(1)

Speaker Recognition Using Artificial Neural Networks Based on Vowel Phonemes

E.F.M.F.Badran , H. Selim

Abstract:

Speaker recognition systems attempt to recognize a speaker by his/her voice through measurements of the specifically individual characteristics arising in the speakers voice. Among transformations of LPC parameters the adaptive component weighted (ACW) cepstrum has been shown to be less susceptible to channel effects than others. Text-independent and text-dependent speaker recognition systems suitable for verification and identification (open set and closed set) are presented, The system is based on locating the vowel phonemes of the test utterance. A preprocessing is applied to the speech signal. The centers of the vowel phonemes are located and identified as speech events using a three-step vowel phoneme locating process. The steps of the locating process are: (1) average magnitude function calculation; (2) vowel phoneme candidates location; and (3) ripple rejection. For each vowel phoneme (20 ms) 10 ACW cepstrum coefficients are calculated and are used as inputs to neural networks and the outputs are accumulated and averaged. The system hardware requirements are a microphone and a round card. The system software written in C++ language for windows. The system was tested with a population of 10 speakers (7 male and 3 female), and the statistics were taken (95.67% for text-dependent verification, 93% for text-dependent identification, 92.2% for text-independent verification and 88.95% for text-independent identification). There tests were done with utterances of one word having one vowel phoneme (20 msec used for recognizing the speaker). A vowel phoneme recognition application is also presented. A limited vocabulary recognition system is developed using vowel phoneme in the limited vocabulary. The feature vectors calculation is the same as in the speaker recognition system the only difference is in the neural network training and size (97.5% of word recognition)

Keywords:

Speaker Recognition Using Artificial Neural Networks Based on Vowel Phonemes

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(2)
-A Novel Robustness Watermarking Scheme for Copyright Proving

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Abstract:

In this paper, a new wavelet-based copyright proving scheme is proposed. The original image is not required for watermark verification. Moreover, the proposed scheme does not introduce any visual quality degradation into the original image; however, a feature is extracted from the original image and mixed with the watermark to generate a secret key. This secret key is registered for verification. The feature of the image is extracted from the approximation subband in the t-level of the wavelet decomposition. A series of experiments are conducted to prove the robustness property. The results show that this proposed scheme has high robustness under heavy image signal processing attacks and geometric transformation attacks. Moreover, the experimental results show that our scheme outperforms recent works in most cases.

Keywords:

Copyright protection , discrete wavelet transform , image watermarking

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(3)

Utilizing Repeated Adjacencies of Vector Quantization Indices in Image Compression

M.F. Abdel-Latif, T.K. Abdel-Hamid, M.M. Doss and H. Selim

Abstract:

Image compression using vector quantization (VQ) results in highly correlated indices. The correlation between these indices is used to reduce the bits needed to represent them. This is done by many index compression algorithms such as the Hu and Chang, search order coding (SOC), and switching tree coding (STC). A new algorithm for VQ index compression is introduced and it utilizes the local statistics of each image and the repeating pattern of its adjacent indices. The proposed algorithm improves the index compression performance of the basic VQ, with a relatively slight increase of complexity.

Keywords:

VQ Index Compression, Lossless Coding, Image Compression, Vector Quantization.

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(4)

A New Image Compression Technique Based on Combining Feedforward Neural Networks and Discrete Cosine Transform

P.E. William, T.K. Abdel-Hamid, M.M. Doss and H. Selim

Abstract:

In this paper, we propose an algorithm for the application of one-hidden layer Feedforward Neural Network (OHL-FNN) to image compression. The algorithm combines OHL-FNN with Discrete Cosine Transform (DCT), here, the neural network learning algorithm performs the compression in a spectrum domain of DCT coefficients, i.e., the OHL-FNN approximates only the DCT coefficients representing the high detailed part of the image, Network parameters are stored in order to recover the image. Results, compared with baseline JPEG algorithm, demonstrate that the new algorithm dramatically increase compression for a given quality; conversely it increases image quality for a given compression ratio.

Keywords:

Image compression, discrete cosine transform, Feedforward Neural Network (FNN)

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(5)

Utilizing Index Usage Map for VQ Index Compression

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Abstract:

In practical vector quantization (VQ) of images, the used pixel block dimensions are kept small to reduce the cost of computations. This in turn results in highly correlated blocks and the corresponding VQ indices will inherit this high correlation. The compression of the basic VQ can be increased through utilising this high correlation of indices by inserting a lossless index compression stage after the VQ stage. A new index compression algorithm is introduced. In this algorithm the 2 dimensional index map is divided into nonoverlapping square blocks. Index usage in each of these blocks is employed to remap (renumber) the reduced number of actually used indices in this block, thus resulting in reduced bit rate expressed in bits/pixel The proposed algorithm reduces the average bit rate by a value depending on the codebook size, namely a reduction of about 32% for codebook size of 64, and down to about 23% for codebook size of 1024. Moreover this algorithm lends itself to being cascaded by other index compression algorithms resulting in increased compression.

Keywords:

VQ Index Compression, Lossless Coding, Image Compression, Vector Quantization.

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(6)

Flagged-modulo coding for robust and low complexity coding

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Abstract:

Two novel speech coders , called flag modulo pulse code modulation GMPCM and enhansened flag modulo pulse code modulation EGMPCM , are presented. The coders are designed for medium and high bit rate applications. These coders use the modulo PCM principle. A flag bit is used to indicate changing in speech segments. The coders have small modulo amplitude and hence better SNR. Dynamic quantization and bit allocation are also used in these coders in effect these coders function as dynamic quantizers. EGMPCM coders use a new technique to reduce the modulo amplitude by modifying the input speech files. Both coders do not use predictors neither in coding nor in decoding stages. The coders depend only upon the actual difference between conp.

Keywords:

Flagged-modulo coding for robust and low complexity coding

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(7)

Adaptive local data and membership based KL divergence incorporating C-means algorithm for fuzzy image segmentation

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Abstract:

In this paper, a fuzzy clustering technique for image segmentation is developed by incorporating a hybrid of local spatial membership and data information into the conventional hard C-means (HCM) algorithm. This incorporation is a threefold procedure. (1) The membership function of a pixel is spatially smoothed in the pixel vicinity. (2) The Kullback-Leibler (KL) divergence between the pixel membership function and the smoothed one is added to the HCM objective function for fuzzification. (3) The resulting fuzzified HCM is regularized by adding a weighted HCM-like function where the original pixel data are replaced by locally smoothed ones. Thereby the weight is proportional to the residual of the locally smoothed membership. This residual decreases when many pixels existing in the pixel vicinity belong to the same cluster. Thus, the weighted distance decreases, allowing the pixel membership to follow the dominant membership in the pixel vicinity. The simulation results of segmenting synthetic, medical and media images have shown that the proposed algorithm provides better performance compared to several previously developed algorithms. For example, in a synthetic image, with added white Gaussian noise having a variance of 0.3, the proposed algorithm provides accuracy, sensitivity and specificity of 92%, 84% and 94.7% respectively, while the algorithm with the closest results provides 81.9% of accuracy, 62.2% of sensitivity and 86.8% of specificity. In addition, the proposed algorithm shows the capability to identify the number of clusters.

Keywords:

Fuzzy C-means Medical image segmentation Local membership information Weighted distances Kullback–Leibler (KL) divergence

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(8)
-Micro Phasor Measurement Unit Phasor estimation by off nominal frequency

George Daoud Hany Selim Mohamed M. AbdelRaheem

Abstract:

Due to the high sampling rate of the power signals by Micro Phasor Measurement Units (PMUs) tight restrictions on computing time of different algorithms are imposed. Especially when dealing with Phasor estimation by off-nominal frequency. Sampling the power signal with a frequency which is a multiple of the instantaneous power frequency is used to fix this problem. To perform this, Gabor Transform is applied to get the update of the instantaneous frequency each cycle. In this paper we propose many time complexity enhancements to allow the Phasor estimation during the limited intersample interval, and we validate our techniques using computer simulation and real implementation on microcontroller. Using the Arm Cortex STM32F407 microcontroller with the floating point core at the maximum clock speed of 168MHz, we managed to perform all needed algorithms in about 43% of the intersample interval, and leaving the rest for data storage, display of relevant information and communication with the Data Center.

Keywords:

Smart Networks; Micro Phasor measurement units; off-Nominal frequency; Phasor estimation

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