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An improved partial Haar dual adaptive filter for rapid identification of a sparse echo channel

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ABSTRACT

Recently, a coupled echo canceller was proposed that uses two short adaptive filters for sparse echo cancellation. The first filter operates in the partial Haar domain and is used to locate the channel's dispersive region; the second filter is then centered around this location to cancel the echo in the time domain. In this paper, we propose feasible solutions to improve the performance of this *partial Haar dual adaptive filter* (PHDAF) in practical applications. These include: (1) alleviating the dependence of the PHDAFs performance on the echo-path impulse response's bulk delay; (2) improving the tracking performance of the PHDAF in response to abrupt changes in the echo path; and (3) extending the original PHDAF structure to support the cancellation of multiple echoes. The proposed algorithmic solutions exploit the Haar transform's polyphase representation and make use of a novel peak tendency estimator (PTE) based on Dezert–Smarandache theory (DSmT). The improved PHDAF is evaluated in terms of its mean-square error (MSE) curves and its mean time to properly locate a dispersive region for different SNRs. Results show that enhanced performance can be obtained using the proposed solutions at a minimal increase in computational cost.

1. Introduction

Line, or network echo is commonplace in today's expanding communications infrastructure. Unlike other types of echo (e.g. acoustic), line echo is sparse: the echo-path impulse response consists of an initial zero or *bulk delay* region, corresponding to the signal round-trip, followed by a non-zero or *dispersive region*, corresponding to the echo arrival. Line echo is usually caused by an impedance mismatch, as occurring in the hybrid circuits used for $\frac{4}{2}$-wire conversion [1]. In voice communications, when the bulk delay between callers exceeds 25 ms or so, the reflected signal is perceived as a distinct echo that can severely impede a conversation. The coding and signal processing functions of digital technologies may introduce delays in excess of 100 ms; while for long distance calls routed via satellites, the propagation delay may reach several 100 ms [2]. Recent advancements, such as Voice over Internet Protocol (VoIP) telephony and xDSL technologies for broadband data transmission, highlight the need to develop better echo cancellers for sparse line echo.

In one of the earliest works on sparse echo cancellation [3], the input and desired signals are bandpass-filtered, decimated, and used by a short adaptive filter to estimate the bulk delay. A second short filter operating at the original sampling rate is centered around the dispersive region to cancel the echo. This way, only two short adaptive filters are required (compared to one long filter), thus reducing overall system complexity. However, the use of bandpass filtering in this approach has important drawbacks. Firstly, in speech applications, it can remove

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important frequency components in the signals, preventing proper convergence of the subsampled adaptive filter. Secondly, bandpass filtering in effect smears the peak of the unknown echo-path impulse response, making it more difficult to correctly locate the dispersive region.

Recent literature is rich in adaptive filtering algorithms that exploit the sparse characteristics of line echo [4]. Most of these algorithms are based on finding ways to determine which filter coefficients are associated with the echo, and then adapting only these coefficients. An adaptive multiple echo (ME) canceller is proposed in [5], which uses a full-length primary adaptive filter in parallel with a group of short secondary adaptive filters. In [6], a two-stage adaptation process is proposed in which the first stage estimates the bulk delay while the second stage adapts filter coefficients using a constrained tap-selection approach [7]. A well-known class of sparse echo cancellers are based on the proportionate normalized least mean squares (PNLMS) algorithm [8] and its variants [9–11]. These algorithms allocate individual step-size gains in proportion to the magnitude of each filter coefficient.

Other recent attempts at improving sparse echo cancellation rely on applying orthogonal wavelet transforms to the input data. It is shown in [12] that the number of adaptive coefficients needed to cancel the echo can be reduced significantly by applying a Haar transform. In [13], the authors propose using a subset of Haar wavelets to detect the significant channel coefficients. By exploiting the hierarchical structure of the dyadic wavelet expansion, these significant coefficients are used to activate wavelets in the remaining Haar subsets that share the same non-zero time-support. The Haar transform is particularly attractive for sparse echo identification: it facilitates the location of the dispersive region through proper selection of its scale/translation parameters, and it is easily amenable to a digital implementation.

Recently, Bershad and Bist [14] have proposed a solution to the sparse echo cancellation problem that combines favourable attributes of [3,13] in a coupled configuration consisting of two short adaptive filters. In this approach, referred to here as the partial Haar dual adaptive filter (PHDAF), the first filter operates on a subset of input Haar coefficients, and is used by a peak delay estimator to locate the echo-path's dispersive region. The second filter is centered around this location to cancel the echo in the time domain. In cases where the bulk delay is large, the PHDAF provides a significant reduction in computational and memory requirements. In addition, by reducing the number of filter taps to an amount necessary to model the dispersive region, the convergence speed of the echo canceller is increased. An improved theoretical model of the LMS algorithm in [14] for low rank systems was recently developed in [15] to better predict the behaviour of first and second moment statistics of the partial Haar adaptive filter. In [16], a partial block wavelet transform is proposed to increase efficiency and improve peak detection of the system in [14]. It is shown that the block transform reduces to using Daubechies' biorthogonal 2.2 spline wavelet, which is claimed to have better properties for estimating a peak's location.

In this paper we identify, and propose feasible solutions to three inherent limitations of the PHDAF for sparse echo cancellation in [14]: (1) dependence of the PHDAFs performance on the echo-path impulse response's bulk delay; (2) degraded tracking performance of the PHDAF in response to abrupt changes in the echo-path impulse response; and (3) limitation of the original PHDAF to a single dispersive region. The proposed algorithmic solutions exploit the polyphase representation of the Haar transform and make use of a novel peak tendency estimator (PTE) based on Dezert–Smarandache theory (DSmT) and fuzzy inference [17,18]. The improved PHDAF is evaluated in terms of its mean-square error (MSE) curves as well as its mean time to properly locate a dispersive region under different SNRs. In experiments using normalized least mean squares (NLMS) for the filter coefficient adaptation, the proposed amendments to the original PHDAF are shown to yield significant performance gains at a minimal increase in computational cost.

This paper is organized as follows: The structure and main equations of the PHDAF in [14] are reviewed in Section 2, along with a discussion of its main limitations. The solutions that we propose to overcome the latter are developed in Section 3. A series of supporting computer experiments is presented in Section 4. Finally, Section 5 concludes the work.

2. Background and problem formulation

2.1. The partial Haar dual adaptive filter

2.1.1. Partial Haar transform

Let $N = 2^J$, where J is a positive integer. The N-dimensional discrete-time Haar wavelet transform can be represented by an $N \times N$ orthogonal matrix $\mathbf{H}$ with entries $h_m(n)$, where the row index m identifies a basis vector and the column index n represents discrete-time [19]. The elements of the first row are equal to $h_0(n) = 1/\sqrt{N}$, while the remaining rows are obtained by scaling and shifting a discrete-time wavelet filter $\psi(n)$ defined as

$$\psi(n) = \begin{cases} +1, & 0 \leqslant n < N/2, \\ -1, & N/2 \leqslant n < N, \\ 0 & \text{otherwise}. \end{cases} \tag{1}$$

Specifically, let $m = 2^j + k$, where $j \in \{0, \ldots, J-1\}$ is the scale index and $k \in \{0, \ldots, 2^j - 1\}$ is the translation index. We have

$$h_m(n) = \alpha_j \psi(2^j n - kN), \tag{2}$$

where the normalization factor $\alpha_j = \sqrt{2^j/N}$. The number of rows with scale index j is equal to 2^j and the corresponding basis vectors have non-overlapping time-support of length $N/2^j$. A *partial* Haar wavelet transform consists of using only a subset of size $q = 2^j$ of Haar basis vectors corresponding to scale index j, to transform a given input data vector. The corresponding transform matrix, denoted $\mathbf{H}_q$, is thus a $q \times N$ submatrix of $\mathbf{H}$,