Record 1 of 50

Title: Recent advances on active noise control: open issues and innovative applications

Author(s): Kajikawa, Y (Kajikawa, Yoshinobu); Gan, WS (Gan, Woon-Seng); Kuo, SM (Kuo, Sen M.)

Source: APSIPA TRANSACTIONS ON SIGNAL AND INFORMATION PROCESSING **Volume:** 1 **Article Number:** e3 **DOI:** 10.1017/ATSIP.2012.4 **Published:** DEC 2012

Abstract: The problem of acoustic noise is becoming increasingly serious with the growing use of industrial and medical equipment, appliances, and consumer electronics. Active noise control (ANC), based on the principle of superposition, was developed in the early 20th century to help reduce noise. However, ANC is still not widely used owing to the effectiveness of control algorithms, and to the physical and economical constraints of practical applications. In this paper, we briefly introduce some fundamental ANC algorithms and theoretical analyses, and focus on recent advances on signal processing algorithms, implementation techniques, challenges for innovative applications, and open issues for further research and development of ANC systems.

Accession Number: WOS:000215262700003

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518
Gan, Woon-Seng	A-5151-2011	0000-0002-7143-1823

ISSN: 2048-7703

Record 2 of 50

Title: Active Noise Control System for Reducing MR Noise

Author(s): Kumamoto, M (Kumamoto, Masafumi); Kida, M (Kida, Masahiro); Hirayama, R (Hirayama, Ryotaro); Kajikawa, Y (Kajikawa, Yoshinobu); Tani, T (Tani, Toru); Kurumi, Y (Kurumi, Yoshimasa)

Source: IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS COMMUNICATIONS AND COMPUTER SCIENCES **Volume:** E94A **Issue:** 7 **Pages:** 1479-1486 **DOI:** 10.1587/transfun.E94.A.1479 **Published:** JUL 2011

Abstract: We propose an active noise control (ANC) system for reducing periodic noise generated in a high magnetic field such as noise generated from magnetic resonance imaging (MRI) devices (MR noise). The proposed ANC system utilizes optical microphones and piezoelectric loudspeakers, because specific acoustic equipment is required to overcome the high-field problem, and consists of a head-mounted structure to control noise near the user's ears and to compensate for the low output of the piezoelectric loudspeaker. Moreover, internal model control (IMC)-based feedback ANC is employed because the MR noise includes some periodic components and is predictable. Our experimental results demonstrate that the proposed ANC system (head-mounted structure) can significantly reduce MR noise by approximately 30 dB in a high field in an actual MRI room even if the imaging mode changes frequently.

Accession Number: WOS:000292619300001

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1745-1337

Record 3 of 50

Title: Head-mounted active noise control system with virtual sensing technique

Author(s): Miyazaki, N (Miyazaki, Nobuhiro); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: JOURNAL OF SOUND AND VIBRATION **Volume:** 339 **Pages:** 65-83 **DOI:** 10.1016/j.jsv.2014.11.023 **Published:** MAR 17 2015

Abstract: In this paper, we apply a virtual sensing technique to a head-mounted active noise control (ANC) system we have already proposed. The proposed ANC system can reduce narrowband noise while improving the noise reduction ability at the desired locations. A head-mounted ANC system based on an adaptive feedback structure can reduce noise with periodicity or narrowband components. However, since quiet zones are formed only at the locations of error microphones, an adequate noise reduction cannot be achieved at the locations where error microphones cannot be placed such as near the eardrums. A solution to this problem is to apply a virtual sensing technique. A virtual sensing ANC system can achieve higher noise reduction at the desired locations by measuring the system models from physical sensors to virtual sensors, which will be used in the online operation of the virtual sensing ANC algorithm. Hence, we attempt to achieve the maximum noise reduction near the eardrums by applying the virtual sensing technique to the head mounted ANC system. However, it is impossible to place the microphone near the eardrums. Therefore, the system models from physical sensors to virtual sensors are estimated using the Head And Torso Simulator (HATS) instead of human ears. Some simulation, experimental, and subjective assessment results demonstrate that the head mounted ANC system with virtual sensing is superior to that without virtual sensing in terms of the noise reduction ability at the desired locations. (C) 2014 Elsevier Ltd. All rights reserved.

Accession Number: WOS:000346849100006

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 0022-460X
eISSN: 1095-8568

Record 4 of 50

Title: An Estimation Method of Parameters for Closed-box Loudspeaker System

Author(s): Nakao, R (Nakao, Rika); Kajikawa, Y (Kajikawa, Yoshinobu); Nomura, Y (Nomura, Yasuo)

Source: IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS COMMUNICATIONS AND COMPUTER SCIENCES **Volume:** E91A **Issue:** 10 **Pages:** 3006-3013 **DOI:** 10.1093/ietfec/e91-a.10.3006 **Published:** OCT 2008

Abstract: In this paper, we propose a method that Uses Simulated Annealing (SA) to estimate the linear and nonlinear parameters of a closed-box loudspeaker system for implementing effective Mirror filters. The nonlinear parameters determined by W. Klippel's method are sometimes inaccurate and imaginary. In contrast, the proposed method can estimate the parameters with satisfactory accuracy due to its use of SA: the resulting impedance and displacement characteristics match those of an actual equivalent loudspeaker. A Mirror filter designed around these parameters can well compensate the nonlinear distortions of the loudspeaker system. Experiments demonstrate that the method can reduce the levels of nonlinear distortion by 5 dB to 20 dB compared to the before compensation condition.

Accession Number: WOS:000260400600042

Conference Title: International Workshop on Smart Info-Media Systems (SISB 2007)

Conference Date: 2007

Conference Location: Bangkok, THAILAND

Conference Sponsors: IEICE, Smart Info Media Syst Tech Grp

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1745-1337

Record 5 of 50

Title: Binaural active noise control using parametric array loudspeakers

Author(s): Tanaka, K (Tanaka, Kihiro); Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: APPLIED ACOUSTICS **Volume:** 116 **Pages:** 170-176 **DOI:** 10.1016/j.apacoust.2016.09.021 **Published:** JAN 15 2017

Abstract: This paper reports the binaural active noise control (ANC) system developed to deal with factory noise. The control points are located in the vicinity of the left and right ears of a worker sitting along the production line. Due to the complicated safety requirements in the factory, secondary sources and error microphones are not allowed to be placed near the worker. Therefore, the proposed ANC system employs the feedforward structure and adopts the parametric array loudspeakers (PALs) as the secondary sources. The PAL is a type of directional loudspeaker that generates a much narrower sound field as compared to the conventional loudspeaker. Once the proposed ANC system has been trained offline, the error microphones can be removed. The performance of the binaural ANC system is successfully demonstrated based on a digital signal processor (DSP) implementation. (C) 2016 Elsevier Ltd. All rights reserved.

Accession Number: WOS:000388156400020

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518
Shi, Chuang		0000-0002-1517-2775

ISSN: 0003-682X

eISSN: 1872-910X

Record 6 of 50

Title: A convolution model for computing the far-field directivity of a parametric loudspeaker array

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA **Volume:** 137 **Issue:** 2 **Pages:** 777-784 **DOI:** 10.1121/1.4906163 **Published:** FEB 2015

Abstract: This paper describes a method to compute the far-field directivity of a parametric loudspeaker array (PLA), whereby the steerable parametric loudspeaker can be implemented when phased array techniques are applied. The convolution of the product directivity and the Westervelt's directivity is suggested, substituting for the past practice of using the product directivity only. Computed directivity of a PLA using the proposed convolution model achieves significant improvement in agreement to measured directivity at a negligible computational cost. (C) 2015 Acoustical Society of America.

Accession Number: WOS:000350024600039

PubMed ID: 25698012

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Shi, Chuang		0000-0002-1517-2775
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 0001-4966

eISSN: 1520-8524

Record 7 of 50

Title: On adaptive covariance and spectrum estimation of locally stationary multivariate processes

Author(s): Niedzwiecki, M (Niedzwiecki, Maciej); Ciolek, M (Ciolek, Marcin); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: AUTOMATICA **Volume:** 82 **Pages:** 1-12 **DOI:** 10.1016/j.automatica.2017.04.033 **Published:** AUG 2017

Abstract: When estimating the correlation/spectral structure of a locally stationary process, one has to make two important decisions. First, one should choose the so-called estimation bandwidth, inversely proportional to the effective width of the local analysis window, in the way that complies with the degree of signal nonstationarity. Too small bandwidth may result in an excessive estimation bias, while too large bandwidth may cause excessive estimation variance. Second, but equally important, one should choose the appropriate order of the spectral representation of the signal so as to correctly model its

resonant structure when the order is too small, the estimated spectrum may not reveal some important signal components (resonances), and when it is too high, it may indicate the presence of some nonexistent components. When the analyzed signal is not stationary, with a possibly time-varying degree of nonstationarity, both the bandwidth and order parameters should be adjusted in an adaptive fashion. The paper presents and compares three approaches allowing for unified treatment of the problem of adaptive bandwidth and order selection for the purpose of identification of nonstationary vector autoregressive processes: the cross-validation approach, the full cross -validation approach, and the approach that incorporates the multivariate version of the generalized Akaike's final prediction error criterion. It is shown that the latter solution yields the best results and, at the same time, is very attractive from the computational viewpoint.

Accession Number: WOS:000403984300001

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 0005-1098

eISSN: 1873-2836

Record 8 of 50

Title: Ultrasound-to-Ultrasound Volterra Filter Identification of the Parametric Array Loudspeaker

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2015 IEEE INTERNATIONAL CONFERENCE ON DIGITAL SIGNAL PROCESSING (DSP) **Pages:** 1-4 **Published:** 2015

Abstract: The Volterra filter is a favorable approach to represent a variety of nonlinear systems, including the parametric array loudspeaker (PAL), which is a weak nonlinear acoustic system to create directional sounds. Using the Volterra filter to model the sound process of the PAL saves the computational cost of solving the nonlinear acoustic equation. In the past studies, the Volterra filter is identified from the audio input to the audio output of the PAL, which is known as the audio-to-audio Volterra filter (A2VF). In this paper, the ultrasound-to-ultrasound Volterra filter (U2VF) is recommended. The experiment results validate that the U2VF outperforms the A2VF, in terms of the robustness to the change of the modulation index in the PAL.

Accession Number: WOS:000380506600001

Conference Title: IEEE International Conference on Digital Signal Processing (DSP)

Conference Date: JUL 21-24, 2015

Conference Location: singapore, SINGAPORE

Conference Sponsors: IEEE Circuits and Syst Singapore Chapter, Imperial Coll London, IEEE, ROSE, NANYANG TECHNOL UNIV Temasek Labs NTU, parisstandford

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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Shi, Chuang		0000-0002-1517-2775

ISBN: 978-1-4799-8058-1

Record 9 of 50

Title: Linearization of loudspeaker systems using a subband parallel cascade volterra filter

Author(s): Furuhashi, H (Furuhashi, Hideyuki); Kajikawa, Y (Kajikawa, Yoshinobu); Nomura, Y (Nomura, Yasuo)

Source: IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS COMMUNICATIONS AND COMPUTER SCIENCES **Volume:** E90A **Issue:** 8 **Pages:** 1616-1619 **DOI:** 10.1093/ietfec/e90-a.8.1616 **Published:** AUG 2007

Abstract: In this paper, we propose a low complexity realization method for compensating for nonlinear distortion. Generally, nonlinear distortion is compensated for by a linearization system using a Volterra kernel. However, this method has a problem of requiring a huge computational complexity for the convolution needed between an input signal and the 2nd-order Volterra kernel. The Simplified Volterra Filter (SVF), which removes the lines along the main diagonal of the 2nd-order Volterra kernel, has been previously proposed as a way to reduce the computational complexity while maintaining the compensation performance for the nonlinear distortion. However, this method cannot greatly reduce the computational complexity. Hence, we propose a subband linearization system which consists of a subband parallel cascade realization method for the 2nd-order Volterra kernel and subband linear inverse filter. Experimental results show that this proposed linearization system can produce the same compensation ability as the conventional method while reducing the computational complexity.

Accession Number: WOS:000248934200017

Conference Title: 21st Symposium on Signal Processing

Conference Date: NOV 15-17, 2006

Conference Location: Kyoto, JAPAN

Conference Sponsors: IEICE, Signal Proc Tech Grp

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1745-1337

Record 10 of 50

Title: LINEARIZATION OF DYNAMIC LOUDSPEAKER SYSTEM USING THIRD-ORDER NONLINEAR IIR FILTER

Author(s): Iwai, K (Iwai, Kenta); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: 2012 PROCEEDINGS OF THE 20TH EUROPEAN SIGNAL PROCESSING CONFERENCE (EUSIPCO) **Book Series:** European Signal Processing

Conference **Pages:** 1970-1974 **Published:** 2012

Abstract: In this paper, we propose a 3rd-order nonlinear IIR filter for compensating for nonlinear distortions of loudspeaker systems. The 2nd-order nonlinear IIR filter based on the Mirror filter is used for reducing nonlinear distortions of loudspeaker systems. However, the 2nd-order nonlinear IIR filter cannot reduce nonlinear distortions at high frequencies because it does not include the nonlinearity of the self-inductance of loudspeaker systems. On the other hand, the proposed filter includes the effect of such self-inductance and thus can reduce nonlinear distortions at high frequencies. Experimental results demonstrate that the proposed filter can realize a reduction by 4 dB more than the conventional filter on inter-modulation distortions at high frequencies.

Accession Number: WOS:000310623800396

Conference Title: 20th European Signal Processing Conference (EUSIPCO)

Conference Date: AUG 27-31, 2012

Conference Location: Bucharest, ROMANIA

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 2076-1465

ISBN: 978-1-4673-1068-0

Record 11 of 50

Title: STATISTICAL-MECHANICAL ANALYSIS OF THE FXLMS ALGORITHM WITH ACTUAL PRIMARY PATH

Author(s): Miyoshi, S (Miyoshi, Seiji); Kajikawa, Y (Kajikawa, Yoshinobu)

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:** 3502-3506 **Published:** 2015

Abstract: A theory that predicts the behaviors of the Filtered-X LMS algorithm was derived by using a statistical-mechanical method. In this paper, the theory is generalized to explain the system behaviors in the case of an actual primary path. In the theory, cross-correlations between the element of a primary path and that of an adaptive filter and autocorrelations of the elements of the adaptive filter are treated as macroscopic variables. Simultaneous differential equations that describe the dynamical behaviors of the macroscopic variables are obtained under conditions in which the tapped-delay line is sufficiently long. The equations are analytically solved to obtain the correlations and finally compute the mean-square error. In order to generalize the theory to the case of an actual primary path, the correlations of the elements of the primary path are absorbed. The generalized theory quantitatively predict the behaviors in the case of an actual primary path.

Accession Number: WOS:000427402903123

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
Miyoshi, Seiji	C-7406-2013	
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1520-6149

ISBN: 978-1-4673-6997-8

Record 12 of 50

Title: Recent Applications and Challenges on Active Noise Control

Author(s): Kajikawa, Y (Kajikawa, Yoshinobu); Gan, WS (Gan, Woon-Seng); Kuo, SM (Kuo, Sen Maw)

Book Group Author(s): IEEE

Source: 2013 8TH INTERNATIONAL SYMPOSIUM ON IMAGE AND SIGNAL PROCESSING AND ANALYSIS (ISPA) **Book Series:** International Symposium on Image and Signal Processing and Analysis **Pages:** 661-+ **Published:** 2013

Abstract: Acoustic noise problems become more and more serious with increasing use of industrial and medical equipment, appliances, and consumer electronics. Active noise control (ANC) has been studied to solve such acoustic noise problems. ANC is a technique based on the principle of superposition, i.e., an antinoise with the same amplitude and opposite phase is generated and combined with an unwanted noise, thus resulting in the cancellation of both noises. However, ANC is still not widely used owing to the effectiveness of control algorithms, and to the physical and economical constraints of practical applications. In this paper, we briefly introduce some fundamental ANC structures, and focus on recent advances on signal processing algorithms, implementation techniques, challenges for innovative applications, and open issues for further research and development of ANC systems.

Accession Number: WOS:000349789200117

Conference Title: 8th International Symposium on Image and Signal Processing and Analysis (ISPA)

Conference Date: SEP 04-06, 2013

Conference Location: Trieste, ITALY

Conference Sponsors: Univ Trieste, Dept Engn & Arch, Univ Zagreb, Fac Elect Engn &d Comp, EURASIP, IEEE Signal Proc Soc, Italy Chapter, DAVe, FONDAZIONE CRTRIESTE, CHIRON, FIMI, ESTECO

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number

Gan, Woon-Seng	A-5151-2011	0000-0002-7143-1823
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1845-5921

ISBN: 978-953-184-194-8; 978-953-184-187-0

Record 13 of 50

Title: STATISTICAL-MECHANICAL ANALYSIS OF THE FXLMS ALGORITHM WITH NONWHITE REFERENCE SIGNALS

Author(s): Miyoshi, S (Miyoshi, Seiji); Kajikawa, Y (Kajikawa, Yoshinobu)

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:** 5652-5656 **Published:** 2013

Abstract: We analyze the learning curves of the FXLMS algorithm using a statistical-mechanical method when the reference signal is not necessarily white. We treat the nonwhite reference signal by introducing the correlation function of the signal to the method proposed in our previous study. Cross-correlations between the element of a primary path and that of an adaptive filter and autocorrelations of the elements of the adaptive filter are treated as macroscopic variables. We obtain simultaneous differential equations that describe the dynamical behaviors of the macroscopic variables under the conditions in which the tapped-delay line is long. We analytically solve the equations to obtain the correlations and finally compute the mean-square error. The obtained theory quantitatively agrees with the results of computer simulations. The theory also gives the upper limit of the step size in the FXLMS algorithm.

Accession Number: WOS:000329611505164

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:

Author	Web of Science ResearcherID	ORCID Number
Miyoshi, Seiji	C-7406-2013	
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1520-6149

ISBN: 978-1-4799-0356-6

Record 14 of 50

Title: THEORETICAL DISCUSSION OF THE FILTERED-X LMS ALGORITHM BASED ON STATISTICAL MECHANICAL ANALYSIS

Author(s): Miyoshi, S (Miyoshi, Seiji); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2012 IEEE STATISTICAL SIGNAL PROCESSING WORKSHOP (SSP) **Pages:** 341-344 **Published:** 2012

Abstract: We theoretically obtain the learning curves of the FXLMS algorithm on the basis of statistical mechanical analysis. Cosines of angles between the coefficient vectors of an adaptive filter, its shifted filters, and an unknown system are treated as macroscopic variables. Assuming that the tapped-delay line is sufficiently long and exactly calculating the correlations between the past tap input vectors and the coefficient vector of the adaptive filter, we obtain simultaneous differential equations that describe the dynamical behaviors of the macroscopic variables in a deterministic form. We analytically solve the equations and show that the obtained theory quantitatively agrees with computer simulations. In the analysis, neither the independence assumption, the small step-size condition, nor the few-taps assumption is used.

Accession Number: WOS:000309943200086

Conference Title: IEEE Statistical Signal Processing Workshop (SSP)

Conference Date: AUG 05-08, 2012

Conference Location: Ann Arbor, MI

Conference Sponsors: IEEE

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Miyoshi, Seiji	C-7406-2013	
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISBN: 978-1-4673-0183-1

Record 15 of 50

Title: Generating dual beams from a single steerable parametric loudspeaker

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu); Gan, WS (Gan, Woon-Seng)

Source: APPLIED ACOUSTICS **Volume:** 99 **Pages:** 43-50 **DOI:** 10.1016/j.apacoust.2015.05.004 **Published:** DEC 1 2015

Abstract: The parametric loudspeaker utilizes an ultrasonic transducer array to transmit a directional sound beam in air based on the parametric array effect. In recent studies, phased array techniques have been applied to achieve controllable directivity patterns or to change the direction of the sound beam. Such a parametric loudspeaker is often referred to as a steerable parametric loudspeaker. In this paper, a dual beam generation method is elaborated. It aims to transmit two sound beams from just one steerable parametric loudspeaker. The two sound beams carries the same audio content to different locations. This dual beam generation method is compatible with the configuration of existing steerable parametric loudspeakers based on phased array techniques. As an algorithm solution, the dual beam generation method readily improves the flexibility of the steerable parametric loudspeaker. (C) 2015 Elsevier Ltd. All rights reserved.

Accession Number: WOS:000358969200005

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Gan, Woon-Seng	A-5151-2011	0000-0002-7143-1823
Kajikawa, Yoshinobu		0000-0002-1735-2518
Shi, Chuang		0000-0002-1517-2775

ISSN: 0003-682X

eISSN: 1872-910X

Record 16 of 50

Title: ADAPTIVE FEEDBACK ANC SYSTEM USING VIRTUAL MICROPHONES

Author(s): Miyazaki, N (Miyazaki, Nobuhiro); Kajikawa, Y (Kajikawa, Yoshinobu)

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:** 383-387 **Published:** 2013

Abstract: In this paper, we apply a virtual sensing technique to a head-mounted ANC system we have already proposed. Adaptive feedback ANC system can reduce noise with periodicity or having narrow band components. However, since quiet zones are formed only at the locations of error microphones, an adequate noise reduction cannot be achieved at the locations where error microphones cannot be placed. In the head-mounted ANC system, the error microphones are placed near the opening of the ear canals. However, the intended locations at which the maximum noise reduction should be achieved are near eardrums. Hence, we try to introduce a virtual sensing technique to a head-mounted ANC system to achieve higher noise reduction. Experimental results demonstrate that the proposed system can achieve higher noise reduction than the conventional system. Moreover, a subjective assessment result demonstrates that the proposed system can also give higher noise reduction to human auditory.

Accession Number: WOS:000329611500078

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:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1520-6149

ISBN: 978-1-4799-0356-6

Record 17 of 50

Title: HEAD-MOUNTED ACTIVE NOISE CONTROL SYSTEM FOR MR NOISE

Author(s): Kida, M (Kida, Masahiro); Hirayama, R (Hirayama, Ryotaro); Kajikawa, Y (Kajikawa, Yoshinobu); Tani, T (Tani, Toru); Kurumi, Y (Kurumi, Yoshimasa)

Book Group Author(s): IEEE

Source: 2009 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, VOLS 1- 8, PROCEEDINGS **Book Series:** International Conference on Acoustics Speech and Signal Processing ICASSP **Pages:** 245-+ **Published:** 2009

Abstract: Recently, magnetic resonance imaging (MRI) devices are used in many medical institutions on the grounds of safety and convenience. An open-configuration MR system is introduced at Shiga University of Medical Science in order to conduct microwave coagulation therapy by using near-real-time MR images. However, this system has a fatal defect. When MRI device works to take images, it also generates serious noises (MR noise). Hence, an operator and other medical staff (ex. nurses and anesthetists) suffer from MR noise and cannot communicate with each other during the operation. In this paper, we therefore propose a head-mounted ANC system in order to reduce the MR noise, and sonic experimental results demonstrate cancellation performance of the system implemented by digital signal processor (DSP).

Accession Number: WOS:000268919200062

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

Conference Date: APR 19-24, 2009

Conference Location: Taipei, TAIWAN

Conference Sponsors: IEEE, IEEE Signal Proc Soc

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1520-6149

ISBN: 978-1-4244-2353-8

Record 18 of 50

Title: Volterra model of the parametric array loudspeaker operating at ultrasonic frequencies

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA **Volume:** 140 **Issue:** 5 **Pages:** 3643-3650 **DOI:** 10.1121/1.4966962 **Published:** NOV 2016

Abstract: The parametric array loudspeaker (PAL) is an application of the parametric acoustic array in air, which can be applied to transmit a narrow audio beam from an ultrasonic emitter. However, nonlinear distortion is very perceptible in the audio beam. Modulation methods to reduce the nonlinear distortion are available for on-axis far-field applications. For other applications, preprocessing techniques are wanting. In order to develop a preprocessing technique with general applicability to a wide range of operating conditions, the Volterra filter is investigated as a nonlinear model of the PAL in this paper. Limitations of the standard audio-to-audio Volterra filter are elaborated. An improved ultrasound-to-ultrasound Volterra filter is proposed and empirically demonstrated to be a more generic Volterra model of the PAL. (C) 2016 Acoustical Society of America.

Accession Number: WOS:000391707700032
PubMed ID: 27908058
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 0001-4966
eISSN: 1520-8524

Record 19 of 50

Title: ON ADAPTIVE SELECTION OF ESTIMATION BANDWIDTH FOR ANALYSIS OF LOCALLY STATIONARY MULTIVARIATE PROCESSES

Author(s): Niedzwiecki, M (Niedzwiecki, Maciej); Ciolek, M (Ciolek, Marcin); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2016 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING PROCEEDINGS **Book Series:** International Conference on Acoustics Speech and Signal Processing ICASSP **Pages:** 4860-4864 **Published:** 2016

Abstract: When estimating the correlation/spectral structure of a locally stationary process, one should choose the so-called estimation bandwidth, related to the effective width of the local analysis window. The choice should comply with the degree of signal nonstationarity. Too small bandwidth may result in an excessive estimation bias, while too large bandwidth may cause excessive estimation variance. The paper presents a novel method of adaptive bandwidth selection. The proposed approach is based on minimization of the cross-validatory performance measure for a local vector autoregressive signal model and, unlike the currently available methods, does not require assignment of any user-dependent decision thresholds.

Accession Number: WOS:000388373405002
Conference Title: IEEE International Conference on Acoustics, Speech, and Signal Processing
Conference Date: MAR 20-25, 2016
Conference Location: Shanghai, PEOPLES R CHINA
Conference Sponsors: Inst Elect & Elect Engineers, Inst Elect & Elect Engineers Signal Proc Soc

Author Identifiers:

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Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1520-6149
ISBN: 978-1-4799-9988-0

Record 20 of 50

Title: Third-Order Nonlinear IIR Filter for Compensating Nonlinear Distortions of Loudspeaker Systems

Author(s): Iwai, K (Iwai, Kenta); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS COMMUNICATIONS AND COMPUTER SCIENCES **Volume:** E98A **Issue:** 3 **Pages:** 820-832 **DOI:** 10.1587/transfun.E98.A.820 **Published:** MAR 2015

Abstract: In this paper, we propose a 3rd-order nonlinear IIR filter for compensating nonlinear distortions of loudspeaker systems. Nonlinear distortions are common around the lowest resonance frequency for electrodynamic loudspeaker systems. One interesting approach to compensating nonlinear distortions is to employ a mirror filter. The mirror filter is derived from the nonlinear differential equation for loudspeaker systems. The nonlinear parameters of a loudspeaker system, which include the force factor, stiffness, and so forth, depend on the displacement of the diaphragm. The conventional filter structure, which is called the 2nd-order nonlinear IIR filter that originates the mirror filter, cannot reduce nonlinear distortions at high frequencies because it does not take into account the nonlinearity of the self-inductance of loudspeaker systems. To deal with this problem, the proposed filter takes into account the nonlinearity of the self-inductance and has a 3rd-order nonlinear IIR filter structure. Hence, this filter can reduce nonlinear distortions at high frequencies while maintaining a lower computational complexity than that of a Volterra filter-based compensator. Experimental results demonstrate that the proposed filter outperforms the conventional filter by more than 2 dB for 2nd-order nonlinear distortions at high frequencies.

Accession Number: WOS:000359464800003

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 1745-1337

Record 21 of 50

Title: MULTI-CHANNEL ANC SYSTEM USING OPTIMIZED REFERENCE MICROPHONES BASED ON TIME DIFFERENCE OF ARRIVAL

Author(s): Hase, S (Hase, Satoru); Kajikawa, Y (Kajikawa, Yoshinobu); Liu, LC (Liu, Lichuan); Kuo, SM (Kuo, Sen M.)

Book Group Author(s): IEEE

Source: 2015 23RD EUROPEAN SIGNAL PROCESSING CONFERENCE (EUSIPCO) **Book Series:** European Signal Processing Conference **Pages:** 305-309 **Published:** 2015

Abstract: Feedforward active noise control (ANC) system using upstream reference signal can reduce various noises such as broadband noise by arranging a reference microphone close to a noise source. However, the performance of ANC system deteriorates if the noise environment such as the arrival direction is changed. This is because of the causality constraint that the unwanted noise propagates to the control point faster than the "antinoise" to cancel the unwanted noise. To solve this problem, we propose an ANC system that estimates the arrival direction of noise using multiple reference microphones placed around the control point, This system uses a tune difference of arrival technique to estimate noise source location and then optimize reference signal. Noise reduction performances are examined through some simulations in this paper.

Accession Number: WOS:000377943800062

Conference Title: 23rd European Signal Processing Conference (EUSIPCO)
Conference Date: AUG 31-SEP 04, 2015
Conference Location: Nice, FRANCE
Conference Sponsors: EURECOM
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 2076-1465
ISBN: 978-0-9928-6263-3

Record 22 of 50
Title: IDENTIFICATION OF THE PARAMETRIC ARRAY LOUDSPEAKER WITH A VOLTERRA FILTER USING THE SPARSE NLMS ALGORITHM
Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)
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:** 3372-3376 **Published:** 2015
Abstract: Volterra filters can be applied to a wide range of nonlinear systems, keeping only the low order kernels to yield a good approximation. The parametric array loudspeaker (PAL), as a weak nonlinear acoustic system, is an attractive directional sound reproduction device. Volterra filters have been adopted in the linearization system of the PAL that efficiently reduces the nonlinear distortion with no need of solving the nonlinear acoustic equation. In this paper, the ultrasound-to-ultrasound Volterra filter is proposed, being inspired by the nonlinear acoustic principle, to provide a better systematic representation of the PAL. Experiment results are presented to prove the effectiveness of the proposed approach, where the sparse NLMS algorithm is carried out in the identification.
Accession Number: WOS:000427402903097
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
Shi, Chuang		0000-0002-1517-2775
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1520-6149
ISBN: 978-1-4673-6997-8

Record 23 of 50
Title: Linearization method based on multiple loudspeaker systems
Author(s): Kajikawa, Y (Kajikawa, Yoshinobu)
Source: ACOUSTICAL SCIENCE AND TECHNOLOGY **Volume:** 32 **Issue:** 5 **Pages:** 220-223 **DOI:** 10.1250/ast.32.220 **Published:** 2011
Accession Number: WOS:000445549100008
Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1346-3969
eISSN: 1347-5177

Record 24 of 50
Title: Acoustic echo cancellation using sub-adaptive filter
Author(s): Ohta, S (Ohta, Satoshi); Kajikawa, Y (Kajikawa, Yoshinobu); Nomura, Y (Nomura, Yasuo)
Book Group Author(s): IEEE
Source: 2007 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING, VOL I, PTS 1-3, PROCEEDINGS **Book Series:** International Conference on Acoustics Speech and Signal Processing (ICASSP) **Pages:** 85-88 **Published:** 2007
Abstract: In this paper, we propose an acoustic echo cancellation (AEC) using a sub-adaptive filter. In the AEC, the step-size parameter of the adaptive filter must be varied according to the situations where a double talk and an echo path change occur. The proposed AEC can appropriately control the step-size parameter even if the double talk and the echo path change simultaneously occur because the optimal step-size parameter can be obtained according to the output of the sub-adaptive filter and the echo path change detector is controlled through the double talk detector. Hence, the proposed AEC can realize superior convergence property to the conventional one. Simulation results demonstrate that the proposed AEC can achieve higher ERLE and faster convergence than the conventional one.
Accession Number: WOS:000249040000022
Conference Title: 32nd IEEE International Conference on Acoustics, Speech and Signal Processing
Conference Date: APR 15-20, 2007
Conference Location: Honolulu, HI
Conference Sponsors: IEEE Signal Proc Soc

ISSN: 1520-6149
ISBN: 978-1-4244-0728-6

Record 25 of 50

Title: Effect of the ultrasonic emitter on the distortion performance of the parametric array loudspeaker

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: APPLIED ACOUSTICS **Volume:** 112 **Pages:** 108-115 **DOI:** 10.1016/j.apacoust.2016.05.013 **Published:** NOV 2016

Abstract: The parametric array loudspeaker (PAL) is a type of directional loudspeaker that utilizes the nonlinear acoustic effect to create the audible sound in an ultrasonic beam. Due to this unusual sound principle, it is inevitable that nonlinear distortion is incurred in the sound transmission of the PAL. Numerous modulation methods aiming to reduce the nonlinear distortion have been developed on the basis of the Berktay's far-field solution, but they often perform in an unexpected manner. The degraded practical performance has been credited to the inaccuracy of the Berktay's far-field solution. In this paper, we demonstrate the effect of the ultrasonic emitter on the distortion performance of the PAL and suggest that the Berktay's far-field solution remains to be a good model equation. (C) 2016 Elsevier Ltd. All rights reserved.

Accession Number: WOS:000380416900012

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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Shi, Chuang		0000-0002-1517-2775

ISSN: 0003-682X
eISSN: 1872-910X

Record 26 of 50

Title: AUTOMATIC GAIN CONTROL FOR PARAMETRIC ARRAY LOUDSPEAKERS

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2016 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING PROCEEDINGS **Book Series:** International Conference on Acoustics Speech and Signal Processing ICASSP **Pages:** 589-593 **Published:** 2016

Abstract: Parametric array loudspeaker (PAL) modulates the audio input on an ultrasonic carrier and relies on airborne nonlinear acoustic effects to generate the audible sound output. The sound output is mainly confined in the beam of the ultrasonic carrier and thus shows a pronounced directivity. There are three parameters that together influence the output volume of a PAL. They are the input level, modulation index, and ultrasound level. In existing PALs, the volume knob is associated with the ultrasound level, while the modulation index is either fixed in the circuit or rarely adjustable by another knob. In this paper, an automatic gain control is proposed to improve the sound quality of the PAL by minimizing the modulation index, maintaining the output-to-input ratio, and ensuring the ultrasound level within the safety range. Simulation and measurement results validate that the proposed approach leads to a reduction in the average total harmonic distortion (THD) level by more than one third for all the tested modulation methods.

Accession Number: WOS:000388373400119

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

Conference Date: MAR 20-25, 2016

Conference Location: Shanghai, PEOPLES R CHINA

Conference Sponsors: Inst Elect & Elect Engineers, Inst Elect & Elect Engineers Signal Proc Soc

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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Shi, Chuang		0000-0002-1517-2775

ISSN: 1520-6149
ISBN: 978-1-4799-9988-0

Record 27 of 50

Title: SYNTHESIS OF VOLTERRA FILTERS FOR THE PARAMETRIC ARRAY LOUDSPEAKER

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2016 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH AND SIGNAL PROCESSING PROCEEDINGS **Book Series:** International Conference on Acoustics Speech and Signal Processing ICASSP **Pages:** 4229-4233 **Published:** 2016

Abstract: The ultrasound-to-ultrasound Volterra filter (U2VF), which was previously proposed to represent the nonlinear response of the parametric array loudspeaker (PAL), identifies the PAL as a nonlinear system that uses ultrasonic signals as its input. It has been proven that the U2VF is a more generic model as compared to the audio-to-audio Volterra filter (A2VF), when the modulation method is adaptive or the input is time varying. However, there is no explicit solution to a linearization system based on the U2VF. Therefore, this paper proposes a synthesis method to extract A2VFs from the U2VF based on the parallel cascade structure. The mature technique of building a linearization system based on the A2VF can hence be adapted to consort with the U2VF.

Accession Number: WOS:000388373404075

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

Conference Date: MAR 20-25, 2016

Conference Location: Shanghai, PEOPLES R CHINA

Conference Sponsors: Inst Elect & Elect Engineers, Inst Elect & Elect Engineers Signal Proc Soc

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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Shi, Chuang		0000-0002-1517-2775

ISSN: 1520-6149

ISBN: 978-1-4799-9988-0

Record 28 of 50

Title: A LINEARIZATION SYSTEM FOR PARAMETRIC ARRAY LOUDSPEAKERS USING THE PARALLEL CASCADE VOLTERRA FILTER

Author(s): Hatano, Y (Hatano, Yuta); Shi, C (Shi, Chuang); Kinoshita, S (Kinoshita, Satoshi); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2015 23RD EUROPEAN SIGNAL PROCESSING CONFERENCE (EUSIPCO) **Book Series:** European Signal Processing Conference **Pages:** 1068-1072 **Published:** 2015

Abstract: The parametric array loudspeaker (PAL) is well known for its ability to radiate a narrow sound beam from a relatively small ultrasonic emitter. Nonlinear distortions commonly occur in the self-demodulated sound of the PAL. Based on the Volterra filter modeling the self-demodulation process of the PAL, a linearization system can be developed for the PAL. However, the computational complexity of the Volterra filter increases dramatically with the tap length. In this paper, the parallel cascade structure is adopted to implement the Volterra filter. The experiment results demonstrate that the computational complexity of the Volterra filter is significantly reduced by using the parallel cascade structure, and based on such an implementation of the Volterra filter, the performance of the linearization system is not compromised.

Accession Number: WOS:000377943800215

Conference Title: 23rd European Signal Processing Conference (EUSIPCO)

Conference Date: AUG 31-SEP 04, 2015

Conference Location: Nice, FRANCE

Conference Sponsors: EURECOM

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Shi, Chuang		0000-0002-1517-2775
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 2076-1465

ISBN: 978-0-9928-6263-3

Record 29 of 50

Title: Multi-channel Feedforward ANC System Combined with Noise Source Separation

Author(s): Kinoshita, S (Kinoshita, Satoshi); Kajikawa, Y (Kajikawa, Yoshinobu)

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:** 379-383 **Published:** 2015

Abstract: In this paper, we examine the effectiveness of Multi-channel feedforward active noise control (ANC) system combined with noise source separation through some simulations. Multi-channel feedforward ANC system can reduce various noise such as the broad-band noise by arranging the reference microphones close to noise sources. However, the performance of ANC system deteriorates when the reference microphones cannot be arranged close to noise sources because each microphone catches undesired noise in addition to desired noise. The proposed system separates noises by using microphone arrays as reference microphones and the separated signals are used as the reference signals of the ANC system. In simulation results, the proposed system can improve the noise reduction performance in case where the ANC system uses two reference microphones and the reference microphones do not need to be arranged close to the noise sources.

Accession Number: WOS:000382954100077

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 30 of 50

Title: ESTIMATION OF NONSTATIONARY HARMONIC SIGNALS AND ITS APPLICATION TO ACTIVE CONTROL OF MRI NOISE

Author(s): Niedzwiecki, M (Niedzwiecki, Maciej); Meller, M (Meller, Michal); Kajikawa, Y (Kajikawa, Yoshinobu); Lukwinski, D (Lukwinski, Dawid)

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:** 5661-5665 **Published:** 2013

Abstract: A new adaptive comb filtering algorithm, capable of tracking the fundamental frequency and amplitudes of different frequency components of a nonstationary harmonic signal embedded in white measurement noise, is proposed. Frequency tracking characteristics of the new scheme are studied analytically, proving (under Gaussian assumptions and optimal tuning) its statistical efficiency for quasi-linear frequency changes. Laboratory tests show that the proposed algorithm can be successfully used for active control of MRI noise.

Accession Number: WOS:000329611505166

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:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1520-6149

ISBN: 978-1-4799-0356-6

Record 31 of 50

Title: An Active Noise Control System Using DXHS Algorithm for MR Noise

Author(s): Kumamoto, M (Kumamoto, Masafumi); Kida, M (Kida, Masahiro); Hirayama, R (Hirayama, Ryotaro); Kajikawa, Y (Kajikawa, Yoshinobu); Tani, T (Tani, Toru); Kurumi, Y (Kurumi, Yoshimasa)

Book Group Author(s): IEEE

Source: 2009 INTERNATIONAL SYMPOSIUM ON INTELLIGENT SIGNAL PROCESSING AND COMMUNICATION SYSTEMS (ISPACS 2009) **Pages:** 69-+ **DOI:** 10.1109/ISPACS.2009.5383900 **Published:** 2009

Abstract: Recently, magnetic resonance imaging (MRI) devices are used in many medical institutions on the grounds of safety and convenience. An open-configuration MR system is introduced at Shiga University of Medical Science in order to conduct microwave coagulation therapy by using near-real-time MR images. However, this system has a fatal defect. The MRI device generates serious acoustic noises (MR noise) during operation. Hence, surgeons and other medical staff (e.g. nurses and anesthetists) suffer from MR noise and cannot communicate with each other during operation. In this paper, therefore, we study an active noise control (ANC) system using delayed-x harmonics synthesizer (DXHS) algorithm in order to reduce the MR noise. Some simulation results demonstrate the effectiveness of the system.

Accession Number: WOS:000289915800018

Conference Title: International Symposium on Intelligent Signal Processing and Communication Systems

Conference Date: DEC 07-09, 2009

Conference Location: Kanazawa, JAPAN

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISBN: 978-1-4244-5015-2

Record 32 of 50

Title: Acoustic echo cancellation using sub-adaptive filter

Author(s): Ohta, S (Ohta, Satoshi); Kajikawa, Y (Kajikawa, Yoshinobu); Nomura, Y (Nomura, Yasuo)

Source: IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS COMMUNICATIONS AND COMPUTER SCIENCES **Volume:** E91A **Issue:** 4 **Pages:** 1155-1161 **DOI:** 10.1093/ietfec/e91-a.4.1155 **Published:** APR 2008

Abstract: In the acoustic echo canceller (AEC), the step-size parameter of the adaptive filter must be varied according to the situation if double talk occurs and/or the echo path changes. We propose an AEC that uses a sub-adaptive filter. The proposed AEC can control the step-size parameter according to the situation. Moreover, it offers superior convergence compared to the conventional AEC even when the double talk and the echo path change occur simultaneously. Simulations demonstrate that the proposed AEC can achieve higher ERLE and faster convergence than the conventional AEC. The computational complexity of the proposed AEC can be reduced by reducing the number of taps of the sub-adaptive filter.

Accession Number: WOS:000255647700030

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 1745-1337

Record 33 of 50

Title: Improvement of the stability and cancellation performance for the active noise control system using the simultaneous perturbation method

Author(s): Tokoro, Y (Tokoro, Yukinobu); Kajikawa, Y (Kajikawa, Yoshinobu); Nomura, Y (Nomura, Yasuo)

Source: IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS COMMUNICATIONS AND COMPUTER SCIENCES **Volume:** E90A **Issue:** 8 **Pages:** 1555-1563 **DOI:** 10.1093/ietfec/e90-a.8.1555 **Published:** AUG 2007

Abstract: In this paper, we propose the introduction of a frequency domain variable perturbation control and a leaky algorithm to the frequency domain time difference simultaneous perturbation (FDTDSP) method in order to improve the cancellation performance and the stability of the active noise control (ANC) system using the perturbation method. Since the ANC system using the perturbation method does not need the secondary path model, it has an advantage of being able to track the secondary path changes. However, the conventional perturbation method has the problem that the cancellation performance deteriorates over the entire frequency band when the frequency response of the secondary path has dips because the magnitude of the perturbation is controlled in the time domain. Moreover, the stability of this method also deteriorates in consequence of the dips. On the other hand, the proposed method can improve the cancellation performance by providing the appropriate magnitude of the perturbation over the entire frequency band and stabilizing the system operation. The effectiveness of the proposed method is demonstrated through simulation and experimental results.

Accession Number: WOS:000248934200009

Conference Title: 21st Symposium on Signal Processing

Conference Date: NOV 15-17, 2006

Conference Location: Kyoto, JAPAN

Conference Sponsors: IEICE, Signal Proc Tech Grp

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
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ISSN: 1745-1337

Record 34 of 50

Title: An acoustic echo cancellation using subadaptive filter

Author(s): Otani, M (Otani, Masayuki); Kajikawa, Y (Kajikawa, Yoshinobu); Nomura, Y (Nomura, Yasuo)

Source: ELECTRONICS AND COMMUNICATIONS IN JAPAN PART III-FUNDAMENTAL ELECTRONIC SCIENCE **Volume:** 90 **Issue:** 2 **Pages:** 9-21 **DOI:** 10.1002/ecjc.20231 **Published:** 2007

Abstract: In this paper, a method using a subadaptive filter playing the supplementary role of acoustic echo cancellation is discussed. Since the updating state of the adaptive filter can be calculated approximately as a characteristic of the subadaptive filter, the step size parameter can be optimally controlled depending on the estimation accuracy. Also, for prevention of echo path variation, we discuss a variable detector in which the variation of the filter coefficients at the time of variation is considered. The proposed method is found to be robust to channel variations and can improve the ERLE by 10 to 15 dB in comparison with the conventional method. (C) 2006 Wiley Periodicals, Inc. Electron Comm Jpn Pt 3, 90(2): 9-21, 2007;

Accession Number: WOS:000241645200002

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 1042-0967

Record 35 of 50

Title: Multichannel feedforward active noise control system combined with noise source separation by microphone arrays

Author(s): Iwai, K (Iwai, Kenta); Kinoshita, S (Kinoshita, Satoshi); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: JOURNAL OF SOUND AND VIBRATION **Volume:** 453 **Pages:** 151-173 **DOI:** 10.1016/j.jsv.2019.04.016 **Published:** AUG 4 2019

Abstract: In this paper, we propose a multichannel active noise control (ANC) system combined with noise source separation by microphone arrays. A multichannel feedforward ANC system uses multiple reference microphones located close to noise sources and can reduce various acoustic noises such as broadband noise. In general, it is difficult to place the reference microphones close to the noise sources owing to physical constraints. In this case, the reference signals, which are the outputs of reference microphones, are highly correlated with each other, the noise reduction performance of the multichannel ANC system is degraded. In other words, the reference microphones detect the highly correlated noises, reducing the noise reduction ability of the ANC system. To solve this problem, a multichannel feedforward ANC system with a noise source separation technique is proposed. This system uses microphone arrays located at the reference microphone positions. The mixed signal is separated from the target noise and other components by noise source separation and the separated signals are used to update the noise control filters. Simulation results demonstrate that the proposed ANC system can separate the target noise and the disturbance noise and effectively reduce the target noise. (C) 2019 Elsevier Ltd. All rights reserved.

Accession Number: WOS:000467388700009

ISSN: 0022-460X

eISSN: 1095-8568

Record 36 of 50

Title: Statistical-Mechanics Approach to Theoretical Analysis of the FXLMS Algorithm

Author(s): Miyoshi, S (Miyoshi, Seiji); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS COMMUNICATIONS AND COMPUTER SCIENCES **Volume:** E101A **Issue:** 12 **Pages:** 2419-2433 **DOI:** 10.1587/transfun.E101.A.2419 **Published:** DEC 2018

Abstract: We analyze the behaviors of the FXLMS algorithm using a statistical-mechanical method. The cross-correlation between a primary path and an adaptive filter and the autocorrelation of the adaptive filter are treated as macroscopic variables. We obtain simultaneous differential equations that describe the dynamical behaviors of the macroscopic variables under the condition that the tapped-delay line is sufficiently long. The obtained equations are deterministic and closed-form. We analytically solve the equations to obtain the correlations and finally compute the mean-square error. The obtained theory can quantitatively predict the behaviors of computer simulations including the cases of both not only white but also nonwhite reference signals. The theory also gives the upper limit of the step size in the FXLMS algorithm.

Accession Number: WOS:000451763000055

ISSN: 1745-1337

Record 37 of 50

Title: Multichannel Feedforward Active Noise Control System with Optimal Reference Microphone SelectorBased on Time Difference of Arrival

Author(s): Iwai, K (Iwai, Kenta); Hase, S (Hase, Satoru); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: APPLIED SCIENCES-BASEL **Volume:** 8 **Issue:** 11 **Article Number:** 2291 **DOI:** 10.3390/app8112291 **Published:** NOV 2018

Abstract: In this paper, we propose a multichannel active noise control (ANC) system with an optimal reference microphone selector based on the time difference of arrival (TDOA). A multichannel feedforward ANC system using upstream reference signals can reduce various noises such as broadband noise by arranging reference microphones close to noise sources. However, the noise reduction performance of an ANC system degrades when the noise environment changes, such as the arrival direction. This is because some reference microphones do not satisfy the causality constraint that the unwanted noise propagates to the control point faster than the anti-noise used to cancel the unwanted noise. To solve this problem, we propose a multichannel ANC system with an optimal reference microphone selector. This selector chooses the reference microphones that satisfy the causality constraint based on the TDOA. Some experimental results demonstrate that the proposed system can choose the optimal reference microphones and effectively reduce unwanted acoustic noise.

Accession Number: WOS:000451302800281

ISSN: 2076-3417

Record 38 of 50

Title: Modified second-order nonlinear infinite impulse response (IIR) filter for equalizing frequency response and compensating nonlinear distortions of electrodynamic loudspeaker

Author(s): Iwai, K (Iwai, Kenta); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: APPLIED ACOUSTICS **Volume:** 132 **Pages:** 202-209 **DOI:** 10.1016/j.apacoust.2017.11.014 **Published:** MAR 2018

Abstract: In this paper, we propose a modified second-order nonlinear infinite impulse response (IIR) filter for equalizing the frequency response and compensating nonlinear distortions of an electrodynamic loudspeaker. A problem of electrodynamic loudspeakers is the generation of nonlinear distortions, which degrade the sound quality. One of the approaches to reducing nonlinear distortions is to use a second-order nonlinear IIR filter. This filter is based on an equivalent circuit model of an electrodynamic loudspeaker and its coefficients are determined by physical parameters. However, it is difficult to compensate nonlinear distortions when a loudspeaker has a high Q factor at the lowest resonance frequency, at which the displacement of the diaphragm and nonlinear distortions become large. The Q factor determines the linear frequency response of the electrodynamic loudspeaker. Although it is necessary to compensate the Q factor of the loudspeaker, a nonlinear IIR filter cannot compensate the Q factor because it does not have a linear filtering feature. In this paper, we propose a modified second-order nonlinear IIR filter that can not only compensate the nonlinear distortions caused by the nonlinearities of the force factor and stiffness but also equalize the frequency response by employing the linear characteristics of the loudspeaker with the desired Q factor. Experimental results show that the proposed filter can compensate the linear and nonlinear distortions more effectively than a conventional filter.

Accession Number: WOS:000423639000020

ISSN: 0003-682X

eISSN: 1872-910X

Record 39 of 50

Title: Automatic Design Support System for Compact Acoustic Devices Using Deep Neural Network

Author(s): Hirai, K (Hirai, Kai); Nakamura, K (Nakamura, Kai); Kajikawa, Y (Kajikawa, Yoshinobu); Iwai, K (Iwai, Kenta)

Book Group Author(s): IEEE

Source: 2018 IEEE 7TH GLOBAL CONFERENCE ON CONSUMER ELECTRONICS (GCCE 2018) **Book Series:** IEEE Global Conference on Consumer Electronics **Pages:** 652-655 **Published:** 2018

Abstract: An appropriate acoustic structure with a desired frequency response is rarely obtained through the acoustic equivalent circuit analysis in the case of compact acoustic devices. Thus, skilled acoustic engineers must design structures based on their know-how, and the time and cost are increased. Therefore, we propose an automatic design support system for compact acoustic devices introducing deep neural network. In the proposed system, the acoustic characteristics of candidate structures are analyzed by the acoustic FDTD method and an optimal candidate is obtained by learned deep neural network. We demonstrate the effectiveness of the proposed system through some comparisons between desired and designed frequency responses.

Accession Number: WOS:000459859500225

Conference Title: IEEE 7th Global Conference on Consumer Electronics (GCCE)

Conference Date: OCT 09-12, 2018

Conference Location: Nara, JAPAN

Conference Sponsors: IEEE

ISSN: 2378-8143

ISBN: 978-1-5386-6309-7

Record 40 of 50

Title: A PARTIAL-UPDATE MINIMAX ALGORITHM FOR PRACTICAL IMPLEMENTATION OF MULTI-CHANNEL FEEDFORWARD ACTIVE NOISE CONTROL

Author(s): Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2018 16TH INTERNATIONAL WORKSHOP ON ACOUSTIC SIGNAL ENHANCEMENT (IWAENC) **Book Series:** International Workshop on Acoustic Signal Enhancement **Pages:** 11-15 **Published:** 2018

Abstract: In the implementation of a multi-channel feedforward active noise control (ANC) system, the standard FxLMS algorithm demands too much computational power to be real-time processed by a low-cost digital processor. Therefore, various algorithms have been proposed to reduce their computational complexity by modifying the cost function and introducing the partial update (P-U). This paper proposes a new P-U minimax algorithm, which minimizes the uniform-norm instead of the 2-norm of the error signals and updates the filtered reference signals partially. The P-U minimax algorithm is compared with existing algorithms based on an experimental setup of a case (1, 4, 8) ANC system. The proposed P-U minimax algorithm is validated to be efficient and outperform the scanning error and mixed-error algorithms with the same degree of computational complexity.

Accession Number: WOS:000458323900003

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 41 of 50

Title: Automatic Speech Translation System Selecting Target Language by Direction-of-Arrival Information

Author(s): Tsujikawa, M (Tsujikawa, Masanori); Okabe, K (Okabe, Koji); Hanazawa, K (Hanazawa, Ken); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2018 26TH EUROPEAN SIGNAL PROCESSING CONFERENCE (EUSIPCO) **Book Series:** European Signal Processing Conference **Pages:** 2315-2319 **Published:** 2018

Abstract: In this paper, we propose an automatic speech translation system that selects its target language on the basis of the direction-of-arrival (DOA) information. The system uses two microphones to detect speech signals arriving from specific directions. The target language for speech recognition is selected on the basis of the DOA. Both the speech detection and target language selection relieves users from operations normally required for individual utterances, without serious increase in computational costs. In a speech-recognition evaluation of the proposed system, 80% word accuracy was achieved for utterances recorded with two microphones that were 40cm distant from speaker positions. This accuracy is nearly equivalent to that in which the time frame and target language of a user's speech are given in advance.
Accession Number: WOS:000455614900465

Conference Title: European Signal Processing Conference (EUSIPCO)

Conference Date: AUG 03-07, 2018

Conference Location: Rome, ITALY

Conference Sponsors: European Assoc Signal Processing, IEEE Signal Processing Soc, ROMA TRE Univ Degli Studi, MathWorks, Amazon Devices

ISSN: 2076-1465

ISBN: 978-90-827970-1-5

Record 42 of 50

Title: Compensation for Nonlinear Distortion of the Frequency Modulation-Based Parametric Array Loudspeaker

Author(s): Hatano, Y (Hatano, Yuta); Shi, C (Shi, Chuang); Kajikawa, Y (Kajikawa, Yoshinobu)

Source: IEEE-ACM TRANSACTIONS ON AUDIO SPEECH AND LANGUAGE PROCESSING **Volume:** 25 **Issue:** 8 **Pages:** 1709-1717 **DOI:** 10.1109/TASLP.2017.2705280 **Published:** AUG 2017

Abstract: The parametric array loudspeaker (PAL) modulates the audio signal on an ultrasonic carrier. When the modulated signal is transmitted in air, an audio beam is created based on the nonlinear acoustic principle. Each modulation method has advantages and disadvantages. The frequency modulation (FM) is favorable for its low cost and high volume, but the tradeoff is its complicated nonlinear distortion, which is difficult to be reduced. In this paper, the Volterra filter is adopted to model the nonlinearity of the FM-based PAL. A novel complex inverse system is devised to effectively reduce the nonlinear distortion. Three practical aspects are addressed. First, the computational complexity of the Volterra filter is reduced by the parallel cascade structure with almost no compromise to the model accuracy. Second, Volterra filters are identified at discrete input levels to treat the nonlinearity that keeps changing with the time-varying audio input. Third, when the input level is high, a separation approach is proposed to refine the identified Volterra filters, which eventually improves the performance of the proposed inverse system.

Accession Number: WOS:000405376400001

ISSN: 2329-9290

Record 43 of 50

Title: Statistical-Mechanical Analysis of LMS Algorithm for Time-Varying Unknown System

Author(s): Ishibushi, N (Ishibushi, Norihiro); Kajikawa, Y (Kajikawa, Yoshinobu); Miyoshi, S (Miyoshi, Seiji)

Source: JOURNAL OF THE PHYSICAL SOCIETY OF JAPAN **Volume:** 86 **Issue:** 2 **Article Number:** 024803 **DOI:** 10.7566/JPSJ.86.024803 **Published:** FEB 15 2017

Abstract: We analyze the behaviors of the least-mean-square algorithm for a time-varying unknown system using a statistical-mechanical method. Cross-correlations between the elements of a primary path and those of an adaptive filter and autocorrelations of the elements of the adaptive filter are treated as macroscopic variables. We obtain simultaneous differential equations that describe the dynamical behaviors of the macroscopic variables under conditions in which the tapped delay line is sufficiently long. We analytically show the existence of an optimal step size. This result is supporting evidence of Widrow et al.' s pioneering work that clarified the trade-off between the noise misadjustment and the lag misadjustment. Furthermore, we obtain the exact solution of the optimal step size in the case of a white reference signal. The derived theory includes the behaviors for a time-constant unknown system as a special case.

Accession Number: WOS:000393183600017

Author Identifiers:

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Miyoshi, Seiji	C-7406-2013	
Kajikawa, Yoshinobu		0000-0002-1735-2518

ISSN: 0031-9015

Record 44 of 50

Title: A Personal Authentication System Based on Pinna Related Transfer Function

Author(s): Higashiguchi, Y (Higashiguchi, Yutaka); Kajikawa, Y (Kajikawa, Yoshinobu); Kita, S (Kita, Shunsuke)

Book Group Author(s): IEEE

Source: 2017 INTERNATIONAL CONFERENCE ON BIOMETRICS AND KANSEI ENGINEERING (ICBAKE) **Pages:** 123-126 **Published:** 2017

Abstract: In the personal authentication using acoustic properties of ear canal (pinna related transfer function) by previous study, the high authentication rate was obtained in the case of using the earphone and the headphone. On the other hand, in the case of using mobile phone, which does not cover the outer ear or canal, the authentication rate is lower than that of other devices. However, the mobile phone does not require any other equipment for measuring acoustic properties because micro speaker and microphone are built in. This is an important point for the highly convenient authentication system using the pinna related transfer function. In this paper, the authors investigate authentication rate using the pinna related transfer function measured by the mobile phone.

Accession Number: WOS:000425893800025

Conference Title: International Conference on Biometrics and Kansei Engineering (ICBAKE)

Conference Date: SEP 15-17, 2017

Conference Location: Kyoto, JAPAN

ISBN: 978-1-5386-3401-1

Record 45 of 50

Title: Effectiveness of Headrest ANC System with Virtual Sensing Technique for Factory Noise

Author(s): Hirose, S (Hirose, Shun); Kajikawa, Y (Kajikawa, Yoshinobu)

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:** 464-468 **Published:** 2017

Abstract: In this paper, we examine the effectiveness of headrest active noise control (ANC) system with the virtual sensing technique for factory noise. In recent years, the development of various industrial devices has caused acoustic noise problems related to factory machines. Although the noise canceling headphones and earphones can reduce factory noise, but they prevent verbal communications and give uncomfortable feeling, so headrest ANC systems becomes one of candidates for this kind of situations. However, the headrest ANC system has a problem that the zone of quiet (ZoQ) cannot be formed around desired locations (e.g. ears) because the ZoQ is generally formed around error microphones located into the headrest which are far from the desired locations. Therefore, we have developed a headrest ANC system with virtual sensing technique. In this paper, we investigate the effectiveness of the proposed headrest ANC system through real world experiments. As a result, it is found that the proposed headrest ANC system can reduce unwanted factory noise more than 30 dB at the desired locations compared with the conventional system (without virtual sensing technique).

Accession Number: WOS:000425879400083

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 46 of 50

Title: Modification of Second-Order Nonlinear IIR Filter for Compensating Linear and Nonlinear Distortions of Electrodynamic Loudspeaker

Author(s): Iwai, K (Iwai, Kenta); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2017 25TH EUROPEAN SIGNAL PROCESSING CONFERENCE (EUSIPCO) **Book Series:** European Signal Processing Conference **Pages:** 2684-2688 **Published:** 2017

Abstract: In this paper, we propose a modified 2nd-order nonlinear IIR filter for compensation of both linear and nonlinear distortions of electrodynamic loudspeakers. A nonlinear IIR filter is a nonlinear compensator and is based on a physical nonlinear model of electrodynamic loudspeakers. However, it is difficult to compensate nonlinear distortions when a loudspeaker has large sharpness at the lowest resonance frequency, at which the displacement of the diaphragm becomes large. Although it is necessary to compensate the sharpness of the loudspeaker, the nonlinear IIR filter cannot compensate the sharpness because it does not have a linear filtering feature. In this paper, we propose a modified 2nd-order nonlinear IIR filter that can compensate not only the nonlinear distortions but also the sharpness by employing the linear characteristics of the loudspeaker with the desired sharpness. Experimental results show that the proposed filter can compensate the linear and nonlinear distortions more effectively than a conventional filter.

Accession Number: WOS:000426986000542

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

ISBN: 978-0-9928-6267-1

Record 47 of 50

Title: Statistical-mechanical analysis of the FXLMS algorithm for multiple-channel active noise control

Author(s): Murata, T (Murata, Tomoki); Kajikawa, Y (Kajikawa, Yoshinobu); Miyoshi, S (Miyoshi, Seiji)

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:** 136-139 **Published:** 2017

Abstract: We analyze the behaviors of the Filtered-X LMS (FXLMS) algorithm for active noise control (ANC). Correlations between the impulse response of an adaptive filter and a primary path are treated as macroscopic variables. To obtain the correlations, we analytically solve the equations and finally compute the MSE. In particular, we analyze the behaviors of multiple-channel ANC. We theoretically show that the MSE is affected by the secondary paths that are not directly connected.

Accession Number: WOS:000425879400023

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

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Miyoshi, Seiji	C-7406-2013	

ISSN: 2309-9402

ISBN: 978-1-5386-1542-3

Record 48 of 50

Title: Theoretical Analysis of LMS Algorithm for Time-Varying Unknown System

Author(s): Ishibushi, N (Ishibushi, Norihiro); Kajikawa, Y (Kajikawa, Yoshinobu); Miyoshi, S (Miyoshi, Seiji)

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: We analyze the behaviors of the LMS algorithm for a time-varying unknown system using a statistical-mechanical method. We obtain simultaneous differential equations that describe the dynamical behaviors of the macroscopic variables under conditions in which the tapped-delay line is sufficiently long. We show the existence of an optimal step size owing to the trade-off between the noise misadjustment and the lag misadjustment. Furthermore, we obtain the exact optimal step size in the case of a white reference signal. The derived theory includes the behaviors for a time-constant unknown system as a special case.

Accession Number: WOS:000393591800199

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

Author Identifiers:

Author	Web of Science ResearcherID	ORCID Number
Miyoshi, Seiji	C-7406-2013	

ISSN: 2309-9402

ISBN: 978-988-14768-2-1

Record 49 of 50

Title: Integrated Direct Sub-band Adaptive Volterra Filter and Its Application to Identification of Loudspeaker Nonlinearity

Author(s): Kinoshita, S (Kinoshita, Satoshi); Kajikawa, Y (Kajikawa, Yoshinobu)

Book Group Author(s): IEEE

Source: 2016 24TH EUROPEAN SIGNAL PROCESSING CONFERENCE (EUSIPCO) **Book Series:** European Signal Processing Conference **Pages:** 11-15 **Published:** 2016

Abstract: In this paper, we propose a novel realization of sub-band adaptive Volterra filter, which consists of input signal transformation block and only one adaptive Volterra filter. The proposed realization can focus on major frequency band, in which a target nonlinear system has dominant components, by changing the number of taps in each sub-band in order to simultaneously realize high computational efficiency and high identification performance. The proposed realization of subband adaptive Volterra filter is applied to the identification of electro-dynamic loudspeaker systems and the effectiveness is demonstrated through some simulations. Simulation results show that the proposed realization can significantly improve the estimation accuracy.

Accession Number: WOS:000391891900003

Conference Title: 24th European Signal Processing Conference (EUSIPCO)

Conference Date: AUG 28-SEP 02, 2016

Conference Location: Budapest, HUNGARY

Conference Sponsors: European Assoc Signal Proc

Author Identifiers:

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ISSN: 2076-1465

ISBN: 978-0-9928-6265-7

Record 50 of 50

Title: Active Noise Control Systems with Simplified Period Aware Linear Prediction Method for MR Noise

Author(s): Sawano, H (Sawano, Hitoshi); Kajikawa, Y (Kajikawa, Yoshinobu)

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 examine the effectiveness of an active noise control (ANC) system with period aware linear prediction (PALP) method and simplified PALP (SPALP) method for MR noise. The PALP method expanding ordinary linear prediction method using delayed signal has high prediction accuracy for periodic signal. We have attempted to apply the ANC system with the PALP method for reducing MR noise which has high periodicity. However PALP method requires two adaptive filters and delay device. Thus the PALP method requires large memory size and computational complexity. For this reason, we simplify the PALP method by removing one adaptive filter with undelayed signal. Simulation results demonstrate that the feedback ANC system with the SPALP method has the same noise reduction performance as the PALP method while saving the computational complexity.

Accession Number: WOS:000393591800153

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

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