Real time evaluation of soft microphones on a local active noise control system

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Systems for the Active Control of acoustic Noise (ANC) rely on measurements of the noise signal in order to produce the required "quiet zone". Soft Microphones (SM) are a virtual microphone alternative already proposed by the authors, employing estimates rather than measurements of the noise signal obtained from room impulse responses based on measurements. Therefore, a single microphone can be used to estimate the noise signal at multiple points. This has already been successfully applied on single microphone noise mapping. In the present work, the SM method is evaluated on a real time local ANC system. The proposed scheme is a Filtered-X Least Mean Square (FXLMS) based control structure, incorporating SM. Following the proposed system design-simulation with Matlab-Simulink, a three-dimensional enclosure (office-simulating) and a real electroacoustic ANC system based on a TI® TMS320C6713 DSP, developed for this purpose, were built in the laboratory to carry out the experiments. Different options of estimating the SM signals and exploiting them into the control structure are experimentally evaluated as to their ability to enlarge the silence zone and their noise reduction performance around the listener ear zone (using a Head and Torso Simulator) and compared to the standard FXLMS solution.

1 Introduction

The production of an acoustic signal of equal amplitude but of opposite phase (180^o phase difference) to the acoustic noise signal at a specific area is the basis of what is known as active noise control (or cancellation), ANC, and the respective area is called "quiet zone". The produced "anti-noise" signal is destructively superimposed to the existing noise signal and the outcome is a reduced noise level at the quiet zone area. Several scientists are dealing with this topic the last years since there is an increased interest though out the world on noise reduction as a health and quality of living parameter.

One of the most interesting ideas on ANC is the use of virtual sensors [1-9] to achieve noise reduction at a location different from the location where the error microphone is physically positioned. This means that quiet zone could be formed at a specific point without the need to have a real microphone there.

The Soft Microphones (SM) are a virtual microphone alternative already proposed by the authors [10-11], using estimates of the transfer functions of acoustical paths within a room, to estimate the signal arriving to the soft microphones positions given the signal measured at the real microphone position. The acoustical paths estimated are those connecting the real microphone to each soft microphone, i.e. supposing an ideal source at real microphone's position and an ideal sensor at each soft microphone's position.

The estimation of the above mentioned acoustical paths is originally proposed to be done by real electroacoustic system measurements of the room impulse response (RIR) between each pair of positions, equalizing the response of the electroacoustic system used for the measurements by deconvolution. The SM idea has already been successfully applied on single microphone noise mapping [10], proving the effectiveness in the frequency domain (octave-band noise level measurements/ mapping). However, for a time-domain real-time process like ANC with SM the direct deconvolution method used in the above application is not adequately accurate.

The deconvolution itself is a serious theoretical and practical problem. The creation of the inverse filter involved with the deconvolution process is not an easy task for a mixed-phase impulse response, like that of a loudspeaker and thus of an electroacoustic measurement system, although it was addressed by many authors [12-20].

Although there are many methods that could be used, including a recently proposed by the authors [21], in the present work the estimations of all the involved acoustical paths have been done by simulating RIR [22], while the involved electroacoustical impulse responses (secondary paths) have been produced by convolution of the measured impulse of the used electroacoustic system anechoic response with the simulated acoustical paths. Impulse response measurements have been done using the MLS method [23]. This way a perfect deconvolution process has been supposed.

In the present work, the main objective is to evaluate the SM method on a real time local ANC system, concentrating to the idea of the estimation of the soft microphone signal by convolving the real microphone signal with the acoustical path from the real microphone to the soft microphone. The control architecture used is a single-channel control feed-forward system based on the popular Filtered-X Least Mean Square (FXLMS). Different options of exploiting the SM signals into the control structure, deviating from the standard FXLMS, are experimentally evaluated as to their ability to enlarge the silence zone and their noise reduction performance around the listener ear zone and compared to the standard FXLMS solution.

The paper is organized as follows: The control structure for both the off-line simulation and the real time implementation are presented in section 2. Section 3 provides details on the experimental setup and section 4 presents the results of the simulations and the measurements. Finally, the conclusions and future research topics are given in section 5.

2 Local ANC structure

The standard FXLMS ANC structure [24] is that of Fig. 1, where one reference microphone, one anti-noise speaker and one error microphone are used. The critical difference from an LMS ANC structure is the use of a pre-filtering of the LMS reference signal using an estimate of the secondary path.

Supposing that two soft microphone positions are considered, having RIR between the position of the real error microphone and their position R_1 and R_2 respectively, the same ANC structure is represented by Fig. 2, where black lines and schemes denote the real error microphone related parts, while the grey ones denote the soft microphone related parts.

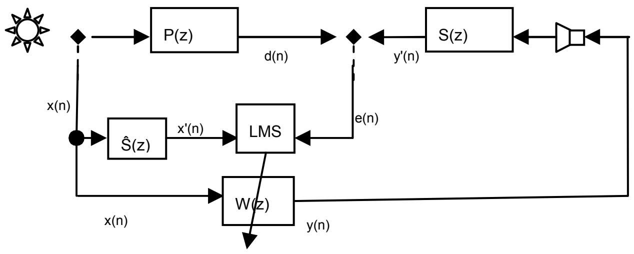


Fig.1 Standard FXLMS.

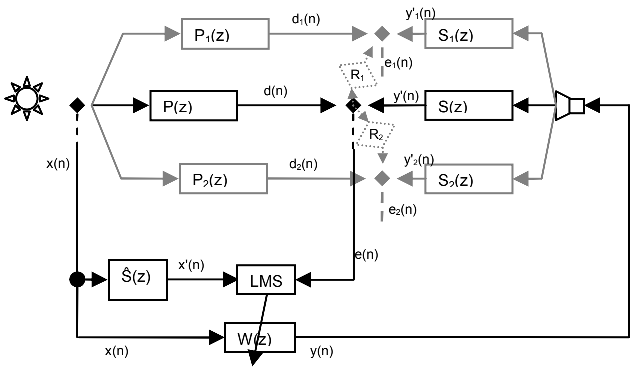


Fig.2 Standard FXLMS with two soft microphones depicted.

The P(z), S(z), and Ŝ(z) are the primary path, the secondary path and the estimate of the secondary path for the real error microphone position, while the $P_i(z)$, $S_i(z)$, and $\hat{S}_i(z)$, are the respective plants/filters for the i-th soft microphone. The same convention holds for the noise reaching the microphones d(n), $d_i(n)$, the anti-noise reaching the microphones y'(n), $y'_i(n)$, and the resulting error signals e(n), $e_i(n)$.

Following the soft microphones idea, we assume that the primary and secondary paths for the soft microphone positions are generally unknown and only the acoustical paths connecting them to the real error microphone, R_i, are known and available either for off-line simulation purposes or for real time calculations. The error signal e(n) is the measured one but the error signals $e_i(n)$ are estimated and their estimates are available for processing.

In order to evaluate the use of soft microphones in local ANC applications, four ANC structure cases are considered and for each one of them the error microphone and soft microphones signals are simulated off-line and measured for the real-time implementation.

Case 1: The standard single-channel FXLMS is considered, where the LMS reference signal is pre-filtered using an estimate of the secondary path to the real error microphone, Ŝ(z), and the cost function used is the mean squared error (acoustic pressure signal) of the real error microphone.

Case 2: A deviation from the standard single-channel FXLMS is considered, where the LMS reference signal is pre-filtered using an estimate of the secondary path to the real error microphone, Ŝ(z), and the cost function used is the mean squared error (acoustic pressure signal)of one of the soft microphones (the i-th).

Case 3: A deviation from the standard single-channel FXLMS is considered, where the LMS reference signal is pre-filtered using an estimate of the secondary path to the i-th error microphone, $\hat{S}_i(z)$, and the cost function used is the mean squared error (acoustic pressure signal) of this soft microphone.

Case 4: A deviation from the standard single-channel FXLMS is considered, where the LMS reference signal is pre-filtered using an estimate of the secondary path to the real error microphone, Ŝ(z), and the cost function used is the sum of the mean squared errors (acoustic pressure signals) of the real and all the soft microphones.

It has to be noted that for the real-time implementation the feedback of the anti-noise signal to the reference microphone has been considered and for the neutralization of this effect the control structure used is using an estimate of the feedback path [24] as illustrated in Fig. 3.

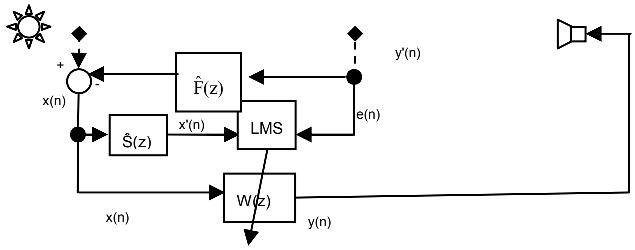


Fig.3 Real-time FXLMS ANC structure.

3 Experimental setup

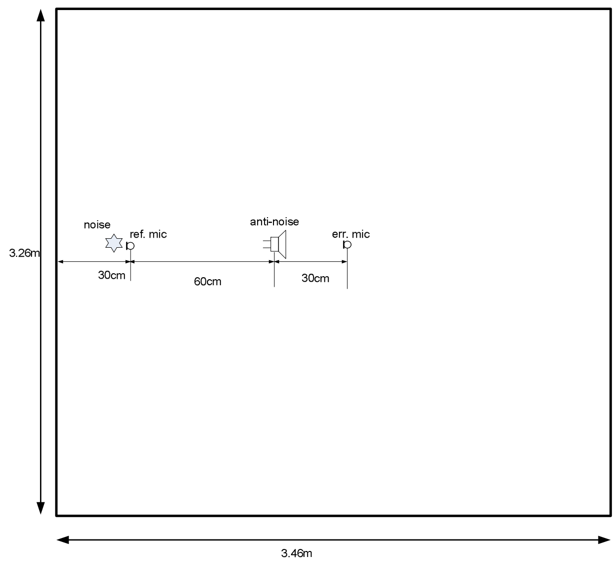


Fig.4 Experimental setup, room dimensions and positioning of the ANC system elements inside the room. All elements have been considered to be at 1m height, room height was 2.86m.

The application environment for the local ANC has been considered to be that of a three-dimensional enclosure, a small rectangular room corresponding to the acoustic conditions found in a regular office. Fig. 4 shows the dimensions of the room and the relative positioning of the ANC system elements inside the room. Five soft microphones (SM) have been considered, two of them between the anti-noise speaker and the error microphone and three of them after the error microphone. SM 1 is the one closer to the anti-noise speaker and SM 5 the most distant one. In cases 2 and 3, of section 2, only SM 5 is used within the ANC structure, while the other soft microphones are only used to estimate (during simulation) or measure (in real-time implementation the noise at their positions. In case 4, all soft microphones are used within the ANC structure.

The experiment could be divided into two phases. The first phase is the simulation phase and the second is the real-time implementation phase.

3.1 Simulation (off-line implementation)

During the simulation phase, the four cases discussed in section 2 were simulated first in Matlab and then in Simulink. The Simulink simulations were considerably faster and provided the basis for the real-time implementation. However, it has to be noted that the Simulink simulation proved to be a more difficult task during the model building phase, since several of the available Simulink blocks presented specific functional particularities. The previous implementation of the ANC system in Matlab proved to be very helpful in resolving Simulink problems. All the necessary acoustical paths for the simulation phase were simulated based on the mirror image source method [22] and the electroacoustical paths (secondary paths) by convolving the corresponding acoustical with the measured anechoic impulse response of the used anti-noise speaker.

3.2 Real-time implementation

The real-time implementation is based on the software development kit (SDK6713) of the TI® TMS320C6713 DSP, which is fully interoperable with Simulink. Starting from the off-line Simulink models, the parts simulating the acoustical and electroacoustical parts were removed and the real-time models were ported to the SDK. Further to the TI hardware (which include ADC and DACs), a two channel power amplifier was used, two microphones with their associated preamplifiers and low-pass filters. The noises were reproduced from one loudspeaker and the anti-noise from another loudspeaker. The power amplifier was driving both loudspeakers. Fig. 5 depicts the HW configuration.

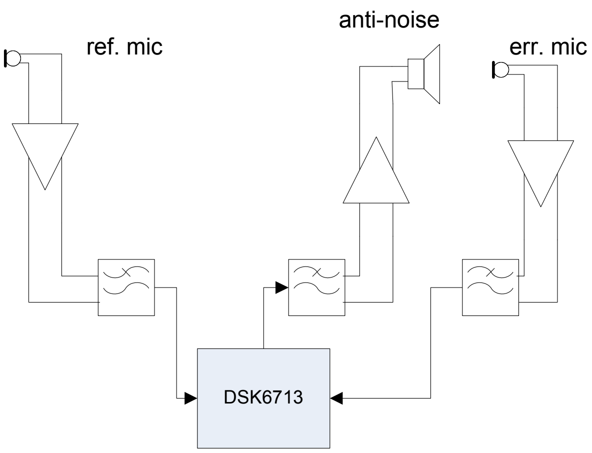
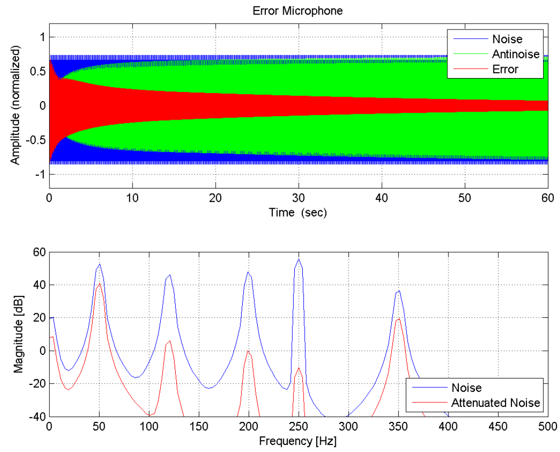


Fig.5 HW configuration for the real-time experiment.

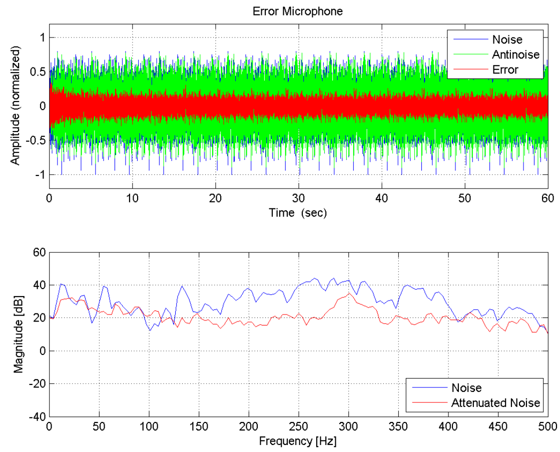
4 Simulation and real-time results

Each one of the four cases presented in section 2 has been simulated and real-time implemented for tree types of band-limited noises, white noise up to 400Hz, a sinus based signal (sum of 5 sinus signals of 50Hz, 120Hz, 200Hz, 250Hz, and 350Hz frequencies and different amplitudes, 0.125, 0.25, 0.125, 0.25, and 0.125 correspondingly) and a real recorded signal from a textile machine (speed-frame). Simulated results presented slightly better performance compared to the measured ones, since during the design phase the acoustic and electroacoustic paths used were simulation-based.

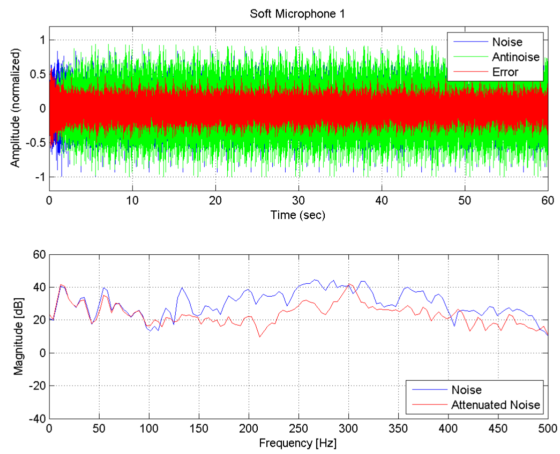
The measured results are presented in the following Fig. 6. Fig. 6(1a) and Fig. 6(1b) show the performance of the standard FXLMS (case 1 of section 2) for the sinus based signal and the real recorded speed-frame (S-F) noise.



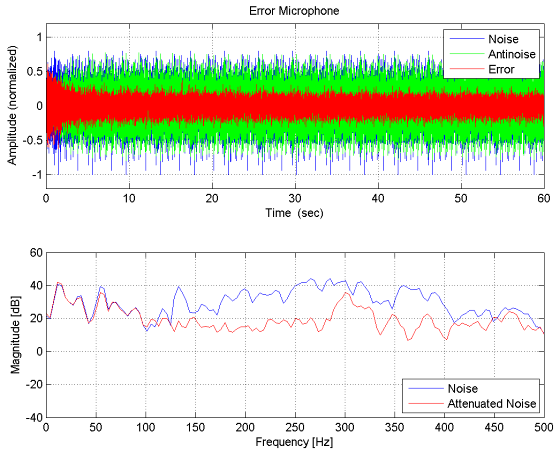
(1a)



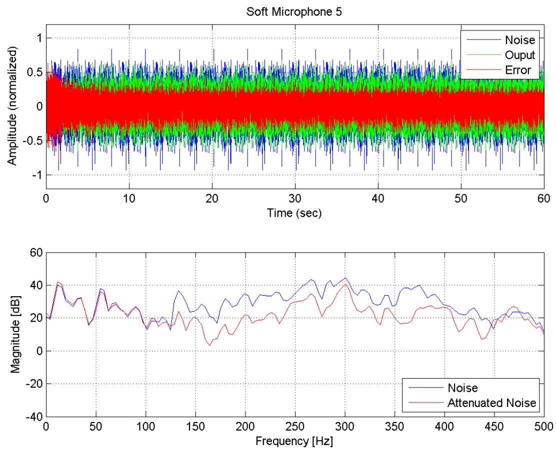
(1b)



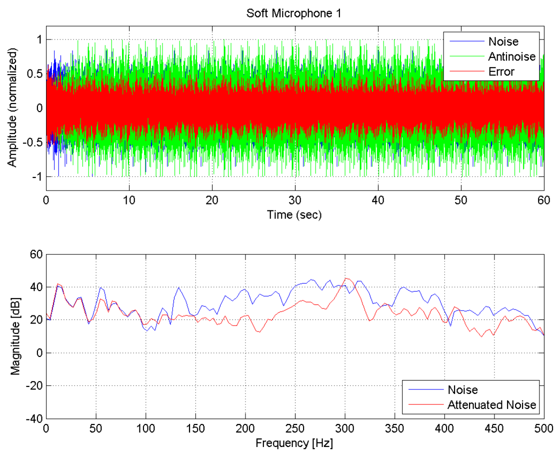
(2a)



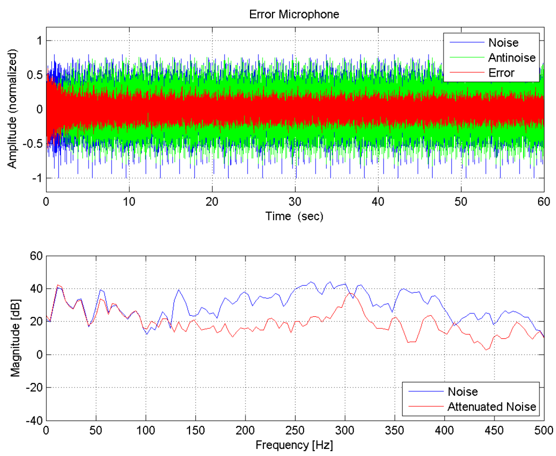
(2b)



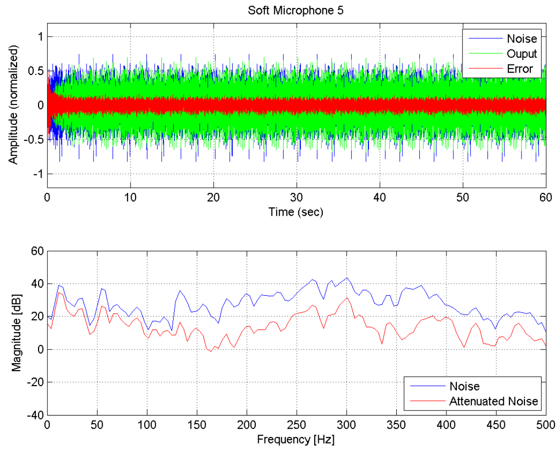
(2c)



(3a)



(3b)



(3c)

Fig.6 Noise reduction measurements of: (1a) case 1 for sinus signals noise, (1b) case 1 for speed-frame (S-F) noise, (2a) case 2 for S-F noise at SM 1, (2b) case 2 for S-F noise at error mic., (2c) case 2 for S-F noise at SM 5, (3a) case 3 for S-F noise at SM 1, (3b) case 3 for S-F noise at error mic., (3c) case 3 for S-F noise at SM 5.

From Fig. 6(2a), 6(2b), 6(2c) is evident that the deviation tested with case 2, although moving the quiet zone towards SM 5, the best noise attenuation is still achieved for the position of the real error microphone. From Fig. 6(3a), 6(3b), 6(3c) is evident that the deviation tested with case 3, moves the quiet zone towards SM 5 and the best noise attenuation is achieved for the specific SM position.

Case 4 results, using the sum of the mean squared errors of the real and all the soft microphones as cost function, presented maximum noise reduction at the real error microphone position and the enlargement of the quiet zone was not obvious. These results were quite similar to the ones of case 1 (standard FXLMS).

5 Conclusion

The use of the soft microphone idea to a single-channel ANC system by slightly deviating from the standard FXLMS, i.e. by using an estimate of the secondary path to the virtual microphone for the pre-filtering of the LMS reference signal, provides significant noise attenuation at the soft microphone's position, for a single soft microphone, and moves the quiet zone from the real microphone position to the soft microphone position. Both off-line estimated and real-time measured results prove this. The relevant deviation of the measured results compared to the simulated is mainly due to that the acoustic and electroacoustic paths were simulated and not measured and estimated.
For a single channel system with one anti-noise speaker and one real microphone, the enlargement of the quiet zone seems to be difficult, but may be achieved with the selection of an appropriate cost function in a future research. The main limiting factor seems to be the existence of just one anti-noise speaker. Future research should also be focused on the use of multiple soft microphones in more complex control structures with multiple anti-noise speakers. The need for multiple real error microphones has also to be investigated.

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References

[1] S. J. Elliott, A. David, "A virtual microphone arrangement for local active sound control", *Proceedings of the 1st International Conference on Motion and Vibration Control*, Yokohama, Japan, 1027-1031 (1992)

[2] J. Garcia-Bonito, S. J. Elliott, C. C. Boucher, "Avirtual microphone arrangement in a practical active headrest", *Proceedings of InterNoise 96*, 1115-1120 (1996)

[3] J. Garcia-Bonito, S. J. Elliott, C. C. Boucher, "Generation of zones of quiet using a virtual microphone arrangement", *J. Acoust. Soc. Am.*, Vol. 101, No. 6, 3498-3516 (1997)

[4] A. Roure, A. Albarrazin. "The remote microphone technique for active noise control", *Proceedings of Active'99*, Fort Lauderdale, USA, 1233-1244 (1999)

[5] S. R. Popovich, "Active acoustic control in remote regions", *US Patent No. 5,701,350* (1997)

[6] B. S, Cazzolato, "An adaptive LMS virtual microphone", *Proceedings of Active'02*, ISVR, Southampton, UK, 105-116 (2002)

[7] J. M. Munn, B. S. Cazzolato, C. D. Kestell, C. H. Hansen, ''Virtual error sensing for active noise control in a one-dimensional waveguide: Performance prediction versus measurement,'' *J. Acoust. Soc. Am.*, Vol. 113, 35–38 (2003)

[8] J. Yuan, "Virtual sensing for broadband noise control in a lightly damped enclosure", *J. Acoust. Soc. Am.*, Vol. 116, No. 2, 934-941 (2004)

[9] C. D. Petersen, R. Fraanje, B. S, Cazzolato, A. C. Zander, C. H. Hansen, "A Kalman filter approach to virtual sensing for active noise control", *Mechanical Systems and Signal Processing*, 22 (2), 490-508 (2008)

[10] M. Rangoussi, S. M. Potirakis, D. Lolos, "Single Microphone Noise Mapping", *Proceedings of Euronoise 2006*, Tampere-Finland (2006)

[11] S. M. Potirakis and M. Rangoussi, "A system development method for global active noise control", *Proceedings of IWSSIP'2005*, Chalkida, Greece, 375-379 (2005)

[12] J.N. Mourjopoulos, "Digital Equalization of Room Acoustics", *J. Audio Eng. Soc., JAES,* Vol. 42, No. 11, 884-900 (1994)

[13] A. Farina, E. Ugolotti, "Spatial equalisation of sound systems in cars by digital inverse filtering", *Proceedings of IOA Conference "Reproduced Sound 13"*, Hydro Hotel, Bowness on Windermere, GB, (1997)

[14] W. Ser, P. Wang, M. Zhang, "Loudspeaker Equalization with Post-Processing", *EURASIP Journal on Applied Signal Processing*, Vol. 2002, Issue 1, 1296–1300 (2002)

[15] S.T. Neely, J.B. Allen, "Invertibility of a room impulse response", *J. Acoust. Soc. Am.*, Vol. 66, 165-169 (1979)

[16] J. Mourjopoulos and al., "A comparative study of least squares and homomorphic techniques for the inversion of mixed phase signals", *Proceedings of the IEEE International Conference on Speech, and Signal Processing (ICASSP '82)*, vol: 7, 1858- 1861 (1982)

[17] S. Fontana, "Deconvolution and Applications to Binaural Technologies", *PhD Thesis*, Nov. 2007, Italo-French University (UIF@ University of Turin)

[18] Y. Lin, D. D. Lee, "Bayesian regularization and non-negative deconvolution for room impulse response estimation" *IEEE Trans. On Signal Processing*, Vol. 54, No. 3, 839- 847 (2006)

[19] E. A. O'Sullivan, C. F. N. Cowan, "Harmonic Inversion of Room Impulse Pesponse", *7th International Conf. on Mathematics in Signal Processing, the Institute of Mathematics and Its Applications* (2006)

[20] T. Collins, "A non-linear technique for room impulse response estimation", *Sixth International Conf. on Digital Audio Effects* (DAFX-03), Queen Mary University of London (2003).

[21] I. Paraskevas, M. Rangoussi, S.M. Potirakis, S. Savvaidis, "Phase spectral processing for improved time-domain soft microphone based noise estimation", *Proceedings of the Acoustics '08, Session: NS00/1,* Paris, France, (2008)

[22] J. B. Allen and D. A. Berkley, "Image method for efficiently simulating small-room acoustics," *J. Acoust. Soc. Am.*, vol. 65, no. 4, 943–950 (1979)

[23] D. Rife, J. Vanderkooy, "Transfer-Function Measurement with Maximum-Length Sequences", *J. Audio Eng. Soc., JAES,* Vol. 6, 419–443 (1989)

[24] S. M. Kuo, D. R. Morgan, "Active noise control: a tutorial review", *Proceedings of the IEEE*, Vol. 87, No. 6, 943-973 (1999)

[25] W-K Tseng, Y-P Chang "A Virtual Microphone Feedback Control System for Active Noise Control", *Proceedings of the IEEE 3rd International Conference on Mechatronics*, *ICM 2006*, 170-175 (2006)

[26] C. D. Kestell, B. S. Cazzolato, C. H. Hansen, "Active noise control in a free field with virtual sensors", *J. Acoust. Soc. Am.*, Vol. 109, No. 1, 232-243 (2001)

[27] C. H. Hansen, X. Qiu, C. Petersen, C. Q. Howard, S. Singh, "Active noise and vibration control system design considerations", *Mechanics*, 26 (2), 37-46 (2007)

[28] J. Diaz, J. M. Eganã, J. Viñolas, ."A local active noise control system based on a virtual-microphone technique for raiway sleeping vehicle applications", *Mechanical Systems and Signal Processing*, Vol. 20, 2259-2276 (2006)

[29] Y. Haneda, "Active noise control with a virtual microphone based on common-acoustical-pole and residue model", *Proceedings of the IEEE International Conference on Acoustics, Speech, and Signal Processing, (ICASSP '02)*, 1877-1880 (2002)