Record 1 of 50

Title: Effects of Imperfect Secondary Path Modeling on Adaptive Active Noise Control Systems

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Source: IEEE TRANSACTIONS ON CONTROL SYSTEMS TECHNOLOGY **Volume:** 20 **Issue:** 5 **Pages:** 1252-1262 **DOI:** 10.1109/TCST.2011.2161762 **Published:** SEP 2012

Abstract: Implementation of adaptive active noise control (ANC) systems requires an estimate model of the secondary path to be uploaded onto digital control hardware. In practice, this model is not necessarily perfect; however, to avoid mathematical difficulties, theoretical analysis of these systems is usually conducted for a perfect secondary path model. This paper conducts a stochastic analysis on performance of Filtered-x LMS (FxLMS)-based ANC systems when the actual secondary path and its model are not identical. This analysis results in a number of mathematical expressions, describing effects of a general secondary path model on stability, steady-state performance and convergence speed of FxLMS-based ANC systems. As a surprising result, it is found that intentional misadjustment of secondary path models can enhance performance of ANC systems in practice. Theoretical results are found to be in a good agreement with the results obtained from numerical analysis. Also, experimental results confirm the validity and accuracy of the theoretical results.

Accession Number: WOS:000305981300010

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1063-6536

Record 2 of 50

Title: Hardware-Software Codesign of Automatic Speech Recognition System for Embedded Real-Time Applications

Author(s): Cheng, O (Cheng, Octavian); Abdulla, W (Abdulla, Waleed); Salcic, Z (Salcic, Zoran)

Source: IEEE TRANSACTIONS ON INDUSTRIAL ELECTRONICS **Volume:** 58 **Issue:** 3 **Pages:** 850-859 **DOI:** 10.1109/TIE.2009.2022520 **Published:** MAR 2011

Abstract: We present a hardware-software coprocessing speech recognizer for real-time embedded applications. The system consists of a standard microprocessor and a hardware accelerator for Gaussian mixture model (GMM) emission probability calculation implemented on a field-programmable gate array. The GMM accelerator is optimized for timing performance by exploiting data parallelism. In order to avoid large memory requirement, the accelerator adopts a double buffering scheme for accessing the acoustic parameters with no assumption made on the access pattern of these parameters. Experiments on widely used benchmark data show that the real-time factor of the proposed system is 0.62, which is about three times faster than the pure software-based baseline system, while the word accuracy rate is preserved at 93.33%. As a part of the recognizer, a new adaptive beam-pruning algorithm is also proposed and implemented, which further reduces the average real-time factor to 0.54 with the word accuracy rate of 93.16%. The proposed speech recognizer is suitable for integration in various types of voice (speech)-controlled applications.

Accession Number: WOS:000287323800015

Author Identifiers:

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Salcic, Zoran		0000-0001-7714-9848
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 0278-0046

Record 3 of 50

Title: Ambient intelligence platform using multi-agent system and mobile ubiquitous hardware

Author(s): Wang, KIK (Wang, Kevin I-K.); Abdulla, WH (Abdulla, Waleed H.); Salcic, Z (Salcic, Zoran)

Source: PERVASIVE AND MOBILE COMPUTING **Volume:** 5 **Issue:** 5 **Pages:** 558-573 **DOI:** 10.1016/j.pmcj.2009.06.003 **Published:** OCT 2009

Abstract: In this paper, a novel ambient intelligence (AmI) platform is proposed to facilitate fast integration of different control algorithms, device networks and user interfaces. This platform defines the overall hardware/software architecture and communication standards. It consists of four layers, namely the ubiquitous environment, middleware, multi-agent system and application layer. The multi-agent system is implemented using Java Agent DEvelopment (JADE) framework and allows users to incorporate multiple control algorithms as agents for managing different tasks. The Universal Plug and Play (UPnP) device discovery protocol is used as a middleware, which isolates the multi-agent system and physical ubiquitous environment while providing a standard communication channel between the two. An XML content language has been designed to provide standard communication between various user interfaces and the multi-agent system. A mobile ubiquitous setup box is designed to allow fast construction of ubiquitous environments in any physical space. The real time performance analysis shows the potential of the proposed AmI platform to be used in real-life AmI applications. A case study has also been carried out to demonstrate the possibility of integrating multiple control algorithms in the multi-agent system and achieving a significant improvement on the overall offline learning performance. (C) 2009 Elsevier B.V. All rights reserved.

Accession Number: WOS:000208396400014

Author Identifiers:

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Wang, Kevin		0000-0001-8450-2558
Salcic, Zoran		0000-0001-7714-9848

ISSN: 1574-1192

eISSN: 1873-1589

Record 4 of 50

Title: On the stability of adaptation process in active noise control systems

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Source: JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA **Volume:** 129 **Issue:** 1 **Pages:** 173-184 **DOI:** 10.1121/1.3514375 **Published:** JAN 2011

Abstract: The stability analysis of the adaptation process, performed by the filtered-x least mean square algorithm on weights of active noise controllers, has not been fully investigated. The main contribution of this paper is conducting a theoretical stability analysis for this process without utilizing commonly used simplifying assumptions regarding the secondary electro-acoustic channel. The core of this analysis is based on the root locus theory. The general rules for constructing the root locus plot of the adaptation process are derived by obtaining root locus parameters, including start points, end points, asymptote lines, and breakaway points. The conducted analysis leads to the derivation of a general upper-bound for the adaptation step-size beyond which the mean weight vector of the active noise controller becomes unstable. Also, this analysis yields the optimum step-size for which the adaptive active noise controller has its fastest dynamic performance. The proposed upper-bound and optimum values apply to general secondary electro-acoustic channels, unlike the commonly used ones which apply to only pure delay channels. The results are found to agree very well with those obtained from numerical analyses and computer simulation experiments. (C) 2011 Acoustical Society of America. [DOI: 10.1121/1.3514375]

Accession Number: WOS:000286944600026

PubMed ID: 21303000

Author Identifiers:

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ISSN: 0001-4966

Record 5 of 50

Title: Filtered weight FxLMS adaptation algorithm: Analysis, design and implementation

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Source: INTERNATIONAL JOURNAL OF ADAPTIVE CONTROL AND SIGNAL PROCESSING **Volume:** 25 **Issue:** 11 **Pages:** 1023-1037 **DOI:** 10.1002/acs.1257 **Published:** NOV 2011

Abstract: In the Filtered-x Least-Mean-Square (FxLMS)-based Active Noise Control (ANC), the convergence speed of the adaptation process has a direct relationship to a scalar parameter, called the step size. There is a theoretical upper-bound for the step size beyond which the system becomes unstable. However, the step size is usually set to a number smaller than its upper-bound in practice. This is because for relatively large step sizes, the adaptation process becomes very sensitive to any non-stationary change in acoustic noise. Owing to this trade-off, real-time implementation of high-performance ANC systems becomes challenging. To overcome this problem, this paper develops a novel ANC algorithm in which a recursive filter compensates for influences of the step size increase on the system performance. It is shown that this filter can efficiently increase the step size upper-bound; consequently, the performance of the system is improved. This improvement is demonstrated using computer simulation. Also, experimental results shows the preference of the proposed algorithm to the traditional FxLMS-based ANC algorithm in practice. Copyright (C) 2011 John Wiley & Sons, Ltd.

Accession Number: WOS:000297016400005

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 0890-6327

Record 6 of 50

Title: Stochastic modelling and analysis of filtered-x least-mean-square adaptation algorithm

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Source: IET SIGNAL PROCESSING **Volume:** 7 **Issue:** 6 **Special Issue:** SI **Pages:** 486-496 **DOI:** 10.1049/iet-spr.2012.0090 **Published:** AUG 2013

Abstract: This study represents a stochastic model for the adaptation process performed on adaptive control systems by the filtered-x least-mean-square (FxLMS) algorithm. The main distinction of this model is that it is derived without using conventional simplifying assumptions regarding the physical plant to be controlled. This model is then used to derive a set of closed-form mathematical expressions for formulating steady-state performance, stability condition and learning rate of the FxLMS adaptation process. These expressions are the most general expressions, which have been proposed so far. It is shown that some previously derived expressions can be obtained from the proposed expressions as special and simplified cases. In addition to computer simulations, different experiments with a real-time control setup confirm the validity of the theoretical findings.

Accession Number: WOS:000326459200005

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1751-9675

eISSN: 1751-9683

Record 7 of 50

Title: Root locus analysis and design of the adaptation process in active noise control

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Source: JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA **Volume:** 132 **Issue:** 4 **Pages:** 2313-2324 **DOI:** 10.1121/1.4746018 **Part:** 1 **Published:** OCT 2012

Abstract: This paper applies root locus theory to develop a graphical tool for the analysis and design of adaptive active noise control systems. It is shown that the poles of the adaptation process performed in these systems move on typical trajectories in the z-plane as the adaptation step-size varies. Based on this finding, the dominant root of the adaptation process and its trajectory can be determined. The first contribution of this paper is formulating parameters of the adaptation process root locus. The next contribution is introducing a mechanism for modifying the trajectory of the dominant root in the root locus. This mechanism creates a single open loop zero in the original root locus. It is shown that appropriate localization of this zero can cause the dominant root of the locus to be pushed toward the origin, and thereby the adaptation process becomes faster. The validity of the theoretical findings is confirmed in an experimental setup which is implemented using real-time multi-threading and multi-core processing techniques. (C) 2012 Acoustical Society of America. [http://dx.doi.org.ezlibproxy1.ntu.edu.sg/10.1121/1.4746018]

Accession Number: WOS:000309650600037

PubMed ID: 23039428

Author Identifiers:

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Abdulla, Waleed		0000-0002-1812-4285

ISSN: 0001-4966

Record 8 of 50

Title: Neonatal EEG signal characteristics using time frequency analysis

Author(s): Abdulla, W (Abdulla, Waleed); Wong, L (Wong, Lisa)

Source: PHYSICA A-STATISTICAL MECHANICS AND ITS APPLICATIONS **Volume:** 390 **Issue:** 6 **Pages:** 1096-1110 **DOI:** 10.1016/j.physa.2010.11.013 **Published:** MAR 15 2011

Abstract: Time-frequency analysis is a way to represent the energy contents of a signal in the joint time-frequency domain. It provides a good visual way to separate the frequency contents of a multi-component signal, and display the changes of these components with respect to time. This paper outlines investigative work on neonatal EEG signals using time-frequency analysis. The Cohen's class distributions are discussed, and kernel optimisation for the Cohen's class distributions is outlined. Segments of EEG with different background continuity states are analysed using a Cohen's class distribution, and their characteristics are discussed. Through this paper, interesting information that offers insight towards the EEG signal can be visualized from the time frequency analysis. (C) 2010 Elsevier B.V. All rights reserved.

Accession Number: WOS:000287271300011

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 0378-4371

Record 9 of 50

Title: A non-invertible cancellable fingerprint construct based on compact minutiae patterns

Author(s): Krivokuca, V (Krivokuca, Vedrana); Abdulla, WH (Abdulla, Waleed H.); Swain, A (Swain, Akshya)

Source: INTERNATIONAL JOURNAL OF BIOMETRICS **Volume:** 6 **Issue:** 2 **Pages:** 125-142 **DOI:** 10.1504/IJBM.2014.060977 **Published:** 2014

Abstract: This paper proposes a new, privacy-preserving fingerprint construct, which consists of a single pattern created from a small subset of minutiae from the corresponding minutiae template. The sparsity of the resulting feature vector ensures that it cannot be used to reconstruct the underlying fingerprint image, while simultaneously allowing for the cancellability of compromised patterns. The proposed construct is, therefore, inherently a non-invertible cancellable fingerprint template protection scheme. Experimental results on the recognition accuracy of patterns consisting of three, four, and five minutiae were very encouraging: an FRR of 0.4% or less was achieved for all three pattern sizes, while the FAR ranged from 2.1% for 5-minutia patterns to 6.5% for 3-minutia patterns. It took about 0.1 seconds to verify a single person, regardless of the pattern size. These results suggest that the proposed fingerprint construct is suitable for deployment in fingerprint-based civilian authentication applications.

Accession Number: WOS:000214648600003

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ISSN: 1755-8301

eISSN: 1755-831X

Record 10 of 50

Title: Voice biometric feature using Gammatone filterbank and ICA

Author(s): Abdulla, WH (Abdulla, Waleed H.); Zhang, YS (Zhang, Yushi)

Source: INTERNATIONAL JOURNAL OF BIOMETRICS **Volume:** 2 **Issue:** 4 **Pages:** 330-349 **DOI:** 10.1504/IJBM.2010.035448 **Published:** 2010

Abstract: Voice biometric feature extraction is the core task in developing any speaker identification system. This paper proposes a robust feature extraction technique for the purpose of speaker identification. The technique is based on processing monaural speech signal using human auditory system based Gammatone Filterbank (GTF) and Independent Component Analysis (ICA). The measures used to assess the robustness to additive noises and speaker identification performance are defined and discussed. The kkn the proposed feature is evaluated in real environments under varying noisy conditions. The proposed feature is benchmarked against the commonly used features such as: MFCC, PLCC, and PLP, and it outperforms them in different noisy environments.

Accession Number: WOS:000214638100002

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ISSN: 1755-8301
eISSN: 1755-831X

Record 11 of 50

Title: A Statistical Inverse Problem Approach to Online Secondary Path Modeling in Active Noise Control

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Kaipio, JP (Kaipio, Jari P.); Nasiri, A (Nasiri, Alireza); Sharifzadeh, H (Sharifzadeh, Hamid); Abdulla, WH (Abdulla, Waleed H.)

Source: IEEE-ACM TRANSACTIONS ON AUDIO SPEECH AND LANGUAGE PROCESSING **Volume:** 24 **Issue:** 1 **Pages:** 54-64 **DOI:** 10.1109/TASLP.2015.2495249 **Published:** JAN 2016

Abstract: This paper recasts the problem of online secondary path modeling in the form of a statistical inverse problem. A statistical and, in particular, a Bayesian approach towards secondary path modeling is developed and the computational issues that emerge from this approach are discussed. All signals and parameters are modeled as random variables and the degree of information concerning them is coded in their probability density functions. An abstract solution is formulated in the form of a probability density function for the secondary path model. For extracting point estimates, common statistical estimation methods are investigated. It is shown that maximum likelihood estimation is not stable; however, Bayesian method of maximum a posteriori gives a reliable solution. An adaptive algorithm is then developed to compute this solution in a computationally efficient manner. This algorithm has three advantages, compared to the traditional secondary path modeling algorithms. First, it does not cause any interference with the main active noise control algorithm. Second, it does not require any additive-noise to be injected into the secondary path. Third, it does not require any off-line initiation. The convergence of the proposed algorithm is analyzed theoretically. The validity of the theoretical results is investigated by using computer simulation. Finally successful integration of the proposed algorithm into a real-time ANC system is reported.

Accession Number: WOS:000364857900005

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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Kaipio, Jari		0000-0002-7392-5149

ISSN: 2329-9290

Record 12 of 50

Title: Speech Recognition System for Embedded Real-time Applications

Author(s): Cheng, O (Cheng, Octavian); Abdulla, W (Abdulla, Waleed); Salcic, Z (Salcic, Zoran)

Book Group Author(s): IEEE

Source: 2009 IEEE INTERNATIONAL SYMPOSIUM ON SIGNAL PROCESSING AND INFORMATION TECHNOLOGY (ISSPIT 2009) **Book Series:** IEEE International Symposium on Signal Processing and Information Technology **Pages:** 118-122 **DOI:** 10.1109/ISSPIT.2009.5407487 **Published:** 2009

Abstract: In this paper a hardware/software co-processing speech recognizer for embedded applications is proposed. The system mainly consists of a softcore processor and a hardware accelerator The accelerator is responsible for GMM emission probability calculation, which is the major computational bottleneck. To alleviate the memory bandwidth issue, the hardware accelerator uses double-buffering, which allows parallel operation of data retrieval and GMM computation. The proposed accelerator is synthesized on an Altera Stratix II FPGA device together with a Nios II softcore processor running at 100MHz. The proposed system is compared with a pure software-based system using test utterances from the Resource Management (RM1) corpus. For a speech utterance length of 2.49s, the decoding time reduces from 6.64s to 2.48s. The real-time factor improves from 2.67 to 1.00. The word accuracy rate of the proposed system on the RM corpus is 93.42%.

Accession Number: WOS:000290365400022

Conference Title: 9th IEEE International Symposium on Signal Processing and Information Technology

Conference Date: DEC 14-17, 2009

Conference Location: Ajman, U ARAB EMIRATES

Conference Sponsors: IEEE Comp Soc

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Author	Web of Science ResearcherID	ORCID Number
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ISSN: 2162-7843

ISBN: 978-1-4244-5949-0

Record 13 of 50

Title: A new time-frequency binary mask estimation method based on convex optimization of speech power

Author(s): Bao, F (Bao, Feng); Abdulla, WH (Abdulla, Waleed H.)

Source: SPEECH COMMUNICATION **Volume:** 97 **Pages:** 51-65 **DOI:** 10.1016/j.specom.2018.01.002 **Published:** MAR 2018

Abstract: In conventional computational auditory scene analysis, the segment segregation and pitch estimation are necessary. However, it is hard to obtain an accurate pitch contour of clean speech for segregating the segments in low signal-to-noise ratio. This often leads to an inaccurate estimation of binary mask and produces artifacts and temporal discontinuity in the enhanced speech. To overcome this problem, we propose in this paper a new estimation method for binary mask based on convex optimization of speech power. In this proposed method, the segment segregation and pitch estimation are excluded and only the speech power of each Gammatone channel is considered as a key cue used for labeling the binary masks. Considering the cross-correlation between the power spectra of noisy speech and noise in each channel, the objective function of speech power is built, and the speech power is solved by the gradient descent method. Accordingly, the time-frequency units of speech and noise are labeled by computing a decision factor derived from the powers of noisy speech, estimated speech and the pre-estimated noise. The erroneous local masks are refined by time-frequency unit smoothing. The objective measurements including segmental signal-to-noise ratio, HIT-False Alarm rate, the percentage of energy loss and the percentage of noise residue and the additional subjective listening test demonstrate the effectiveness of the proposed method.

Accession Number: WOS:000427341500006
ISSN: 0167-6393
eISSN: 1872-7182

Record 14 of 50

Title: Active Noise Control in Three Dimensions

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Source: IEEE TRANSACTIONS ON CONTROL SYSTEMS TECHNOLOGY **Volume:** 22 **Issue:** 6 **Pages:** 2150-2159 **DOI:** 10.1109/TCST.2014.2301457 **Published:** NOV 2014

Abstract: Available acoustical signal processing algorithms for active noise control (ANC) can only attenuate noise at a finite number of points. Accordingly, they are unable to create continuous quiet zones in 3-D space. This paper proposes a methodology for developing a family of acoustical signal processing algorithms that are able to control sound in three dimensions. The theoretical analysis is carried out for spherical quiet zones; however, the proposed methodology can apply to quiet zones with arbitrary shapes. It is assumed that there is no reflection while sound propagates in media. An arbitrary frequency spectrum for the noise to be canceled is considered. An optimal control system for 3-D ANC is then derived. Also, an efficient method for the implementation of the proposed system is developed. The implementation cost of the proposed system is comparable with that of conventional ANC systems. In addition to the simulation results, different experiments with an experimental ANC system show the validity of the theoretical results.

Accession Number: WOS:000345576300007

Author Identifiers:

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Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1063-6536
eISSN: 1558-0865

Record 15 of 50

Title: Theoretical framework for stochastic modeling of FxLMS-based active noise control dynamics

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Book Group Author(s): IEEE

Source: 2012 ASIA-PACIFIC SIGNAL AND INFORMATION PROCESSING ASSOCIATION ANNUAL SUMMIT AND CONFERENCE (APSIPA ASC) **Book Series:** Asia-Pacific Signal and Information Processing Association Annual Summit and Conference **Published:** 2012

Abstract: There have been several contributions on theoretical modeling of FxLMS-based active noise control systems; however, when it is intended to derive elegant closed-form expressions for formulating dynamical behaviors of these systems, a number of simplifying assumptions regarding the acoustic noise, the actual secondary path and its model have to be used. This paper develops a dynamic model for FxLMS-based ANC systems, considering a general stochastic acoustic noise and a general secondary path. Also, an arbitrary secondary path model, which is not necessarily a perfect model, is considered. The main distinction of this model is that previously-derived dynamic models can be resulted in from it as special cases.

Accession Number: WOS:000319456200259

Conference Title: Annual Summit and Conference of the Asia-Pacific-Signal-and-Information-Processing-Association (APSIPA)

Conference Date: DEC 03-06, 2012

Conference Location: Hollywood, CA

Conference Sponsors: Asia Pacific Signal & Informat Proc Assoc (APSIPA)

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ISSN: 2309-9402
ISBN: 978-1-4673-4863-8

Record 16 of 50

Title: Perceptual evaluation of audio watermarking using objective quality measures

Author(s): Lin, YQ (Lin, Yiqing); Abdulla, WH (Abdulla, Waleed H.)

Book Group Author(s): IEEE

Source: 2008 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING, VOLS 1-12 **Book Series:** International Conference on Acoustics Speech and Signal Processing (ICASSP) **Pages:** 1745-1748 **DOI:** 10.1109/ICASSP.2008.4517967 **Published:** 2008

Abstract: This paper proposes objective quality measures adopted in speech processing for perceptual quality evaluation of audio watermarking Different from using an auditory perception model that mimics human auditory system, objective quality measures are introduced as alternative approach to perceive the dissimilarities caused by audio watermarking. After embedding the watermark into a variety of audio signals from EBU database, we calculate the distance between the watermarked and host signals in terms of several well-developed quality measures. For correlation analysis, subjective listening tests and a commercial evaluation tool PEMO-Q are also used to grade the differences. Pearson correlation coefficients reveal that the investigated quality measures, especially Weighted Spectral Slope (WSS) measure, correspond well with reference ratings. Moreover, quality measures run much faster than PEMO-Q. The results indicate that objective quality measures can be the perceptual quality predictors for audio watermarking.

Accession Number: WOS:000257456701098

Conference Title: 33rd IEEE International Conference on Acoustics, Speech and Signal Processing

Conference Date: MAR 30-APR 04, 2008

Conference Location: Las Vegas, NV

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 1520-6149

ISBN: 978-1-4244-1483-3

Record 17 of 50

Title: On Rayleigh distance and absorption length of parametric loudspeakers

Author(s): Farias, F (Farias, Felipe); Abdulla, W (Abdulla, Waleed)

Book Group Author(s): IEEE

Source: 2015 ASIA-PACIFIC SIGNAL AND INFORMATION PROCESSING ASSOCIATION ANNUAL SUMMIT AND CONFERENCE (APSIPA) **Book Series:** Asia-Pacific Signal and Information Processing Association Annual Summit and Conference **Pages:** 1262-1265 **Published:** 2015

Abstract: This paper investigates the propagation of primary and secondary waves produced by parametric loudspeakers of different sizes. The theory of the parametric acoustic array describes the nonlinear interaction of waves to be confined to the near-field, but the nonlinearities may remain over the far-field, producing different results. Four simulations were done to compare the performance of loudspeakers with different Rayleigh distances. Based on the simulations, a new design was proposed to overcome the mismatching of absorption length and Rayleigh distance. Control over these distances is proposed as an improvement of the performance of the parametric loudspeaker, considering the required scope and proper application.

Accession Number: WOS:000382954100239

Conference Title: Asia-Pacific-Signal-and-Information-Processing-Association Annual Summit and Conference (APSIPA ASC)

Conference Date: DEC 16-19, 2015

Conference Location: Hong Kong, PEOPLES R CHINA

Conference Sponsors: Asia Pacific Signal & Informat Proc Assoc

ISSN: 2309-9402

ISBN: 978-988-14768-0-7

Record 18 of 50

Title: Recognition accuracy of the new fingerprint construct based on a compact minutiae pattern

Author(s): Krivokuca, V (Krivokuca, Vedrana); Abdulla, W (Abdulla, Waleed)

Source: INTERNATIONAL JOURNAL OF BIOMETRICS **Volume:** 7 **Issue:** 2 **Pages:** 170-189 **DOI:** 10.1504/IJBM.2015.070929 **Published:** 2015

Abstract: Instead of using the entire minutiae template to generate a protected fingerprint template, recently a non-invertible cancellable fingerprint construct based on a 3-5 minutiae pattern was proposed as a safer alternative. This paper investigates the recognition accuracy attainable by this new fingerprint construct. It is found that using five samples of a person's reference fingerprint and allowing for a maximum of three authentication attempts provides a genuine user with the best chance of being successfully authenticated. An evaluation of the FAR and FRR in this scenario demonstrates that the new fingerprint construct can be tuned to suit the performance and security requirements of different applications by adjusting the pattern size and matching thresholds. The fingerprint construct is then modified to improve its ability to discriminate between genuine users and impostors. Compared to other non-invertible fingerprint template protection schemes, the performance of the modified fingerprint construct is found to be favourable.

Accession Number: WOS:000214650200005

Author Identifiers:

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Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1755-8301

eISSN: 1755-831X

Record 19 of 50

Title: Intra-class Variance Among Multiple Samples of the Same Person's Fingerprint in a Cooperative User Scenario

Author(s): Krivokuca, V (Krivokuca, Vedrana); Abdulla, W (Abdulla, Waleed)

Edited by: Fred A; DeMarsico M; Tabbone A

Source: PATTERN RECOGNITION APPLICATIONS AND METHODS, ICPRAM 2014 **Book Series:** Lecture Notes in Computer Science **Volume:** 9443 **Pages:** 77-92 **DOI:** 10.1007/978-3-319-25530-9_6 **Published:** 2015

Abstract: A significant challenge in the development of automated fingerprint recognition algorithms is dealing with intra-class variance among multiple samples of the same fingerprint. A major contributor to this intra-class variance is the inconsistency with which a finger is presented to the fingerprint scanner across multiple authentication attempts. This paper investigates the consistency of cooperative users in placing their finger on a typical fingerprint scanner, in terms of the amount of translation and rotation of the finger on the scanner surface and the percentage of reference minutiae that are present in the query fingerprint during each authentication attempt. A database of 800 fingerprint samples from 100 cooperative participants was collected for this purpose. Analysis of this database resulted in a median horizontal translation of 13 pixels (0.66 mm), a median vertical translation of 17 pixels (0.86 mm), a median rotation of 2 degrees, and a median minutiae repeatability of 96.1 %.

Accession Number: WOS:000374104100006

Conference Title: 3rd International Conference on Pattern Recognition Applications and Methods (ICPRAM)

Conference Date: MAR 06-08, 2014

Conference Location: Angers, FRANCE

Conference Sponsors: Inst Syst & Technologies Informat, Control & Commun, ACM Special Interest Grp Appl Comp, Assoc Advancement Artificial Intelligence, Portuguese Assoc Pattern Recognit

ISSN: 0302-9743

ISBN: 978-3-319-25530-9; 978-3-319-25529-3

Record 20 of 50

Title: Research Trends in Optical Spectrum for Honey Analysis

Author(s): Noviyanto, A (Noviyanto, Ary); Abdulla, W (Abdulla, Waleed); Yu, W (Yu, Wei); Salcic, Z (Salcic, Zoran)
Book Group Author(s): IEEE

Source: 2015 ASIA-PACIFIC SIGNAL AND INFORMATION PROCESSING ASSOCIATION ANNUAL SUMMIT AND CONFERENCE (APSIPA) **Book Series:** Asia-Pacific Signal and Information Processing Association Annual Summit and Conference **Pages:** 416-425 **Published:** 2015

Abstract: The high demand for genuine honey leads to fraud practices in the market which have disadvantaged top graded genuine honey production. The conventional chemical analysis procedures are usually used to ensure the quality and authenticity of honey. Yet, some drawbacks, such as time-consuming, laborious, invasive and required complex sample preparation, in the chemical approaches make the optical spectrum based honey analysis an advantageous alternative method. This paper reports a comprehensive survey of peer-reviewed articles in honey analysis using spectroscopy techniques. The technologies, features, and preprocessing and prediction methods from the observed articles have been discussed to give an overview about optical spectrum approaches for honey analysis. This paper quickly introduces reseachers to modern honey analysis research.

Accession Number: WOS:000382954100084
Conference Title: Asia-Pacific-Signal-and-Information-Processing-Association Annual Summit and Conference (APSIPA ASC)
Conference Date: DEC 16-19, 2015
Conference Location: Hong Kong, PEOPLES R CHINA
Conference Sponsors: Asia Pacific Signal & Informat Proc Assoc
ISSN: 2309-9402
ISBN: 978-988-14768-0-7

Record 21 of 50

Title: Adaptive signal processing algorithms for creating spatial zones of quiet

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.)

Source: DIGITAL SIGNAL PROCESSING **Volume:** 27 **Pages:** 129-139 **DOI:** 10.1016/j.dsp.2014.01.001 **Published:** APR 2014

Abstract: Available adaptive active noise control (ANC) algorithms can only minimize the noise level at a point that an error microphone is placed. Consequently, a zone of quiet around this microphone is produced as a byproduct. However, they cannot technically control or, even, monitor the noise level within the zone of quiet unless they use several sensors. They cannot also control the shape and the extension of the quiet zone by using only the error microphone. This paper develops a signal processing framework for the derivation of adaptive ANC algorithms that can directly create a controllable zone of quiet in monochromatic noise fields using a single error microphone. It is shown that by adding a filter to the standard ANC structure, a controllable zone of quiet is created. The transfer function of this filter is obtained using an accurate mathematical analysis. It is also shown that the extension of the zone of quiet can be controlled by tuning this filter. The implementation of the proposed system requires no additional hardware, rather than those required for traditional ANC systems. The validity of the results are discussed by using numerical analysis. Also, the performance of the proposed system is practically verified. (C) 2014 Elsevier Inc. All rights reserved.

Accession Number: WOS:000334822800012

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Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1051-2004
eISSN: 1095-4333

Record 22 of 50

Title: Sparse Approximations of 3D Mesh Geometry Using Frames as Overcomplete Dictionaries

Author(s): Krivokuca, M (Krivokuca, Maja); Abdulla, WH (Abdulla, Waleed H.); Wunsche, BC (Wuensche, Burkhard C.)
Book Group Author(s): IEEE

Source: 2013 IEEE INTERNATIONAL CONFERENCE ON COMPUTER VISION WORKSHOPS (ICCVW) **Pages:** 660-667 **DOI:** 10.1109/ICCVW.2013.91 **Published:** 2013

Abstract: This paper presents a novel method for creating a frame, to be used as an overcomplete dictionary for the progressive compression of 3D mesh geometry. The frame is computed from redundant linear combinations of the eigenvectors of a mesh Laplacian matrix, and atoms are selected by a Matching Pursuit algorithm. Experimental results show that a sparser representation of a given mesh geometry can be obtained with the frame than by decomposition of the mesh geometry onto an orthogonal basis. The proposed frame also has other desirable properties, including directionality and orientability of the atoms, and the ability to be applied directly to a manifold mesh with arbitrary topology and connectivity type.

Accession Number: WOS:000349847200089
Conference Title: IEEE International Conference on Computer Vision Workshops (ICCVW)
Conference Date: DEC 01-08, 2013
Conference Location: Sydney, AUSTRALIA
Conference Sponsors: IEEE, CVF, IEEE Comp Soc, APRS, Australiasn Natl Univ, NICTA, FACE++, Natl Robot Engn Ctr, Google, Disney Res, nVIDIA, Raytheon BBN Technologies, Facebook, Adobe, Kitware, OMRON, SRI Int

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Abdulla, Waleed		0000-0002-1812-4285

ISBN: 978-1-4799-3022-7

Record 23 of 50

Title: Lips tracking biometrics for speaker recognition

Author(s): Abdulla, WH (Abdulla, Waleed H.); Yu, PWT (Yu, Paul W. T.); Calverly, P (Calverly, Paul)

Source: INTERNATIONAL JOURNAL OF BIOMETRICS **Volume:** 1 **Issue:** 3 **Pages:** 288-306 **DOI:** 10.1504/IJBM.2009.024275 **Published:** 2009

Abstract: A novel approach to extract successful biometrics from mouth visual images is presented in this paper. Visual features are extracted from a sequence of images of speakers' lips while speaking. These features consist of the shape and intensity of pixels around the edge of the lips as well as their dynamics. The features are extracted by using particle filters technique to track the movements of the lips. The lips tracker shows adequate accuracy and ability to maintain lock in different speaking scenarios. Speaker models based on these features are built using Gaussian Mixture Models (GMM) trained through the Expectation-Maximisation (EM) algorithm. Satisfactory results are obtained for text-independent speaker recognition carried out on a video database of 35 individuals. A recognition rate of 82.8% for speaker identification and equal error rate of 18% for speaker verification are achieved using this technique.

Accession Number: WOS:000214605200003

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1755-8301

eISSN: 1755-831X

Record 24 of 50

Title: Multiple Scrambling and Adaptive Synchronization for Audio Watermarking

Author(s): Lin, YQ (Lin, Yiqing); Abdulla, WH (Abdulla, Waleed H.)

Edited by: Shi YQ; Kim HJ; Katzenbeisser S

Source: DIGITAL WATERMARKING, PROCEEDINGS **Book Series:** Lecture Notes in Computer Science **Volume:** 5041 **Pages:** 440-453 **Published:** 2008

Abstract: Imperceptibility, robustness and security are the vital considerations in the design of any audio watermarking scheme for copyrights protection. In this paper, a secure and robust audio watermarking scheme involving multiple scrambling and adaptive synchronization is proposed. To prevent the unauthorized detection, the new scheme integrates multiple scrambling operations into the embedding process. That is, encrypting the watermark with a coded-image and randomly selecting certain subbands for the embedding process. Moreover, the detection utilizes adaptive synchronization to enhance the robustness under some destructive de-synchronization attacks, like random samples cropping/inserting, pitch-invariant time stretching, and tempo-preserved pitch shifting. Theoretical analysis and simulation results have revealed that the proposed scheme is self-secured indeed and also immune to a wide range of severe attacks.

Accession Number: WOS:000263827600033

Conference Title: 6th International Workshop on Digital Watermarking

Conference Date: DEC 03-05, 2007

Conference Location: Sun Yat sen Univ, Guangzhou, PEOPLES R CHINA

Conference Sponsors: Natl Sci Fdn China, Korea Inst Informat Secur & Cryptol

Conference Host: Sun Yat sen Univ

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 0302-9743

ISBN: 978-3-540-92237-7

Record 25 of 50

Title: Robust speaker modeling using perceptually motivated feature

Author(s): Abdulla, WH (Abdulla, Waleed H.)

Source: PATTERN RECOGNITION LETTERS **Volume:** 28 **Issue:** 11 **Pages:** 1333-1342 **DOI:** 10.1016/j.patrec.2006.11.018 **Published:** AUG 1 2007

Abstract: This paper introduces a novel method to extract robust features for text-independent speaker identification from short utterances. This method is perceptually motivated and inspired by the perceptual linear prediction (PLP) technique. The new feature is called perceptual log area ratio (PLAR). It is perceptual in the sense that it depends on notions from psychoacoustics where the robustness can be assured. Also, the log area ratio is an effective feature for recognizing speakers as it embodies the geometry and dynamics of the vocal tract, which are very much person-dependent. This research thus focuses on providing a reliable vocal biometric from speakers, which can be used effectively with full-band and telephone-band speech in noisy environments. Intensive performance analysis has been performed to benchmark the proposed method against the commonly-used features using different databases in different noisy environments. In almost all usable cases the PLAR proved its superiority over the commonly-used features such as MFCC and LPCC. (c) 2007 Elsevier B.V. All rights reserved.

Accession Number: WOS:000247807500009

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 0167-8655

Record 26 of 50

Title: Multi-agent system with hybrid intelligence using neural network and fuzzy inference techniques

Author(s): Wang, KIK (Wang, Kevin I-Kai); Abdulla, WH (Abdulla, Waleed H.); Salcic, Z (Salcic, Zoran)

Edited by: Okuno HG; Ali M

Source: NEW TRENDS IN APPLIED ARTIFICIAL INTELLIGENCE, PROCEEDINGS **Book Series:** Lecture Notes in Artificial Intelligence **Volume:** 4570 **Pages:** 473-+ **Published:** 2007

Abstract: In this paper, a novel multi-agent control system incorporating hybrid intelligence and its physical testbed are presented. The physical testbed is equipped with a large number of embedded devices interconnected by three types of physical networks. It mimics a ubiquitous intelligent environment and allows real-time data collection and online system evaluation. Human control behaviours for different physical devices are analysed and classified into three categories. Physical devices are grouped based on their relevance and each group is assigned to a particular behaviour category. Each device group is independently modelled by either fuzzy inference or neural network agents according to the behaviour category. Comparative analysis shows that the proposed multi-agent control system with hybrid intelligence achieves significant improvement in control accuracy compared to other offline control systems.

Accession Number: WOS:000248621400047

Conference Title: 20th International Conference on Industrial, Engineering and Other Applications of Applied Intelligent Systems

Conference Date: JUN 26-29, 2007

Conference Location: Kyoto, JAPAN

Conference Sponsors: Int Soc Appl Intelligence, Kyoto Univ, Grad Sch Informat, AAAI, ACM SIGART, CSCSI, SCEIO, ENNS, HIS, IEICE, INNS, IPSJ, JSAI, RSJ, TAAI, Texas State Univ, San Marcos

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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Wang, Kevin		0000-0001-8450-2558

ISSN: 0302-9743

ISBN: 978-3-540-73322-5

Record 27 of 50

Title: Multi-agent software control system with hybrid intelligence for ubiquitous intelligent environments

Author(s): Wang, KIK (Wang, Kevin I-Kai); Abdulla, WH (Abdulla, Waleed H.); Salcic, Z (Salcic, Zoran)

Edited by: Indulska J; Yang LT; Cao J; Ma J; Ungerer T

Source: UBIQUITOUS INTELLIGENCE AND COMPUTING, PROCEEDINGS **Book Series:** Lecture Notes in Computer Science **Volume:** 4611 **Pages:** 1046-+ **Published:** 2007

Abstract: In this paper, a novel ubiquitous intelligent environment platform and its multi-agent control system are presented. The platform named Distributed Embedded Intelligence Room (DEIR) has been constructed with embedded sensors, actuators and computing devices. All the devices are interconnected using five different physical networks. This platform aims to facilitate realistic data collection and online system performance evaluation. The multi-agent control system incorporating two machine learning algorithms, fuzzy inference and decision tree, has been designed to conform to DEIR architecture. Devices to be controlled are classified based on their possible output states and modelled separately by fuzzy inference agents and decision tree agents in the system. The multi-agent control system with hybrid intelligence shows 11% improvements on overall control accuracy and 84% improvements on learning time compared to its predecessor control system. The vast improvement on computational time shows suitability of the approach towards real-time, embedded applications.

Accession Number: WOS:000248246800102

Conference Title: 4th International Conference on Ubiquitous Intelligence and Computing

Conference Date: JUL 11-13, 2007

Conference Location: Hong Kong Polytech Univ, Hong Kong, PEOPLES R CHINA

Conference Host: Hong Kong Polytech Univ

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Wang, Kevin		0000-0001-8450-2558
Abdulla, Waleed		0000-0002-1812-4285
Salcic, Zoran		0000-0001-7714-9848

ISSN: 0302-9743

ISBN: 978-3-540-73548-9

Record 28 of 50

Title: Distributed embedded intelligence room with Multi-agent cooperative learning

Author(s): Wang, KIK (Wang, Kevin I-Kai); Abdulla, WH (Abdulla, Waleed H.); Salcic, Z (Salcic, Zoran)

Edited by: Ma J; Jin H; Yang LT; Tsai JJP

Source: UBIQUITOUS INTELLIGENCE AND COMPUTING, PROCEEDINGS **Book Series:** LECTURE NOTES IN COMPUTER SCIENCE **Volume:** 4159 **Pages:** 147-156 **Published:** 2006

Abstract: In this paper, a novel Multi-agent control system with fuzzy inference learning and its physical testbed are presented. In the Multi-agent system, distributed controlling, monitoring and cooperative learning are achieved through ubiquitous computing paradigm. The physical testbed named Distributed Embedded Intelligence Room (DEIR) is equipped with a fair amount of embedded devices interconnected in three types of physical networks, namely LonWorks network, RS-485 network and IP network. The changes of environment states and user actions are recorded by software agents and are processed by fuzzy inference learning algorithm to form fuzzy rules that capture user behaviour. With these rules, fuzzy logic controllers can perform user preferred control actions. Comparative analysis shows our control system has achieved noticeable improvement in control accuracy compared to the other offline control system.

Accession Number: WOS:000240542600015

Conference Title: 3rd International Conference on Ubiquitous Intelligence and Computing

Conference Date: SEP 03-06, 2006

Conference Location: Wuhan, PEOPLES R CHINA
Conference Sponsors: Huazhong Univ Sci & Technol, Natl Sci Fdn China, Int Federat Informat Proc, IEEE Comp Soc, Lecture Notes Comp Sci Springer
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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Abdulla, Waleed		0000-0002-1812-4285

ISSN: 0302-9743
ISBN: 3-540-38091-4

Record 29 of 50
Title: Automatic detection and segmentation of optic disc and fovea in retinal images
Author(s): Chalakkal, RJ (Chalakkal, Renoh Johnson); Abdulla, WH (Abdulla, Waleed Habib); Thulaseedharan, SS (Thulaseedharan, Sinumol Sukumaran)
Source: IET IMAGE PROCESSING **Volume:** 12 **Issue:** 11 **Pages:** 2100-2110 **DOI:** 10.1049/iet-ipr.2018.5666 **Published:** NOV 2018
Abstract: Feature extraction from retinal images is gaining popularity worldwide as many pathologies are proved having connections with these features. Automatic detection of these features makes it easier for the specialist ophthalmologists to analyse them without spending exhaustive time to segment them manually. The proposed method automatically detects the optic disc (OD) using histogram-based template matching combined with the maximum sum of vessel information in the retinal image. The OD region is segmented by using the circular Hough transform. For detecting fovea, the retinal image is uniformly divided into three horizontal strips and the strip including the detected OD is selected. Contrast of the horizontal strip containing the OD region is then enhanced using a series of image processing steps. The macula region is first detected in the OD strip using various morphological operations and connected component analysis. The fovea is located inside this detected macular region. The proposed method achieves an OD detection accuracy over 95% upon testing on seven public databases and on our locally developed database, University of Auckland Diabetic Retinopathy database (UoA-DR). The average OD boundary segmentation overlap score, sensitivity and fovea detection accuracy achieved are 0.86, 0.968 and 97.26% respectively.
Accession Number: WOS:000454356600022
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Chalakkal, Renoh Johnson	Q-3565-2017	0000-0003-3452-5633

ISSN: 1751-9659
eISSN: 1751-9667

Record 30 of 50
Title: A CONVEX OPTIMIZATION APPROACH FOR TIME-FREQUENCY MASK ESTIMATION
Author(s): Bao, F (Bao, Feng); Abdulla, WH (Abdulla, Waleed H.)
Book Group Author(s): IEEE
Source: 2017 IEEE WORKSHOP ON APPLICATIONS OF SIGNAL PROCESSING TO AUDIO AND ACOUSTICS (WASPAA) **Book Series:** IEEE Workshop on Applications of Signal Processing to Audio and Acoustics **Pages:** 31-35 **Published:** 2017
Abstract: In this paper, we propose a new time-frequency mask method for computational auditory scene analysis (CASA) based on convex optimization of the binary mask. In the proposed method, the pitch estimation and segment segregation in conventional CASA are completely replaced by the convex optimization of speech power. Considering the cross-correlation between the power spectra of noisy speech and noise in each of a Gammatone filterbank channel, the objective function of speech power used for convex optimization is built. The speech power is estimated by gradient descent method. Thus, the time-frequency units dominated by speech and noise are labeled by comparing the powers of noisy and estimated speech, and noise. The erroneous local masks are also removed by using the Teager energy of the estimated speech and time-frequency unit smoothing. The results from the average segmental signal-to-noise ratio improvement, HIT-False Alarm rate and subjective test show that the performance of the proposed method outperforms the reference methods.
Accession Number: WOS:000426939000008
Conference Title: IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)
Conference Date: OCT 15-18, 2017
Conference Location: New Paltz, NY
ISSN: 1931-1168
ISBN: 978-1-5386-1632-1

Record 31 of 50
Title: AUTOMATIC SEGMENTATION OF RETINAL VASCULATURE
Author(s): Chalakkal, RJ (Chalakkal, Renoh Johnson); Abdulla, W (Abdulla, Waleed)
Book Group Author(s): IEEE
Source: 2017 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING (ICASSP) **Book Series:** International Conference on Acoustics Speech and Signal Processing ICASSP **Pages:** 886-890 **Published:** 2017
Abstract: Segmentation of retinal vessels from retinal fundus images is the key step in the automatic retinal image analysis. In this paper we propose a new unsupervised automatic method to segment the retinal vessels from retinal fundus images. Contrast enhancement and illumination correction are carried out through a series of image processing steps followed by adaptive histogram equalization and anisotropic diffusion filtering. This image is then converted to a gray scale using weighted scaling. The vessel edges are enhanced by boosting the detail curvelet coefficients. Optic disk pixels are removed before applying fuzzy C-mean classification to avoid the misclassification. Morphological operations and connected component analysis are applied to obtain the segmented retinal vessels. The performance of the proposed method is evaluated using DRIVE database to be able to compare with other state-of-art supervised and unsupervised methods. The overall segmentation accuracy of the proposed method is 95.18% which outperforms the other algorithms.
Accession Number: WOS:000414286201014

Conference Title: IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)
Conference Date: MAR 05-09, 2017
Conference Location: New Orleans, LA
Conference Sponsors: IEEE, Inst Elect & Elect Engineers, Signal Proc Soc
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Chalakkal, Renoh Johnson	Q-3565-2017	0000-0003-3452-5633
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1520-6149
ISBN: 978-1-5090-4117-6

Record 32 of 50
Title: EFFICIENT FXLMS ALGORITHM WITH SIMPLIFIED SECONDARY PATH MODELS
Author(s): Ardekani, IT (Ardekani, Iman T.); Sharifzadeh, H (Sharifzadeh, Hamid); Rehman, S (Rehman, Saeed); Abdulla, WH (Abdulla, Waleed H.)
Book Group Author(s): IEEE
Source: 2015 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING (ICASSP) **Book Series:** International Conference on Acoustics Speech and Signal Processing ICASSP **Pages:** 609-613 **Published:** 2015

Abstract: This paper extends the existing work on the root locus analysis of FxLMS algorithm by considering secondary path modeling errors. Rules for sketching FxLMS root locus are set out. An analytic convergence condition is then derived from the root locus plot. A deliberately-misaligned secondary path model is proposed to be used as the data preparation filter in the FxLMS algorithm. The proposed filter increases the computational efficiency of the algorithm, without changing its convergence behavior. The theoretical results are verified in practice by busing an experimental system.

Accession Number: WOS:000427402900122
Conference Title: 40th IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)
Conference Date: APR 19-24, 2015
Conference Location: Brisbane, AUSTRALIA
Conference Sponsors: IEEE, Inst Elect & Elect Engineers Signal Proc Soc
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 1520-6149
ISBN: 978-1-4673-6997-8

Record 33 of 50
Title: Fingerprint template protection scheme based on partial minutiae patterns: a comprehensive non-invertibility analysis
Author(s): Krivokuca, V (Krivokuca, Vedrana); Abdulla, W (Abdulla, Waleed)
Source: INTERNATIONAL JOURNAL OF BIOMETRICS **Volume:** 7 **Issue:** 4 **Pages:** 326-353 **DOI:** 10.1504/IJBM.2015.076136 **Published:** 2015

Abstract: A crucial property of an effective fingerprint template protection scheme is non-invertibility, which ensures that the original fingerprint template cannot be recovered from its secured counterpart. Since it is extremely difficult to design a function that achieves a high degree of non-invertibility, it is unsafe to use an entire fingerprint template as the input to any such function. Most existing fingerprint template protection mechanisms, however, do exactly this. One scheme that stands apart from the rest, in that its protected template originates from only a small portion of the minutiae template, is a new fingerprint construct based on partial minutiae patterns. This paper presents a comprehensive analysis of the non-invertibility of this scheme. It is found that the new fingerprint construct has extremely strong non-invertibility and good resistance to a Record Multiplicity Attack in practice. We also propose a modification to improve the method's resistance to this type of attack.

Accession Number: WOS:000214652100002
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1755-8301
eISSN: 1755-831X

Record 34 of 50
Title: STABILITY OF RESIDUAL ACOUSTIC NOISE VARIANCE IN ACTIVE CONTROL OF STOCHASTIC NOISE
Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, W (Abdulla, Waleed)
Book Group Author(s): IEEE
Source: 2013 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING (ICASSP) **Book Series:** International Conference on Acoustics Speech and Signal Processing ICASSP **Pages:** 379-382 **Published:** 2013

Abstract: This paper concerns about the theoretical stability of the adaptation process performed by the Filtered-x Least Mean Square (FxLMS) algorithm in active control of acoustic noise. A dynamic model for the Variance of Residual Acoustic Noise (VRAN) is developed and it is shown that the stability of this model is a sufficient condition for the stability of the adaptation process. The basic rules governing the VRAN root locus are developed, based on which an upper-bound for the adaptation step-size is derived. This upper-bound can apply to a general case with an arbitrary secondary path, unlike the traditional upper-bound used in adaptive filter theory, which was derived only for pure delay secondary paths.

Accession Number: WOS:000329611500077
Conference Title: IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)

Conference Date: MAY 26-31, 2013
Conference Location: Vancouver, CANADA
Conference Sponsors: Inst Elect & Elect Engineers, Inst Elect & Elect Engineers Signal Proc Soc
Author Identifiers:

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ISSN: 1520-6149
ISBN: 978-1-4799-0356-6

Record 35 of 50

Title: Noise Robust Speech Activity Detection

Author(s): Abdulla, WH (Abdulla, Waleed H.); Guan, Z (Guan, Zhou); Sou, HC (Sou, Hou Chi)

Book Group Author(s): IEEE

Source: 2009 IEEE INTERNATIONAL SYMPOSIUM ON SIGNAL PROCESSING AND INFORMATION TECHNOLOGY (ISSPIT 2009) **Book Series:** IEEE International Symposium on Signal Processing and Information Technology **Pages:** 473-477 **DOI:** 10.1109/ISSPIT.2009.5407509 **Published:** 2009

Abstract: An efficient noise robust feature is presented to track the speech activity in noisy environments. Speech is modeled by one class of 16 phone-like Gaussian mixtures while noises are modeled by 15 classes of 6 mixtures each. The feature vector used is a concatenation of carefully selected coefficients from MFCC, LPCC, and their first and second derivatives. A finite state machine and energy validation components are proposed as post-processor for the GMM classifier to rectify the misclassified speech segments. The demonstrated speech activity detection system based on our feature detects reliably both speech and non-speech segments. The designed frame work has been benchmarked against the commercially available codecs G.729, GSM-EFR, MR1, and MR2. Results show the proposed technique outperforms all these commonly used techniques under various SNR levels and in different noisy environments.

Accession Number: WOS:000290365400085

Conference Title: 9th IEEE International Symposium on Signal Processing and Information Technology

Conference Date: DEC 14-17, 2009

Conference Location: Ajman, U ARAB EMIRATES

Conference Sponsors: IEEE Comp Soc

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 2162-7843

ISBN: 978-1-4244-5949-0

Record 36 of 50

Title: Speech enhancement by multi-channel crosstalk resistant adaptive noise cancellation

Author(s): Zeng, QN (Zeng, Qingning); Abdulla, WH (Abdulla, Waleed H.)

Book Group Author(s): IEEE

Source: 2006 IEEE International Conference on Acoustics, Speech and Signal Processing, Vols 1-13 **Book Series:** International Conference on Acoustics Speech and Signal Processing (ICASSP) **Pages:** 485-488 **Published:** 2006

Abstract: A novel Multi-channel Crosstalk Resistant Adaptive Noise Cancellation (MCRANC) algorithm is presented in this paper to enhance noise carrying speech signals. The algorithm would permit locating the microphones in close proximity as it cancels out the crosstalk effect. Results have indicated that this method outperforms the commonly used techniques in the sense of SNR improvement and speech intelligibility. A SNR improvement of 17.8dB using MCRANC keeping highly intelligible speech was achieved in our experiments versus 9.1dB using Multi-channel ANC (MANC) with far less speech quality.

Accession Number: WOS:000245559901004

Conference Title: 31st IEEE International Conference on Acoustics, Speech and Signal Processing

Conference Date: MAY 14-19, 2006

Conference Location: Toulouse, FRANCE

Conference Sponsors: IEEE Signal Proc Soc

ISSN: 1520-6149

ISBN: 978-1-4244-0468-1

Record 37 of 50

Title: Gammatone Auditory Filterbank and Independent Component Analysis for Speaker Identification

Author(s): Zhang, YS (Zhang, Yushi); Abdulla, WH (Abdulla, Waleed H.)

Book Group Author(s): ISCA

Source: INTERSPEECH 2006 AND 9TH INTERNATIONAL CONFERENCE ON SPOKEN LANGUAGE PROCESSING, VOLS 1-5 **Pages:** 2098-2101 **Published:** 2006

Abstract: Feature extraction is the key procedure when aiming at robust speaker identification. The most commonly used feature extraction techniques work successfully only in clean or matched environments. Accurate speaker identification is made difficult due to a number of factors, with handset/channel mismatch and environmental noise being the most prominent. This paper presents a novel technique which based on Gammatone filterbank (GTF) and independent component analysis (ICA). The presented method first relies on the Gammatone filterbank to emulate the human cochlea frequency resolution. By using ICA, it extracts the dominant components from these frequency banks. The extracted features emphasis the difference in the statistical structures among the speakers, which can model the distribution of the individuals. Compared to the commonly used techniques, such as linear predictive cepstral coefficients (LPCC), Mel-frequency cepstrum coefficients (MFCC) and perceptual linear predictive (PLP), the proposed method is more robust to additive noises and yields higher recognition rate in mismatch environments in a text-independent speaker identification system.

Accession Number: WOS:000269965901262

Conference Title: 9th International Conference on Spoken Language Processing/INTERSPEECH 2006
Conference Date: 2006
Conference Location: Pittsburgh, PA
Conference Sponsors: Int Speech Commun Assoc
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285
ISBN: 978-1-60423-449-7

Record 38 of 50
Title: Segmentation and calibration of hyperspectral imaging for honey analysis
Author(s): Noviyanto, A (Noviyanto, Ary); Abdulla, WH (Abdulla, Waleed H.)
Source: COMPUTERS AND ELECTRONICS IN AGRICULTURE **Volume:** 159 **Pages:** 129-139 **DOI:** 10.1016/j.compag.2019.02.006 **Published:** APR 2019
Abstract: Hyperspectral imaging as a fast and non-invasive method for honey analysis has great potential to overcome the drawbacks of the conventional chemically based assessment methods. This paper discusses segmentation and calibration techniques to acquire accurate and consistent spectra in the implementation of hyperspectral imaging. These essential techniques have not yet been fully investigated despite they have significant implications over the honey analysis accuracy. Those techniques were developed and assessed under reflectance and transmittance sensing modes. The developed segmentation strategies (manual and automatic) followed by a selective average function demonstrated a reliable spectra extraction of honey samples. The developed calibration technique using a dynamic reference followed by normalisation successfully corrected the extracted spectra from distortions caused by variations in temperatures and lighting powers; also it greatly reduced the effect of spatial heterogeneity. The proposed segmentation and calibration techniques ensure the repeatability of spectral information acquisition which is very important for further processing to develop machine learning and statistically based prediction models.
Accession Number: WOS:000463308400015
ISSN: 0168-1699
eISSN: 1872-7107

Record 39 of 50
Title: A New Ratio Mask Representation for CASA-Based Speech Enhancement
Author(s): Bao, F (Bao, Feng); Abdulla, WH (Abdulla, Waleed H.)
Source: IEEE-ACM TRANSACTIONS ON AUDIO SPEECH AND LANGUAGE PROCESSING **Volume:** 27 **Issue:** 1 **Pages:** 7-19 **DOI:** 10.1109/TASLP.2018.2868407 **Published:** JAN 2019
Abstract: In the computational auditory scene analysis method, the ideal ratio mask or alternatively the ideal binary mask is the key point to reconstruct the enhanced signal. The ratio mask in its Wiener filtering or its square root form is currently considered. However, this kind of ratio mask overlooked one important issue. It does not exploit the inter-channel correlation (ICC) in the noisy speech, noise, and clean speech spectra. Thus, in this paper, we first propose a novel ratio mask representation by utilizing the ICC. In this way, we adaptively reallocate the power ratio of the speech and noise during the construction of ratio mask, thus, more speech and noise components are retained and masked at the same time, respectively. Second, the channel-weight contour based on the equal loudness hearing attribute is adopted to revise this new ratio mask in each Gammatone filterbank channel. Finally, the revised ratio mask is effectively used to train a five-layer structured deep neural network. Experiments show that the proposed ratio mask performs better than the conventional ratio mask representation and other series of enhancement algorithms in terms of speech quality, intelligibility, and spectral distortion under different signal to noise ratio conditions using six types of noises.
Accession Number: WOS:000446326100001
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285
ISSN: 2329-9290

Record 40 of 50
Title: Noise masking method based on an effective ratio mask estimation in Gammatone channels
Author(s): Bao, F (Bao, Feng); Abdulla, WH (Abdulla, Waleed H.)
Source: APSIPA TRANSACTIONS ON SIGNAL AND INFORMATION PROCESSING **Volume:** 7 **Article Number:** e5 **DOI:** 10.1017/ATSIP.2018.7 **Published:** MAY 15 2018
Abstract: In computational auditory scene analysis, the accurate estimation of binary mask or ratio mask plays a key role in noise masking. An inaccurate estimation often leads to some artifacts and temporal discontinuity in the synthesized speech. To overcome this problem, we propose a new ratio mask estimation method in terms of Wiener filtering in each Gammatone channel. In the reconstruction of Wiener filter, we utilize the relationship of the speech and noise power spectra in each Gammatone channel to build the objective function for the convex optimization of speech power. To improve the accuracy of estimation, the estimated ratio mask is further modified based on its adjacent time-frequency units, and then smoothed by interpolating with the estimated binary masks. The objective tests including the signal-to-noise ratio improvement, spectral distortion and intelligibility, and subjective listening test demonstrate the superiority of the proposed method compared with the reference methods.
Accession Number: WOS:000434322700001
ISSN: 2048-7703

Record 41 of 50
Title: A NOVEL TRAINING TARGET OF DNN USED FOR CASA-BASED SPEECH ENHANCEMENT
Author(s): Bao, F (Bao, Feng); Abdulla, WH (Abdulla, Waleed H.)
Book Group Author(s): IEEE

Source: 2018 16TH INTERNATIONAL WORKSHOP ON ACOUSTIC SIGNAL ENHANCEMENT (IWAENC) **Book Series:** International Workshop on Acoustic Signal Enhancement **Pages:** 346-350 **Published:** 2018

Abstract: Efficient training target plays a vital role in any training model for speech enhancement. In the Computational Auditory Scene Analysis method based on Deep Neural Networks, the ideal ratio mask or square-root ideal ratio mask is usually considered as the effective training target. In this paper, we propose a novel training target method for speech enhancement. This new training target takes into consideration the inter channel correlations of the power spectra of the noisy speech, clean speech and noise to more efficiently retain speech components and mask the noise components. Additionally, the channel-weight contour based on the equal loudness hearing attribute is introduced to revise the training target in each Gammatone channel to make the resynthesized signal more suitable for hearing characteristics. Moreover, the training target is smoothed to further improve the accuracy. Experiments show that using the proposed training target achieves better performances in terms of speech quality and intelligibility.

Accession Number: WOS:000458323900070

Conference Title: 16th International Workshop on Acoustic Signal Enhancement (IWAENC)

Conference Date: SEP 17-20, 2018

Conference Location: Tokyo, JAPAN

Conference Sponsors: Ichimura Fdn New Technol, Tateisi Sci & Technol Fdn, Yahoo! Japan, Google, Microsoft, Adobe, Hitachi, Dialog Semiconductor, Support Ctr Adv Telecommun Technol Res, mhacoustics, RION, IEEE, IEEE Signal Proc Soc, EiC, Informat Proc Soc Japan

ISSN: 2639-4316

ISBN: 978-1-5386-8151-0

Record 42 of 50

Title: Progressive Compression of 3D Mesh Geometry Using Sparse Approximations from Redundant Frame Dictionaries

Author(s): Krivokuca, M (Krivokuca, Maja); Abdulla, WH (Abdulla, Waleed Habib); Wunsche, BC (Wunsche, Burkhard Claus)

Source: ETRI JOURNAL **Volume:** 39 **Issue:** 1 **Pages:** 1-12 **DOI:** 10.4218/etrij.17.0116.0509 **Published:** FEB 2017

Abstract: In this paper, we present a new approach for the progressive compression of three-dimensional (3D) mesh geometry using redundant frame dictionaries and sparse approximation techniques. We construct the proposed frames from redundant linear combinations of the eigenvectors of a combinatorial mesh Laplacian matrix. We achieve a sparse synthesis of the mesh geometry by selecting atoms from a frame using matching pursuit. Experimental results show that the resulting ratedistortion performance compares favorably with other progressive mesh compression algorithms in the same category, even when a very simple, sub-optimal encoding strategy is used for the transmitted data. The proposed frames also have the desirable property of being able to be applied directly to a manifold mesh having arbitrary topology and connectivity types; thus, no initial remeshing is required and the original mesh connectivity is preserved.

Accession Number: WOS:000394359500001

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 1225-6463

eISSN: 2233-7326

Record 43 of 50

Title: Speech Spectrum Analysis Based. On Higher Order Crossings

Author(s): Abdulla, WH (Abdulla, Waleed H.); Wang, RL (Wang, Ruili)

Book Group Author(s): IEEE

Source: 2017 4TH INTERNATIONAL CONFERENCE ON SIGNAL PROCESSING AND INTEGRATED NETWORKS (SPIN) **Pages:** 19-22 **Published:** 2017

Abstract: In this paper, we propose a simple technique for extrapolating the spectra of speech signal based on real zero crossings and higher order crossings (HOC). In this approach, the speech signal is first subjected to a sequence of difference/sum filtering. 'then counts of zero crossings are calculated over short overlapped frames to obtain a sequence of counts. This sequence represents the dominant frequency components of the speech signal over the analysis frames. The HOC based spectral characteristics are found to be distinct for each utterance. To evaluate the discrimination performance of the HOC counts, we consider them as feature vectors to train a speech recognition system. We found that the recognition rate achieved by using the HOC feature is comparable to what can he achieved by the commonly used mel frequency cepstral coefficients, which is more computationally expensive. This paper would pave the road for more extensive work on the HOC features and variants as applied to speech signals.

Accession Number: WOS:000426076800005

Conference Title: 4th International Conference on Signal Processing and Integrated Networks (SPIN)

Conference Date: FEB 02-03, 2017

Conference Location: Amity Univ, Noida, INDIA

Conference Sponsors: Amity Sch Engn & Technol, IEEE, IEEE UP Sect

Conference Host: Amity Univ

ISBN: 978-1-5090-2797-2

Record 44 of 50

Title: Signal Power Estimation Based on Convex Optimization for Speech Enhancement

Author(s): Bao, F (Bao, Feng); Abdulla, WH (Abdulla, Waleed H.)

Book Group Author(s): IEEE

Source: 2017 ASIA-PACIFIC SIGNAL AND INFORMATION PROCESSING ASSOCIATION ANNUAL SUMMIT AND CONFERENCE (APSIPA ASC 2017) **Book Series:** Asia-Pacific Signal and Information Processing Association Annual Summit and Conference **Pages:** 483-487 **Published:** 2017

Abstract: Although the state-of-the-art single-channel speech enhancement algorithms has gained an appreciable suppression of the noise and improved speech quality to some extent, they are not mostly oriented toward improving the intelligibility of noise-degraded speech signal. In this paper, we propose a novel noise masking method based on convex optimization of the clean speech power to target the enhanced signal intelligibility. In our proposed method, the speech signal is analyzed in Gammatone domain instead of Short-Time Fourier Transform domain. The clean speech power is estimated by the convex

optimization algorithm. The ratio mask used to resynthesize speech is estimated by the powers of the estimated clean speech and noise. The combination of the binary mask and ratio mask contributes to a further improvement of the performance. We evaluated the enhancement results for the proposed method using objective and subjective measures. We show that the performance of the proposed approach outperforms the reference methods.

Accession Number: WOS:000425879400087

Conference Title: 9th Annual Summit and Conference of the Asia-Pacific-Signal-and-Information-Processing-Association (APSIPA ASC)

Conference Date: DEC 12-15, 2017

Conference Location: Kuala Lumpur, MALAYSIA

ISSN: 2309-9402

ISBN: 978-1-5386-1542-3

Record 45 of 50

Title: Honey Dataset Standard Using Hyperspectral Imaging for Machine Learning Problems

Author(s): Noviyanto, A (Noviyanto, Ary); Abdulla, WH (Abdulla, Waleed H.)

Book Group Author(s): IEEE

Source: 2017 25TH EUROPEAN SIGNAL PROCESSING CONFERENCE (EUSIPCO) **Book Series:** European Signal Processing Conference **Pages:** 473-477 **Published:** 2017

Abstract: Hyperspectral imaging has been rarely investigated for honey analyses, on the contrary to the optical spectroscopy which is widely investigated. The essential missing component to kick start this research is a standard honey hyperspectral images, called hypercubes, dataset. This paper proposes a systematic procedure for the preparation of the first honey hypercube dataset using hyperspectral imaging. Moreover, a scalable and flexible dataset module is introduced to ease the interaction between raw hypercube data and machine learning software. The developed dataset greatly benefits researchers to progress on the research of honey analysis including constituents prediction and types classification using hyperspectral imaging and machine learning.

Accession Number: WOS:000426986000096

Conference Title: 25th European Signal Processing Conference (EUSIPCO)

Conference Date: AUG 28-SEP 02, 2017

Conference Location: GREECE

Conference Sponsors: European Assoc Signal Proc, IEEE Signal Proc Soc, Inst Commun & Comp Syst, Natl Tech Univ Athens, Plaisio, Mitsubishi Electr Res Lab, MathWorks, Blue Star Ferries, Moto Island Bicycle Rentals

ISSN: 2076-1465

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Record 46 of 50

Title: A Noise Masking Method with Adaptive Thresholds based on CASA

Author(s): Bao, F (Bao, Feng); Abdulla, WH (Abdulla, Waleed H.)

Book Group Author(s): IEEE

Source: 2016 ASIA-PACIFIC SIGNAL AND INFORMATION PROCESSING ASSOCIATION ANNUAL SUMMIT AND CONFERENCE (APSIPA) **Book Series:** Asia-Pacific Signal and Information Processing Association Annual Summit and Conference **Published:** 2016

Abstract: In this paper, we propose a novel noise masking method based on Computational Auditory Scene Analysis by using an adaptive factor. Although it has succeeded in the field of speech separation and speech enhancement to some extent, the usage of fixed thresholds used for segregation and labeling heavily affects the processing performance. Focusing on this issue, the proposed method utilizes the Normalized Cross-Correlation Coefficients between the power spectra of noisy speech and pure noise to find an adaptive threshold, so that the pitch contour and Time-Frequency units can be obtained more accurately. Then, a revised algorithm is used to smooth the current binary mask value by checking the Time-Frequency units within adjacent frames and neighbor channels around the current Time-Frequency unit in order to remove the erroneous local masks. Two kinds of Signal to Noise Ratio test results show that the performance of the proposed method outperforms conventional spectral subtractive, Wiener Filtering and Computational Auditory Scene Analysis methods.

Accession Number: WOS:000393591800208

Conference Title: Asia-Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA)

Conference Date: DEC 13-16, 2016

Conference Location: Jeju, SOUTH KOREA

ISSN: 2309-9402

ISBN: 978-988-14768-2-1

Record 47 of 50

Title: Cancellability and diversity analysis of fingerprint template protection scheme based on compact minutiae pattern

Author(s): Krivokuca, V (Krivokuca, Vedrana); Abdulla, W (Abdulla, Waleed)

Source: INFORMATION SECURITY JOURNAL **Volume:** 25 **Issue:** 1-3 **Pages:** 109-123 **DOI:** 10.1080/19393555.2016.1179375 **Published:** 2016

Abstract: As fingerprints continue toward ubiquity in human recognition applications, growing fingerprint databases will pose an increasingly greater risk of irreversible identity theft in the event of a database breach. Consequently, more focus is being placed on researching new and effective ways of securing fingerprint templates during database storage. Recently, a new fingerprint template protection scheme, based on representing a fingerprint by a sparse 3-, 4-, or 5-minutiae pattern, has been proposed. The most important advantage of this method over other fingerprint template protection schemes is that it employs only a small number of identifying features in the creation of the protected template, such that it is impossible to recover the original fingerprint even if the protected template is compromised. In this article, we present a thorough analysis to demonstrate that this new fingerprint construct also boasts impressive cancellability and diversity properties. Cancellability allows for the replacement of a compromised template with a new template from the same fingerprint, and diversity enables a person to enroll into multiple applications using the same fingerprint without the prospect of being tracked across the different applications.

Accession Number: WOS:000381692500009

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ISSN: 1939-3555

eISSN: 1939-3547

Record 48 of 50

Title: Noninvertible fingerprint transforms: Categorization of design mechanisms and discussion of evaluation techniques

Author(s): Krivokuca, V (Krivokuca, Vedrana); Abdulla, W (Abdulla, Waleed)

Source: INFORMATION SECURITY JOURNAL **Volume:** 25 **Issue:** 4-6 **Pages:** 261-279 **DOI:** 10.1080/19393555.2016.1228129 **Published:** 2016

Abstract: The explosive growth in fingerprint technologies within the past decade has seen the emergence of a dedicated field of research into securing fingerprint templates during storage in a database. While new fingerprint template protection techniques are often broadly classified as belonging to the well-known salting, noninvertible transforms, key binding, or key generation categories, methods within each category are currently lacking a sense of organization. This article aims to fill this gap by proposing a categorization of noninvertible fingerprint transforms based on their design mechanisms. Our survey of the current literature in this field reveals two prominent types of approaches, so we classify existing noninvertible fingerprint transforms into two main categories: perturbation-based and histogram-based. We also discuss the evaluation techniques used to assess the robustness of noninvertible fingerprint transforms in the literature. These contributions will serve to help researchers find their bearing in the growing fingerprint template protection field, thereby encouraging a deeper understanding of the field and faster progress in the development of more effective fingerprint template protection schemes.

Accession Number: WOS:000391009200009

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Abdulla, Waleed		0000-0002-1812-4285

ISSN: 1939-3555

eISSN: 1939-3547

Record 49 of 50

Title: Joint optimization on decoding graphs using minimum classification error criterion

Author(s): Abdelhamid, AA (Abdelhamid, Abdelaziz A.); Abdulla, WH (Abdulla, Waleed H.)

Source: APSIPA TRANSACTIONS ON SIGNAL AND INFORMATION PROCESSING **Volume:** 3 **Article Number:** e6 **DOI:** 10.1017/ATSIP.2014.5 **Published:** 2014

Abstract: Motivated by the inherent correlation between the speech features and their lexical words, we propose in this paper a new framework for learning the parameters of the corresponding acoustic and language models jointly. The proposed framework is based on discriminative training of the models' parameters using minimum classification error criterion. To verify the effectiveness of the proposed framework, a set of four large decoding graphs is constructed using weighted finite-state transducers as a composition of two sets of context-dependent acousticmodels and two sets of n-gram-based languagemodels. The experimental results conducted on this set of decoding graphs validated the effectiveness of the proposed framework when compared with four baseline systems based on maximum likelihood estimation and separate discriminative training of acoustic and language models in benchmark testing of two speech corpora, namely TIMIT and RM1.

Accession Number: WOS:000215262900006

Author Identifiers:

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Abdulla, Waleed		0000-0002-1812-4285

ISSN: 2048-7703

Record 50 of 50

Title: Remote FxLMS Algorithm for Active Control of Sound in Remote Locations

Author(s): Ardekani, IT (Ardekani, Iman Tabatabaei); Abdulla, WH (Abdulla, Waleed H.); Rehman, SU (Rehman, Saeed Ur)

Book Group Author(s): IEEE

Source: 2014 ASIA-PACIFIC SIGNAL AND INFORMATION PROCESSING ASSOCIATION ANNUAL SUMMIT AND CONFERENCE (APSIPA) **Book Series:** Asia-Pacific Signal and Information Processing Association Annual Summit and Conference **Published:** 2014

Abstract: Available adaptive control algorithms can only create a silent point at the locations of an error microphone. In this case, the error microphone occupies a part of the zone of quiet established around the silent point, resulting in an ineffective use of space in the zone of quiet. This paper develops a novel adaptive control algorithm for active control of sound in remote locations rather than the locations of error microphones. It establishes a link between the performance of the FxLMS adaptive algorithm in the digital control domain and the performance of the active noise control mechanism in the acoustic domain. The transfer function of an optimal controller that can minimize the acoustic pressure in a location away from the error microphone is first derived. For the realization of the proposed controller, an adaptive algorithm, called Remote FxLMS is then developed. Different computer simulations are used to show the efficiency of the proposed algorithm.

Accession Number: WOS:000392861900041

Conference Title: Annual Summit and Conference of Asia-Pacific-Signal-and-Information-Processing-Association (APSIPA)

Conference Date: DEC 09-12, 2014

Conference Location: Angkor, CAMBODIA

Conference Sponsors: Asia Pacif Signal & Informat Proc Assoc

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