

ACOUSTIC CANCELLATION OF ENGINE NOISE BY FAST ADAPTIVE IIR FILTERING

E. Masgrau, J.A. Rodríguez-Fonollosa

Dpto. Teoría de la Señal y Comunicaciones. U.P.C.
E.T.S.I. Telecomunicación. Apdo. 30.002
08080 Barcelona, Spain

ABSTRACT

Digital control of acoustic noise is an application area of digital signal processing with increasingly interest along the last years. This work tackles the reduction of the noise inside a motorcar cabin using a loudspeakers array driven by fast adaptive filtering of a reference noise source. Firstly, we propose an adaptive method for the deconvolution of the, in general, non-minimum phase radiation path by using an additional filter. Secondly, we propose the use of some fast adaptive IIR lattice algorithms. They are very adequate for the very long impulse response of the acoustic system. Finally, a real scene is tested and the obtained results are analyzed.

INTRODUCTION

Passive industrial silencers work like acoustical low pass filters in noisy environments and, therefore, they work badly at low frequencies. Thus, digital acoustic noise cancellation has centred high interest and attention in the last years. Other applications related with acoustic cancellation, such as acoustic equalization of rooms, are object of broad analysis at present. The cabin of a motorcar represents a common scene for interesting and promising application of the referred acoustical processing.

In the figure 1 is shown a Widrow's cancellation scheme applied to an acoustic scene, i.e., a cabin car. The reference noise source is the motor of the car. It generates the primary noise inside the cabin across a complex acoustic and mechanic path. It can be modeled in a very simplified way with a pole-zero transference function $G(z)$. The reference noise is used to feed one or several adaptive filters $W_m(z)$ in order to obtain an estimation of the primary noise. This estimation is then radiated in opposed-phase by the secondary sources or loudspeakers and the primary noise cancellation is obtained.

* This work has been supported by PRONTIC Grant nº 105/88

In the figure 2 a monochannel signal model of the acoustic system is shown. An additional

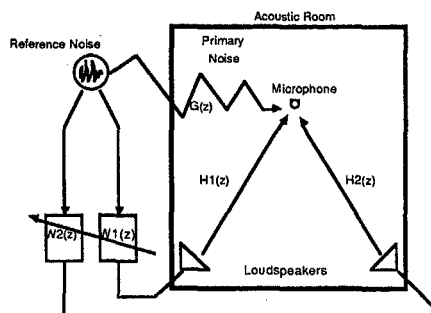


Fig. 1. Cancellation scheme

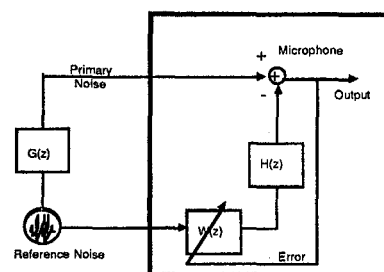


Fig. 2. Signal model.

microphone records the cancellation residual noise and it drives the adaptive filter adaptation. The theoretical optimum transfer function $W(z)$ is $G(z)/H(z)$, where $G(z)$ is the above referred noise path model and $H(z)$ is the radiation path model of the secondary source. This latter includes the loudspeaker response and the antialiasing and the interpolator filters response corresponding to A/D and D/A converters in the secondary paths. Generally, a good approximation of $W(z)$ can be obtained with a FIR filter, depending on the zeroes location of $H(z)$.

Unfortunately, the acoustic radiation path $H(z)$ usually has nonminimum phase character, and therefore, the optimum $W(z)$ filter becomes unstable. A possible solution consists in the use of an additional loudspeaker or secondary source [1], such as is shown in figure 1. In this case, the inverse filtering is carried out by the following equation:

$$H_1(z) W_1(z) + H_2(z) W_2(z) = G(z) \quad (1)$$

where $H_1(z)$ and $H_2(z)$ must be relatively prime in the unit circle of z -plane. That is, they do not have any common zero outside or on the unit circle. On the contrary, the above problem is reproduced. Therefore, a good location selection for the secondary sources is a key factor in this context. The $G(z)$ acoustic model presents, in general, a pole-zero structure and, in this case, a long impulse response for $W_m(z)$ is required; therefore, it is expected that the choice of IIR filters will be more adequate. Unfortunately, the typical IIR adaptive algorithms present a slow speed of convergence due to the multimodal error surface, and they can become unstable in the adaptation process.

In this paper, we use some fast IIR lattice algorithms developed by the authors [2,3]. These algorithms work very better than the classical IIR algorithms, specially in nearly instability conditions, that is, when the magnitude of some lattice parcor coefficient is near to the unity.

In the steady-state, the converged $W_m(z)$ adaptive filters and its corresponding radiation paths $H_1(z)$ can be exchanged. Thus, the filter

inputs are the filtered reference noises across the $H_1(z)$ paths, named $r_{1m}(n)$. The covariance matrix of this filtered signals defines the optimum Wiener solution of the $W_m(z)$ [4]. As it is well known, the eigenvalues spread of this matrix determines in part the convergence properties of the least squares adaptive algorithms. However, this statistic does not describe properly the convergence of the adaptive algorithms. It is only true if the convergence time-scale of these algorithms is very slower than the introduced delays of the radiation paths. This convergence dependence on the magnitude delay will be shown clearly later. In the adaptive algorithms, the correction terms of the coefficients update are determined by the filtered data $r_{1m}(n)$ instead of by its input signals [4].

MULTICHANNEL-MULTIERROR ADAPTIVE ALGORITHMS

The adaptive algorithms considered in this work are the multichannel-multierror extensions of the corresponding scalar algorithms, and they are described in detail in reference [5]. We consider a transversal FIR algorithms, the Normalized LMS, and several lattice IIR algorithms.

The multichannel-multierror LMS has been studied in reference [4]. The NLMS here used introduces the known normalization of the step-size by a variance estimation of the filtered data $r_{1m}(n)$.

The lattice IIR algorithms are the multichannel extensions of the scalar ones and they are described in [2,3,6]. They present a improved convergence with respect to the IIR direct form due to the known uncoupling properties of the lattice structure. Also, they make use of a corrected gradient term based in the inverse Hessian matrix. Many lattice IIR algorithms can be defined according to the accurate of the Hessian matrix calculation [6].

RESULTS

The empirical performance analysis of the proposed algorithms has been carried out in a simulated environment of a real scene. It consists in a cancellation system like that of the figure 1 working with real signals obtained in a motorcar

cabin operating in a anecoic room. Only the radiation paths $H_1(z)$ have been simulated. The primary and reference noises were recorded in an analogic tape, low pass filtered to 1,7 KHz and sampled to 4 KHz. Later, they were decimated to 500 Hz sampling frequency and the band pass signal was up to 230 Hz. It is the frequency range of the second order boom noise of the motor, the more harmful component of the cabin noise (typical frequencies range of 70-200 Hz corresponding to engine speeds between 2100 rpm and 6000 rpm).

In-phase motor and cabin signals were recorded to 2000, 3000, 4000, 5000 and 6000 rpm. After the digital sampling and processing above mentioned, they were used in the simulated cancellation scheme.

As a preliminar analysis, four cases have been considered. Three monochannel-monoerror cases with radiation paths $H(z)=1, z^{-5}$ and $(1+4z^{-5})$, and a bichannel-monoerror case with radiation paths $H_1(z)=z^{-5}$ and $H_2(z)=1+4z^{-5}$ have been analyzed. The first case is the ideal null delay. It determines the best cancellation performance. The rest of the cases present a nonminimum phase character.

For the second order cancellation analysis we have filtered selectively the non-cancelled (primary) noise and the cancelled cabin noise by means a running 32-points FFT. Then, the convergence time and the steady-state cancellation ratio were measured over the FFT output corresponding to the second order frequency. Also, the global cancellation in the full band (0-230 Hz) was calculated.

The convergence parameters of the algorithms were chosen to provide the same (roughly)

convergence speed and a good cancellation ratios. As a preliminar performance analysis, the obtained results for 3000 rpm using the NLMS FIR and a lattice IIR algorithms are exposed in the Table I. The convergence time (T_C) in samples, the second order cancellation ratio (RSO) in dB and the full band cancellation (RB) in dB are shown. These cancellation ratios are calculated over 2500 samples beginning from the sample 500. The lattice IIR algorithm uses the LAR (Log Area Ratio) coefficients and a simplified gradient term where the Hessian matrix is diagonal (LAR-DH) [6]. The FIR order is 10 and the IIR order is 5 (5 zeroes and 5 poles).

In the Figure 3 is shown the second order energy evolution of the cabin (primary) noise and the cancelled cabin noise in the convergence period for the LAR-DH IIR lattice algorithm (case 4 of the Table I). In the Figure 4 is shown the steady-state spectra of the both primary and the cancelled noise for the same case. It can be observed the high second order cancellation ratio and the good tracking of another high noise component.

From the Table I the following facts can be remarked: the NLMS algorithm tracks very well the second order component of the noise, specially in the two first case; however, it provides a low cancellation ratio over the full band. The LAR-DH IIR lattice algorithm provides high second order cancellations and it improves the NLMS full band cancellation in 1,5-2 dB for the nonminimum phase $H(z)$ cases. Also, the use of an additional adaptive filter in the nonminimum phase cases is shown very efficient in both NLMS and lattice-IIR algorithms. In both cases, the full band noise cancellation ratio is improved in 1,5-2 dB.

	T_C	RSO	RB	T_C	RSO	RB	T_C	RSO	RB	T_C	RSO	RB
FIR	50	20	6	100	14	3	50	16	3,5	50	16	4,5
NLMS												
IIR	50	18	7	100	12	3	50	18	5	50	18	6,5
LATTICE												
	$H(z)=1$			$H(z)=z^{-5}$			$H(z)=1+4z^{-5}$			$H_1(z)=z^{-5}, H_2(z)=1+4z^{-5}$		

Table I. Cancellation performance for NLMS FIR and LAR-DH IIR lattice algorithm.

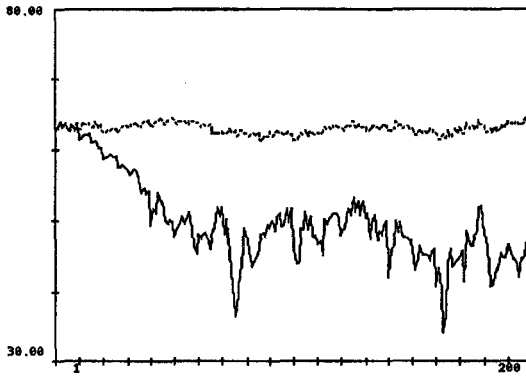


Figure 3.- Second order energy evolution of the primary noise (dashed line) and the cancelled primary noise (solid line) for the LAR-DH IIR lattice algorithm.

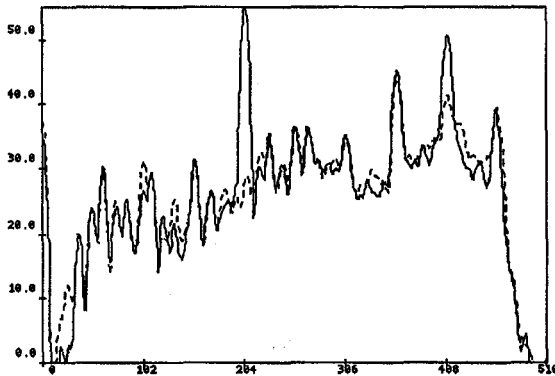


Figure 4.- Steady state spectra of both the primary noise (solid line) and the cancelled primary noise (dashed line) for the same case shown in figure 3.

CONCLUSIONS

In this paper, the acoustic noise reduction inside a motorcar cabin is tackled. We propose to use an additional adaptive filter to the nonminimum phase radiation path deconvolution. Also, we introduce some fast adaptive IIR lattice algorithms to improve the acoustic model tracking. Preliminary results have shown some large improvements in the noise cancellation using the proposed techniques. The additional filter provides an improvement of about 2 dB in the full band cancellation. The IIR lattice filter increases 1,5-2 dB the full band cancellation. By using both techniques we observe a improvement

of about 3 dB. Also, the second order cancellation is improved in 2 dB in this case.

Acknowledgements

The authors acknowledge the assistant of the R. Leyva in obtaining the experimental results and the collaboration of the Laboratorio de Acústica of Seat-Volkswagen at Martorell (Spain) in getting the real signals.

REFERENCES

- [1] M. Miyoshi, Y. Kaneda. "Inverse filtering of room acoustics". Trans. on ASSP, Vol. 36, pp. 145-152, February 1988.
- [2] J.A. Rodríguez-Fonollosa, E. Masgrau. "A new algorithm for adaptive IIR filtering based on the log-area-ratio parameters". Proc. EUSIPCO-90, Barcelona (Spain), September 1990.
- [3] J.A. Rodríguez-Fonollosa, E. Masgrau. "Spectral sensitivity and convergence rate in adaptive IIR filtering". IEEE Proc. ISCAS'90, pp. 1959-1962, New Orleans (USA), May 1990.
- [4] S.J. Elliot et al. "A multiple error LMS algorithm and its application to the active control of sound and vibration". Trans. on ASSP, Vol. 35, No. 10, pp. 1423-1434, October 1987.
- [5] E. Masgrau, J.A. Rodríguez-Fonollosa. "Multichannel-multierror adaptive algorithms for acoustic control noise". To be published.
- [6] J.J. Shynk. "On Lattice-form algorithms for adaptive IIR filtering". IEEE Proc. ICASSP'88, pp. 1547-1557. New York (USA), 1988.