

LETTER

An Effective Acoustic Feedback Cancellation Algorithm Based on the Normalized Sub-Band Adaptive Filter

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SUMMARY In this letter, an effective acoustic feedback cancellation algorithm is proposed based on the normalized sub-band adaptive filter (NSAF). To improve the confliction between fast convergence rate and low misalignment in the NSAF algorithm, a variable step size is designed to automatically vary according to the update state of the filter. The update state of the filter is adaptively detected via the normalized distance between the long term average and the short term average of the tap-weight vector. Simulation results demonstrate that the proposed algorithm has superior performance in terms of convergence rate and misalignment.

key words: sub-band adaptive filters, acoustic feedback cancellation, hearing aids, variable step size

1. Introduction

Digital hearing aids are widely used by hearing-impaired people, and it is mainly used to improve the speech intelligibility and quality. However, the speech quality is susceptible to the acoustic feedback, especially when the acoustic feedback forms howling. In order to minimize the effect of the acoustic feedback on the hearing aids, many acoustic feedback cancellation (AFC) algorithms were proposed in the literature [1], [2]. In particular, the normalized least mean square (NLMS) algorithm is the most popular one to estimate the feedback path due to the simplicity of operation [3]. However, the NLMS algorithm intrinsically suffers from slow convergence rate when the input signal is colored [4]. The sub-band adaptive filter (SAF) algorithms were presented to solve the problem [5]. In conventional SAF algorithms, each sub-band has its own sub-filter and adaptation loop, which decreases the convergence rate of the SAFs due to the aliasing and band edge effects. To solve the structural problems, the Normalized SAF (NSAF) algorithm based on minimum disturbance was derived to improve the convergence rate while remaining the computational efficiency [6]. Recently, the SAF algorithms have become the subject of intensive investigation [7], [8]. For the SAF algorithms, it is inevitable to compromise between fast convergence rate and low misalignment. Typically, a

large step size benefits fast convergence rate but fails to derive low misalignment. On the contrary, a small step size is helpful to low misalignment but not to fast convergence rate. Therefore, several variable step size SAF (VSS-SAF) algorithms were developed to improve the confliction between fast convergence rate and low misalignment [9], [10]. Furthermore, VSS-SAF algorithm was raised for AFC in hearing aids [11]. However, these approaches still fail to capture the low misalignment as well as fast convergence rate, etc.

Based on the above analysis, a novel variable step size normalized sub-band adaptive filter (VSS-NSAF) algorithm for AFC in hearing aids is proposed, aiming to further improve the confliction between fast convergence rate and low misalignment to obtain faster convergence rate and lower misalignment than the conventional algorithms. In contrast to the traditional VSS-SAF algorithm, the step size of the proposed algorithm adaptively varies according to the update state of the filter. Simulation results validate the effectiveness of the proposed algorithm for AFC in hearing aids.

2. VSS-NSAF Algorithm for Acoustic Feedback Cancellation

2.1 Acoustic Feedback Cancellation System Model Based on VSS-NSAF

The block diagram of the proposed AFC system based on VSS-NSAF algorithm is illustrated in Fig. 1. A delay-less NSAF structure is utilized in the system because the delay in currently available hearing aids is typically no more than 10ms [12]. Sub-band signal processing is remained within an auxiliary loop in parallel with the main signal path from the microphone to the receiver of the adaptive system, avoiding the delay produced in the adaptive procedure. In the AFC system, the forward path $G(z)$ maps the microphone signal $d(n)$ to the receiver signal $u(n)$, where $d(n)$ is the sum of external signal $x(n)$ and the feedback signal $p(n)$. In the adaptive loop, both $d(n)$ and $u(n)$ are partitioned into N sub-bands signals by means of analysis filters $H_i(z)$, and they are denoted as $d_i(n)$ and $u_i(n)$, respectively. The sub-band signals are then decimated by a factor N , and $F(z)$ denotes the feedback path. The tap weights of the filter updated via the VSS-NSAF algorithm are directly copied to a full-band filter $W(z)$. The estimation of the feedback signal $p(n)$ is produced by the VSS-NSAF and then subtracted from the microphone signal $d(n)$.

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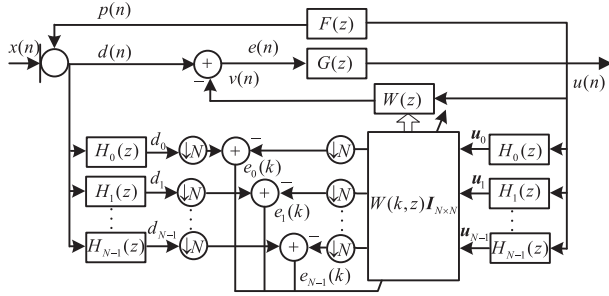


Fig. 1 Acoustic feedback cancellation system based on VSS-NSAF algorithm

In the sub-band AFC system model depicted in Fig. 1, $W(k, z)\mathbf{I}_{N \times N}$ denote a set of parallel adaptive filters with identical transfer function $W(k, z) = \sum_{n=0}^{M-1} w_n(k)z^{-n}$, where $\mathbf{I}_{N \times N}$ represents the $N \times N$ identity matrix, and M is the length of the filter $W(k, z)$. The tap weights of the filter are given as

$$\mathbf{w}(k) = [w_0(k), w_1(k), \dots, w_{M-1}(k)]^T \quad (1)$$

where the superscript T denotes matrix transposition. The sub-band error signal $\mathbf{e}(k) = [e_0(k), e_1(k), \dots, e_{N-1}(k)]^T$ is expressed as

$$\mathbf{e}(k) = \mathbf{d}(k) - \mathbf{U}^T(k)\mathbf{w}(k) \quad (2)$$

where $\mathbf{d}(k) = [d_0(k), d_1(k), \dots, d_{N-1}(k)]^T$, $\mathbf{U}(k) = [\mathbf{u}_0(k), \mathbf{u}_1(k), \dots, \mathbf{u}_{N-1}(k)]$, $\mathbf{u}_i(k) = [u_i(kN), u_i(kN-1), \dots, u_i(kN-M+1)]^T$, $i = 0, 1, \dots, N-1$. The tap weights of the filter $W(k, z)$ are iteratively updated by

$$\mathbf{w}(k+1) = \mathbf{w}(k) + \mu(k)\mathbf{U}(k)\mathbf{\Lambda}^{-1}(k)\mathbf{e}(k) \quad (3)$$

where $\mu(k)$ denotes the step size at instant k , $\mathbf{\Lambda}(k)$ is a diagonal matrix of normalization factor, $\mathbf{\Lambda}(k) = \text{diag}[\lambda_0(k), \lambda_1(k), \dots, \lambda_{N-1}(k)]$. According to the principle of minimal disturbance [13] and the diagonal assumption [14], $\mathbf{\Lambda}(k)$ can be written as

$$\mathbf{\Lambda}(k) = \text{diag}[\mathbf{U}^T(k)\mathbf{U}(k) + \alpha\mathbf{I}] \quad (4)$$

i.e., $\lambda_i(k) = \mathbf{u}_i^T(k)\mathbf{u}_i(k) + \alpha$, $i = 0, 1, \dots, N-1$, where α is a small positive constant to avoid possible division by zero. Substituting (4) into (3), the update formula of the tap weights can be expressed as

$$\mathbf{w}(k+1) = \mathbf{w}(k) + \mu(k) \sum_{i=0}^{N-1} \mathbf{u}_i(k) [\mathbf{u}_i^T(k)\mathbf{u}_i(k) + \alpha]^{-1} e_i(k) \quad (5)$$

The tap weights of the filter are updated once for every N input samples $u_i(n)$, where $kN = n$.

2.2 Step Size Control Strategy

In AFC system presented in Fig. 1, a variable step size for NSAF is proposed to improve the convergence rate and

lower the misalignment. The step size varies according to the update state of the filter. The update state of the filter is detected by the normalized distance between the long term average $\mathbf{A}_l(n)$ and the short term average $\mathbf{A}_s(n)$ of the tap-weight vector. Each parameter is estimated using a one-pole low pass filter with an averaging constant. The constant of the long term average is λ_l , $0 < \lambda_l < 1$. $\mathbf{A}_l(n)$ is defined as

$$\mathbf{A}_l(n) = \lambda_l \mathbf{A}_l(n-1) + (1 - \lambda_l) \mathbf{w}(n) \quad (6)$$

The constant of the short term average is λ_s , $0 < \lambda_s < \lambda_l < 1$. $\mathbf{A}_s(n)$ is defined as

$$\mathbf{A}_s(n) = \lambda_s * \mathbf{A}_s(n-1) + (1 - \lambda_s) \mathbf{w}(n) \quad (7)$$

A distance vector $\mathbf{D}(n)$ between the long term average and the short term average is expressed as

$$\mathbf{D}(n) = \mathbf{A}_l(n) - \mathbf{A}_s(n) \quad (8)$$

The normalized distance is expressed as

$$K(n) = \mathbf{D}^T(n)\mathbf{D}(n) / (\mathbf{A}_l^T(n)\mathbf{A}_l(n) + \varepsilon) \quad (9)$$

where ε is a small positive number to avoid the possible division by zero. If the long term average differs significantly from the short term average, $K(n)$ is large. This indicates that the update state is convergent. On the contrary, if the long term average differs slightly from the short term average, $K(n)$ is close to zero. This indicates that the update state is steady. Therefore, the update state can be categorized into convergent state, transitional state and steady state by imposing two different thresholds δ_1 and δ_2 to $K(n)$.

Different step sizes are adopted in different update states. An initialized large step size is adopted in convergent state to guarantee fast convergence rate. While in transitional state, a declining ladder-like step size is employed to further lower the misalignment. The ladder-like step size is given as

$$\mu(n+1) = \mu_0 / 2^{\lfloor (n-n_0)/5M \rfloor} \quad (10)$$

where $\lfloor \bullet \rfloor$ denotes rounding towards minus infinity, n_0 is the number of the sample when the update state transforms from convergent state to transitional state. A small step size benefits the convergence performance so that a low misalignment near the optimum can be obtained, hence, the step size in steady state is given as

$$\mu(n+1) = \mu(n_1) \quad (11)$$

where n_1 is the number of the sample when the update state transforms from transitional state to steady state. There are basically two advantages to adopt the above step size. On the one hand, high misalignment induced by a large step size can be lowered in steady state. On the other hand, slow convergence rate induced by a small step size can be accelerated in convergent state.

2.3 Computational Complexity

In this section, the computational complexity of the conventional NLMS [4], NSAF [6], INSFAF [8], VSS-SSAF [10] and the proposed algorithm will be briefly compared. The

computational complexity of the NLMS algorithm is proportional to M and denoted as $O(M)$, while the complexities of the NSAF, INSAF, VSS-SSAF and the proposed algorithm are all $O(M)+O(NL)$. It is noteworthy that the computational complexity of the proposed algorithm increases slightly in comparison to NLMS algorithm and is generally in the same order compared with other SAF algorithms. Due to the parallel processing of analysis filters, the time cost of the proposed algorithm in the hardware operation will be decreased. Thus its time complexity will be slightly higher than that of the conventional NLMS algorithm.

3. Experimental Results

3.1 Calculation of Normalized Distance between Long Term Average and Short Term Average

Kates' work showed that the performance of AFC was impacted by room reflection in hearing aids. If the filter is not long enough to include the time delays corresponding to the major room reflections, the performance of the AFC will not be improved [15]. Therefore, the long acoustic feedback path [1] resampled to 8k Hz is utilized in our experiment.

In the first simulation, the relationship between the number of sub-bands and the misalignment of the AFC system based on NSAF algorithm is displayed. A linear FIR model with 1001 tap weights is employed for AFC. The performance of the AFC system is evaluated by the misalignment which is defined as [16]

$$mis(n) = 20 \log 10 \frac{\|\hat{f}(n) - f\|_2}{\|f\|_2} \quad (12)$$

where f represents the true feedback path, \hat{f} represents the estimated feedback path. The input signal is obtained by filtering a white, zero mean, Gaussian random sequence through a first order system $G(z) = 1/(1 - 0.95z^{-1})$. The forward gain G is set to 45. A delay of 1ms is inserted in the forward path to decorrelate the couple between the microphone and the receiver. Considering the trade-off between the convergence rate and the misalignment, the step size μ is set to 0.2. Other parameters in the NSAF algorithm are set to $\alpha = 1 - (N/8L)$, $L = 8N$ [5]. The same parameter which will be used in other algorithms is set to the same value as the outlined. Quadrature mirror filter is utilized as the prototype filter. The analysis filters are cosine modulated versions of the prototype filter. Figure 2 shows the misalignment curves obtained by ensembling averaging over 50 independent trials. It is worth noting that the misalignment increases at first and then decreases with the increasing of the number of sub-bands. Considering the system performance and computational complexity, the number of sub-bands is chosen as $N=32$.

The second simulation illustrates the normalized distance between the long term average and the short term average. To obtain fast convergence rate, the initial step size μ_0 is set to 1. The parameters used to compute the normalized distance are $\lambda_l = 0.99$, $\lambda_s = 0.75$, and the detailed illustra-

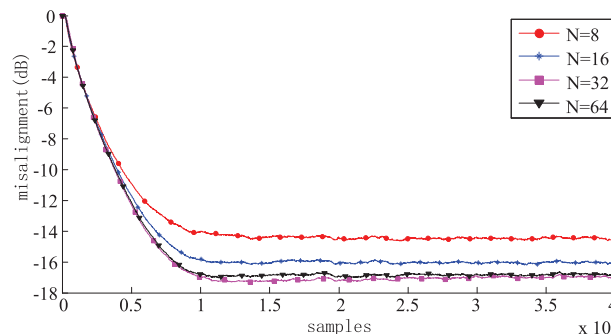


Fig. 2 Misalignment curves with different number of sub-bands

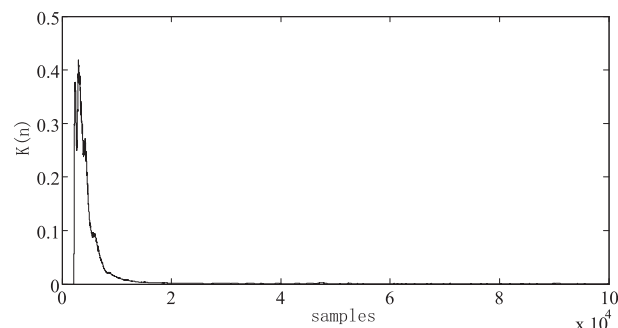


Fig. 3 Normalized distance between long term average and short term average

tion is given in the Sect. 3.2. As shown in Fig. 3, the normalized distance curve increases firstly and then decreases. Hence, it can be divided into three parts by using two thresholds δ_1 and δ_2 . If the normalized distance is greater than δ_1 , it implicates that the filter is in convergent state. If the normalized distance is smaller than δ_2 , it implicates that the filter is in steady state. Otherwise, the filter is in transitional state.

3.2 Performance Analysis of Acoustic Feedback Cancellation Algorithms

In this experiment, the performance of the the NLMS [4], NSAF [6], INSAF [8], VSS-SSAF [10] and the proposed algorithm for AFC in hearing aids are compared. For a fair comparison, the step sizes of the NLMS, NSAF and INSAF algorithm are all set to 0.2, and all of the initial step sizes μ_0 of VSS-SAF algorithm are set to 1 for fast convergence rate. The parameter P in INSAF algorithm is set to 2 [8]. In VSS-SSAF algorithm, the lower bound of the step size μ_L is set to 10^{-5} [10]. The parameters to be determined for the proposed algorithm are λ_l , λ_s , δ_1 and δ_2 . Considering that different feedback path leads to different maximum of $K(n)$, convergence rate and misalignment, the respective thresholds of δ_1 and δ_2 are constructed as $\max(K(n))/(\beta M)$ and $\max(K(n))/(\gamma M)$ in order to detect the update of the filter, where $\gamma > 0$, $0 < \beta < \gamma$. The parameters λ_l , λ_s , β and γ is determined by the following steps. Firstly, the parameter γ is specified as 1 based on the amount of preliminary experiments with different input speech and simulated feed-

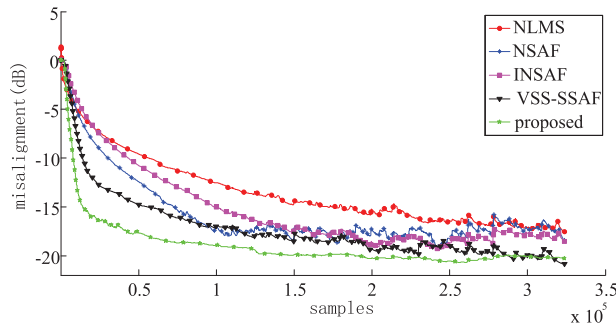


Fig. 4 Misalignment curves of NLMS, NSAF, INSAF, VSS-SSAF and proposed algorithm

back paths. Secondly, with the fixed γ , the possible range of values of parameters λ_l , λ_s and β are exhausted by the numerical mesh grid method. Finally, the optimal parameters for AFC system are determined as $\lambda_l = 0.99$, $\lambda_s = 0.75$ and $\beta = 0.02$ based on the average misalignment. The maximum of $K(n)$, i.e., $\max(K(n))$, is obtained by a sliding window including M samples. It is the maximum of the last window when the maximum in the present window starts to decrease.

From the misalignment curves in Fig. 4, it is clear that the NLMS algorithm displays a slow asymptotic convergence after a fast initial convergence because the input signal has a large spectral dynamic range. However, the SAF algorithms display consistently faster convergence rate than that of the NLMS algorithm due to the reduced spectral dynamic range. Among the family of SAF algorithms, the INSAF algorithm has lower misalignment but slower convergence rate than the NSAF algorithm because the INSAF algorithm uses the past tap weights during updating, and the VSS-SSAF algorithm has faster convergence rate and lower misalignment than the NSAF algorithm. It is noteworthy that the proposed VSS-NSAF algorithm has the fastest convergence rate and the lowest misalignment compared with the conventional algorithms. This is due to the adaptive control to the step size according to the update state of the filter. In addition, the running time of each algorithm is tested on a 2.4 GHz Pentium 4 machine using MATLAB R2014a. The time complexity of the NLMS, NSAF, INSAF, VSS-SSAF and our proposed algorithm for processing 20s speech are 32.53s, 53.24s, 82.26s, 85.03s and 86.02s, respectively. This indicates that although the computational complexity of the proposed algorithm increases, it still has comparable results compared with other competing algorithms.

4. Conclusions

In this letter, an effective AFC algorithm based on the NSAF for hearing aids is proposed. The fundamental idea of the new algorithm is to adjust the step size according to the update state of the filter. The update state of the filter is determined by the normalized distance between the long term average and the short term average. Simulation results confirm that the proposed algorithm achieves smaller misalignment as well as faster convergence rate than other competing al-

gorithms although the computational complexity increases slightly.

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