A Combined Variable Step Size Strategy for Two Microphones Acoustic Feedback Cancellation using Proportionate Algorithms

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***Abstract*—This paper proposes to use a combination of the improved proportionate normalized least mean square (IPNLMS) algorithm and a new algorithm called combined-step-size IPNLMS (CSS-IPNLMS) algorithm in the two microphone closed-loop feedback cancellation technique. The simulation results indicate an improvement in convergence speed of the proposed algorithms as compared to previously suggested methods, especially for the case of the normal feedback path.**

I. INTRODUCTION

The well-known acoustic feedback problem of the hearing aids is caused by the loudspeaker signal that leaks into the microphone, could cause howling and reduces the maximum stable gain. The use of prediction filters for acoustic feedback cancellation (AFC) have been proposed in [1]-[3]. In these methods, the microphone and loudspeaker signals were prefiltered by an inverse signal model and then fed into the adaptive filter in feedback canceller path [1]-[3]. As a result, the bias problem in estimation of feedback coefficients is reduced, but the performance depends a lot on the accuracy of estimation of the incoming signal model.

A two microphone feedback cancellation scheme for behind the ear (BTE) hearing aids has been proposed (AFC-2mics) in [4]-[6]. The estimation of the first incoming signal is made using the second microphone signal. The error signal of the two-microphone structure is used by the adaptive filters. It was shown in [4]-[6] that the AFC-2mics has obtained better maximum stable gain (MSG) and misalignment if compared to the Prediction Error Method (PEM).

The Least Mean Square (LMS), the normalized LMS (NLMS), the affine projection (AP) algorithm and their versions have been widely used for various applications [7] – [13]. The application of AP and their fast variants for AFC in hearing aids so far has been limited due to lack of performance improvement in one microphone settings [13].

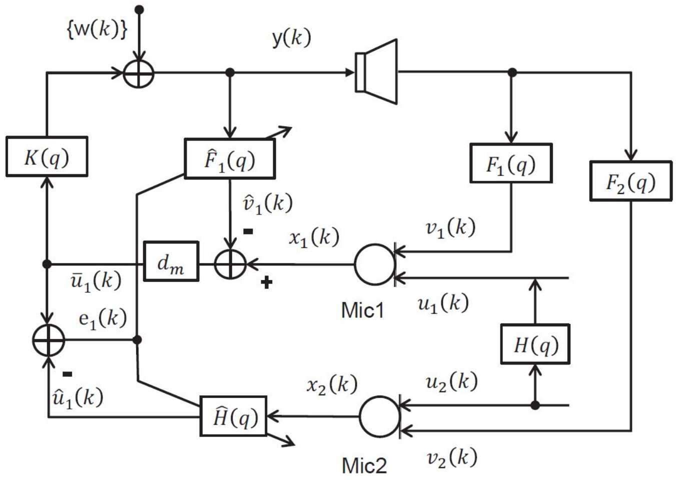
An approach for computational complexity reduction and robustness against impulsive noise is to use the affine projection sign algorithm (APSA) and its variants [14]-[17]. A promising approach was to combine a fast filter with a slow filter through the combined-step-size APSA (CSS-APSA) [18]. The larger step size led to a fast convergence rate while the smaller step size led to low steady-state misadjustment. The modified sigmoidal activation function was used for the variable mixing factor for the CSS-APSA [18]. However, the performance of the sign algorithms is not very good on AFC scenario [19]. Several papers have pointed to the superior performances of the proportionate algorithms for the AFC scenario [19]-[21]. In this paper we propose to keep the computational complexity low and focus on the use of the IPNLMS based algorithms. The same ideas from [18] are used in order to derive a new algorithm called CSS-IPNLMS algorithm. This algorithm is used to adapt the coefficients of the second microphone filter, $H(q)$, and derive the step size employed by the first microphone adaptive filter. The misalignment and the maximum stable gain (MSG) performance of the proposed combined variable step size (CVSS) version, the AFC-2mics-IPNLMS-CVSS, is compared with that of the AFC-2mics-NLMS and the AFC-2mics-NLMS using the variable Gaussian step-size (VGSS) approach proposed in [22].

This paper has four sections. Section II presents the proposed system and describes the application of the CSS-IPNLMS and IPNLMS in the AFC-2mics approach. Section III shows the simulation results and performance comparison with previously proposed solutions. Section IV provides the conclusions of the paper.

II. THE PROPOSED APPROACH AND ALGORITHM

Fig.1 illustrates the AFC using two microphones and one loudspeaker [4]. First microphone is placed in the ear (mic1), while the second microphone (mic2) is behind the ear. The condition $|F_2(q)| \ll |F_1(q)|$ is met, where q is the time shift operator, $F_1(q)$, $F_2(q)$ are transfer functions of the first and second feedback paths, respectively [6].

If k is the discrete-time variable, the feedback paths are $\mathbf{f}_1(k)$ and $\mathbf{f}_2(k)$, respectively. Their coefficients are $f_1(k-i)$, $i=0:L_f-1$ and $f_2(k-i)$, $i=0:L_f-1$, respectively, where L_f is FIR's length. The coefficients of the first adaptive filter $\hat{\mathbf{f}}_1(k)$ are $\hat{f}_1(k-i),\ i=0:L_f-1$ while $H(q)$ is a L_h – dimensional FIR filter whose elements are $h_i,\ i=0:L_h-1$ (see Fig. 1).



a.

Fig. 1. AFC-2mics [21]

It can be seen from Fig. 1 that $x_1(k)$ is the sum of first incoming signal, $u_1(k)$, and the first feedback signal $v_1(k)=F_1(q)y(k)$, while $x_2(k)$ is the sum of second incoming signal, $u_2(k)$, and the second feedback signal $v_2(k)=F_2(q)y(k)$ [4]-[6],[17]. The gain in the forward path is $|K|$ and d_m is a small delay [4]. The error signal $e_1(k)=x_1(k)-\hat{v}_1(k)-\hat{u}_1(k)$ controls the estimation of $F_1(q)$ and $H(q)$ [4]-[6].

In this paper we propose to use different adaptive algorithms for updating the coefficients of the adaptive filters. The IPNLMS algorithm [23] is employed to update $\hat{F}_1(q)$ as follows:

$$\hat{\mathbf{f}}_1(k)=\hat{\mathbf{f}}_1(k-1)+\frac{\mu(k)\mathbf{A}(k-1)\mathbf{y}(k)e_1(k)}{\mathbf{y}^T(k)\mathbf{A}(k-1)\mathbf{y}(k)+\delta_{IPNLMS}} \tag{1}$$

where $\mu(k)$ is a step-size, δ_{IPNLMS} is a regularization parameter, and the elements of $\mathbf{y}(k)$ collect previous $L_{\hat{f}}$ samples of $y(k)$, i.e. $y(k-i),\ i=0:L_{\hat{f}}-1$ and

$$\mathbf{A}(k-1)=diag\left\{a_0(k-1),a_1(k-1),...,a_{L_{\hat{f}}-1}(k-1)\right\}. \tag{2}$$

The diagonal matrix, $\mathbf{A}(k-1)$, allocates higher values for larger coefficients and smaller values for lower coefficients. The coefficients of the $\mathbf{A}(k-1)$ are computed as follows

$$a_l(k)=\frac{1-\alpha}{2L_{\hat{f}}}+(1+\alpha)\frac{\left|\hat{f}_l(k)\right|}{2\left\|\hat{\mathbf{f}}(k)\right\|_1+\varepsilon} \tag{3}$$

where $-1\le\alpha<1$, $0\le l<L_{\hat{f}}-1$ and ε is a small constant. The added computational complexity to the NLMS algorithm is small and proportional with the length of the filter.

The vector $\mathbf{x}_2(k)$ collects $L_{\hat{h}}$ previous samples of $x_2(k)$, i.e. $x_2(k-i),\ i=0:L_{\hat{h}}-1$.

For the second filter another diagonal matrix is computed, i.e.

$$\mathbf{B}(k-1)=diag\left\{b_0(k-1),b_1(k-1),...,b_{L_{\hat{h}}-1}(k-1)\right\}, \tag{4}$$

whose coefficients are computed as follows

$$b_l(k)=\frac{1-\alpha}{2L_{\hat{h}}}+(1+\alpha)\frac{\left|\hat{h}_l(k)\right|}{2\left\|\hat{\mathbf{h}}(k)\right\|_1+\varepsilon}, \tag{5}$$

where $0\le l<L_{\hat{h}}-1$.

The CSS scheme described in [18] combines two step sizes (a large step size μ_1 and a small one μ_2). The combined step size [18] is

$$\mu(k)=\left[\mu_1\varphi(k)+\mu_2\left(1-\varphi(k)\right)\right], \tag{6}$$

where the variable mixing factor $\varphi(k)$, is updated by using a sigmoidal activation function as follows [18]

$$\varphi(k)=C\left(1+e^{-\gamma(k)}\right)^{-1}-(C-1)/2, \tag{7}$$

where C is a positive constant higher than 1. If we denote $\chi=\ln\left((C+1)/(C-1)\right)$, the parameter $\gamma(k)$ is limited in value as

$$\gamma(k)=\begin{cases}-\chi & \text{if } \gamma(k)<-\chi\\ \chi & \text{if } \gamma(k)>\chi\\ \gamma(k) & \text{otherwise}\end{cases}, \tag{8}$$

and updated by using the gradient descent method in order to minimize the square error. After some calculations, following the same principles from [18], [24], [25] the following updating formula is obtained

$$\begin{aligned}\gamma(k)&=\gamma(k-1)+\mu_\gamma\left[\varphi(k-1)\left(1-\varphi(k-1)\right)+\varepsilon\right]\\&\times(\mu_1-\mu_2)\frac{e(k)e(k-1)\mathbf{x}_2^T(k)\mathbf{B}(k-2)\mathbf{x}_2(k-1)}{\mathbf{x}_2^T(k-1)\mathbf{B}(k-2)\mathbf{x}_2(k-1)+\delta_{IPNLMS}},\end{aligned} \tag{9}$$

where μ_γ is a step size parameter. Finally, the update equation of the weight coefficient vector for the second filter $\hat{H}(q)$ using the proposed CSS-IPNLMS algorithm is

$$\begin{aligned}\hat{\mathbf{h}}(k)&=\hat{\mathbf{h}}(k-1)+\mu(k)\mathbf{B}(k-1)\mathbf{x}_2(k)\\&\times\left(\mathbf{x}_2^T(k)\mathbf{B}(k-1)\mathbf{x}_2(k)+\delta_{IPNLMS}\right)^{-1}e(k).\end{aligned} \tag{10}$$

III. Simulation Results

In order to implement simulations, the same feedback path characteristics measured for both scenarios of normal and closest feedback paths from [22] are used. The real speech which is constructed by concatenating real male and female speech patterns extracted from Noizeus database [22] is used as the speech source. The MSG and misalignment (MisAL) are used to evaluate the performance of the AFC system. They are defined as follows [22]

$$\mathrm{MSG} = 20\log_{10}\left(\min_{\omega}\frac{1}{\left|F(\omega)-e^{-j\omega d_{fb}}\hat{F}(\omega)\right|}\right), \quad (11)$$

$$\mathrm{MisAL} = 20\log_{10}\frac{\int_0^{\pi}\left|F(\omega)-e^{-j\omega d_{fb}}\hat{F}(\omega)\right|d\omega}{\int_0^{\pi}\left|F(\omega)\right|d\omega}, \quad (12)$$

where d_{fb} is a delay in the feedback canceller's path.

In the simulations a white Gaussian noise, SNR = 30 dB is used as probe noise. For the first 40 seconds the normal feedback path was used while for the next 40 seconds the closest feedback path was employed.

The following parameters are used as in [22]: the delay $d_k = 32$ samples, d_{fb} =16 samples, $|K|$=30 dB, d_m =1, $L_{\hat{h}}$ =10, L_f =38, $L_{\hat{f}}$ =22 and the sampling rate f_s =16 kHz. The characteristics of measured normal and closest feedback paths used in this paper are shown in Fig. 2.

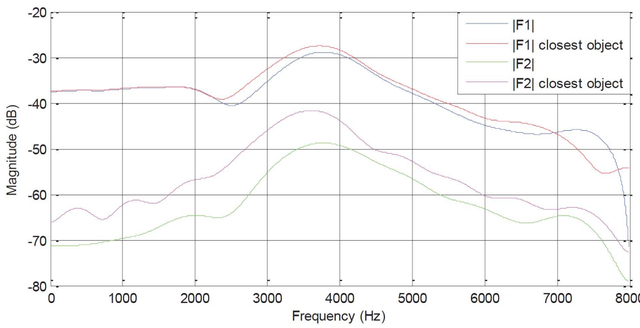


Fig. 2. Characteristics of measured feedback paths [22]

Fig. 3 illustrates the MSG comparison for different step sizes of the AFC-2mics-IPNLMS algorithm for both situations with/without probe noise. Two step-size values were used $\mu = 0.00025$ and $\mu = 0.0005$, respectively. The following parameters were used for the IPNLMS based algorithms $\delta_{IPNLMS} = \varepsilon = 10^{-8}$ and $\alpha = -0.5$. It can be noticed from Fig. 3 that the step size μ = 0.0005 achieves better MSG performance for both situations and most of the time regardless if the probe noise is used or not. The average MSG improvement for both situations is about 1.4 dB. This step-size value is used for the AFC-2mics-IPNLMS for the following simulations.

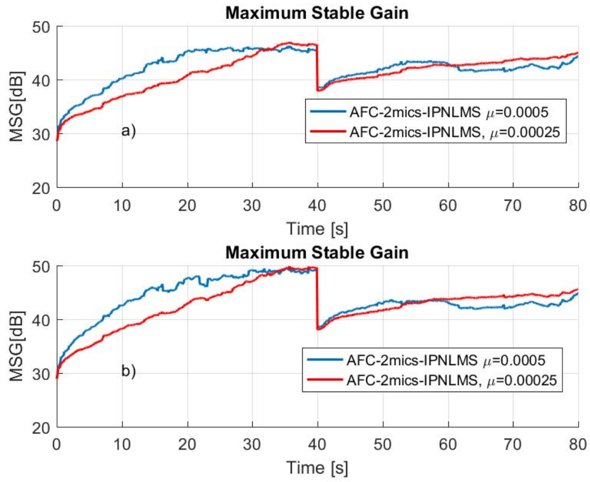


Fig. 3. a) MSG performance of AFC-2mics-IPNLMS without probe noise for two step values; b) MSG performance of AFC-2mics-IPNLMS with probe noise for two step values;

Fig. 4 illustrates the MSG comparison for a different minimum step size of the AFC-2mics-IPNLMS-CSS algorithm for both situations with/without probe noise. Two step-size values were used for μ_2 in both cases, i.e., μ_2 = 0.00025 and μ_2 = 0.0005, respectively. In the case without probe noise we used μ_1 = 0.0015, while μ_1 = 0.001 in case of the example with probe noise. The limiting constant was C = 4 and $\mu_\gamma = 100$. It can be noticed from Fig. 4 that the step size μ_2 = 0.00025 leads to better average MSG performance for both situations. The average improvement over that of using μ_2 = 0.0005 is about 0.5 dB. This value is used for the next performance comparison of the investigated algorithms.

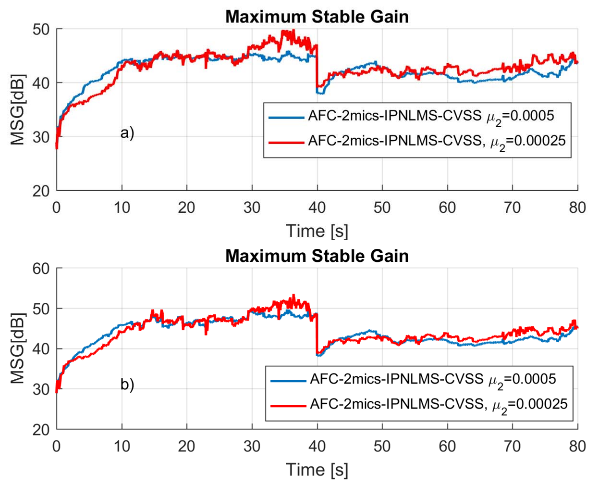


Fig. 4. a) MSG performance of AFC-2mics-IPNLMS-CVSS without probe noise for two step-size values; b) MSG performance of AFC-2mics-IPNLMS-CVSS with probe noise for two step-size values.

The effect of the C value on the MSG and misaligment performance of the AFC-2mics-IPNLMS-CVSS algorithm is shown in Table 1. It can be seen that the best average MSG and misalignment performance is obtained for $C = 4$. Therefore, this value is used for the next simulations.

TABLE I. THE AVERAGE VALUES OF MSG AND MISALIGNMENT FOR DIFFERENT C VALUE

C	0.1	1	2	4	10
Average MSG	42.31	42.31	42.31	**43.72**	43.64
Average Misalignment	-13.12	-13.12	-13.12	**-14.5**	-14.36

The misalignment and MSG performance of the AFC-2mics-NLMS, AFC-2mics-IPNLMS, AFC-2mics-IPNLMS-CVSS and AFC-2mics-NLMS-VGSS algorithms are shown in Fig. 5 (without probe noise) and Fig. 6 (with probe noise). The step sizes for AFC-2mics-NLMS-VGSS are the same with those of AFC-2mics-IPNLMS. It can be seen from Figs. 5 and 6 that the AFC-2mics-IPNLMS-CVSS algorithm has faster convergence and better performance especially for the first normal feedback path. When the feedback path becomes stronger (corresponding to the closest feedback path) the proposed algorithm has an initial better MisAL and MSG than the AFC-2mics-NLMS. When the probe noise is injected into the loudspeaker signal, it further decouples the loudspeaker and incoming signal. The proposed method still has faster convergence than the competing algorithms in the case of probe noise with SNR = 30dB, especially for the case of normal feedback path. However, in the case of closest feedback path there is a small performance loss on MisAL and MSG compared to the competing algorithms around time 52s-68s when the incoming signal is the male speech that follows after the female speech.

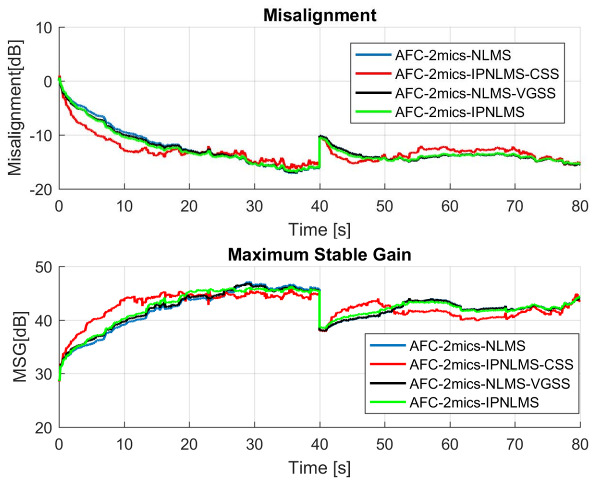


Fig. 5. Performance of AFC-2mics-NLMS, AFC-2mics-IPNLMS, AFC-2mics-IPNLMS- CVSS and AFC-2mics-NLMS-VGSS without probe noise.

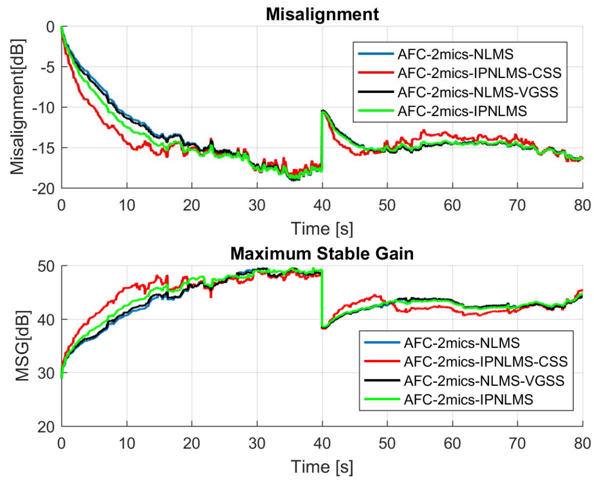


Fig. 6. Performance of AFC-2mics-NLMS, AFC-2mics-IPNLMS, AFC-2mics-IPNLMS-CVSS and AFC-2mics-NLMS-VGSS with probe noise.

Future work will be focused in investigating the suitability of the method from [26] for our two-microphone approach.

IV. CONCLUSIONS

In the investigated two-microphone approach for acoustic feedback cancellation, the IPNLMS algorithm and the proposed CSS-IPNLMS algorithm were chosen to adapt the first microphone adaptive filter $\hat{F}_1(q)$ and the second microphone adaptive filter $\hat{H}(q)$, respectively. The variable step size is determined by the proposed CSS-IPNLMS algorithm. It is shown that the proposed solution for the AFC-2 mics is able to obtain better initial convergence rate and stable solution in case of normal feedback path than previously suggested AFC-2mics-NLMS or AFC-2mics-NLMS-VGSS solutions.

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