A Variable Step Size Modified Decorrelated NLMS Algorithm for Adaptive Feedback Cancellation in Hearing Aids

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Abstract **- This paper presents a new algorithm for adaptive feedback cancellation (AFC) suitable for hearing aids. A variable step size scheme is added to a step size decorrelated NLMS algorithm. It is shown that the proposed algorithm has increased robustness and stability for both fixed and variable gain cases.**

Keywords **– variable step size, decorrelated NLMS algorithm, adaptive feedback cancellation, modified pseudo affine projection**

I. INTRODUCTION

Acoustic feedback is a common problem of the hearing aids, because the sound quality is deteriorated due to the feedback signal resonating in the closed loop. Many adaptive feedback cancellation (AFC) techniques have been proposed in order to minimize the effect of the feedback on the hearing aids [1-6]. The adaptation control is difficult due to the correlated input and feedback signals that could lead to a biased filter and severe signal distortion at the hearing aid output [4]. The algorithms for AFC should provide a compromise between fast convergence speed, and low steady-state level. Another requirement is a low computational complexity and good sound quality. The most used algorithm for AFC is the normalized least mean square (NLMS) algorithm [7-8]. It is simple to implement, but it also has slow convergence speed, especially for colored inputs. The recursive least square algorithms have fast convergence speed, but their complexity is too high and instability issues frequently arise. The performance of the affine projection algorithm [9] lies between that of NLMS and RLS. However, it involves a matrix inversion in the weight update equation and many fast versions were proposed (e.g. [10-14]). The application of the pseudo affine projection (PAP) algorithm [14] and its versions based on GS method for AFC [3] led to a solution with small numerical complexity and close performance to that of the AP algorithm. It is known that there is a bias in the estimate of feedback path in case of AFC due to correlation between the input and output signals of the hearing aids. There are several techniques that reduce the bias [15] (e.g. by adding probe signal to output [2], inserting the de-correlation filter [7] or time delays in the forward or filter path [16]). The gain for compensating the hearing loss should be applied as per hearing loss in frequency [15]. As a result, the AFC algorithms should be evaluated with frequency-dependent compensation gains, not only with fixed gains. In [15] an inverse gain filter (IGF) before the update of adaptive feedback canceller was used. The previous instability problem of PAP [3] caused by the delayed estimate of the linear prediction (LP) coefficients was solved using IGF and the LP coefficients were estimated from the input of the hearing aids. The robustness and stability of the Modified PAP (MPAP) algorithm was proved in [15].

It is known that there is a strong similarity between AP/PAP algorithms and the NLMS algorithm with decorrelated filters because they exhibit the same structure (e.g. [11] and [17]). The decorrelation filters are forward predictor error filters with their coefficients being matched with the correlation properties of the speech signal [17]. The decorrelated NLMS (D-NLMS) algorithms have been investigated in [18] and [19], while the use of a variable step size LMS algorithm was proposed in [20]. Our simulations have shown that the D-NLMS algorithm is not stable in case of AFC configurations for hearing aids. In this paper we propose to use a new variable step size scheme and decorrelate both input (output of hearing aids) and the error signal. The new algorithm called Variable Step Size Modified Decorrelated NLMS (VSS-MDNLMS) has improved steady-state behavior and stability. It is shown that the impact of decorrelation on both input and error on the signal quality is similar to that of MPAP.

The paper is organized as follows: Section II describes the adaptive feedback canceller based on the VSS-MDNLMS algorithm. Simulation results that compare the proposed algorithm with MPAP and NLMS are presented in Section III. Finally, the conclusions are presented in Section IV.

II. THE PROPOSED ALGORITHM

The adaptive feedback cancellation system (Fig.1) based on VSS-MD-NLMS algorithm is similar with that based on MPAP algorithm [15]. We note by L the filter length, $\mathbf{w}(n)=\left[w(n-1),...,w(n-L+1)\right]^T$ is the filter weight vector $e(n)$ is the error signal, $d(n)$ is the desired signal, $y(n)$ is the primary input signal, and $\mathbf{y}(n)=\left[y(n-1),...,y(n-L+1)\right]^T$ is the input signal vector. $\mathbf{A}(z)$ is the linear prediction [15] block and $\left|\mathbf{G}_m(z)\right|^{-1}$ is the inverse-gain filter (IGF) where $\mathbf{G}_m(z)$ is a minimum-phase system whose magnitude response is the same as that of the compensation system. The transfer function of the feedback path is $\mathbf{W}(z)$.

It is well known that the weight update equation for the NLMS algorithm [21] is

$$\mathbf{w}(n+1)=\mathbf{w}(n)+\frac{\mu e(n)\mathbf{y}(n)}{\mathbf{y}^T(n)\mathbf{y}(n)+\delta} \qquad (1)$$

where δ is a regularization factor.

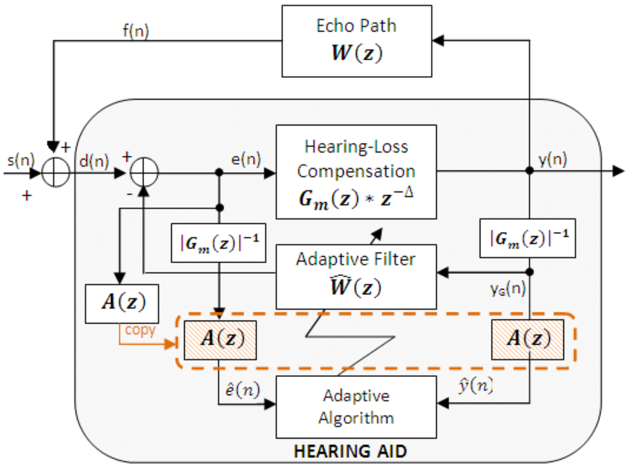

Figure 1. The adaptive feedback canceller system

According to [22] the filter update recursion for a decorrelated NLMS algorithm (DNLMS) is:

$$\mathbf{w}(n+1)=\mathbf{w}(n)+\frac{\mu e(n)\hat{\mathbf{y}}(n)}{\hat{\mathbf{y}}^T(n)\hat{\mathbf{y}}(n)+\delta} \qquad (2)$$

where $\hat{\mathbf{y}}(n)$ is the decorrelated signal vector obtained as $\hat{\mathbf{y}}(n)=\left[\hat{y}(n),...,\hat{y}(n-L+1)\right]^T$, $\hat{y}(n)$ is the M order prediction error of the input signal $y(n)$, $\hat{y}(n)=y(n)-\mathbf{a}_M^T(n)\mathbf{y}_M(n)$ and $\mathbf{a}_M(n)$ is the estimate coefficients of the forward predictor (LP) at time n. As described in [19] and [21] the Wiener Filter obtained from $\{\mathbf{y}(n), e(n)\}$ is the same with the one obtained from their corresponding decorrelated values $\{\hat{\mathbf{y}}(n), \hat{e}(n)\}$.

According to [15] a delay in estimation of the coefficients of LP might cause instability as well as a performance degradation of the adaptive filter in the AFC configuration. To overcome this, an early LP is necessary. It can be easily observed that $y_G(n)=e(n-\Delta)$. Therefore we can modify the standard DNLMS algorithm by removing the delay in LP and using e(n) signal instead of $y_G(n)$. The new algorithm is termed Modified DNLMS (MDNLMS) algorithm.

The filter update recursion for the MDNLMS algorithm is:

$$\mathbf{w}(n+1)=\mathbf{w}(n)+\frac{\mu\hat{e}(n)\hat{\mathbf{y}}(n)}{\hat{\mathbf{y}}^T(n)\hat{\mathbf{y}}(n)+\delta} \qquad (3)$$

It has many similarities with MPAP because it shares a similar approach [15] and [17-19]. Also it has almost similar numerical complexity with MPAP. The only difference is that the MPAP uses a weighted lattice predictor for updating the error vector of PAP [15], while MDNLMS computes a lattice predictor for updating its error vector.

Additionally, we propose to use an empirically found variable step size formulas inspired from the idea of [23] and given by the Eqs. 4 – 7:

$$\mu_c(n)=\mu_{max}*\left|1-\frac{\hat{\sigma}_d(n)}{\hat{\sigma}_e(n)+\xi}\right| \qquad (4)$$

$$\mu(n)=\begin{cases}\mu_{\max}, & \text{if } \mu_c(n)>\mu_{\max}\\ \mu_{\min}, & \text{if } \mu_c(n)<\mu_{\min}\\ \mu_c(n), & \text{otherwise}\end{cases} \qquad (5)$$

where $\mu_{\max}$ and $\mu_{\min}$ are the maximum and minimum allowed step sizes respectively, ξ is a small constant that avoids division by zero. The estimated variances from Eq. 4 are computed recursively as follows:

$$\hat{\sigma}_d^2(n)=\lambda\hat{\sigma}_d^2(n-1)+(1-\lambda)d^2(n) \qquad (6)$$

$$\hat{\sigma}_e^2(n)=\lambda\hat{\sigma}_e^2(n-1)+(1-\lambda)e^2(n) \qquad (7)$$

where λ is a constant close to one, computed as in [23] and the initial values are $\hat{\sigma}_e^2(0)=\hat{\sigma}_d^2(0)=0$. The bounds of the step size are chosen by trials in order to have a compromise between a minimum tracking ability, stability in the steady-state region and fast convergence. The proposed algorithm using the above VSS scheme is termed VSS-MDNLMS.

III. Simulation Results

The performance of the proposed algorithm for AFC was compared with that of NLMS and MPAP algorithms. The feedback path was modeled as a FIR with 64 coefficients and the adaptive filter had 64 coefficients too. Two cases were tested: a constant 25 dB gain and frequency-dependent gain. For the latter case, the gain is set to 10 dB under 0.5 kHz, to 15 dB between 0.5 kHz and 1 kHz, 20 dB between 1 kHz and 2 kHz, 25 dB between 2 kHz and 4 kHz, and 30 dB between 4 kHz and 8 kHz. The feedback path was changed after half of the input sequence length. The misalignment was used in order to measure the performance of the algorithms. The smoothing factor for the lattice algorithms was set to 0.992 and $\delta = 0.001$. The step size values were chosen in order to have a stable behavior and similar initial convergence speed for the investigated cases: $\mu_{\max} = 0.005$, $\mu_{\min} = 0.0005$, and $\mu_{MPAP} = \mu_{NLMS} = 0.004$.

For Figs. 2 and 3 an AR noise was generated by passing a white noise through a 10th order all-pole filter and the SNR of the background noise was 20 dB. The feedback echo change was simulated by inserting five very small values at the beginning and shifting the coefficients of the feedback path.

It can be seen that the misalignment performance of the VSS-MDNLMS algorithm is comparable to that of the MPAP algorithm. Also, the NLMS algorithm behaves better in the variable gain case than in the fixed gain case.

For Figs. 4 and 5 the AR noise was replaced by a speech sequence. Similar conclusions can be drawn regarding the behavior of MPAP and VSS-MDNLMS algorithms for both fixed and variable gain cases. It can be noticed that MPAP and VSS-MDNLMS algorithms obtains a lower misalignment values in case of fixed gain that in the case of variable gain case. Unlike for the AR noise case, the NLMS algorithm behaves better in the variable gain case than in the fixed gain case when applying a speech sequence. Fig. 6 shows the error signal, the input speech signal and the computed step sizes for the VSS-MDNLMS algorithm for the considered case. Similar conclusions can be obtained for different SNR values and confirm that the VSS-MDNLMS algorithm has increased robustness and stability for both fixed and variable gain cases.

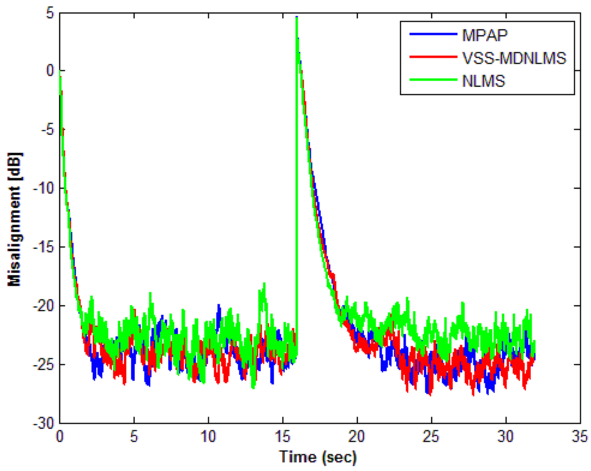

Figure 2. Misalignment curves for MPAP, VSS-MDNLMS and NLMS algorithms for the variable gain case and AR noise

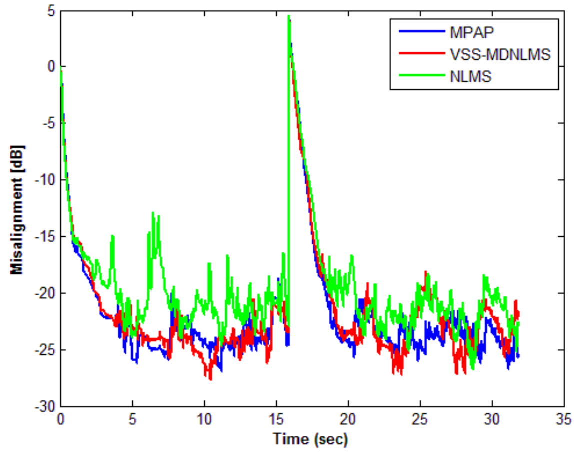

Figure 3. Misalignment curves for MPAP, VSS-MDNLMS and NLMS algorithms for the fixed gain case and AR noise

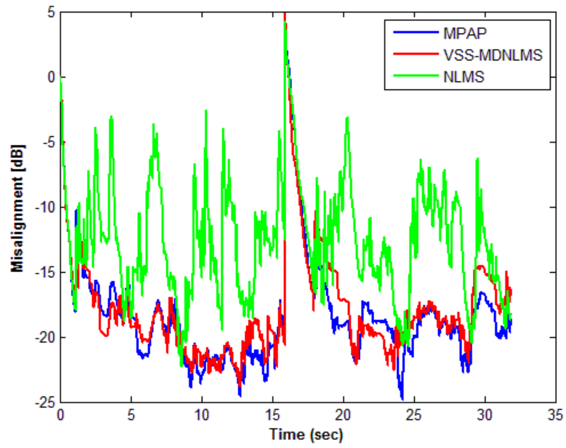

Figure 4. Misalignment curves for MPAP, VSS-MDNLMS and NLMS algorithms for the variable gain case and speech sequence

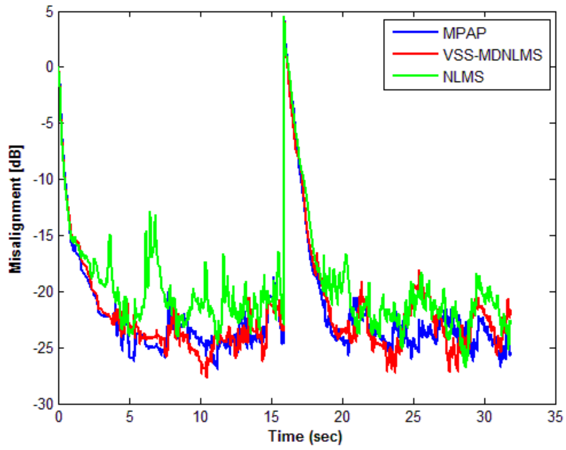

Figure 5. Misalignment curves for MPAP, VSS-MDNLMS and NLMS algorithms for the fixed gain case and speech sequence

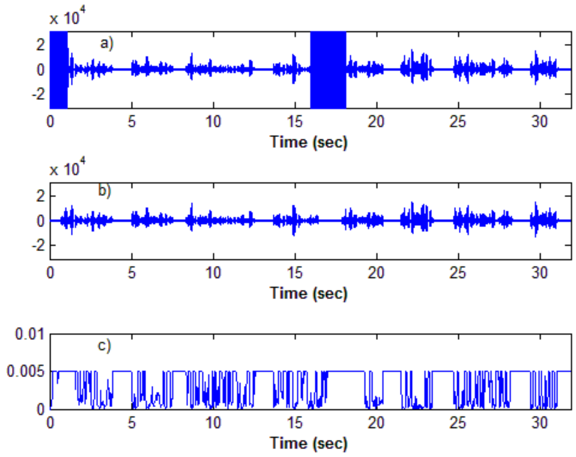

Figure 6. a) The error signal; b) the input signal; c) the step size of the VSS-MDNLMS algorithm

Future work will be focused in finding a better VSS scheme, in investigating the effect of the LP order on the VSS-MDNLMS based hearing aid performance and its PESQ-MOS score. Also, the suitability of some sign algorithms (e.g. [24]) for AFC will be investigated.

IV. CONCLUSIONS

In this paper, we propose a new adaptive feedback cancellation algorithm with reduced numerical complexity for digital hearing aids. The VSS-MDNLMS algorithm it is based on the D-NLMS algorithm and uses a variable step size scheme. Simulation results showed that the proposed algorithm has comparable convergence characteristics with MPAP.

ACKNOWLEDGMENT

This work was supported by a grant of the Romanian National Authority for Scientific Research, CNCS - UEFISCDI, project number PN-II-ID-PCE-2011-3-0097 and and by the Sectorial Operational Program "Human Resources Development" 2007-2013 of the Romanian Ministry of Labour, Family and Social Protection through the Financial Agreement POSDRU/107/1.5/S/76813.

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How to cite this paper if used in research publications:

M. Rotaru, F. Albu, H. Coanda, "A Variable Step Size Modified Decorrelated NLMS Algorithm for Adaptive Feedback Cancellation in Hearing Aids, " in *Proc. of ISETC 2012*, Timisoara, Romania, 15-16 November 2012, pp. 263-266.

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