A NEW TWO-SENSOR ACTIVE NOISE CANCELLATION ALGORITHM

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ABSTRACT

In this paper we develop a new approach to active noise cancellation using two microphones. The primary microphone is placed at the point where noise cancellation is desired and the secondary microphone is placed at some other point. The cancelling signal is generated as the output of a two-input/single-output FIR filter which is driven by the outputs of the secondary *and* the primary microphones. The output of the primary microphone is not only used to adjust the coefficients of this filter, but is also used as an input to the filter itself.

1 INTRODUCTION

Unwanted acoustic noise is a by-product of many industrial processes and systems. In active noise cancellation (ANC), one introduces a secondary noise source to generate an acoustic field that interferes destructively with the unwanted noise, and thereby attenuates it [4,5,6]. Typically, measurements of the unwanted noise field are made at one or more locations using single or multiple sensors, and the cancelling field is generated based on these measurements. These sensors are usually divided into two groups: the primary sensor and the secondary sensors. The primary sensor is located at the point where noise cancellation is desired, and the secondary sensors are located at other points to provide information about the unwanted noise at the primary sensor. In most existing ANC systems, the cancelling signal is derived only from the outputs of the secondary sensors. In this paper, it is shown that the performance of the ANC system can be significantly improved by generating the cancelling signal based on the outputs of the secondary sensors *and* the primary sensor. Our approach is to construct the cancelling signal as a linear combination of the past values of the outputs of the primary and the secondary sensors, and we use a two-input/single-output LMS algorithm to find the linear combination that results in maximum noise attenuation.

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In this paper, the above two-sensor algorithm is developed and its performance is compared to the LMS algorithm proposed in [6] and to the single-sensor algorithm proposed in [2]. Specifically, these three algorithms were applied to noise recordings made using a set of headphones worn inside the cabin of a propeller aircraft. The primary sensor was placed inside the headphones and the secondary sensor was attached to the outside of the headphones. Assuming that the transfer function between the cancelling speaker and the primary sensor was a pure delay of about 40 μseconds, the new algorithm was able to attenuate the overall noise power 15dB more than the LMS algorithm proposed in [6] and 3dB more than the single-sensor algorithm proposed in [2].

2 EXISTING ANC ALGORITHMS

A block diagram for a generic ANC system is depicted in Fig. 1. In this figure, $G(z)$ represents the transfer function from the input to the cancelling speaker to the output of the primary sensor. The signal $n(t)$ is the unwanted noise at the primary sensor; $c(t)$ is the cancelling signal at the primary sensor; $e(t) = c(t) + n(t)$ is the residual signal measured by the primary sensor; and $s(t)$ is the output of the secondary sensor. Throughout this paper we will make the assumption that there is no feedback from the cancelling speaker to the secondary microphone. This assumption is justified in the case of noise cancelling headphones with since the cancelling speaker is located inside the headphones and the secondary microphone is attached to the outside of the headphones. $b1$ and $b2$ are binary switches (equal to one or zero) that determine what

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signals are used to generate the cancelling signal. We will assume in the reminder of this paper that $G(z)$ is known and causal with $g(0) = 0$, where $g(t)$ is the impulse response associated with $G(z)$. The objective of the ANC system is to minimize $E\{e^2(t)\}$ (here $E\{.\}$ stands for the expected value).

Noise cancellation systems based on the LMS algorithms are currently the most widely used two-sensor systems for ANC. In the two-sensor ANC algorithm presented in [6] the cancelling signal is derived from the output of the secondary sensor alone (this corresponds to $b1 = 0$ and $b2 = 1$ in Fig. 1). In this system $r(t)$ is generated by applying an appropriate FIR filter to the output of the secondary sensor $s(t)$. The output of the primary sensor, $e(t)$, is only used to adjust the coefficients of this FIR filter based on an LMS algorithm. Note that after an initial adaptation period, the coefficients of this FIR filter are no longer changing and $e(t)$ is not used at all. Essentially, this system cancels that part of the noise $n(t)$ which is correlated with the output of the secondary sensor. The ANC system proposed in [1] operates on the same principle except that it tries to account for the possible feedback from the cancelling signal to the secondary sensor; hence, the block diagram for this system corresponds to setting $b1 = 0$ and $b2 = 1$ in Fig. 1.

In the single sensor algorithm presented in [2], the cancelled noise at the primary sensor is measured and the output of this sensor alone is used to generate the cancelling signal (this corresponds to $b1 = 1$ and $b2 = 0$ in Fig. 1). This algorithm is derived assuming that $G(z)$ is a pure delay of k samples. With this assumption, the authors show that the cancelling signal that minimizes the power of $e(t)$ is simply the negative of the k-step ahead predicted value of $n(t)$.

3 ADAPTIVE NOISE CANCELLATION ALGORITHM

In this paper we present a new two-sensor ANC algorithm which uses the outputs of the primary and the secondary sensors to generate the cancelling signal (i.e. $b1 = 1$ and $b2 = 1$ in Fig. 1). Our approach is to construct the cancelling signal, $c(t)$, as a linear combination of the past values of $n(t)$ and $s(t)$. Specifically, the cancelling signal is generated as the output of a two-input/single-output FIR filter whose inputs are $s(t)$ and $n(t)$, and a stochastic gradient procedure is used to continuously adjust the coefficients of this filter so that $E\{e^2\}$ is minimized. Although $n(t)$ is not directly measured, it can be easily computed from the measurements of $e(t)$ by simply subtracting the known cancelling signal $c(t)$ from $e(t)$.

The block diagram for our ANC system is depicted in Fig. 2. In this figure, W is a two-input/single-output FIR filter whose coefficients are adjusted to minimize the mean-squared power of the residual signal $e(t)$,

$$e(t) = n(t) + r(t) * g(t). \tag{1}$$

In Fig. 2, W is implemented as the sum of two parallel single-input/single-output FIR filters of length L, i.e.

$$r(t) = \sum_{k=0}^{L-1} w_1^k(t)n(t-k) + \sum_{k=0}^{L-1} w_2^k(t)s(t-k), \tag{2}$$

where $w_1^k(t)$ and $w_2^k(t)$ $k = 0, \ldots, L-1$ are respectively the coefficients of the FIR filter w_1 and w_2 at time "t".

Using (2) we can rewrite (1) as

$$e(t) = n(t)+w_1(t)*[n(t)*g(t)]+w_2(t)*[s(t)*g(t)]. \tag{3}$$

Next, we define the vector of filter coefficients at time t as

$$\underline{\theta}(t) = [w_1^0(t), \ldots, w_1^{L-1}(t), w_2^0(t), \ldots, w_2^{L-1}(t)] \tag{4}$$

and two signals $v_1(t)$ and $v_2(t)$ as

$$v_1(t) = n(t) * g(t) \qquad\qquad v_2(t) = s(t) * g(t). \tag{5}$$

Using these definitions, we can rewrite (3) as

$$e(t) = n(t)+\sum_{k=0}^{L-1} w_1^k(t)v_1(t-k)+\sum_{k=0}^{L-1} w_2^k(t)v_2(t-k) \tag{6}$$

The following stochastic gradient procedure is used to continuously adjust $\underline{\theta}(t)$ so that $E\{e^2(t)\}$ is minimized:

$$\underline{\theta}(t+1) = \underline{\theta}(t) - \mu\nabla e^2(t), \tag{7}$$

where μ is the step-size of the stochastic gradient algorithm, and $\nabla e^2(t)$ is the gradient of $e^2(t)$ with respect to $\underline{\theta}$ evaluated at $\underline{\theta} = \underline{\theta}(t)$.

The gradient in (7) can be easily calculated from the measured signals $n(t)$ and $s(t)$ [6]. Noting that $\nabla e^2(t) = 2e(t)\nabla e(t)$, we see that this calculation reduces to determining $\nabla e(t)$. From (6), we then get

$$\begin{aligned} \nabla e(t) &= [v_1(t), \ldots, v_1(t-L+1), \qquad (8) \\ &\quad\; v_2(t), \ldots, v_2(t-L+1)] \\ &= \underline{V}(t), \qquad (9) \end{aligned}$$

where $\underline{V}(t) = [v_1(t), \ldots, v_1(t-L+1), v_2(t), \ldots, v_2(t-L+1)]$. It is important to note that at time "t", $\underline{V}(t)$ can be computed according to (5) from the known signals $s(t)$ and $n(t)$ and the known impulse response $g(t)$.

In summary, the update equation for the filter coefficients that minimize $E\{e^2(t)\}$ is

$$\underline{\theta}(t+1) = \underline{\theta}(t) - 2\mu e(t)\underline{V}(t), \tag{10}$$

where

$$\begin{aligned} \underline{V}(t) &= [v_1(t), \ldots, v_1(L-1), \\ &\quad v_2(t), \ldots, v_2(L-1)] \quad (11) \\ v_1(t) &= n(t) * g(t) \quad (12) \\ v_2(t) &= s(t) * g(t) \quad (13) \end{aligned}$$

4 SIMULATION RESULTS

The performance of the two-sensor algorithm of (10)-(13) was compared to that of the LMS algorithm presented in [6] and the single-sensor algorithm presented in [2]. Specifically, the three algorithms were applied to noise recordings made using a set of headphones worn inside the cabin of a propeller aircraft. The primary sensor was placed inside the headphones and the secondary sensor was attached to the outside of the headphones. In all the simulations, $G(z)$ (the transfer function from the cancelling speaker to the primary sensor) was assumed to be a pure delay of one or more samples. The results of applying these three algorithms to the recorded data are shown in figure 3. The x-axis in this figure indicates the delay between the cancelling speaker and the primary sensor.

As is clear from these results, the new two-sensor algorithm significantly outperforms the LMS algorithm. It is also interesting to note that if the delay between the cancelling speaker and the primary microphone is small, the performance improvement from use of the two-sensor algorithm over the single-sensor algorithm in [2] is relatively small. In a typical noise cancelling headphone application, the distance between the cancelling speaker and the primary microphone is on the order of 1 or 2 centimeters, corresponding to delay of about 40-80 μseconds, which might well not justify the incorporation of the secondary sensor. In other applications with longer delays, the importance of the secondary sensor increases.

The results in figure 3 imply that for short delays the secondary sensor does not provide significant additional information, beyond that provided by the primary sensor, about the future values of the noise at the primary sensor. The comparison in figure 3 between the single sensor algorithm and the LMS algorithm suggests that the future values of the noise at the primary sensor are more correlated with their past values than with the past values of the output of the secondary sensor.

It is important to remember that these results are only valid for the special case of noise cancelling headphones with the primary sensor located inside the headphones and the secondary sensor attached to the outside of the headphones. For example in a duct application with the secondary sensor located close to the source of the noise, one would expect that the output of the secondary sensor would contain significant additional information beyond that provided by the primary sensor. However, in many applications, such as the noise cancelling headphones, it is not practical to place the secondary sensor near the source of noise.

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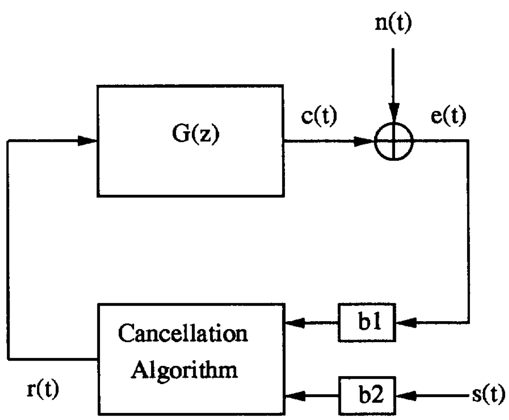

Figure 1: Generic ANC system.

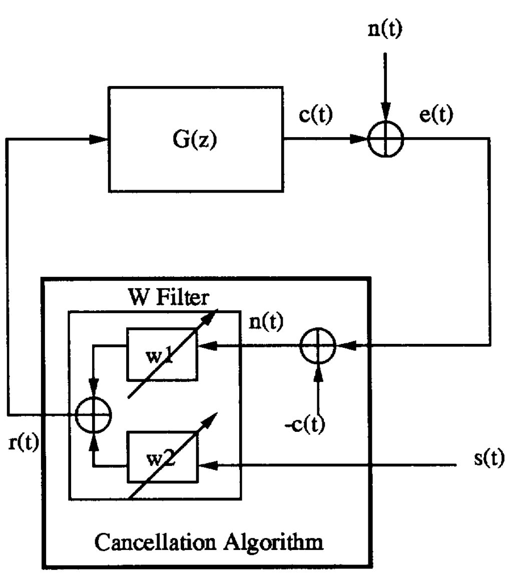

Figure 2: The new two-sensor ANC system.

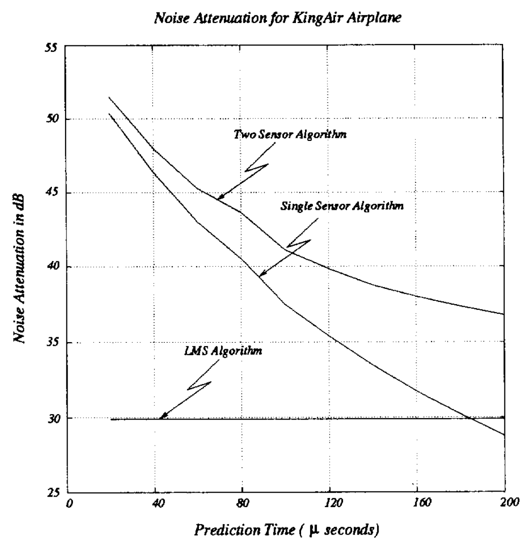

Figure 3: Attenuation obtained using three different ANC algorithms.