

Exponentially Weighted Step Size NLMS Adaptive Filter Based on the Statistics of a Room Impulse Response

Shoji Makino, *Member, IEEE*, Yutaka Kaneda, *Member, IEEE*, and Nobuo Koizumi

Abstract—This paper proposes a new normalized least-mean-squares (NLMS) adaptive algorithm with double the convergence speed, at the same computational load, of the conventional NLMS for an acoustic echo canceller. This algorithm, called the ES (exponentially weighted step size) algorithm, uses a different step size (feedback constant) for each weight of an adaptive transversal filter. These step sizes are time-invariant and weighted proportional to the expected variation of a room impulse response. The algorithm is based on the fact that the expected variation of a room impulse response becomes progressively smaller along the series by the same exponential ratio as the impulse response energy decay. As a result, the algorithm adjusts coefficients with large errors in large steps, and coefficients with small errors in small steps. A transition formula is derived for the mean-squared coefficient error of the proposed algorithm. The mean step size determines the convergence condition, the convergence speed, and the final excess mean-squared error. The algorithm is modified for a practical multiple DSP structure, so that it requires only the same amount of computation as the conventional NLMS. The algorithm is implemented in a commercial acoustic echo canceller and its fast convergence is demonstrated.

I. INTRODUCTION

AN ACOUSTIC echo canceller can overcome the acoustic feedback that interferes with teleconferencing and hands-free telecommunication. It adaptively identifies the transfer function between a loudspeaker and a microphone, and then produces an echo replica which is subtracted from the real echo.

Various adaptive algorithms are applicable to an acoustic echo canceller. The recursive least-squares (RLS) algorithm [1] provides fast convergence at the price of a high computational load. Recently developed fast RLS algorithms still require excessive computation [2], [3]. The least-mean-squares (LMS) algorithm [4], [5], on the other hand, is robust and simple. The normalized LMS (NLMS) algorithm [6], whose convergence speed is independent of input signal power, is widely used in commercial acoustic echo cancellers [7], [8]. However, the major drawback of the LMS and NLMS algorithms is their slow convergence. For example, the mean-squared error in the NLMS takes 2 s to converge for a white noise input signal and 10 s for speech for 8-kHz sampling rate and a filter with an order of 4000. Therefore, there is a

strong need to increase the convergence speed of the LMS and NLMS.

A step size parameter (feedback constant), used in many gradient-type adaptive algorithms, controls the convergence rate of the filter coefficients but also determines the final excess mean-squared error from the Wiener solution. Therefore, both a time-varying step size and a time-varying matrix-form step size [9] have been introduced to obtain fast convergence in the transient state and a small excess mean-squared error in the steady state. These time-varying step size algorithms, however, require complicated control of the step size. On the other hand, the convergence speed of the conventional NLMS with a time-invariant step size has a maximum, attained when the step size is unity for white noise [10].

Knowledge of the room impulse response is rarely used in conventional algorithms. An adaptive algorithm suited to the variation of an acoustic echo path is expected to improve convergence. In this paper, the room impulse response was measured repeatedly, and the impulse response variation was studied to determine its statistical characteristics. Based on the results, the ES (exponentially weighted step size) algorithm is proposed.

This paper is organized as follows. A brief review of acoustic echo cancellers and some conventional adaptive algorithms are given in Section II. The new adaptive algorithm is derived in Section III. Section IV discusses the properties of the proposed algorithm, followed by the modifications for a practical acoustic echo canceller in Section V. The real-time experimental results are shown in Section VI. Section VII summarizes the paper.

II. ACOUSTIC ECHO CANCELLERS AND CONVENTIONAL ADAPTIVE ALGORITHMS

A. Configuration of an Acoustic Echo Cancellor

The configuration of an acoustic echo canceller is shown in Fig. 1. The echo canceller identifies the transfer function of the acoustic echo path, *i.e.*, the impulse response $\mathbf{h}(k)$ between the loudspeaker and the microphone, where $\mathbf{h}(k) = [h_1(k), h_2(k), \dots]^T$ and $h_1(k), h_2(k), \dots$ represent coefficients of the impulse response at discrete time k . Since the impulse response $\mathbf{h}(k)$ varies as a person moves and varies with the environment, an adaptive filter $\hat{\mathbf{h}}(k)$ is used to identify $\mathbf{h}(k)$. Usually, $\hat{\mathbf{h}}(k)$ is a finite impulse-response (FIR) filter

Manuscript received May 6, 1991; revised May 5, 1992.

The authors are with NTT Human Interface Laboratories, Musashino-shi, Tokyo, 180 Japan.

IEEE Log Number 9203886.