A Hybrid Active Noise Cancelling with Secondary Path Modeling

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Abstract—This paper presents a hybrid active noise cancelling (ANC) system, to overcome the acoustic feedback present in most feedforward ANC system, together with an on line secondary path estimation scheme. The ANC system provides a suitable solution to two important problems present in most ANC systems. The acoustic feedback is reduced using two FIR adaptive filters with feedforward and feedback configurations, respectively; while the secondary path modeling is carried out using a modification of the on line estimation method proposed by Akhtar. Computer simulation results show that the combination of the feedback and feedforward stages provides a fairly good acoustic feedback cancellation, while the secondary path estimation method performs fairly well, providing both a fairly good noise cancelation performance.

I. Introduction

The need to eliminate unwanted noise is increases, as the people tolerance to the perception of sounds generated by industrial equipment, appliances decreases, because some general properties of these signals are unpleasant for most people. Generators are just a few examples of processes or equipments that produce signals nuisance to human ear. Mechanical vibrations produced by engines in operation, digging machinery and electricity. While methods for mitigating these unwanted sounds already have been proposed, most of them based on passive elements, they offer a poor response to low frequency sounds, because the wavelength of the signal is longer than the size of the muffler liabilities. The relevance in the treatment of low-frequency sounds is that they produce fatigue and loss of concentration, thus affecting the people performance, machinery and equipment present. That is because low-frequency sounds produce very intense vibrations that can fracture structures during very long periods of exposure. In addition, the ANC systems must respond to time varying environments and time varying frequency characteristics of the primary noise which must be tracked to get an acceptable noise cancellation performance:

Consider the ANC shown in Figs. 1 and 2 which are similar to the well known adaptive noise canceling proposed by Widrow and Stearns [2], although there are two important

differences. Firstly in an ANC the cancellation process is carried out in the acoustic domain, while in adaptive noise canceling it is carried out in the electrical domain, doing it a more difficult task; and second the ANC output is filtered by a system S(z), as shown in Figs. 2 and 3, which produces a delay of the filter output with respect to its input. S(z), known as secondary path, which represent the effect of filters, A/D and D/A converters, loudspeakers, microphone and acoustic path between canceling loudspeaker and microphone must be estimated in order to avoid performance degradation.

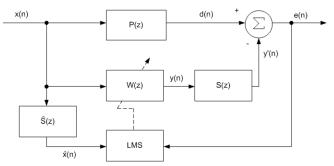


Figure 1 Feedforward ANC System with FXLMS Algorithm

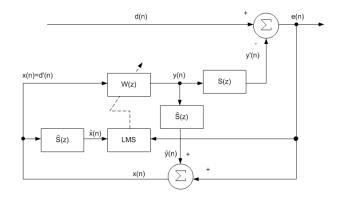


Figure 2 Feedback ANC System with FXLMS Algorithm

Two ANC configurations are widely used: The feedforward and feedback configurations, shown in Figs. 1

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and 2, respectively, both of them with advantage and disadvantages depending on the noise signal and environment characteristics. The ANC with feedforward structure is able to handle both, narrow band and wide band noise, however in many cases the canceling signal produced by the ANC also reach the input microphone. This fact produces significant performance degradation. On the other hand the feedback ANC structure does not present this kind of distortion because this structure does not use any input microphone, as shown in Fig 2. However, because this structure generates its own input signal though a linear prediction operation, its performance degrades when the correlation among its samples weakens [3]. Because the feedforward ANC structure can be used to solve almost any noise cancelling problem, important efforts have been done to solve the feedback distortion problem. Among them the hybrid structures, which combine the properties of both realization forms, appear to be a desirable alternative [4]-[6].

As mentioned before, the filter output signal y(n) reach the cancelling point to generate the output error, e(n), through the so called secondary path, S(z), which takes in consideration the digital to analog converter, reconstruction filters, the loudspeaker, amplifier, the acoustic from the loudspeaker to the microphone error, the error microphone, and analog to digital converter. Because the presence of S(z) causes that the input and the output error signals of adaptive algorithm be out of phase, to avoid distortion S(z) must be estimated in order that both, the filter output error and input signals be filtered by, in theory, the same system. There are two techniques for estimating the secondary path which are: offline and the

online secondary path modeling. The first one is carried out using a Feedforward system where the plant now is S(z) and the coefficients of the adaptive filter are the estimated secondary path. This approach performs very well when S(z) is time invariant. However this situation is seldom present in practice. When the secondary path is time varying, an online modeling approach must be used. Because an accurate estimation of S(z) is very important, several approaches have been proposed in the literature, among them the Eriksson [7] and Akhtar [8], in which a white noise sequence is used for S(z) estimation, are two of the more widely used.

This paper presents a hybrid structure similar to that proposed in [4]-[6] which consists of a feedforward structure, used to estimate the noise path, and a feedback structure, used to cancel the feedback acoustic noise. To avoid distortion due to the time varying conditions of S(z), a secondary path estimation algorithm based on the Akhtar method [8] is proposed.

II. PROPOSED ANC STRUCTURE

Figure 3 shows the bock diagram of proposed hybrid ANC structure with online on-line secondary path modeling. Proposed hybrid ANC structure consists of a feedforward stage, W(z), which is used to estimate the noise path, P(z), and a predictive structure, M(z), which is used to cancel the distortion due to the acoustic feedback path, F(z). The main idea is that, because the samples of feedback distortion are strongly correlated among them, they can be predicted.

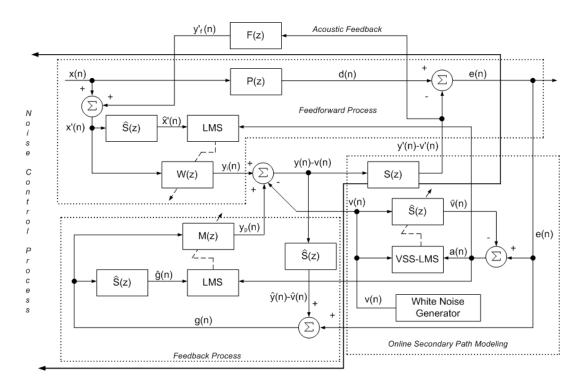


Figure 3 Hybrid ANC structure with secondary path madeling

As shown in Fig. 3 the same signal, a(n), is used as the error signal to update the adaptive filter, W(z), which corresponds to the Feedforward stage used to identify the noise path, as well as to update the linear predictive filter M(z), which intends to cancel the distortion produced by the feedback propagation from the cancelling loudspeaker to the input microphone thorough the system F(z); and to estimate $\hat{S}(z)$ that represents the online secondary path modeling adaptive filter. From Fig. 3, we can see that the error signal of all the ANC system is given by:

$$e(n) = d(n) + [v(n) - y(n)] * s(n)$$
 (1)

where d(n) is the desired response, v(n) is a white noise signal, s(n) is the finite impulse response of the secondary path and y(n) is the secondary path input signal used to produce the acoustic noise that achieves the attenuation of the primary noise. Thus from Fig. 3 it follows that

$$y(n) = y_i(n) + y_n(n) \tag{2}$$

$$y_i(n) = \overline{\mathbf{w}}^{\mathrm{T}}(n)\overline{\mathbf{x}}'(n) \tag{3}$$

$$\overline{\mathbf{w}}(n) \equiv [w_0(n), w_1(n), ..., w_{l-1}(n)]^{\mathrm{T}}, \tag{4}$$

$$\overline{\mathbf{x}}'(n) \equiv [\mathbf{x}'(n), \mathbf{x}'(n-1), ..., \mathbf{x}'(n-L+1)]^{\mathrm{T}}$$
 (5)

$$x'(n) = x(n) + y'_{f}(n) - v'_{f}(n)$$
 (6)

Here (6) denotes the reference signal that already considers the effects of the acoustic feedback. Thus taking in account the acoustic feedback consideration it follows that:

$$y'_{f}(n) = y'(n) * f(n)$$
 (7)

$$v'_{f}(n) = v'(n) * f(n)$$
 (8)

Both equations (7) and (8) contains f(n), the finite impulse response of the acoustic feedback filter; moreover y'(n) and v'(n) are the signals that have already been filtered by S(z). Next consider the predictive stage whose output signal is give by

$$y_n(n) = \overline{\mathbf{m}}^{\mathrm{T}}(n)\overline{\mathbf{g}}(n) \tag{9}$$

$$\overline{\mathbf{m}}(n) \equiv [m_0(n), m_1(n), ..., m_{M-1}(n)]^{\mathrm{T}}$$
 (10)

$$\overline{g}(n) \equiv [g(n), g(n-1), ..., g(n-M+1)]^{T}$$
 (11)

$$g(n) = e(n) + \hat{y}(n) - \hat{v}(n)$$
 (12)

$$\hat{v}(n) = v(n) * \hat{s}(n) \tag{13}$$

$$\hat{\mathbf{y}}(n) = \mathbf{y}(n) * \hat{\mathbf{s}}(n) \tag{14}$$

From (13) and (14) it follows that during the estimation of ANC coefficients vectors it is necessary to used the signals

y(n) and v(n) once both of them already have been filtered by the estimated secondary path $\hat{S}(z)$. Thus an accurate estimation of $\hat{S}(z)$ is required. To this end Akhtar method will be used.

The advantages of using the Akhtar's method [6], [7] for the secondary path modeling in proposed ANC system are reflected in the VSS-LMS algorithm that allows the modeling process to selects initially a larger step size, $\mu_{e}(n)$ which decreases to a minimum value according with the decreasing of [d(n)-y'(n)]. If the filter W(z) is slow in reducing [d(n) - v'(n)], then the step size may remain to small value for more time. Furthermore, the signal $a(n) = e(n) - \hat{v}(n)$ is the same error signal use to update all the adaptive filters involved in the ANC system, W(z), M(z) and $\hat{S}(z)$. The reason to use this signal is that for W(z), [v'(n)-v(n)] < v'(n) compared with the Eriksson's method. So when $\hat{S}(z)$ converges, that is $\hat{S}(z) \approx S(z)$, ideally $v'(n) \approx v(n) \Rightarrow v'(n) - v(n) \rightarrow 0$. Thus, using the Akhtar method [8] for secondary path estimation, the proposed hybrid structure in updated as follows:

$$\overline{\mathbf{w}}(n+1) = \overline{\mathbf{w}}(n) + \mu_{w} \hat{\overline{\mathbf{x}}}(n) [d(n) - y'(n)] + \mu_{w} \hat{\overline{\mathbf{x}}}(n) [v'(n) - \hat{v}(n)]$$
(15)

$$\overline{\overline{m}}(n+1) = \overline{\overline{m}}(n) + \mu_m \overline{\hat{g}}(n) [d(n) - y'(n)] + \mu_m \overline{\hat{g}}(n) [v'(n) - \hat{v}(n)]$$
(16)

$$\overline{\hat{\mathbf{s}}}(n+1) = \overline{\hat{\mathbf{s}}}(n) + \mu_s \overline{\mathbf{v}}(n) [\mathbf{v}'(n) - \hat{\mathbf{v}}(n)]$$

$$+ \mu_s \overline{\mathbf{v}}(n) [d(n) - \mathbf{y}'(n)]$$
(17)

Although (13) shows that when $\hat{S}(z)$ converges, the whole control noise process of the system is not perturbed by the estimation process of $\hat{S}(z)$, it is significant to identify that the online secondary path modeling is degraded by the perturbation of $\eta(n) = \mu_z \overline{V}(n) [d(n) - y'(n)]$.

III. SIMULATION RESULTS

This section presents the evaluation of proposed system by computer simulation using synthetic as well as actual acoustic noise signals.

Figure 4 and 5 shows the convergence performance of proposed ANC structure when it is required to cancel a narrow band noise signal that consists of 4 sinusoidal signals of frequencies 100, 200, 400, 600 Hz. A zero mean uniform white noise is added with SNR of 20dB, and a zero mean uniform white noise of variance 0.005 is used in the modeling process. Here in the iteration 1000 it is performed an abrupt change con the secondary path. The order that P(z), W(z) and S(z) and F(z) was in all cases equal to 20. Figures 6 and 7 shows the convergence performance of proposed structure when is required to cancel an actual noise motor signal. A zero mean uniform white noise of variance 0.005 is used in the modeling process. In iteration 600 it is performed an

abrupt change on the secondary path. Simulation results show that proposed ANC performs fairly well when required to cancel both synthetic and actual noise signals.

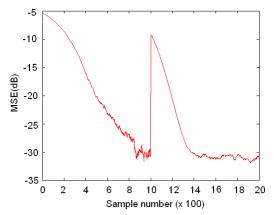


Figure 4 Convergence performance of secondary path estimation algorithm, when the noise to be cancelled is a narrow band noise. In iteration 1000 there is an abrupt change on S(z).

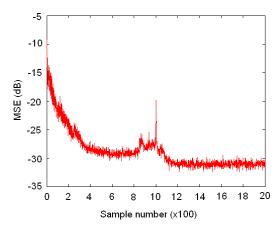


Figure 5 Convergence performance of proposed ANC when the noise to be cancelled is a narrow band noise. In iteration 1000 there is an abrupt change on S(z)

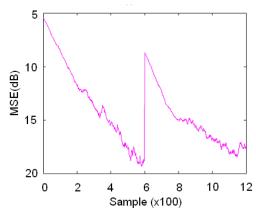


Figure 6 Convergence performance of secondary path estimation algorithm, when the noise to be cancelled is a narrow band noise. In iteration 600 there is an abrupt change on S(z).

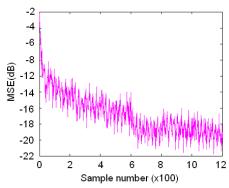


Figure 7 Convergence performance of proposed ANC when the noise to be cancelled is a narrow band noise. In iteration 600 there is an abrupt change on S(z)

IV. SIMULATION RESULTS

This paper proposed a hybrid ANC system that combines the feedforward and feedback structures, updated using the FxLMS, to improve the performance of ANC in presence of acoustic feedback distortion. This fact together with the online secondary path modeling allows the adaptation of the system to for any kind of secondary path change (gradual ideally). Computer simulations show that proposed system provides an improved performance, at somewhat increased computational cost because the Akhtar's online secondary path method, but this method compensates the noise control process for the feedforward and feedback stages. A VSS-LMS algorithm is used in the secondary path modeling to achieve the fastest optimal solution of the modeling filter. Computer simulation demonstrate the effectiveness of the proposed system

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