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Performance analysis of under-modelling stereophonic acoustic echo cancellation by adaptive filtering LMS algorithm ☆

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ABSTRACT

This paper addresses the field of stereophonic acoustic echo cancellation (SAEC) with adaptive filtering algorithms. In SAEC applications, using the least mean square (LMS) algorithm, it is usually assumed that the lengths of the adaptive filters are equal to that of the unidentified system responses. Although, in many realistic situations, under-modelled lengths adaptive filters, whose lengths are less than that of the unidentified systems (under-modelled systems), are employed, and analysis results for the exact modelled stereophonic LMS algorithm are not automatically appropriate to the under-modeled lengths. In this paper, we present a statistical analysis of the under-modeled stereophonic LMS algorithm. Exact expressions and deterministic recursive equations to the mean coefficients behavior of the adaptive LMS filters are derived to completely characterize and assess the performances (transient and steady-state) of the under-modeling stereophonic LMS algorithm. The expected theoretical behaviour is compared with Monte Carlo simulations and practical experimental results, showing a very good agreement.

1. Introduction

Acoustic echo cancellers (AEC's) are necessary in applications such as mobile phones, hands-free telephony, speakerphones and for communication systems (i.e. audio and video conferencing) in order to reduce or fully cancel the echoes phenomenon which make worse the quality of connections and communications. Theoretically, stereophonic acoustic echo cancellation (SAEC) can be viewed as a simple generalisation of the usual single-channel acoustic echo cancellation principle to the two channel case [1–3]. The purpose of echo canceller is to identify the receiving room echo paths and subtract an estimated replica of the echo, thereby achieving cancellation. An adaptive filter is used to identify the echo paths. The output of the adaptive filter, which is an estimate of the echo signal, can be used to reduce undesirable echoes [4–7].

The SAEC systems allow to have far better sound quality and sound localisation than what has been provided before. The improvements in quality are brought by increasing the signal bandwidth and also by adding more audio channels to the system. This last fact spurred the need for multi-channel acoustic echo cancellers [8].

The two-channel SAEC application is most attractive since only complexity issues differ for the more general multi-channel case. A basic scheme for SAEC is sketched in Fig. 1, where we illustrate the concept with a transmission room on the left and a receiving room on the right. The transmission room is sometimes referred to as the far-end and the receiving room as the near-end. As depicted in Fig. 1, the echo is due to acoustic coupling between the loud-speakers and microphones in the receiving room. In this scheme, the acoustic echo paths $\mathbf{h}_1$ and $\mathbf{h}_2$ in the local room are modelled by adaptive FIR filters $\tilde{\mathbf{h}}^{v1}(n)$ and $\tilde{\mathbf{h}}^{v2}(n)$, from which their added outputs produces an estimate $\hat{y}$ of the true echo y. Indeed, the physical impulse responses

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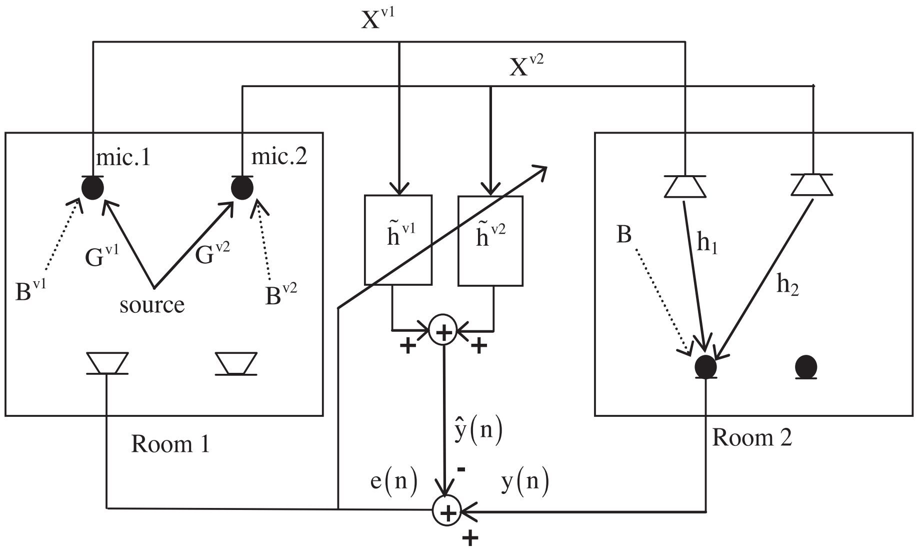

Fig. 1. Schematic diagram of a stereophonic echo canceller. Two adaptive filtering algorithms are used between Room 1 and Room 2 where: Room 1 is the transmitting room and Room 2 is the receiving room.

$\mathbf{h}_1$ and $\mathbf{h}_2$ are of infinite length; nevertheless it is assumed that the filters $\mathbf{h}^{v1}$ and $\mathbf{h}^{v2}$ are "sufficiently long", in the sense that the tails of $\mathbf{h}_1$ and $\mathbf{h}_2$ not modelled by $\tilde{\mathbf{h}}^{v1}(n)$ and $\tilde{\mathbf{h}}^{v2}(n)$ have low energy and thus can be neglected. Speaking in the sequel of "true" impulse responses means that we only consider the first parts of $\mathbf{h}_1$ and $\mathbf{h}_2$ which contain most of the energy, and which are assumed to be of the same size L as the model filters $\tilde{\mathbf{h}}^{v1}(n)$ and $\tilde{\mathbf{h}}^{v2}(n)$.

In SAEC for teleconferencing, we have a fundamental problem of the possibility to identify the true impulse responses of the acoustic echo paths. This problem arises from the correlation between the two signals picked up in the remote room in this request. SAEC is fundamentally different from traditional mono echo cancellation. A SAEC, straightforwardly implemented, not only would have to track changing echo paths in the receiving room but also in the transmission room. Thus, a generalisation of the mono AEC in the stereo case does not result in satisfactory performance. The problems of SAEC were first described in [3] and later on in [4]. The fundamental problem is that the two channels may carry linearly related signals which in turn may make the normal equations, to be solved by the adaptive algorithm, singular. This implies that there is no unique solution to the equation but an infinite number of solutions and it can be shown that all solutions (but the physically true one) depend on the transmission room. As a result, intensive studies have been made of how to handle this properly [8].

Generalisation of the solution to the normal equations in a more practical sense was addressed in references [4,5,8]. It was explained that in practice, the problem is not actually singular but extremely ill-conditioned due to the fact that the length of the adaptive filter is shorter than the echo paths of the transmission room. Furthermore, in practice, the transmission room is not completely stationary, i.e. smooth continuous changes exist, which slightly improves the situation by making the problem somewhat less ill-conditioned [9,10]. A complete theory of non-uniqueness and characterisation of the SAEC solution was presented in [11,12]. It is shown that the only solution to the non-uniqueness problem is to reduce the correlation between the stereo signals and an efficient low complexity method for this purpose was also given in [11–13].

In [14], the authors present a combination of mono and stereo echo cancellation which has the benefit of lower complexity than a pure stereo solution. Currently, attention has been focused on the investigation of other methods that decrease the cross-correlation between the channels in order to get well-behaved estimates of the echo paths [15]. The main problem is how to reduce the correlation sufficiently without affecting stereo perception and sound quality. Early examples of SAEC implementations can be found in [16–18]. These proposed solutions were presented before the theory and limitations of SAEC were fully understood, and were mainly based on the use of a single adaptive filter for each return channel. Recently, several methods and technique are proposed to solve this problem in the time domain as in [19–23], and also in the frequency domain as proposed in [24–26]. The performance of the SAEC is strictly affected by the choice of algorithm more than in the monophonic case. This is easily recognised since the performance of most adaptive algorithms depends on the condition number of the input signal covariance matrix.

In SAEC applications, the condition number is very high, and algorithms such as the LMS or the NLMS that do not take the coherence between the input signals into account, converge very slowly to the theoretical solution. It is consequently very interesting to study multi-channel adaptive filtering algorithms. A framework for multi-channel adaptive filtering can be found in references [1–8,27].

In this paper, we focus our interest on the case where the length of the adaptive LMS filters, employed with the SAEC application in the receiving room, are less than the length of the real filters. We propose a new study of this case, in terms of the mean behaviour convergence of the coefficient error vectors, and show theoretical results that are very close to the performed ones.