



A new pre-whitening transform domain LMS algorithm and its application to speech denoising

Laid Chergui, Saad Bouguezel*

Laboratoire de Croissance et Caractérisation de Nouveaux Semiconducteurs, Département d'Electronique, Faculté de Technologie, Université Ferhat Abbas Sétif-1, Sétif 19000, Algeria

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ABSTRACT

In this paper, we propose a new pre-whitening transform domain LMS algorithm. The main idea is to introduce a pre-whitening using a simple finite impulse response decorrelation filter of order one before applying the transform to reinforce its decorrelation. The resulting algorithm has the advantage of using any transform even with low decorrelation. This advantage can be exploited to consider transforms having lower computational and structural complexities than those of the classical transforms. For this purpose, we also investigate the use of other transforms, namely the parametric Fourier and Hartley transforms. This investigation is accomplished by studying the eigenvalue spreads obtained by a given parametric transform and then finding the value of the parameter corresponding to the minimum eigenvalue spread, which is equivalent to the best mean square error (MSE) convergence behavior. This approach provides new attractive transforms for the proposed algorithm. Moreover, we consider the adaptive speech denoising as an application to evaluate the performance of the proposed algorithm. The comparisons between the proposed and conventional algorithms for different transforms are performed in terms of the computational complexity, MSE convergence speed, reached steady state level, residual noise in the denoised signal, steady state excess MSE, misadjustment and output SNR.

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1. Introduction

The least mean square (LMS) algorithm is the mostly used in the adaptive filtering for its simplicity and robustness [1]. However, it has a slow convergence in the case of highly correlated input signals [2,3]. This is due to the fact that the autocorrelation matrix of the input signal has a large eigenvalue spread. To overcome this problem by reducing the eigenvalue spread, whitening or decorrelated adaptive algorithms have been proposed for time-domain LMS [4–6]. In [4], the authors proposed a joint decorrelation of both the input and error signals. The decorrelation is achieved in the time domain using an adaptive decorrelation filter based on the concept of prediction. The resulting decorrelated normalized LMS (NLMS) structure has the advantage of improving the mean square error (MSE) convergence speed and steady state compared to the conventional LMS and NLMS algorithms. The decorrelation of the input signal can also be achieved by an orthogonal transformation followed by power normalization [7]. These two operations have led to a new configuration named

* Corresponding author.

E-mail addresses: chergui.laid@gmail.com (L. Chergui), bouguezel_saad@yahoo.com (S. Bouguezel).

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transform-domain LMS (TDLMS) adaptive filters, which outperform the time-domain LMS adaptive filters in terms of the MSE convergence speed and steady state [8–10]. The orthogonal transforms such as the discrete Fourier transform (DFT), the discrete Hartley transform (DHT) and the discrete cosine transform (DCT) have been used in the TDLMS [7]. The resulting adaptive filters have been named as DFT–LMS, DHT–LMS and DCT–LMS, respectively [11]. The convergence speed of these filters depends on the used transform [8,11,12]. In general, the DCT–LMS filter presents a convergence performance better than those of the DFT–LMS and DHT–LMS filters [11]. This is mainly due to the fact that the DCT is suboptimal in terms of decorrelation [13,14]. Therefore, it is highly desirable to use in the TDLMS adaptive filters a transform having better decorrelation. However, the existing transforms are fixed and hence it is not possible to increase their decorrelation ability. Since the introduction of a decorrelation in the time-domain LMS has brought interesting improvements, it is important to investigate the introduction of such a decorrelation in the TDLMS. However, to the best of author's knowledge, this investigation has not been reported in the literature.

One of the applications of LMS adaptive filters is speech denoising. It is a very crucial operation to enhance the quality and intelligibility of the voice and reduce communication fatigue in modern communication systems such as mobile phones, hands-free