Available online at www.sciencedirect.com
ScienceDirect
ELSEVIER
February 2015, 22(1): 6–10
www.sciencedirect.com/science/journal/10058885
The Journal of China Universities of Posts and Telecommunications
http://jcupt.xsw.bupt.cn

Adaptive feedback cancellation based on variable step-size affine projection for hearing aids

Qin Haijuan[1] (✉), Zhang Linghua[1,2]

1. College of Telecommunication & Information Engineering, Nanjing University of Posts and Telecommunications, Nanjing 210003, China
2. Jiangsu Provincial Engineering Research Center of Telecommunications and Network Technology, Nanjing 210003, China

Abstract

A new variable step-size (VSS) affine projection algorithm (APA) (VSS-APA) was proposed for adaptive feedback cancellation suitable for hearing aids. So, a nonlinear function between step-size and estimation error is established and automatically adjusted according to the change of the estimation error, which leads to low misalignment and fast convergence speed. Analysis of the proposed algorithm offers large capacities in converging to the objective system. Simulation shows that the proposed algorithm achieves lower misalignment and faster convergence speed compared to fixed step-size APA and conventional adaptive algorithms.

Keywords adaptive feedback cancellation, VSS, APA, adaptive filter, hearing aids

1 Introduction

Acoustic feedback is an important problem of the hearing aids since it causes screeching sounds, and makes the hearing aids users uncomfortable [1]. It limits the maximum usable gain of hearing aids and, in turn, degrades overall performance. To overcome this problem, many adaptive feedback cancellation technologies have been proposed, in which the entire effect of the feedback was eliminated by estimating the acoustic feedback path [2]. The least mean square (LMS) is the simplest algorithm, but its convergence speed is too slow. Much improved algorithms based on LMS were proposed to solve the problem [3].The normalized least mean square (NLMS) algorithm has been widely used for hearing aids thanks to its efficiency and simplicity [4]. To achieve better performance, an improved algorithm for NLMS was proposed in this article. A VSS-NLMS algorithm which adjusts the step-size according to the power of estimate error was proposed by Rotaru [5]. A sparse impulse response has only a small percentage of coefficients active with most coefficients insignificant or zero. Acoustic feedback path is included in this kind of sparse impulse response [6]. So the proportionate NLMS (PNLMS) algorithm and improved PNLMS algorithms were proposed based on Refs. [7–8].

However, the problem of convergence speed has not been fully resolved. The convergence speed of NLMS algorithm and its improved algorithms is not very fast, especially for colored input signals. The recursive least square algorithm has fast convergence speed but its complexity is too high and instability issues frequently arise [2], it's not practical in hearing aids. The APA was proposed. In contrast to NLMS, which updates the weight vector based on the current input vector, APA updates the weight based on K input vectors [9]. APA reuses the input signal to increase the convergence speed and lower the misalignment. Its superior performance can still be contained when the input signal is speech sequence. Similar to the NLMS algorithm, APA must choose appropriate step-size to meet the conflicting requirements of fast convergence and low misalignment. When the step-size is large, the convergence speed is fast but the misalignment is large. And the misalignment is small when the step-size is small, but the convergence speed is

Received date: 25-06-2014
Corresponding author: Qin Haijuan, E-mail: qinhaijuan119@163.com
DOI: 10.1016/S1005-8885(15)60618-5

slow [10–12].

A new nonlinear function between step-size and estimation error to lower the misalignment was derived. It adjusts the step-size according to the estimation error. Simulations show that the proposed algorithm achieves faster convergence speed and lower final misalignment compared to traditional adaptive methods and fixed step-size APA, especially for colored input signals.

2 VSS-APA for hearing aids

A general system model of adaptive feedback cancellation is depicted in Fig. 1. It is seen that $s(n)$ is the input signal and includes original voice $x(n)$ and near-end noise $v(n)$. $f(n)$ is set as the echo signal. $\boldsymbol{w}^{\mathrm{o}}$ is a column vector for the impulse response of an unknown system that we want to estimate. We note L as the filter length, $\boldsymbol{w}(n)=[w(n-1), w(n-2), ..., w(n-L)]^{\mathrm{T}}$ is the filter weight vector. $d(n)$ is the desired signal, $e(n)$ is the error signal.

Consider the data $\{d(n)\}$ and $\{e(n)\}$ that arise from the model

$$d(n)=\boldsymbol{u}^*(n)\boldsymbol{w}^o+v(n) \tag{1}$$

$$e(n)=d(n)-y(n)=\boldsymbol{u}^*(n)\boldsymbol{w}^{\mathrm{o}}-\boldsymbol{u}^*(n)\boldsymbol{w}(n)+v(n) \tag{2}$$

where $\boldsymbol{u}(n)$ denotes the $L\times 1$ input vector, and * denotes the Hermitian transpose.

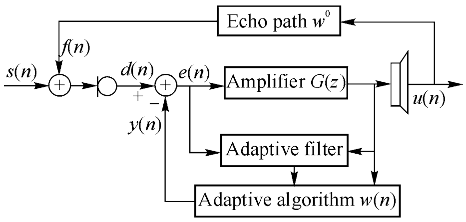

Fig. 1 The adaptive feedback cancellation system

2.1 Conventional APA

The APA updates the weight vector based on K most recent input vectors. Similar to the well-known NLMS algorithm, a step-size is used to control the rate of convergence and the misalignment.

The weight update equation for the APA algorithm is:

$$\boldsymbol{w}(n+1)=\boldsymbol{w}(n)+\mu\boldsymbol{U}(n)[\boldsymbol{U}^*(n)\boldsymbol{U}(n)]^{-1}\boldsymbol{e}(n) \tag{3}$$

where

$$\boldsymbol{U}(n)=[\boldsymbol{u}(n),\boldsymbol{u}(n-1),...,\boldsymbol{u}(n-K+1)]$$

$$\boldsymbol{d}(n)=[d(n),d(n-1),...,d(n-K+1)]$$

$$\boldsymbol{e}(n)=\boldsymbol{d}(n)-\boldsymbol{U}^*(n)\boldsymbol{w}(n)$$

$\boldsymbol{U}(n)$ denotes an input data matrix, $\boldsymbol{e}(n)$ and $\boldsymbol{d}(n)$ represent error output vectors and primary input, respectively. μ is the step-size.

It can be easily noticed that, for K=1, the NLMS algorithm can be obtained.

A small step-size will ensure small misalignment but result in slow convergence speed. On the other hand, a large step-size will result in fast convergence speed but large misalignment. The aim of a VSS-APA is to try to solve the conflicting requirement of fast convergence speed and low misalignment.

2.2 Conventional VSS-APA

The conventional VSS-APA choose μ such that this choice guarantees that the mean-square deviation will undergo the largest decrease from iteration ($n-1$) to iteration n. The conventional VSS-APA becomes

$$\mu(n)=\mu_{\max}\frac{\|\tilde{p}(n)\|^2}{\|\tilde{p}(n)\|^2+C} \tag{4}$$

$$\tilde{p}(n)=a\tilde{p}(n-1)+(1-a)\boldsymbol{U}^*(n)(\boldsymbol{U}(n)\boldsymbol{U}^*(n))^{-1}\boldsymbol{e}(n) \tag{5}$$

where C is a positive constant. $\tilde{p}(n)$ is an intermediate variable.

When $\|\tilde{p}(n)\|^2$ is large, $\mu(n)$ tends to $\mu_{\max}$. On the other hand, when $\|\tilde{p}(n)\|^2$ is small, the step-size becomes small. Thus depending on $\|\tilde{p}(n)\|^2$, $\mu(n)$ varies between 0 and $\mu_{\max}$.

2.3 A new VSS-APA

The conventional VSS-APA is so complex that it isn't suitable in hearing aids. In the following, we derives a new VSS-APA which is proven to be efficient for hearing aids.

According to Eq. (1) and Eq. (2)

$$\boldsymbol{d}(n)=\boldsymbol{U}^*(n)\boldsymbol{w}^{\mathrm{o}}+\boldsymbol{v}(n) \tag{6}$$

$$\boldsymbol{e}(n)=\boldsymbol{d}(n)-\boldsymbol{U}^*(n)\boldsymbol{w}(n) \tag{7}$$

The update recursion Eq. (3) can be written in terms of the weight-error vector, $\tilde{\boldsymbol{w}}(n)=\boldsymbol{w}^{\mathrm{o}}-\boldsymbol{w}(n)$, as

$$\tilde{\boldsymbol{w}}(n+1)=\tilde{\boldsymbol{w}}(n)-\mu(n)\boldsymbol{U}(n)[\boldsymbol{U}^*(n)\boldsymbol{U}(n)]^{-1}\boldsymbol{e}(n) \tag{8}$$

Multiply both sides by $\boldsymbol{U}^*(n)$,

$$\boldsymbol{U}^*(n)\tilde{\boldsymbol{w}}(n+1)=\boldsymbol{U}^*(n)\tilde{\boldsymbol{w}}(n)-\mu(n)\boldsymbol{e}(n) \tag{9}$$

From Eq. (6) and Eq. (7), Eq. (9) can be written as

$$\boldsymbol{U}^*(n)\tilde{\boldsymbol{w}}(n+1)=[1-\mu(n)]\boldsymbol{U}^*(n)\tilde{\boldsymbol{w}}(n)-\mu(n)\boldsymbol{v}(n) \tag{10}$$