm achieves better performance in PSNR and subjective quality than conventional recovery methods of motion information, such as average method and median filter.

Optimization of Filter Banks for Subband Coding Using Orthogonal and Biorthogonal Wavelet Transforms

Yu HE* Ryuji KOHNO* Hideki IMAI**

*Division of Electrical and Computer Engineering, Yokohama National University, Yokohama, Japan
**Institute of Industrial Science, University of Tokyo, Tokyo, Japan

Abstract

This paper presents new algorithms for computing subband coding gain of two-dimensional multi-layer sub-band coding systems which satisfy perfect reconstruction conditions. Design optimization is performed for the orthogonal and biorthogonal wavelet filters used in the two-dimensional multi-layer filter bank. An input signal-driven filter bank is also proposed in the case of biorthogonal wavelet filters with the optimal filter coefficients depending on the characteristics of the input signal. The improvement of the designed optimal filter bank is demonstrated by experiments using SIDBA standard static images.

Incorporating Nonlinear Contractive Functions into the Fractal Coding

H. Lin and A.N. Venetsanopoulos
Department of Electrical and Computer Engineering, University of Toronto
Toronto, Ont., Canada, M5S IA4

Abstract
In this paper, we introduce nonlinear contractive functions into the fmctal coding to solve the decoding divergent problem. Statistical analyses of the fractal codes are given. The results show that the method produces a visually better decoded image while the decoding is about twice faster.

High Speed Iterated Transformation Theory Based Coding using Wavelet Transformation

Masatomo SAEKI*, Yoshinori TAKEUCHI** and Hiroaki KUNIEDA*

*Tokyo Institute of Technology
Tokyo, Japan
**Tokyo University of Agriculture & Technology
Koganei, Japan

Abstract

This paper proposes a new high speed algorithm for Iterated Transformation Theory Based Coding(ITTBC) [1]. ITTBC is one of the efficient coding algorithm which uses the fractal feature of image. However, the ITTBC requires the vast amount of computation of its pattern matching/searching process. We propose an algorithm which utilizes wavelet transformation to reduce the time required for searching process. The advantage of the proposed algorithm is demonstrated by computer simulations.

Session 9: Fuzzy-logic Based Processing

A Study on Tracking Center Frequency of Nonstationary Signal using Master-Slave Filter Bank and Fuzzy Model

Jae-whan Lim* Hyung-sig Youn** Jae-seung Choi* Sang-hui Park*
*Dept. of Electrical Engineering in Yonsei University
**Korea Aerospace Research Institute

Abstract

In this paper the method using master-slave filter bank with series-parallel structure and fuzzy model is proposed to track the center frequency. of signal whose center frequency is varying but spectral configuration and frequency bandwidth is rarely varying with time. When conventional filter bank is used to track the center frequency of signal, the frequency resolution of filter must be increased to enhance tracking accuracy. But it results in the decrease of time resolution. The proposed filter bank that two band notch filters and fuzzy model are connected to conventional filter bank has enhanced tracking accuracy without time resolution decreased. The frequency-tracking experiments of sinusoidal signal are performed using conventional filter bank and the proposed filter bank. The results show that the latter has enhanced tracking performance.

Fuzzy Reasoning of Local Threshold Values for Extracting Features

Sung-Mok Cho, Sung-Mo Koo, and Jin-Ho Cho
Dept. of Electronics, Kyungpook Nat'l. Univ., 702-701, Taegu, Korea

Abstract

An algorithm which is to select the thresholds for extracting the significant features without human intervention as perceived by human beings is described. It is based on the fuzzy theory. To do this, we define two measures as input fuzzy variables: contrast and local brightness and threshold as output variable. Combining nine if-then rules generates a threshold value, which is assigned by a membership function to detect edge for each pixel. The input membership functions reflect local information and the output membership function represents global property of processed image. Therefore, this approach not only preserve broad range of edges but also is independent of processed images. To support the validity of the proposed method, we will present several experimental results of real images as well as synthetic image. It is also expected that the proposed algorithm provides a satisfactory improvement in the performance in the edge detection process by using conventional edge operators as the contrast measure of the proposed technique.

A study on Fuzzy Vector Quantizer Mapping in Speech Synthesis for Speaker Adaptation

Lee Jin-Yi , You Jae-Taek, Lee Kwang-Hyung
Department of Eleetromcs Engineering, Soong Sil University
1-1 Sangdo 5-dong Dongjack-gu, Seoul, 156-743, Korea
E-mail:ljy@dsp.soongsil.ac.kr

ABSTBACT

In this paper, we propose a speaker-adaptive speech synthesis method using a mapped codebook designed by fuzzy mapping. Competitive learning neural networks are used to design both input speaker's and reference speaker's codebook. We used a fuzzy VQ mapping to design the mappod codebook. The fuzzy VQ mapping replaces a codevector preserving its fuzzy membership function. The codevector correspondence histogram is obtained from accumulating the vector correspondence along the DTW optimal

path. Using each histngram as a weighting function, the mapped codebook is defined as a linear combination of the reference speaker's vectors.

Speech synthesis is performed as follows: input speaker's speech is fuzzy vector quantized by the mapped codebook, and then FCM arithmatic is used to synthesize the speech adapted to input speaker. The speaker adaption experiments are carried out using speech of a male in his thirties, speech of a male in his twenties, and speech of a female in her twenties, as input speaker's speech, and speech of a other female in her twenties, as reference speaker's speech. Speech used in experiments are sentences / anyoung hasira nika/ and /good morning/. As a results of experiments, we have listened a synthesized speech adapted to input speaker.

Poster Session (I)

Adaptive Beamforming Technique for Arrays with Element Failure

Ki M. Kim, Il W. Cha, and Dae H. Youn
Department of Electronic Eng., Yonsei Univ.
134 Shinchon-dong, Sudaemoon-ku, Seoul, KOREA (120-749)

ABSTRACT

In the real world, one operates with an array which has faulty elements. Element failure produces an elevated sidelobe level and fails to reject the interference signals in an adaptive beamformer. In this paper, we present the adaptive beamforming algorithm for array with element failure. The proposed method minimizes the array output power subject to a set of linear constraints which maintain the frequency response in the look direction and force the weights of the inoperative elements to zero. The simutafion results illustrate that the presented method outperforms the conventional method in terms of the capability of interferences suppression.

Adaptive Eigenspace Estimation Algorithm Using Gram-Schmidt Structure

Ha-Jong Sung, Young-Cheol Park, Il-Whan Cha
Depamnent of Electronic Eng., Yonsei Univ.
134 Shinchon-dong Sudaemoon-ku, Seoul, Korea (120-749)

Abstract

An eigenspace estimation method employing the IPM (Inverse Power Iteration) and GS(Gram-Schmidt) structure is proposed in this paper. In the proposed method, the matrix inversion required to implement the IPM is avoided by using the Cholesky decomposition characteristics, and the Cholesky factor is obtained from the GS filter which orthogonalizes the input signal. The eigenspace estimation method develop in this paper makes use of the systolic characteristics of the GS structure to produce a structure for simultaneous realization of the Cholesky decomposion of the inverse matrix and the IPM. In addition, a modified IPM using an adaptive forward-backward GS filter is proposed, and improved performance over that using the adaptive forward GS filter is shown.

An Efficient Adaptive Eigenspace-based Algorithm for Direction Estimation and Tracking

Lena Chang
Department of Merchant Marine, National Taiwan Ocean University, Taiwan, R.O.C.

ABSTRACT

This paper presents an efficient eigencomponents updated algorithm via an iterative technique which can alleviate the computation complexity in the eigendecomposition. The proposed algorithm requires only

O(M) multiplications in each iteration and can converge faster than the LMS-type algorithms. Moreover, the proposed algorithm is applied to estimate and track the angles of arrival of the radiation sources. Simulation results confirm the performance improvement of the proposed eigencomponents updated algorithm.

Tracking Center Frequency of Seaway Signal Using A New Master-Slave Filter Bank

Hyong-sig Youn** Jae-whan Lim* Jae-seung Choi* Sang-hui Park*
* *Korea Aerospace Research Institute
*Dept. of Electrical Engineering in Yonsei University

Abstract

A new filter bank is proposed in order to track center frequency of non stationary signals. The filter bank proposed here has a master and a slave filter bank. The two banks are connected in series-parallel. The master filter bank which is made up of conventional filter bank detects the center frequency of the signal roughly. And the performance for tracking the center frequency is greatly improved by slave filter bank which is based on energy-difference estimator without increasing filter order. The frequency tracking experiments of sinusoidal signals and the elimination experiments of seaway disturbances are performed using conventional filter bank and proposed filter bank. And the performance of frequency-tracking of seaway signal using proposed method is better than that of conventional filter banks.

Instantaneous Frequency Estimation with Modified Trench's Method

Shiunn-Jang Chern and Lin-Chung Tsai
National Sun Yat-Sen University, Department of Electrical Engineering
Kaohsiung, 80424, Taiwan, R.O.C.
Tel:886-7-5316171 ext.4017,
Fax:886-7-5615137
E-mail: chern@ee.nsysu.edu.tw

Abstract

In this paper, a new algorithm for instantaneous frequency estimation (IFE) is developed. The modified Trench's method along with the Bauer-Fike theorem is proposed for solving the principal eigenvalues of the Hermitian Toeplitz autocorrelation matrix for IFE. Here for IFE, first, the modified Trench's method is employed for solving the principal eigenvalues of the autocorrelation matrix formed by the initial block of data with length N. When a new data is received, the Bauer-Fike theorem is then applied to update the new eigenvalues based on the previous obtained eigenvalues. The advantage of this approach is that each individual eigenvalue can be updated independently. Therefore, it can be implemented by the parallel structure such that the computation time can be reduced. From the simulation results, we found that the presented method can resolve two chirped signal with close center frequencies.

The Adaptive Equalizer with Tentative Decision of the Viterbi Decoder

Han Jong KIM, Jong Moon CHUNG, Young Jo LEE, Chang Eon KANG

Dept. of Electronic Engineering, Yonsei University
134 Sudaernoon Gu, Seoul 120 749, Korea

Abstract

The common receiver structure for modems employing trellis coded modulation uses an adaptive equalizer to prevent intersymbol interference resulting from channel distortion, and a Viterbi decoder to decode the equalized signal. In this paper, a new equalizer structure which combines the decision

feedback equalizer with the Viterbi decoder is proposed to improve the performance. To provide accurate data to the feedback filter, this equalizer uses the recently stored data in the Viterbi decoder. Namely, the data in the feedback filter are not shifted like the conventional decision feedback equalizer. In this case error propagation does not occur and adaptation becomes more accurate resulting an improvement in performance.

A Real-Time Isolated Word Recognition System

Yun-Seok Cho and Hwang-Soo Lee

Department of Information and Communication Engineering
Korea Advanced Institute of Science and Technology, Seoul Campus
P.O.BOX 201, Cheongryang, Seoul, KOREA 130-650

Abstract

This paper describes the implementation of a realtime isolated word recognition system based on the hidden Markov model (HMM). The speech recognition system consists of a host computer (PC), a DSP board, and a prototype Viterbi scoring board. The DSP board extracts feature vectors of speech signal. The Viterbi scoring board, which has been implemented using three field-programmable gate array (FPGA) chips, performs the Viterbi algorithm for HMM-based speech recognition. At the clock rate of 10 MHz, the system can update about 100,000 states within a single frame of 10 ms.

NEW METHOD FOR MODELING COXPLEX SYSTEMS BY CONVERSION OF THE LAPLACE TRANSFORM TO THE Z-TRANSFORM

EUGENIA RASSKAZOVA PENN

Russian Industrial Training Movement, Inc.
191 King's Highway, Milford, CT 06460, USA

LEO SCHELOVANOV

St. Petersburg University of Telecommunications
61 Moyka nab, St. Petersburg, Russia, 191065

ABSTRACT

This paper introduces a method to simplify the modeling of complex systems. A matrix is developed to simplify the evaluation of coefficients of the recurrent procedure to model complex analog systems. The authors demonstrate that diverse devices are accurately modeled, and develop a validation process for recurrent models for linear and non-linear networks. The technique requires much less computation than traditional methods utilizing differential equations. The technique also allows the determination of the probability characteristics of random signals generated by analog systems of any complexity.

Adaptive Equalization for Reduction of Nonlinearity in High-Density Recording Channels

Joo-Sin Son, Won-Gi Jeon*, Yong-Soo Cho*, Yong~Hoon Lim**, and Dae-Hee Youn***
**Dept. of Electronic Eng., ChungAng Univ., Seoul, Korea*
***Dept. of Electronic Eng., Yonsei Univ., Seoul, Korea*

Abstract

In this paper, a structure for a nonlinear adaptive equalizer is discussed to reduce nonlinearity in digital high-density recording systems. We propose a nonlinear adaptive decision feedback equalizer which can reduce the nonlinear intersymbol interference increasing with highdensity recording systems, and compare its performance with the RAM-DFE which is designed to remove nonlinear intersymbol interference existing in postcursor part. By observing the output SNR of each equalizer applied to recording channels with three different densities, we confirm that the nonlinear adaptive decision feedback equalizer performs the best in the general case where nonlinear intersymbol interference exists in both precursor and postcursor parts.

Poster Session (II)

New Recurrence Formulae for Convergence Analysis of Adaptive Filters Using the Sign-Sign Algorithm

Shin 'ichi Koike

NEC Corporation
7-1, Shiba 5-chome, Minato-ku
Tokyo, 108-01 Japan
Telephone:+81 3 3798 9525
Facsimile:+81 3 3798 9532

ABSTRACT

This paper derives a new set of recurrence formulae, which can describe the expected convergence process of adaptive filters using the stochastic gradient Sign-Sign Algorithm with Gaussian reference and Gaussian additive noise signals. In the analysis, the correlation between the signum of the error signal and that of the reference signal is calculated, based on the fact that they are jointly Gaussian distributed given the values of the "errors at the taps", or the tap weight errors. Further taking the expectation with respect to the "errors at the taps" completes the recurrence formulae for updating the, mean and the covarianee of the "errors at the taps". Theoretical expression for the Mean Squared Error (MSE) after convergence is also derived from the recurrence formulae.

The analysis is then applied to an echo canceller, where simulation and theoretical calculation are performed and compared for examples with different parameters. The results show an excellent agreement between experiment and theory, and prove the usefulness of the proposed formulae.

Variable Length Adaptive Algorithm with Optimum Convergence Factor Based on Structural Subband Decomposition

In- Hyoung Boob and Cheo I H. Kang
Dept. of Electronic comm. Engineering
Kwangoon University
447-1 Wolgye-Dong, Nowon-Gu
Seoul 139- 701, Korea

Abstract

In this paper we describe the new adaptive algorithm. In this algorithm the order of adaptive fiter and convergence factor are changed automatically based on the structural subband decomposition of input signal during the course of adaptation process. By this means, the proposed algorithm results in both fast convergence and good steady-state perromance in spite of very large eigenvalue spread of input signal and time-varying system environments. The proposed algorithm is evaluated by computer simulation for system identification in various circumstances and the results show that the proposed algorithm is very efficient compared with the conventional algorithims (such as normalized LMS, RLS).

Realization of Two-Dimensional Recursire and Non-Recursive Adaptive Filters Using Parallel Separable Structures

Masayuki Kawarnata, Tomotaka Yoshie and Tatsuo Higuchi
Graduate School of Information Sciences Tohoku University
Aoba, Aramaki, Aoba-ku, Sendal 980-77, Japan
E-mail: kawamata@higuchi.ecei.tohoku.ac.jp

Abstract

This paper proposes parallel separable (PS) two-dimensional (2-D) adaptive filters. Their adaptation algorithms are formulated and their computational complexities are estimated. They are applied to 2-D identification for comparison of the proposed adaptive filter and a conventional 2-D adaptive filter. A PS 2-D recursive adaptive filter and a *PS* 2-D non-recursive adaptive filter are realized in the form of the 2-D adaptive line enhancer, and they are applied for restoration of noisy images.

Evolutionary Digital Filters
—adaptive digital filters based on evolutionary strategies of organisms—

Masayuki Kawamata, Masahide Abe and Tatsuo Higuchi

Graduate School of Information Sciences
Tohoku University, Sendai 980-77 Japan
E-mail: kawamata@higuchi.ecei.tohoku.ac.jp

Abstract

Tiffs paper proposes evolutionary digital filters as a new concept for adaptive digital filters. Evolutionary digital filters are adaptive filters which are controlled by adaptive algorithm based on the evolutionary strategies of organisms. An evolutionary digital filter consists of many linear but time-variant inner digital filters which correspond to organisms. The output of the evolutionary digital filter is the output of an inner filter of which fitness is maximum at every iteration. The adaptive algorithm controls and changes the coefficients of inner filters using asexual reproduction method or sexual reproduction method. The adaptive algorithm is inherently parallel because of the natural independence of evolutionary operations. An example for interference canceling shows the effectiveness of evolutionary digital filters.

Eigenspace-Based Adaptive Multiple Linear Constraint Beamformering

Jung-Lang Yu and Chien-Chung Yeh
Department of Electrical Engineering, National Taiwan University
Taipei, Taiwan, R.O.C.

Abstract

The generalized eigenspace-based beamformer (GEIB) is presented here which utilizes the eigenstructure of the correlation matrix to enhance the performance of the multiple linear constrained beamformer (MLCB). The weight vector of the GEIB is found by projecting the MLCB weight vector onto a vector subspace constructed from the eigenstructure of the correlation matrix. The GEIB and the MLCB have the same responses to the desired signal and the interferers. But the GEIB weight vector has a smaller norm and, therefore, generates a lower output noise power. By properly constructing the vector subspace, each of the original linear constraints can be preserved or not preserved by the GEIB as desired. The cost of preserving a linear constraint is to get more output noise power. Computer simulations are also presented to demonstrate the merits of the GEIB.

A Consideration on Block LMS-Newton Algorithm and Its Performance

Masanori KIMOTO*, Toshihiro FURUKAWA**

*Graduate School, Fukuoka Institute of Technology,
Department of Management Engineering, Fukuoka Institute of Technology,
3-30-1 Wajiro-Higashi Higashiku, Fukuoka-shi, 811-02, Japan,
Tel: +81-92-606-3131 ext.2390
Fax: +81-92-607-6691, +81-92-606-8923

Abstract

Adaptive algorithm especially play the important role in adaptive signal processing, and various techniques has been proposed until up to now. Among them, LMS-Newton algorithm is well known. LMS-Newton algorithm can have rapid convergence speed no matter how input signal are correlated. But it has too many computational requirements to be implemented with hardware structure. This paper presents a Block LMS-Newton algorithm, it is derived by introducing the block adaptive signal processing to LMS-Ncwton algorithm. Compared with LMSNewton algorithm, the proposed algorithm is expected to reduce the computational requirements and fast convergence speed.

A Study on the Speech Recognition using Pitch Synchronous LPC Cepstrum-VQ and HMM

Nam-Kon Ko, Hyung-Seuk Kim, Jin-Yi Lee, Kwang-Hyung Lee
Department of Electronic Engineering
Soong Sil Uinversity, Seoul, Korea
email:mergury@logos.soongsil.ac.kr

ABSTRACT

In this paper we propose a phoneme recognition model of the Korean speech by using the pitch synchronous LPC cepstrum-VQ and HMM. The LPC cepstrum coefficient have been obtained by synchronizing the analysis frame per a pitch. The pitch synchronization method reduces the analysis time and the effect of the pitch pulses.

The LPC cepstrum of the Input speech signal is vector quantized by using the reference codebook of the LPC cepstrum coefficient. This codebook has already been designed with the FCM clustering algorithm. The LPC cepstrom codevector's address in codebook is applied to the HMM algorithm, and then recognized phoneme by phoneme.

As the speech signal vary slowly with time, the HMM techniclue is, in general, ellective in its modeling in time. The performance evaluation of the proposed method has been done for both inside training data and outside training data.

As the simulation results. it can be shown that the proposed recognition method has improved about 3[%] in recognition ratio than the pitch asynchronous method.

Voice Transmission Using Adaptive Filter Bank

SungHoon HONG and SouGuil ANN
Department of Electronics Engineering
Seoul National University
San 56-1, Shillim-dong, Kwanak-gu
Seoul 151-742, Korea

ABSTRACT

In this paper, we propose the new speech analysis technique for applications to low bit rate speech coding. We employ the adaptive algorithm which is modified version of Kalman filter. This algorithm split error covanance matrix of the Kalman algorithm into signal subspace and noise subspace using reduced-rank SVD(singular value decomposition), and then update vocal-tract parameters by applying adaptive algorithm which assumes residual characteristics as normal mixture density. This proposed algorithm gives more accurate time-varying vioce parameters. And, we propose adaptive filter bank approach as a new excitation modeling method which can represent various source characteristics. In this model, excitation signals are assumed to be the output of any filter driven by impulse like signals, and excitation signals are assumed to be pulses that are spread in the noisy environment. We split estimated voce source waveforms into sub-banded versions using filter bank concept, where each filter bank coefficients can be obtained by proceeding ML(maximum likelihood) estimation: 1) Filter coefficients are estimated to make error signals to impulse-like patterns. 2) Estimated impulse responses in each sections are modeled by codebook index which represent expected subband source patterns. and 3) Weights and time-shift parameters of codebook source pattern are estimated. This adaptive filter bank shows that the filtered signal can be seen that the peaks are remarkably impulse-like and the resolution is significantly increased.

A Reduced Lattice Filter Structure for Acoustic Echo Cancellation

Jae Ha Yoo, Sung Ho Cho **, and Dae Hee Youn**
*Department of Electronic Eng., Yonsei Univ. Shinchon-dong, Sudaemoon-ku, Seoul, Korea
**Department of Electronic Eng., Hanyang Univ. Daehak-dong, Ansan, Kyunggi-Do, Korea

Abstract

The least mean square(LMS) algorithm is now widely used in variety of applications due to its simplicity. When this algorithm is employed in a transversal filter structure, however, the convergence speed of the algorithm is very sensitive to the eigenvalue spread ratio of the input autocorrelation matrix. Therefore, if the input signal is a highly correlated signal like a speech signal, the algorithm suffers from slow convergence. The joint lattice filter which is the lattice-based joint process estimator using the LMS algorithm, shows fast convergence. When the impulse response is very long like an acoustic echo canceller, however, real time implementation is very difficult because of the excessive computations. If the input signal is well modelled by the P-th order autoregmssive(AR) process, the reflection coefficients after the P-th stage are closely zero. So, the joint lattice filter can be implemented with transversal filter structure after the P-th stage.

In this paper, we propose this new joint lattice filter that we call a reduced lattice filter. Using the reduced lattice filter, we are able to achieve the performance as good as that of the joint lattice filter, while maintaining the complexity as low as that of the transversal filter. Simulation results are presented to test the performance of the reduced lattice filter.

A Robust Adaptive Algorithm and Its Performance Analysis with Contaminated-Gaussian Noise

Senng Chan Bang and Souguil Ann

Dept. of Electronics Engineering, Seoul National University, Seoul 151-742, Korea

ABSTRACT

We introduce a steepest descent linear adaptive algoritlun, the proportion-sign algoritlun (PSA), using the notion of M-estimator to make the least mean square (LMS) algorithm robust to impulsive interference occuring in the desired response. Its performance analysis is presented when the signals are from zero-mean jointly stationary Gaussian processes and the additive noise to the desired response is from a zero-mean stationary contminated-Gaussian (CG) process which is usually used to represent impulsive interference. Since a special case of the PSA becomes the LMS algorithm, the analysis of the LMS is also obtained as a by-product. By adding a minimal amount of computational complexity, the PSA improves to some degree the convergence speed over the LMS algorithm without overly degrading the steady-state error performance for Gaussian noise. In addition, it has the properties of robustness to impulsive interference occuring in the desired response while the LMS algoritlun is vulnerable to it. Computer simulations are used to demonstrate the validity olf our analysis and the robustness of the PSA compared with the LMS algorithm.

Poster Session (III)

Optimal Design of Multi-Dimensional Recursive Digital Filters Using a Simple Genetic Algorithm

Masayuki Kawamata, Jun Irnakubo, and Tatsuo Higuchi

Department of System Information Sciences
Graduate School of Information Sciences
Tohoku University
Aoba, Aramaki, Aoba-ku, Sendai 980-77
Japan
email: kawamata@higuchi.ecei.tohoku.ac.jp

Abstract

This paper proposes an optimal design method of multi-dimensional recursive digital filters (MDDFs) based on a simple genetic algorithm (SGA) for frequency-domain specifications. Design problem is formulated both in general class and in separable-denominator (SD) class. Parameters of the transfer function of a M-DDF are encoded to bit strings in order to apply a SGA to the design. The proposed method has flexibility for change of specifications of M-DDFs and for error functions used in optimization. The solution obtained by the proposed method can be also a good initial point for traditional optimization methods, for example, the DFP method. The effectiveness of the proposed method is demonstrated by design examples.

An Intelligent Automatic Frequency Control Method Using Discrete Wavelet Transform

Shinsuke Hara and Norihiko Morinaga

Department of Communication Engineering, Faculty of Engineering, Osaka University,
2-1, Yamada-oka, Suita-shi, Osaka 565 Japan
Tel/Fax:+81-6-879-7738/+81-6-876-1923
E-Mail:hara@comm.eng.osaka-u.ac.jp

Abstract

The automatic frequency control method using the Discrete Wavelet Transform (DWT AFC) has an intelligent characteristic of *"observation and control according to frequency error"*, which makes it possible to achieve both fast initial acquisition and accurate tracking. Furthermore, it can make the irreducible frequency error arbitrarily small.

This paper discusses the frequency acquisition performance of the DWT AFC using three different mother wavelets; the modified Haar, modified [1,3,3,1] and Gabor functions.

Adaptive IIR Filtering with Noise Suppression

MIN-SOO PARK and WOO-JIN SONG

Department of Electrical Engineering, Pohang University of Science & Technology,
Pohang, Kyungbuk, Republic of Korea

Abstract

The adaptive digital filtering is widely used in the communication and control systems for obtaining desired system characteristics. Although IIR filter structures are more efficient than FIR filter structures, the conventional filter coefficient update algorithms for adaptive IIR fillers have serious deficiencies.\ The stochasfic-gradient based conventional adaptive IIR algorithms such as the PLR algorithm or the LMSEE algorithm may produce coefficient estimates that are unstable or biased, and recursive Gauss-Newton methods such as the simplified RPE algorithm or the RLSEE algorithm may exhibit excessive computational complexity in addition. More recent algorithms like the SHARF algorithm and lhe BRLE algorithm try to solve the problems of the estimation bias and the system unstabilily but instead created the new problem of determining the proper value of user-selected parameters which greatly affect the algorithm performance.

This paper presents a new algorithm that can overcome these problems by using the combined form of the regressor vector and the estimation error. The new algorithm, called NSLE algorithm, gradually suppresses the noise components in the LMSEE coefficient update equation as much as possible since the noise signal is the main cause of the estimation inaccuracy. The computer simulation results also demonstrate that the performance of lhe proposed algorithm is better than those of the conventional algoritms.

Feasibility Condition on Band Selection for a Derived Design of Non-uniform Cosine Modulated Filter Banks

Jeong-Jin Lee
Elec. Telecom. Research Inst.
Daejon, 305400, Korea

Byeong Gi Lee
Seoul National University
Seoul, 151-742, Korea

Abstract

Feasibility conditions on band selection is established for a non-uniform CMF bank design which is derived in association with a uniform CMF bank. In this derived non-uniform CMF bank design, each constituent non-uniform filter is formed by merging the relevant uniform filters in the associated

urnform CMF bank. The feasibility of the derived design is investigated in terms of the band selectivity of the non-uniform filters, the alias cancellation among the non-uniform analysis and synthesis filters, and the distortion function of the overall filter bank. The investigations are done in a rigorous manner via three proved properties, which are finally combined into a main theorem. The theorem provides the feasibility condition that the proposed design method is applicable if, and only if, the upper band edge frequency of each non-uniform filter is an integral multiple of the bandwidth of the corresponding band.

Iterative Thresholded Lowpass Filter for Blocking Effect Removal

Sangho Kim, Haemook Jung, Byunguk Lee, Kyuwhan Chang, Shiryong Yu
Team-2, Video Research Center, Daewoo Elec. Co. Ltd.
C.P.O. Box 8003, Chung-gu, Seoul, Korea
Tel: +82-2-751-1666, Fax: +82-2-753-6168
Email: skim@taurus.dwe.co.kr(128.134.85.31)

Abstract

In this paper, we propose a postprocessing melhod that neatly removes blocking effect but retains visually important image details and edges. The iterative thresholded lowpass filter is basically a low pass filter whose ouput depends on three variable elements. i.e. iteration number, threshold and passband width. The threshold sets the limit to which the ouput of the proposed filter can differ from original input regardless of the iteration number. With this regulalion of threshold, the iterative thresholded lowpass filler can retain most of the image details while smoothing the blocking effect. The other two variable elements, i.e. iteration number and passband width, can determine the convergence speed of the proposed filler.

Signal Recovery from Wavelet Threshold Crossings

Makoto Nakashizuka, Hisakazu Kikuchi, Ikuo Ishii and Hideo Makino
Faculty of Engineering, Niigata University, Japan
Tel: + 81-25-262-7431
Fax: +81-25-263-3174
E-mail: mako@info.eng.niigata-u.ac.jp

abstract

This paper presents signal reconstruction from wavelet threshold-crossings by using convex projection. We define the wavelet threshold crossings as the irregular sampling of a discrete dyadic wavelet transform. The recontruction is done by the projection between a convex which characterized by the threhold-crossings and a linear space of all possible wavelet transform. The wavelet threshold-crossings convey the same information as the quantized wavelet transform. The reconstruction precision of the thresholdcrossing is superior to the inverse wavelet transform of the quantized wavelet transform. Image reconstruction from the wavelet threshold crossings is demonstrated.

Polynomial Matrix Decomposition for Use in LBR Two-Pair Realizations

Seung Joo Maeng, Byeong Gi Lee
Department of Electronics Engineering
Seoul National University, Seoul, Korea

Il-Taek Lim
GoldStar Central Rearch Lab.
Woomyeon-Dong, Seocho-Gu,
Seoul, Korea

Abstract

In this paper we present a polynomial matrix decomposition procedure such that, given a polynomial matrix $V(z)$ which is paraconjugate hermitinn matrix of normal rank r and positive semidefinite on the unit circle of z-plane, we can determine a polynomial matrix $M(z)$ which satisfies the relation $V(z) = M(z)M(z)$.

Also, we discuss how to apply this polynomial matrix decomposition in realizations of MIMO LBR two-pairs.

On the Convergence Behavior of a Filtered-X LMS Adaptive Active Noise Canceler

Kang Seung Lee*, Jae Chon Lee**, and Dae Hee Youn*
*Dept. of Electronic Engineering, Yonaei University, Seoul, Korea
**Dept. of Systems Engineering, Ajou Univeraity, Seoul, Korea

Abstract

Application of the Filtered-X LMS adaptive filter to active noise cancellation requires to estimate the transfer characteristics between the output and the error signal of the adaptive canceler. In this paper, we derive an adaptive cancellation algorithm and analyze its convergence behavior when the acoustic noise is assumed to consist of multiple sinusolds. The results of the convergence analysis of the Filtered-X LMS algorithm indicate that the effects of parameter estimation inaccuracy on the convergence behavior of the algorithm are characterized by two distinct components; Phase estimation error and estimated magnitude. In particular, the convergence of the FHtered-X LMS algorithm is shown to be strongly affected by the accuracy of the phase response estimate. Simulation results of the algorithm are presented which support the theoretical convergence analysis.

An Optimal Implementation of Digital Filters on Multiple Pipelined Processors +

Chun Pyo Hong and JongJung Woo*'*

**Dept. of Computer Science, Taegu University Jinryang-Myun, Kyttngsan-Gun, Kyungpook, KOREA*
***Dept. of Computer Science, Sungshin Women's University 249-1, Dongsun-Dong, Sungbuk-Gu, Seoul, KOREA*

Abstract

This paper presents a set of techniques to automatically find rate optimal or near rate optimal implementations of digital filters on multiple pipelined processors, in which digital filters are represented by recursive shift-invariant flow graphs. In such case, the problem to be addressed is the scheduling of multiple instruction streams which controls all of the pipeline stages. The goal of an automatic scheduler in this context is to rearrange the order of instructions such that they are executed with minimum iteration period between successive iteration of defining flow graphs. Since the node execution times in defining flow graphs are deterministic, this research addresses compile time scheduling.

Design of Arbitrarily Rotated Frequency Response in Two-Dimensional Digital Filters

Jiemei ZHU and Nozomu HAMADA

Department of Electrical Engineering, Faculty of Science and Technology, Keio University, Yokohama, Japan

Abstract

The aim of this paper is to present a design method of the zero phase two-dimensional (2-D) finite impulse response (FIR) filters having rotated frequency response of a given simple prototype FIR filter response. The first step of our approach is to design a prototype response by the filter with the impulse response whose domain is restricted within a circular region. This restriction prevents rotated impulse array

to going out from the rectangular region. Then, the rotation of prototype impulse response, which is given no more than on the rectangular lattice grid, causes the same rotation of prototype frequency response.

The second step is the interpolation of the impulse response on the grids in the sense of the reconstruction of 2-D bandlimited discrete signals. Previous other methods for the rotation and the interpolation are compared with the proposed method. The obtained results for fan-type filter approximation by the proposed method show its validity.

Design Method of 2-D Lattice Digital Filters Using the Genetic Algorithm

Masahiko KISHIDA and Nozomu HAMADA

Department of Electrical Engineering, Faculty of Science and Technology,
Keio University, Yokohama 223, Japan
Tel:+81-045-563-1141 ext.3360
Fax: +81-045-563-2773
E-mail:kishida@tkhm.elec.keio.ac.jp

Abstract

This paper proposes design method of 2-D lattice digital filters using the Genetic Algorithm(GA). By using the GA, they are derived directly from the desired frequency response, and 2-D stable nonseparable-denominator transfer functions can be obtained.

In our method, ARMA transfer functions are designed by two algorithms. One is to optimize AR and MA parameters simultaneously, and the other is to search an optimal AR parameters with fixed MA parameters, and then search an optimal MA parameters separately. Both are done by the GA, and the simulation shows that they give the simillar results.

Design of FIR Linear Phase Filters with Discrete Coefficients Using Hopfield Neural Networks

Xi ZHANG, Member, IEEE, and Hiroshi IWAKURA, Member, IEEE
Department of Communications and Systems
University of Electro Conmmnications
1-5-1 Chofugaoka, Chofu-shi, Tokyo 182 JAPAN
Tel: 0081 424 83 2161 EX3131
Fax: 0081 424 84 6890
E-mail: zhangxi@qr.cas.uec.ac.jp

Abstract

A novel method is presented for designing FIR linear phase filters with discrete coefficients using Hopfield neural networks. The proposed procedure is based on the minimization of the energy function of the Hopfield neural network, and can produce a good solution to the design of FIR linear phase filters with discrete coefficients. A design example is presented to demonstrate the effectiveness of the proposed method.

Poster Session (IV)

Partial Orthogonalization based Adaptive Interference Suppression for Direct-Sequence CDMA Systems

(Edward) Kwang Bok Lee

Land Mobile Products Sector Research Lab.,Motorola Inc.,USA

Abstract:

A conventional matched filter receiver in Direct Sequence Code Division Multiple Access (DS/CDMA) systems suffers from the near-far problem, and relies on a power control meclmism and good partial cross-correlation properties of the spreading sequence waveforms to minimize multiple-access interference. In this paper, rapid converging adaptive interference suppression algorithms for DS/CDMA systems are presented. These adaptive algorithms, based on a partial orthogonalization preprocessing, do not assume a priori knowledge on the spreading sequences and relative signal power levels of interfering signals, nor do they require the identification of signal and noise components after preprocessing. The convergent rate of the proposed algorithms depends on the level of an orthogonalization preprocessing. The proposed algorithms with a partial orthogonalization converge much faster than the algorithms without a partial orthogonalization.

An Efficient Speech Recognition Algorithm for LSP-based Speech Transmission Parameters

Hong-Kook Kim*, Seung-Ho Choi**, and Hwang-Soo Lee*

*Core Technology Research Center
Samsung Advanced Institute of Technology
P.O.BOX 111, Suwon, Korea 440-600

**Dept. of Information and Communication Eng., KAIST
P.O.BOX 201, Cheongryang, Seoul, Korea 130-650

Abstract

In this paper, we propose an efficient speech recognizer at the receiver side of a digital telephone or mobile communication network using quantized LSP frequencies from a low bit-rate speech coder as spectral information of speech. Using the proposed speech recognizer, we can recognize speech signal from digital telephone handsets at the switching office instead of recognizing speech in each handset. Thus, we can offer various services in a convenient and economical way. To this, end, we first perform LSP-based speech recognition experiments and then we obtain the performance of the proposed speech rccognition system.

A VLSI Architecture for the Alternative Subsampling-Based Block Matching Algorithm

Hae-Kwam Jung, Chun-Pyo Hong,** Jin-Soo Choi,* and Yeong-Ho Ha'*

**Dept. of Electronic Engineering, Kyungpook National University, Taegu, Korea*
***Dept. of Computer Science, Taegu University, Kyungpook, Korea*

Abstract

A VLSI architecture of the block matching algorithm based on the alternative subsampling method for the motion estimation is proposed. The alternative subsampling method reduces the computational complexity by subsampling the number of pixels used to estimate motion vectors, whereas conventional methods limit the number of locations searched. Simulation results show that the performance of this

method is very close to full search algorithm. For subsampling factor of N, this approach can achieve approximately N/2 times of calculation with additional small overhead associated with address generator and temporary buffer. In addition, this architecture has about a half silicon area compared to Yang's architecture.

Speech Enhancement in the Subband Domain

Wen-Rong Wu, Po-Cheng Chen, and *Chih-Song Wang*
Dept. of Comrnunication Engineering
National Chiao Tung Univeristy, Hsin-Chu, Ta/wan *ROC*

Abstract

The main goal of speech enhancement is to filter speech contaminated by additive white or colored noise. Conventional apporach is to model speech as a high order time varying AR process and apply the Kalman filter. However, the conventional approach has two problems. First, the high order Kalman filter, which involves intensive matrix computation, is difficult to be implemented. Second, the estimate of AR parameters from noisy speech is not trivial. In this paper, we propose new approaches to solve these problems. We model subband signals of speech as AR processes and then apply Kalman filter to filter noise. Since the lower order AR models can be used for the subband signals, we can apply a lower order Kalman filter to avoid the intensive computation. To solve the second problem, we propose new adaptive algorithms for white or colored noise respectively. The new algorithm is also compared with norrealized LMS algorithm. Simulation results show that the speech enhancenent. in subband domain has much better performance than that in fullband domain.

A Multi-DSP Architecture: Application to Detection and Generation of MF/DTMF Signaling for Digital Voice Server

Agus Trihandoyo, Keli Li and Kun-Mean Hou
Heudiasyc CNRS URA 817 Universite de Technologie de Compiegne
B.P. 649, 60200 Compiegne Cedex, FRANCE
E-mail: atrihand@hds.univ-compiegne.fr

Abstract

This paper presents a multi DSP architecture for telecom applications, based on TMS320C31 DSPs. It is the kernel of an interactive digital voice server operating on the European standard time division multiplexing (TDM) line of 2.048 Mbps.

A global description of the methodology for analyzing the whole system and the approach to a specific requirement of telecom signaling that leads us to a multi DSP solution will be presented. The various stages in the development of a prototype are delineated. Some of the issues involved in incorporating MF/DTMF decoding and generation in our architecture are discussed by taking advantage of the current advances in computer hardware technology. Frequency detection relies generally on the spectral representation of the signal. Fast Fourier Transform (FFT) is an efficient method to do such a transformation. However, we will show how this technique is not quite adapted to solve our problem. We put forth the use of Goertzel filter which is very efficient in terms of machine-cycle. And we make a proposal for a modified algorithm, which integrates an automatic gain control (AGC) and an adaptive start point detection.

The system speed, the accuracy, the capability of handling multi-channel, and the flexibility of the system are the main advantages of the implemented system.

Error Tolerant Video Transmission using FEC and Multi-level Coding

Martyn J. Riley, Iain E.G. Richardson
Robert Gordon University, Aberdeen, Scotland

Abstract

Emerging communication networks based on the Asynchronous Transfer Mode (ATM)[1] are designed to support the transport of real-time compressed video traffic. At times of congestion, buffer overflow can occur, and cells are lost. Compressed video traffic is extremely sensitive to data loss. Recently Forward Error Correction (FEC) has been suggested[2] as a means of recovering ATM cell loss during the transmission of compressed video. We investigate video compressed using two different operating modes of the JPEG International Standard[3]. We consider the basedine and the Progressive Spectral Selection (PSS) mode of the JPEG standard. The PSS JPEG algorithm allows video to be compressed into a number of separate spatial frequency bands or scans, each scan consisting of a different sequence of coded Discrete Cosine Transform (DCT) coefficients. We have shown[4] that each scan exhibits a different tolerance to cell loss. Therefore, we add varying amounts of FEC parity information to each of the scans depending on their sensitivity to cell loss. We compare the PSS JPEG algorithm, with varying quantities of FEC on each scan, to the base-line JPEG algorithm, with a constant amount of FEC added across all spatial frequencies. We perform an experiment to transmit a number of coded video sequences resulting from real traffic streams through a simple simulated ATM network which introduces a high level of cell loss. We reanimate the received traffic streams and perform subjective quality evaluation.

Variable-Length Coding of Quantized DCT Coefficients using Adaptive Huffman Codebook Selection

Byeungwoo Jeon, Juha Park, and Jechang Jeong

Signal Processing Lab.
Samsung Electronics Co.
416 Maetandong, Paldalgu
Suwon, Korea

Tel: +82-331-200-3188
Fax: +82-331-200-3195
Email: bj@rnd.sec.samsung.co.kr

ABSTRACT

In this paper, we propose a Huffman coding method for image sequence compression which can adaptively choose the best codeword table among the given set of tables so that maximum possible bit saving can be achieved. Experimental results show the bit saving dependence upon different codeword tables and verify effectiveness of the proposed method of adaptive codeword table selection over the current MPEG 2 coding method.

Signal Restoration of Broad Band Speech Using Adaptive Digital Filter

Hiroshi Yasukawa
NTT Optical Network Systems Laboratories
Nippon Telegraph and Telephone Corporation, Yokosuka, 238-03 Japan

Abstract

This paper focuses on restoring band limited speech signals to obtain highly quality speech by reproducing the upper band speech power. Two methods that offer simple and easy implementation are introduced. One method is automatic level control. The other introduces frequency domain adaptive digital filtering to broaden band limited signals into wide band signals, namely we find the optimum inverse transfer function

to convert band limited signals back into the original broad band signals. Implementation of the system and its performance are discussed.

Concurrent Edge Detection and Prominent Edge Enhancement Using a Cooperative Neural Network

Joon-Hyung Jeon and Heung-Moon Choi
Department of Electronics, Kyungpook National University,
1370 Sangyuk-dong Bukgu Teagu, 702-701, Korea
pdp@ee.kyungpook.ac.kr

Abstract

In this paper, we proposed an efficient algorithm in which edge detection, prominent edge enhancement and binarization, and orientation detection can be done concurrently by using cooperative neural network. We applied 4 cooperative neural masks to each pixel, and we also give a threshold bound in the neural network to find out the prominent edge accurately. In contrast to the conventional edge detection algorithm, the proposed algorithm does not require the postproceasing such as thresholding for the binarization or feature extraction for image recognition. The experiments are conducted on various complex 256 X 256 images, which have small grey level differences in 256-grey level, and the results show that prominent edge enhancement and binanzation , and orientation detection can be done concurrently without postprocessing. And the proposed algorithm have good characteristics for the important feature detection and noise removal, and the results of the experiment show that it is efficient for neural network based target recognition system under study which highly requires those characteristics.