

ACTIVE NOISE SYSTEMS FOR REDUCING OUTDOOR NOISE

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An ANC system prototype designed for stationary noise control linked to traditional noise barriers was initially developed by authors in the past years. The encouraging results initially obtained have shown that ANC systems are feasible and should be further investigated to improve their performances. Nevertheless, FXLMS algorithm, considered in the first prototype architecture, is known to be affected by convergence problems that generally arise when a system fails to properly identify the noise to be cancelled; this process may (and often does) yield an instable system, where the control algorithm tries to catch-up its own sound, increasing the overall sound pressure level on the targets as in “avalanche” effect, potentially leading to damage fragile instrumentation. Hence, instability is unacceptable in practical applications and must be avoided or prevented. With this respect, in this manuscript some more robust alternative ANC algorithms have been investigated and experimentally verified.

Keywords: active noise control, stability.

1. Introduction

Active noise control (ANC) techniques are nowadays becoming more and more refined and reached a high level of maturity that allowed integration in modern acoustic devices, mainly targeted to hearing aids, headphones and propagation of noise in ducts [1, 2]. Only in recent years applications considering open field scenarios have been developed in practice (open spaces, ambient noise propagation [3, 4]), being such situations affected by weather phenomena, randomly moving sources and, circumstantially, time-varying emission spectrum. ANC techniques exhibit the attractive property to be effectively complementary to passive solutions for low-medium frequency range (20-500Hz). Indeed, the relatively long period of oscillation of low frequencies relaxes the real-time requirements for the control system. The main idea behind the scene is that sound can cancel sound. Specifically, given the annoying acoustic pressure field in a space position, an ANC system generates an identical field having opposite phase by means of acoustic one or more acoustic transducers (loudspeakers), in order to obtain a null acoustic pressure. The potential of ANC techniques make them very attractive to solve practical situations where traditional interventions are not adequate. Typical examples are represented by man-made disturbances, such as sounds generated by manufacturing plants, vehicle transit, etc. The interest at international level towards these issues is also witnessed by European efforts in funding research projects and feasibility studies aiming at increasing, for instance, acoustical comfort in urban and peri-urban areas [5, 6].

In the last years, efforts in the development of ANC systems have been made by authors investigating and implementing specific ANC systems aiming to tackle stationary noise [7, 8] and pulse noise [9, 10].

Mainly referring to the stationary noise, adaptive ANC algorithms are the most attractive ones because they continuously carry out such an adjustment and can catch up a time-varying noise. The most notable example of adaptive algorithm for ANC systems is the Filtered-X Least Mean Square (FXLMS) [11], which has been deeply investigated in the literature [12] and applied in practical scenarios [8]. FXLMS algorithm has been shown to be particularly effective in case of stationary and quasi-stationary noise, that is, when the time scale of the noise variations is greater than the adaptation time of the system; variants of the algorithm have been also proposed for the abatement of noise following particular statistical distributions that lack of second order moments [13-16]. Nevertheless, FXLMS algorithm is known to be affected by convergence problems [1] that generally arise when a system fails to properly identify the noise to be cancelled; this process may (and often does) yield an instable system, where the control algorithm tries to catch-up its own sound, increasing the overall sound pressure level on the targets as in “avalanche” effect, potentially leading to damage fragile instrumentation. Hence, instability is unacceptable in practical applications and must be avoided or prevented. With this respect, some more robust alternative ANC algorithms have been proposed in the literature [1, 11, 12], by considering an approach inspired to FXLMS algorithm and enforcing a better control over the convergence of the system.

To the best of authors’ knowledge, only simulation-based evaluations between FXLMS algorithm and its robust variants have been presented in the literature so far. In this manuscript, a comparison is proposed by considering the case of single channel ANC systems in a real outdoor scenario, considering both pure tone and multi tone stationary noise. The paper is structured as follows: in Section 2, the basis of FXLMS algorithm is summarized; the description of the robust alternatives is presented in Section 3; in Section 4 the experimental results obtained by the implemented ANC systems are illustrated and, finally, the conclusions are reported in Section 5.

2. Filtered-X Least Mean Square Algorithm

The FXLMS algorithm belongs to the class of feed-forward control systems [1]. This class of controlling systems relies on three components: “reference” microphones, control sources (loudspeakers) and “error” microphones. In Fig. 1 the basic scheme of a single channel feed-forward system is depicted. The first microphone, named “reference”, is placed after the noise source at a distance $d_{r,c}$ from the rear of the control source and it is used to acquire the incoming noise that has to be cancelled. The other microphone, named “error”, is placed in front of the cone of the control source at a distance $d_{c,e}$, in the point where the acoustic cancellation is measured. The control algorithm pilots the output of the control source by processing the signal at the reference microphone and measuring the effects on the error microphone. The control setups in many applications and show good performances for narrow-band low frequency (< 500 Hz) noise signals.

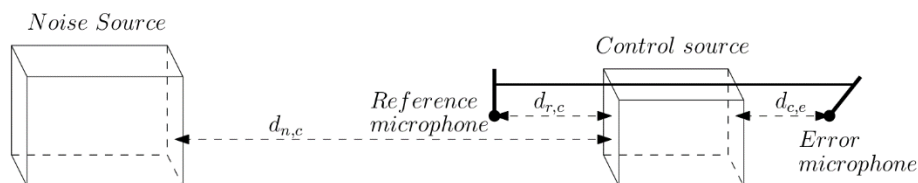


Figure 1: Reference setup for the single channel ANC device. $d_{n,c}$ is the distance between the noise source and the rear of the control source; $d_{r,c}$ and $d_{c,e}$ are respectively the distances of the reference microphone and the error microphone from the nearest side of the control source.

The previous scheme can be generalized to a multichannel scenario, by considering an assembly of properly spaced multiple single channel ANC devices that cooperate in order to create a broader cancellation front [8]. The distance between control sources is usually set in order to be not greater than half of the shortest wavelength of interest [17].

The description of a system based on the FXLMS algorithm is provided in the following considering the general multichannel case. Let us denote nC as the total number of control sources, nR the total number of reference microphones and nE the total number of error microphones. Even if a natural extension of the single channel system would imply $nC=nR=nE$, the following argumentation it is generally valid for arbitrary $nC \neq nR \neq nE$. Let's define and indexing of the listed elements, being c the index of the control sources, r the index of reference microphones and e the index of error microphones. In Fig. 2 the general block diagram of the FxLMS algorithm is reported.

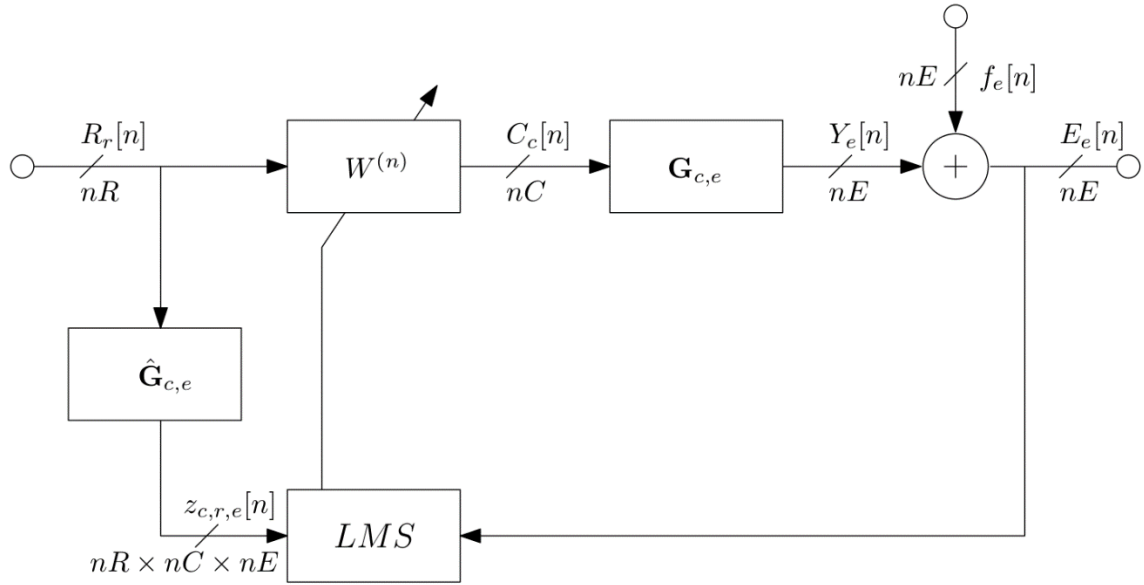


Figure 2: Block diagram for the FxLMS control algorithm.

The control block $W^{(n)}$ represents the finite impulse response (FIR) of the FxLMS filter of order I at discrete time step n . For each n , the block takes as an input all the reference signals $R[n]$ from the reference microphones and produces the signals $C[n]$ to be sent to the control sources. The filter coefficients are periodically updated by the adaptive algorithm in order to minimize the acoustic pressure level $E[n]$ measured on error sensors. Both E and R are used in order to compute the new coefficients, and requires an estimate of all the *secondary paths* [11], that is, all the impulse response $G_{c,e}$ between the c -th control source and the e -th error microphone, for each combination of these elements. In the scheme, $\hat{G}_{c,e}$ is a FIR filter of order K and is an estimate of the actual $G_{c,e}$.

Let $R_r[n]$ be the reference signal at time n measured on the r -th reference microphone; accordingly, let us define $C_c[n]$ as the control signal for the c -th source and $E_e[n]$ the error signal on the e -th error microphone. The error signal that has to be minimized is the sum of the noise $f_e[n]$ coming from the noise source and the signal coming from the control sources. The block $W^{(n)}$ contains I coefficients for each combination of reference microphone and control source, defined as $w^{(n)}[i]$ (with $i \in [0, I-1]$).

In the above framework, the signal $C_c[n]$ is obtained through the following procedure:

$$C_c[n] = \sum_{r=1}^{nR} \sum_{i=0}^{I-1} w_{c,r}^{(n)}[i] R_r[n-i]. \quad (1)$$

The updating of the coefficients $W^{(n)}$ is ruled by the following expression [18]:

$$w_{c,r}^{(n+1)}[i] = w_{c,r}^{(n)}[i] + \mu \sum_{e=1}^{nE} z_{c,r,e}[n-i]E_e[n]. \quad (2)$$

In Eq. 2 $\mu \in \mathbb{R}^+$ is a multiplicative coefficient commonly called *step* in the literature. The coefficients $W^{(n)}$ depend on μ , so this parameter is critical for convergence and stability. There exist methods to automatically compute, even in an adaptive manner, the coefficient μ ; in Section 3 we provide alternative algorithms that exploit this opportunity in order to improve both convergence and stability.

In Eq. (3) the elements $z_{c,r,e}[n]$ represent the signal $R_r[n]$ modified by the secondary path impulse response $\hat{G}_{c,e}$ and it is defined by:

$$z_{c,r,e}[n] = \sum_{k=0}^{K-1} \hat{G}_{c,e}[k] R_r[n-k], \quad (3)$$

with $G_{c,e}[k]$ being the k -th coefficient of the FIR filter of order K .

3. Improving Convergence and Stability

In case of optimal estimation of the secondary path, it has been proved [11] that stability and convergence of a single channel FXLMS algorithm are approximately achieved if

$$\mu < \frac{1}{(I + \Delta)P_z}, \quad (4)$$

where Δ and P_z are the delay of the secondary path (in time steps) and the power of the signal z , respectively. On the other hand, the greater μ , the lower is the time of convergence, which should always be much greater than the time variations of the noise in order to obtain an effective cancellation system. Thus, stability and time of convergence are trade-off parameters.

Unfortunately, in practical applications, other factors impair the performance and the actual upper bound on μ has to be lowered in order to prevent a divergent response. For instance, a portion of spectrum where the background noise level is noticeably higher than the one generated by the noise source may severely influence the stability of the system; similarly, quantization noise due to finite arithmetic implementation of the controller may prevent the system to converge.

The first considered alternative to FXLMS is the *Leaky* FXLMS algorithm [1] that aims to “enforce” a decrement of the magnitude of filters’ coefficients; the update phase of coefficients described in Eq. 2 is replaced by Eq. 5 (single channel is considered for sake of clarity)

$$w^{(n+1)}[i] = \nu w^{(n)}[i] + \mu z[n-i]E[n], \quad (5)$$

being $0 < \nu \leq 1$ a leakage coefficient. The rationale behind the Leaky FXLMS relies on the fact that containing filter’s energy necessarily prevents instability. Nevertheless, it can be shown that the lower the leakage coefficient, the farther the convergence of the system from the optimum, that is, stability is exchanged with cancellation performance.

The second considered solution is the *Normalized* FXLMS algorithm [19]. It is based on a different concept, that is, introducing a time variant step $\mu(n)$ in the classic FXLMS algorithm. The idea is to compensate for the power of the reference signal, avoiding sudden divergence behaviour, without penalty for the speed of convergence. By considering a constant factor α , $0 < \alpha \leq 2$, we define

$$\mu^{(n)} = \frac{\alpha}{I\hat{P}_z^{(n)}}, \quad (6)$$

being $\hat{P}_z^{(n)}$ an estimation of the power of the z signal at time step n . The performance of the Normalized FXLMS algorithm in terms of speed of convergence are strongly dependent on how such an estimation is performed. In the specific case of estimation on rectangular moving window on the last I samples, by inserting Eq. 6 in Eq. 2, we obtain

$$w^{(n+1)}[i] = w^{(n)}[i] + \alpha \frac{z[n-1]E[n]}{\sum_{n=0}^{I-1} |z[n]|^2}. \quad (7)$$

Because the denominator in Eqs. 6-7 can be arbitrary small in case of small power of reference signal, in practice a lower bound on $\hat{P}_z^{(n)}$ is adopted in order to limit $\mu^{(n)}$ from above.

The last considered solution is the so named *Correlation* FXLMS algorithm [20]. It is based on the fact that, when the system converges, that the statistical correlation between the error E and the modified reference signal z must be zero, namely

$$\rho = \mathbf{E}[E[n]z[n]] = 0. \quad (8)$$

Hence, the time varying step μ can be accordingly tuned. When the correlation is high, this procedure assumes that the cancelling system is not working enough and the step must be high in order to accomplish major adjustments; on the contrary, it has to be small when the correlation is close to zero. Thus, the step is given by

$$\mu^{(n)} = \alpha \hat{\rho}^{(n)}, \quad (9)$$

where

$$\hat{\rho}^{(n)} = \beta \hat{\rho}^{(n-1)} + (1 - \beta) E[n]z[n], \quad (10)$$

being α and β a constant factor and a smoothing factor, respectively.

4. Results

Experimental results have been carried out by considering the setup summarized in Fig. 1, by setting the following distances: $d_{n,c} = 6$ m, $d_{r,c} = 5.3$ m and $d_{c,e} = 0.4$ m. Both the noise source and the control source have been realized by means of loudspeakers, whereas a pair of low cost microphones have been used as reference and error microphones. All components have been put on hard ground in outdoor environment, as illustrated in Fig. 3. An extra measurement by using an high accuracy microphone placed at a distance of 2 cm from the error microphone has been performed, in order to check the sound pressure level (SPL). The FXLMS algorithm and its variants (briefly referred to as Leaky, Normalized and Correlation) have been implemented from scratch on a Field-Programmable Gate Array (FPGA) controller.

A first set of experiments has been carried out by measuring the attenuations obtained in the case of pure tonal noise source at 200, 300, 400 and 500 Hz, by considering two different original SPL on the error microphone, i.e. 75 and 85 dB. The obtained results are reported in Table 1. The FXLMS and Leaky algorithms outperform the other ones up to 500 Hz; in this specific case, the Normalized exhibits a 2-3 dB margin, but an overall performance reduction of all tested systems occurred. Interestingly, we observed that the performance of each algorithm is barely influenced by the original SPL.

The second set of tests has been performed in a similar fashion by considering a broader band noise in two frequency ranges, 200-300 Hz and 200-400 Hz, respectively. Specifically, in both scenarios a multi tonal signal has been generated, setting an interval of 20 Hz between adjacent tones; furthermore, two different original SPL on the high accuracy error microphone, i.e. 70 and 80 dB, have been tested. In Table 2, the measured attenuations are reported; it can be observed that the FXLMS algorithm dramatically outperform the variants in the former case, but instability arises as the noise level increases. On the other hand, stable responses have been verified for the variants. As a general trend, a performance loss is registered when increasing the noise bandwidth, but Normalized and Correlation exhibit a noticeable immunity to the increment of the original SPL.

Table 1: Attenuations (dB) measured on a single tone by means of different algorithms.

Algorithm	Original SPL = 75 dB				Original SPL = 85 dB			
	200 Hz	300 Hz	400 Hz	500 Hz	200 Hz	300 Hz	400 Hz	500 Hz
FXLMS	27.7	27.5	27.7	15.3	27.9	28.9	28.9	15.2
Leaky	28.3	28.5	27.2	15.8	28.5	28.8	28.4	15.8
Normalized	23.3	26.2	23.8	17.5	23.7	26.3	24.2	18.5
Correlation	20.9	23.6	25	15.1	21.0	19.5	26.5	14.7

Table 2: Attenuations (dB) measured on multiple tones by means of different algorithms.

Algorithm	Original SPL = 70 dB		Original SPL = 80 dB	
	200-300 Hz	200-400 Hz	200-300 Hz	200-400 Hz
FXLMS	21.3	29.2	unstable	unstable
Leaky	19.8	16.7	4.2	4.4
Normalized	17.4	12.6	16.7	12.1
Correlation	14.8	12.4	20.0	10.2



Figure 3: Experimental setup.

5. Conclusions

Active noise control systems represent promising alternatives to passive solutions for the reduction of low-frequency noise in outdoor contexts due to their relatively lower dimensions, costs and environmental impact. Nevertheless, stability and convergence of commonly adopted control algorithms are important issues for practical applications and tests on real scenarios are required. In this paper we proposed a comparison on-the-field between the Filtered-X Least Mean Square algorithm and some variants that are theoretically more stable. Some preliminary tests in the case of a single control channel were carried out and showed that the original algorithm exhibits a superior cancellation performance on simple single tone noise (up to 29 dB), but it suffers of stability problems whenever a broad band frequency noise characterized by high sound pressure levels has to be faced. In this specific case, robust variants should be preferred since they exhibit stable responses. In order to provide a complete overview of the benchmark, the most interesting comparison between multiple channels active noise control systems are expected to be set up in the next months.

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