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Separation of Multiple Signals in Hearing Aids by Output Decorrelation and Time-Delay Estimation

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Abstract

Sensori-neural hearing impaired listeners have difficulty in separating multiple signals or perceiving speech in background noise and hearing aids are widely used to enhance the desired signal. In this paper we present a simple two-input system to separate two signals using output decorrelation where the filter coefficients are adaptively estimated to minimise the correlation of the output signals. In particular we show that an optimal filter can be designed and the convergence behaviour can be greatly improved by using a time-delay estimation technique. The algorithm was implemented to successfully separate two signals and the results are presented.

1. Introduction

Hearing impaired listeners have great difficulty in perceiving a signal in the presence of background noise or many unwanted signals. Consequently there has been a growing interest in developing algorithms for digital hearing aids to enhance the desired signal [1,2]. Although various single-input systems (monaural aids) have been useful in some cases, multi-microphone techniques provide better enhancement by exploiting the spatial features of the signal sources. In this paper we consider the problem of separating a mixture of two independent signals from two observed signals (a binaural aid). In [3] we presented a simple linear adaptive decorrelator to achieve separation when the received signals are a scalar mixture and in-phase at the inputs. However, in many cases the signals between the microphones are not in-phase and algorithms must be developed to separate the signals in the presence of unknown and arbitrary transfer functions which will be determined by the spatial location of the sources and room characteristics. We consider a signal model where the received signals have arbitrary gain and delay as shown below:

$$\begin{aligned}x_1(k) &= s_1(k) + h_1(k) \otimes s_2(k) \\x_2(k) &= s_2(k) + h_2(k) \otimes s_1(k)\end{aligned}\quad (1)$$

where $s_1(k)$ and $s_2(k)$ are mutually uncorrelated independent signals, h_i s are the unknown transfer functions of the signals between the two inputs and \otimes is the convolution operator.

Although this is a common problem in many engineering applications, the search for a successful algorithm still continues. In [4] the adaptive noise canceller was proposed when one of the signals is identical in both inputs. However, the schemes failed to produce successful separation when the signals were not identical due to the well known *signal-leakage* in the reference input. In [5] Van Compernelle *et. al.* proposed an adaptive decorrelator to achieve separation under a strict causality assumption on h_i s. The algorithm proposed in this paper is similar to [5] but is capable of producing rapid convergence and requires far fewer coefficients to achieve separation. Furthermore no assumptions are made about the transfer functions of the signals and the proposed scheme is therefore capable of performing satisfactorily in most realistic scenarios. In particular we propose a time-delay estimation technique to determine the (unknown) delays of the signals between the inputs and the adaptive decorrelation algorithm is employed to estimate the optimal values (weights) of the corresponding coefficients.

2. The Algorithm

The block diagram of the proposed filtering scheme is shown in Fig. 1. The output signals are given by

$$\begin{aligned}y_1(k) &= x_1(k) + g_1(k) \otimes x_2(k) \\y_2(k) &= x_2(k) + g_2(k) \otimes x_1(k)\end{aligned}\quad (2)$$

where g_i s are the FIR filters to be estimated.

It can be shown [6] that signal separation can be guaranteed if

$$\begin{aligned}g_1(i) &= -h_1(i) & \forall i \\g_2(i) &= -h_2(i) & \forall i\end{aligned}\quad (3)$$

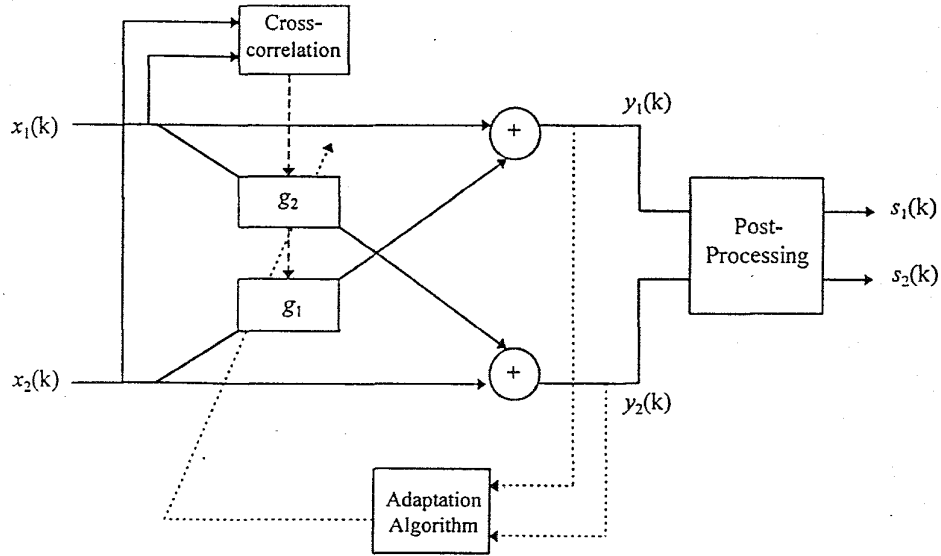


Fig. 1. The Block diagram of the proposed separation algorithm

However, the transfer functions h_1 and h_2 are not known *a-priori* and must be determined only from the received signals $x_1(k)$ and $x_2(k)$. In [3] we showed that the optimal parameters of a scalar mixture can be estimated by output decorrelation. A similar principle was adopted here to determine the parameters adaptively using the following update equations:

$$\begin{aligned} g_1^{k+1}(i) &= g_1^k(i) + \mu_1 y_1(k) y_2(k-i) \\ g_2^{k+1}(j) &= g_2^k(j) + \mu_2 y_2(k) y_1(k-j) \end{aligned} \quad (4)$$

where μ_1 and μ_2 are the stepsizes that control the convergence and stability of the adaptive scheme. It can be shown that the filters will converge to the optimal solution given by Eq. (3) when the output signals are decorrelated (*i.e.* $E[y_i(k)y_j(k-l)] = 0$) and will lead to signal separation. Here each filter coefficient is affected only by the corresponding cross-correlation lag and not any others. This one-to-one linkage between individual cross-correlation lags and the coefficients is only partially true and is not maintained when all the other coefficients are not optimal. From Eq. (2) we can see that the output signals are determined by all the filter coefficients and a non-optimal coefficient will directly affect the correlation estimates at all lags. Therefore, the gradient estimates become noisy and the convergence behaviour is severely degraded. Furthermore, since the h_i s are unknown, the choice of filter length will be arbitrary and an over-modelling of the transfer functions will add redundant filter coefficients, which leads to severe degradation in convergence behaviour. Therefore, we propose to estimate the optimal number of coefficients and the corresponding delays by using a simple time-delay estimation technique. This will result in an optimal filter configuration and lead to superior convergence characteristics due to less noise in the gradient estimates (correlation estimates).

2.1 Optimal Filter Length

Time-delay estimation is a standard problem in many signal processing and communication applications and many well-known algorithms exist [7]. We propose to use the cross-correlation between the received signals $x_1(k)$ and $x_2(k)$ to estimate the delays and subsequently the optimal number of coefficients required for separation. The cross-correlation function, $r_{x_1 x_2}(p)$, for lag p is given by

$$r_{x_1 x_2}(p) = E[x_1(k)x_2(k-p)] \quad (5)$$

Substituting Eq. (1) in Eq. (5) and taking the expectation gives (since $s_1(k)$ and $s_2(k)$ are mutually uncorrelated)

$$r_{x_1 x_2}(p) = \sum_i h_1(i) r_{s_2 s_2}(p-i) + \sum_j h_2(j) r_{s_1 s_1}(p+j) \quad (6)$$

where $r_{s_i s_i}(p)$ is the autocorrelation of the signal s_i . If we assume that $s_1(k)$ and $s_2(k)$ are stationary white signals of constant power, then

$$r_{s_i s_i}(p) = \delta(p) \sigma_i^2 \quad i = 1, 2 \quad (7)$$

where $\delta(p)$ is the delta function and σ_i^2 denotes the signal power. Now Eq. (6) can be rewritten as

$$r_{x_1 x_2}(p) = \sum_i h_1(i) \delta(p-i) \sigma_2^2 + \sum_j h_2(j) \delta(p+j) \sigma_1^2 \quad (8)$$

Therefore, it follows from Eq. (8)

$$r_{x_1 x_2}(p) = \begin{cases} \sum_i h_1(i) \delta(p-i) \sigma_2^2 = h_1(p) \sigma_2^2 & \forall p \geq 0 \\ \sum_i h_2(i) \delta(p+i) \sigma_1^2 = h_2(p) \sigma_1^2 & \forall p < 0 \end{cases} \quad (9)$$

Eq. (9) reveals that the cross-correlation function of a particular lag is determined only by the corresponding delay terms in the transfer function. Hence, the peaks in the cross-correlation function can be used to estimate the signal delays between the two inputs. The resulting filter g_i s will contain only the necessary delay terms and will lead to an optimal length filter. This approach can also be viewed as setting *a-priori* the necessary g_i terms to zero, thus eliminating the noise introduced by the adjustment of these unwanted delay terms. Furthermore, significant power savings and speed improvements can be achieved in the hardware implementation by this reduced complexity of the filter.

3. Simulation Results

The performance of this approach was investigated with two simulated AR signals and the following signal model was considered for the purposes of simulation.

$$\begin{aligned} x_1(k) &= s_1(k) + 0.6s_2(k-3) \\ x_2(k) &= s_2(k) + 0.8s_1(k-1) + 0.4s_1(k-5) \end{aligned}$$

Fig. 2 shows the cross-correlation function for various lags and it can be seen that the peaks at lags (-5, -1 and 3) clearly correspond to the actual delays of the transfer function as expected. Therefore, the optimal filter length is one for g_1 and two for g_2 , requiring adaptation of the second (one unit delay) and sixth taps of the filter.

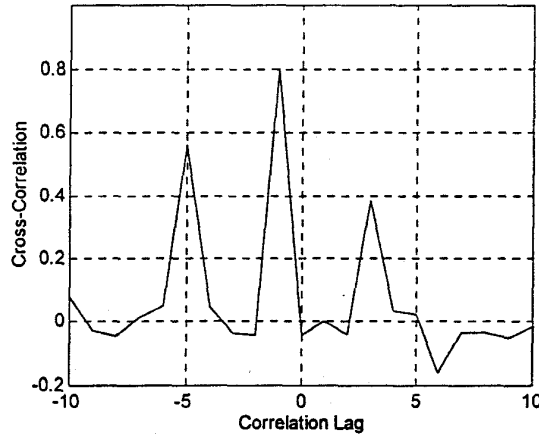


Fig. 2. The cross-correlation function of the received signals, $r_{x_1 x_2}$

The convergence behaviour of the proposed scheme was investigated and a significant difference in convergence speed was observed. The convergence rate of the proposed scheme was improved by a factor of 10 compared to a scheme where all the terms of a sixth order FIR filter was considered to model the transfer functions. Fig. 3 and Fig. 4 show the convergence behaviour of the separation algorithm. The improvements in convergence speed achieved by our proposed scheme is evident from these plots. The convergence properties of an adaptive hearing aid is very critical in real-time applications and thus the proposed approach is ideal for practical implementations.

Fig. 4 and Fig. 5 show one of the received signals and the reconstructed signal using this approach. It can be seen that the reconstructed signal matches the desired signal very accurately.

4. Conclusions

We presented a simple scheme for signal separation in hearing aids using output decorrelation and time-delay estimation techniques. The algorithm does not make any assumptions about the signal transfer functions and is capable of separating the signals even when they are not in-phase and the results demonstrate the separation capability of the proposed scheme. The convergence behaviour of the algorithm was considerably improved by estimating the optimal number of taps in the filter by

using a time-delay estimation technique. The scheme is very simple and requires fewer coefficients than existing schemes and will be very suitable for real-time implementation in digital hearing aids.

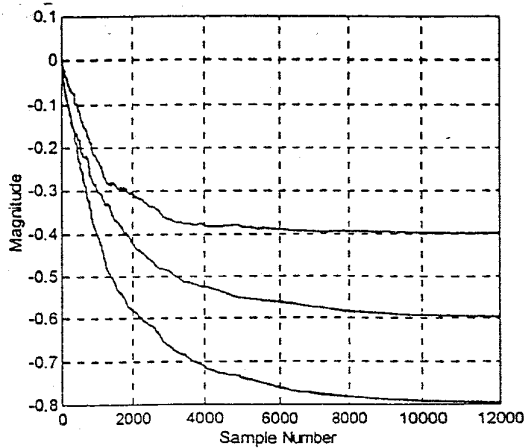


Fig. 3. The convergence of the separation algorithm with two 6th order filters.

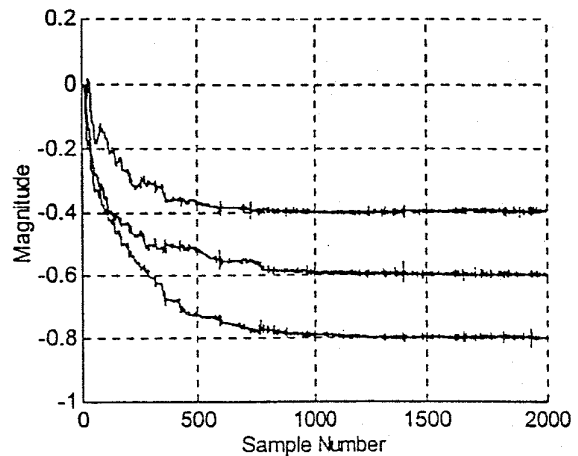


Fig. 4. The convergence behaviour of the proposed algorithm.

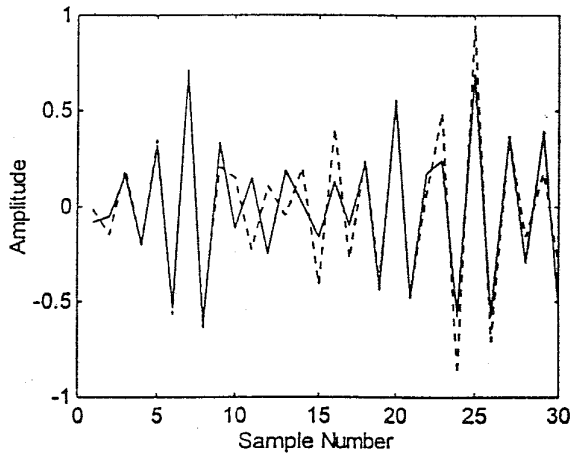


Fig. 5. The original signal (solid) and a received mixture signal $x_1(k)$ (dashed)

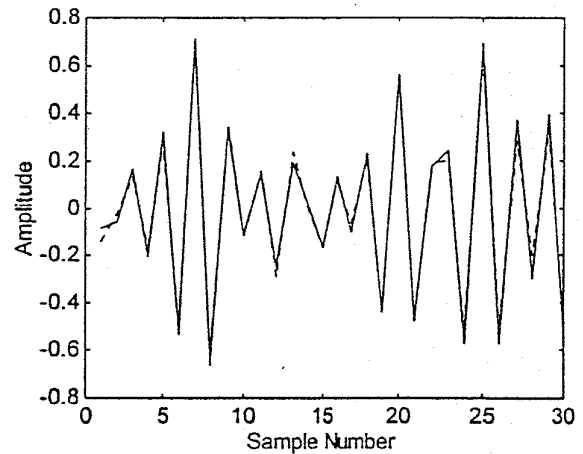


Fig. 6. The reconstructed signal (dashed) and the original signal (solid)

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