Crosstalk Reduction Using a New Adaptive Noise Canceller

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Abstract: – This paper introduces a new adaptive noise canceller (ANC) to improve the system performance in the presence of crosstalk. The proposed ANC consists of three microphones and two adaptive filters that automatically adjust their impulse responses through least mean-square (LMS) algorithms. Two microphones are used to represent the original speech signal and the reference noise input. The third microphone is used to provide a signal that is processed through the first adaptive filter to cancel the signal crosstalk leaking from the primary input into the reference input. The proposed ANC is simulated using different noise power levels for both stationary and nonstationary noise environments. Simulation results, carried out using a real speech, clearly demonstrate the significant achievements of the proposed ANC in minimizing the signal distortion and reverberation.

Key-Words: Adaptive filtering, Crosstalk reduction, LMS algorithm, Noise cancellation, Signal leakage.

1 Introduction

An important operation in voice communication systems involves the extraction of noise from the desired speech. This problem arises in many situations, such as airplanes, helicopters, and automobiles where acoustic noise is added to speech. Although the single microphone approach for noise canceling can be achieved using Wiener and Kalman filtering, the two-microphone approach using adaptive filtering is a more powerful technique for that purpose. The strength of the adaptive noise cancellers lies in the fact that no prior knowledge of the speech signal or the corrupting noise is required. However, a correlation between the noise that corrupts the speech and the noise in the reference input (adaptive filter input), is necessary for the adapting least mean-square (LMS) algorithm to remove the noise from the primary input signal.

A typical adaptive noise canceller (ANC), shown in Fig.1, is composed of two inputs: primary input and reference input. The primary input **d** consists of the original speech signal, **S**, corrupted by an additive noise **v**. The noise source is represented by **g**, and the transmission path from the noise source to the primary input is represented by the low pass filter, h. The input to the adaptive filter is the reference signal **g** that is correlated with **v**, but uncorrelated with **S**. The effectiveness of the ANC depends on how much **v** and **g** are correlated. The filter weights **w** are adapted by means of an LMS-based algorithm to minimize the power in the output signal. This minimization is achieved by processing **g** via the adaptive filter to provide an estimate of **v**, ($\mathbf{y} = \hat{\mathbf{v}}$), and then subtracting it from **d** to get **e**. Thus, at the kth iteration:

$$\mathbf{e}(k) = \mathbf{S}(k) + \mathbf{v}(k) - \mathbf{y}(k) \tag{1}$$

Many two-microphone ANCs have been proposed in the literature [1-5] using LMS-based algorithms that alter the step-size of the update equation to improve the tracking ability of the algorithm and its speed of convergence as well. In all these ANCs, it was assumed that there are no signal components leaking into the reference input. The presence of these signal components (also called signal crosstalk or signal leakage) at the reference input is a practical concern because it causes cancellation of a part of the original speech signal at the input of the ANC,

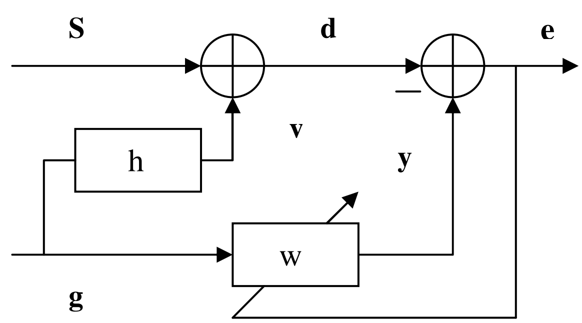

Fig. 1: A conventional ANC with no signal leakage.

and results in severe signal distortion and low signal to noise ratio at the output of the ANC. The magnitude of this distortion depends on the signal to noise ratios at the primary and reference inputs. Several techniques were proposed in the literature to enhance the system performance in this case of signal leakage (see [6], [7]). High computational complexity is associated with these algorithms.

In the present work, we propose a new ANC that uses two adaptive filters and three microphones instead of two as in a typical ANC. The third microphone provides a signal that is an attenuated replica of the desired signal. That signal is processed through the first adaptive filter to cancel the signal components leaking into the reference input. The second adaptive filter is used to cancel the noise at the input of the ANC.

2 Proposed ANC

Figure 2 shows a block diagram of the proposed ANC. The first microphone represents the speech signal and the second microphone represents a mixture of noise and signal components leaking from the first microphone through a channel with impulse response h_3. These signal components cause distortion in the recovered speech at the output of a conventional ANC.

To solve this problem we introduce a third microphone to provide a signal that is an attenuated replica of the original speech. This signal is processed by the first adaptive filter (w_1) to produce a crosstalk-free noise at its output. This noisy signal, with almost no leakage of the speech, is processed through the second adaptive filter to cancel the noise at the input of ANC, and accordingly produces the recovered speech. It is assumed that microphones 2 and 3 are placed farther apart such that there is no signal crosstalk leaking from the first into the second.

Many LMS-based adaptation algorithms could be used in the ANCs including the standard and normalized LMS algorithms [8], [9]. However, we prefer using the LMS adaptation algorithm which was also in one of our previous works [10] for its superiority over other algorithms. In that algorithm, the weight update recursion is given by

$$\mathbf{w}(k+1) = \mathbf{w}(k) + \frac{\alpha}{\varepsilon + \|\mathbf{e}(k)\|^2} e(k)\ \mathbf{v}(k) \tag{2}$$

where

$$\|\mathbf{e}(k)\|^2 = \sum_{n=0}^{k-1} |\mathbf{e}(k-n)|^2 \tag{3}$$

is the squared norm of the error vector $\mathbf{e}(k)$, estimated over its entire updated length k, α is an adaptation constant, and $\mathbf{v}$ is the input of the filter and is replaced by $\mathbf{v_3}$ in the first adaptive filter and by $\mathbf{v_2}$ in the second. ε is a small positive number, added to avoid a data over-flow error when $\|\mathbf{e}(k)\|^2$ becomes too small [11]. This proposed algorithm was shown to have a small number of computations [10].

The performance of the adaptive noise canceller may be described in terms of the excess mean-square error (EMSE) or misadjustment M. The EMSE at the k^{th} iteration is defined by

$$\text{EMSE}(k) = \frac{1}{L} \sum_{j=0}^{L-1} |\mathbf{ee}(k-j)|^2 \tag{4}$$

where $\mathbf{ee}(k) = \mathbf{e}(k) - \mathbf{S}(k)$ is the excess (residual) error, k is the sample (iteration) number, and L is the number of samples used to estimate the EMSE. The effect of L is just to smooth the plot of EMSE.

The steady-state EMSE (EMSE_{ss}), estimated by averaging EMSE(k) in (4) over k after the algorithm has reached steady-state condition, is defined by

$$\text{EMSE}_{ss} = \left(\frac{1}{K-P}\right) \sum_{k=P}^{K-1} \text{EMSE}(k) \tag{5}$$

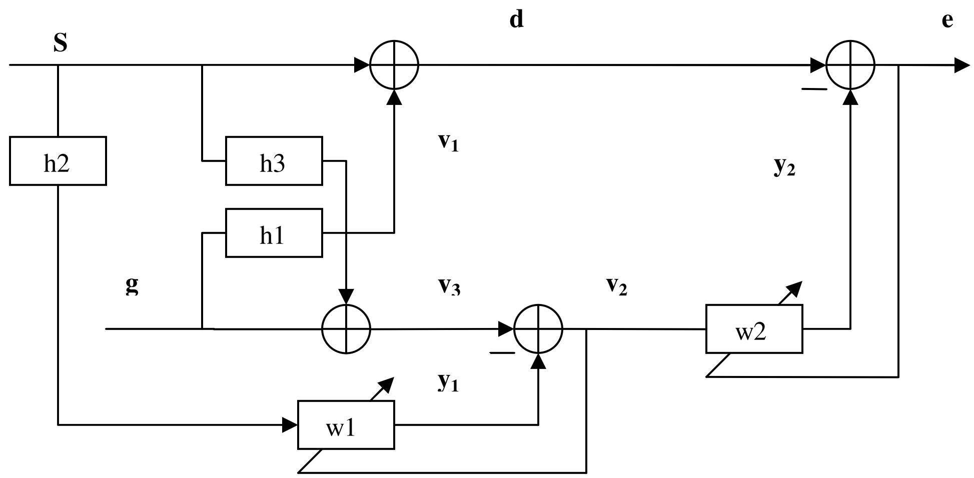

Fig. 2: Proposed ANC for signal leakage problem

where K is the total number of samples of the speech signal, and P is the number of samples after which the algorithm reaches steady-state condition.

The *misadjustment* M, a normalized mean-square error, is defined [8] as the ratio of the steady-state excess MSE to the minimum MSE.

$$M = \frac{EMSE_{ss}}{MSE_{min}} \qquad (6)$$

where MSE_{min} equals the power of the original clean speech signal, S, averaged over samples at which the algorithm is in steady-state ($k \geq P$) and is given by

$$MSE_{min} = \left(\frac{1}{K-P}\right)\sum_{k=P}^{K-1} \left|\mathbf{S}(k)\right|^2 \qquad (7)$$

Computer simulations were accomplished by using a real speech and different noise power levels for both stationary and nonstationary noise environments. The simulations show performance superiority of the proposed ANC in decreasing signal distortion, reverberation and consequently, producing small values of EMSE.

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3 Simulation Results

The simulations of the proposed ANC were carried out using a male native speech saying “sound editing just gets easier and easier” and sampled at a sampling frequency of 11.025 kHz. The number of bits per sample is 8 and the total number of samples is 33000, or 3 sec of real time. The simulation results are presented for stationary and nonstationary environments. For the stationary case, the noise ***g*** was assumed to be zero mean white Gaussian with three different variances as shown in Table 1. For nonstationary case, the noise was assumed to be zero mean white Gaussian with a variance that increases linearly from $\sigma^2_{g_{min}}$ =0.001 to three different maximum values $\sigma^2_{g_{max}}$ as demonstrated in Table 2. In all simulations, the following values of parameters were used: L=200, P=2000, ε =0.0001, N_1=N_2=12, α_1=2, and α_2=0.1. The values of α were selected as a compromise between fast rate of convergence and good tracking capability with most important concern to have a high rate of convergence in the first adaptive filter (α_1=2) and good tracking capability in the second (α_2=0.1). The impulse responses of the three IIR low pass filters used in the simulations, are: h_1=[1.5 −0.5 0.1], h_2=[1.5 −0.4 0.1], and h_3=[3 −1.2 0.3].

Figure 3 illustrates the performance of our proposed ANC in canceling the signal leakage at the output of the first adaptive filter for the case in which σ^2_g =0.001 as shown in Table 1. From top to bottom, that figure shows the original speech (S),

Table 1: Comparison of the $EMSE_{ss}$ and M of the proposed and conventional ANCs for stationary noise case.

Stationary white zero-mean noise **g**	Conventional ANC		Proposed ANC	
	Steady -State EMSE (dB)	M %	Steady -State EMSE (dB)	M %
$\sigma_g^2 = 0.001$	–22.7	53.9	–45.9	0.3
$\sigma_g^2 = 0.01$	–23.7	42.0	–38.0	1.6
$\sigma_g^2 = 0.1$	–26.1	24.6	–31.9	6.4

Table 2: Comparison of the $EMSE_{ss}$ and M of the proposed and conventional ANCs for non-nonstationary noise case.

Nonstationary white noise **g** $\sigma_{g\ min}^2$ = 0.0001	Conventional ANC		Proposed ANC	
	Steady -State EMSE (dB)	M %	Steady -State EMSE (dB)	M %
$\sigma_{g\ max}^2 = 0.001$	–22.4	57.1	–48.3	0.2
$\sigma_{g\ max}^2 = 0.01$	–22.5	55.7	–42.0	0.6
$\sigma_{g\ max}^2 = 0.1$	–24.5	35.6	–38.2	1.5

combination of noise and signal leakage (v_3), and the error signal of the first adaptive filter (v_2) which is the noise free of signal leakage.

A comparison of the proposed ANC with the conventional ANC for both stationary and nonstationary noise environments is shown in Tables 1 and 2. The adaptation constants of the LMS algorithms used in both ANCs were selected to achieve a compromise between small EMSE and high initial rate of convergence for a wide range of noise variances. From these Tables, improvements of up to 26dB in $EMSE_{ss}$ of the proposed ANC over the conventional one were achieved. It is worthwhile to note that if the noise variance increases, the performance of the conventional ANC becomes better as illustrated in Tables 1 and 2. This is expected because increasing noise power results in a

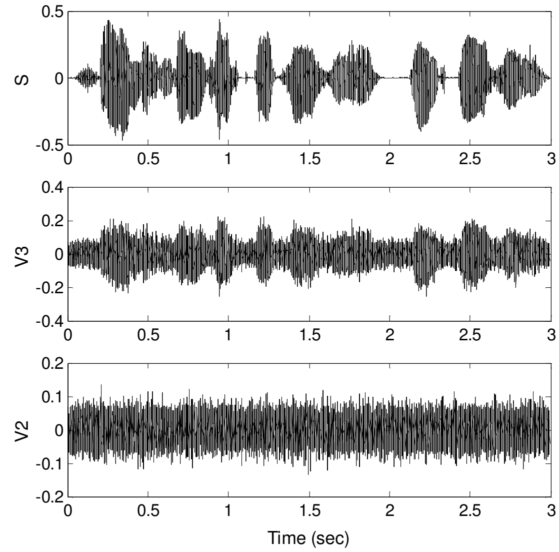

Fig.3: Cancellation of crosstalk at the output of the first adaptive filter of the proposed ANC. From top to bottom: Original clean speech (S), noise corrupted with crosstalk (V_3), and cross-talk- free noise (V_2). See Fig.2. (σ_g^2 = 0.001, Table 1).

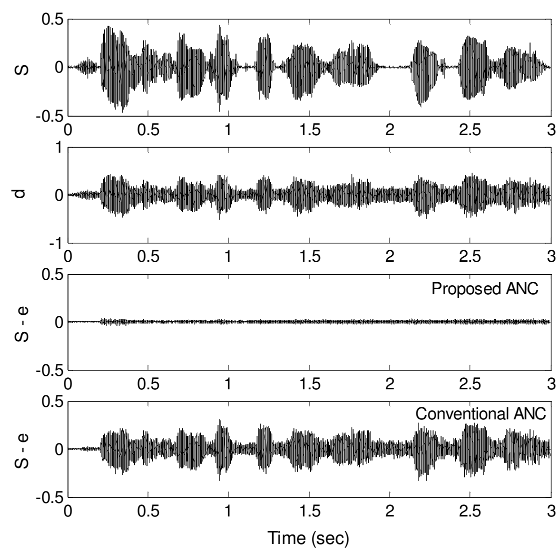

Fig.4: Performance comparison between the proposed and conventional ANCs. From top to bottom: Original clean speech (S), noise corrupting speech (d), residual (excess) error of the proposed ANC, and residual error of the conventional ANC ($\sigma_{g\ max}^2$ = 0.01, Table 2).

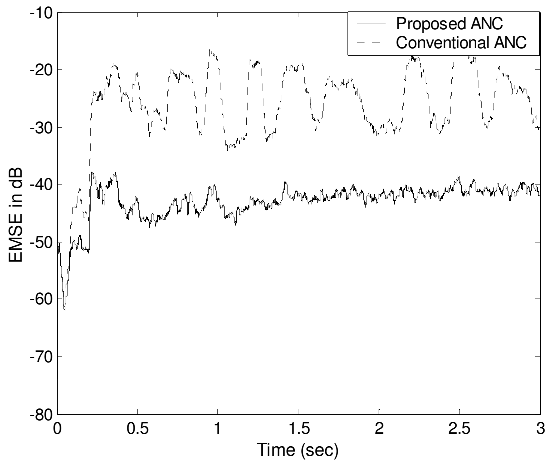

Fig.5: EMSE of the proposed and conventional ANCs ($\sigma_g{}^2{}_{max} = 0.01$, Table 2).

less significant effect of the signal leakage at the reference input.

Figures 4 and 5 provide more illustrations of the significant achievements of the proposed ANC over the conventional one for the nonstationary noise case in which $\sigma_g{}^2{}_{max} = 0.01$ (Table 2). From top to bottom, Fig.4 shows the speech signal (S), the noise corrupting speech (d), and the residual error (S−e) of the proposed ANC and (S-e) of the conventional ANC. The effect of increasing the variance of the noise on the processed speech is clearly shown in the second plot of Fig.4 (plot of d). Fig.5 shows the plot of EMSE for both ANCs. The improvements of the proposed ANC are clearly evident.

4 Conclusions

In this paper we presented a new ANC that corrects the cross talk leaking from the primary channel to the reference channel. The proposed ANC consists of three microphones and two adaptive filters that use LMS-based adaptation algorithms. The leaking signal is cancelled by the first adaptive filter using an attenuated replica of the speech signal provided by a third microphone. Accordingly, the output of the first adaptive filter is a crosstalk-free noise and this noise is cancelled through the second adaptive filter. Compared with the conventional ANC in both stationary and nonstationary noise environments, the proposed ANC demonstrates superior performance in minimizing signal distortion, reverberation and EMSE.

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