Integrated Feedback and Noise Reduction Algorithm In Digital Hearing Aids Via Oscillation Detection

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Abstract

In this paper, an integrated feedback and noise reduction scheme in hearing aids is developed. The technique presented is based on the adaptive feedback cancellation (AFC) and general sidelobe canceller (GSC) with a band-limited adaptation method to better the convergence behavior of both AFC and GSC. The band pass pre-filter is applied to AFC and the band stop pre-filter is applied to GSC to increase the portion of desired signal. An oscillation detector based on the zero crossing rates of the autocorrelation of sub-band signals is designed to calculate the center frequency of the oscillation to make the band-limited adaptation more robust. Convergence analysis and computer simulation illustrate that the proposed algorithm performs effectively to reduce the feedback and noise.

Index Terms: Feedback cancellation, digital hearing aids, general sidelobe canceller, oscillation detection

1. Introduction

Hearing aids are commonly used to amplify the input signal to compensate the deficit in auditory pathways. An undesired oscillation or howling is usually generated by the existence of the feedback path which is mainly caused the vent of the hearing aid. It can be efficiently solved by creating an internal feedback path to predict the feedback signal. A typical digital hearing aid includes a microphone, a forward path gain, a receiver, and an internal adaptive filter to simulate the feedback path. To better the sound quality and convergence behavior, many methods were developed such as prediction error method (PEM) [1] which utilizes linear predictive coding (LPC) filters to whiten the speech signal and band-limited adaptive feedback canceller with a filtered-X least mean-square (LMS) algorithm [2] [3]. In contrast to the conventional wide-band feedback cancellation method, the band-limited adaptive filter is constrained to operate only in those regions known as oscillation frequencies, thus the filter would be more efficient in canceling oscillation, and introduce less distortion than the wide-band approach. Accordingly, the pre-filter is made to be band-limited whose center frequency can be configured by an oscillation detector through real-time calculation.

Apart from the oscillation which is brought by feedback path, the interfering noises also bring many problems to the hearing aids such as the performance degradation of the adaptive feedback cancellation and sound quality deterioration. To solve these problems, especially in the circumstance of microphone array application, algorithms such as general sidelobe canceller are usually applied and have been proved to be effective [4] [5].

2. Integrated feedback and noise reduction with band-limited adaptation

The block diagram of an integrated band-limited feedback and noise reduction structure is shown in Figure 1.

![Figure 1: The block diagram of integrated band-limited feedback and noise reduction algorithm](image)

The frequency domain normalized least mean-square (FNLMS) algorithm is applied to AFC and GSC in this section, the explicit explanation of the details of Figure 1 is shown in Table 1.

<table>
<thead>
<tr>
<th>Module/Signal</th>
<th>Explanation</th>
<th>Size</th>
</tr>
</thead>
<tbody>
<tr>
<td>$W(k)$</td>
<td>AFC</td>
<td>$1 \times M$</td>
</tr>
<tr>
<td>$R(k)$</td>
<td>Feedback path</td>
<td>$N \times M$</td>
</tr>
<tr>
<td>$M(k)$</td>
<td>Fixed beamformer</td>
<td>$N \times M$</td>
</tr>
<tr>
<td>$B$</td>
<td>Blocking matrix</td>
<td>$N \times N - 1$</td>
</tr>
<tr>
<td>$C(k)$</td>
<td>Interference canceller</td>
<td>$N - 1 \times M$</td>
</tr>
<tr>
<td>$L(k)$</td>
<td>Band pass pre-filter</td>
<td>$1 \times M$</td>
</tr>
<tr>
<td>$L'(k)$</td>
<td>Band stop pre-filter</td>
<td>$1 \times M$</td>
</tr>
<tr>
<td>$x(n)$</td>
<td>Output of hearing aid</td>
<td>$1 \times M$</td>
</tr>
<tr>
<td>$d(n)$</td>
<td>Desired signal of AFC</td>
<td>$1 \times M$</td>
</tr>
</tbody>
</table>

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\[ e_s(n) \] error signal of AFC, \( 1 \times M \)  
\[ e(n) \] error signal of GSC, \( 1 \times M \)  
\[ s(n) \] input speech signal, \( N \times M \)  
\[ v(n) \] interfering noise, \( N \times M \)  
\[ q(n) \] output of \( B(k) \), \( N \times (N-1) 	imes M \)  
\[ G(k) \] forward path, \( 1 \times M \)  
\[ LMS \] least mean-square operation.

The power frame of error signal \( e \) in frequency domain is converted to diagonal matrix as follows:

\[
E^{(e)} = \text{diag} \begin{bmatrix} e(nM+1) \\
\vdots \\
e((n+1)M) \end{bmatrix}
\tag{1}
\]

Where \( F \) denotes the Fourier operator, \( E^{(e)} \) denotes the result of \( E^{(e)} \) after \( \text{diag}(\cdot) \) operation. It can be deduced as

\[
E^{(e)} = \left( \sum_{i=1}^{N} R_i (M_i - (BC^{(e)})) - W^{(e)} \right) X^{(e)}
\tag{2}
\]

\[
+ \sum_{i=1}^{N} (M_i - (BC^{(e)}))V^{(e)} + S^{(e)}
\]

Where the subscript \( i \) is \( R_i \), represents \( i \)-th row of \( R \).

To make ideal feedback cancellation and noise reduction, we can conclude the following two equations,

\[
\sum_{i=1}^{N} R_i (M_i - (BC^{(e)})) - W^{(e)} = 0
\tag{3}
\]

\[
\sum_{i=1}^{N} (M_i - (BC^{(e)}))V^{(e)} = 0
\tag{4}
\]

It can be observed from Equation (2) that the GSC also takes the feedback signal as a kind of interference, and steers a null to the direction of the receiver by adapting interference canceller. The following equation will be satisfied,

\[
\sum_{i=1}^{N} R_i (BC^{(e)}), X^{(e)} = 0
\tag{5}
\]

Under this assumption, (3) can be rewritten as

\[
\sum_{i=1}^{N} R_i M_i - W^{(e)} = 0
\tag{6}
\]

The updating process of AFC and interference canceller can be depicted as follows

\[
W^{(e+1)} = W^{(e)} + \mu \left[ L^T \left( E^{(e)} X^{(e)} + \delta I \right)^2 \right]^{-1}
\tag{7}
\]

\[
C^{(e+1)} = \delta I + \mu \left[ L^T \left( E^{(e)} Y^{(e)} + \delta I \right)^2 \right]^{-1}
\tag{8}
\]

Where \( 1 \leq m \leq N-1 \), \( \delta_i \) and \( \delta \) are constants.

Assuming \( W \) is the weight error of \( W \), as a result it can be defined according to (6),

\[
\bar{W}^{(e)} = W^{(e)} - \sum_{i=1}^{N} R_i M_i
\tag{9}
\]

Substitutes (9) to (7), the following equation can be derived,

\[
\bar{W}^{(e+1)} = (I - A_1) \bar{W}^{(e)} + A_2
\tag{10}
\]

Where,

\[
A_1 = \mu \left[ L^T \left( E^{(e)} X^{(e)} + \delta I \right)^2 \right]^{-1}
\tag{11}
\]

\[
A_2 = \mu \left[ L^T \left( E^{(e)} Y^{(e)} + \delta I \right)^2 \right]^{-1}
\tag{12}
\]

represents the steady-state error of \( W \). Noting that the energy of \( X \) far outweigh that of \( V \) and \( S \) in the band of oscillating frequencies, \( A_1 \) can be reduced by choosing \( L \) to be a bandpass filter whose pass band locates around the center frequency of oscillation. Besides, for the reason of \( A_1 \) increases with \( L X^{(e)} \), it can be maximized by configuring the center frequency of pass band of \( L \) with the detected frequency of oscillation when the bandwidth of the pass band of \( L \) is fixed.

A similar analysis can be applied to the adaptive filter \( C(k) \), assuming \( \bar{P} \) is the estimating error of GSC, therefore it can be defined according to (4),

\[
\bar{P}^{(e)} = \sum_{i=1}^{N} (M_i - (BC^{(e)}))
\tag{13}
\]

Substitutes (13) to (8), the following equation can be derived,

\[
\bar{P}^{(e+1)} = (I - A_1) \bar{P}^{(e)} - A_2
\tag{14}
\]

Where

\[
A_1 = \mu \left[ L^T \left( \sum_{i=1}^{N} (S^{(e)} + V^{(e)}), + \sum_{i=1}^{N} (B R_i X^{(e)}) \right]^2 \right]^{-1}
\tag{15}
\]

\[
A_2 = \mu \left[ L^T \left( \sum_{i=1}^{N} (R_i X^{(e)} + S^{(e)} + V^{(e)}), + \sum_{i=1}^{N} (B R_i X^{(e)}), \right)^2 \right]^{-1}
\tag{16}
\]

represents the steady-state error of GSC, and \( B = B B^T \).

In contrast with the analysis of \( A_1 \) above which have similar form, \( A_2 \) need to be increased by getting rid of the
components of $X$ of the oscillating frequencies. Therefore, $L$ is chosen to be a band-stop filter whose stop band locates around the center frequency of oscillation. Besides, for the reason of $A_i$ increases with $L X_{W}^{o}$, it can be minimized by configuring the center frequency of stop band of $L$ with the detected frequency of oscillation when the bandwidth of the stop band of $L$ is fixed.

3. Oscillation detection

From the analysis above, we can conclude that the performance of both feedback cancellation and noise reduction will be improved by using an effective oscillation detector. The oscillation frequencies identifying scheme based on the Nyquist criteria applied in [2] can be interpreted as applying pseudo-random noise to calculate the impulse response of the open loop system in the fitting procedure of hearing aids. However, it fails to consider that the variance of the shape and peak of feedback path in frequency domain according to the environment of application.

In this paper, we propose a novel oscillation detection method which monitors the oscillation that is caused by the variation of feedback path and calculates the exact frequency at which the oscillation happens.

$$\eta = \frac{f_o \cdot z_i}{2 \cdot M}$$  \hspace{1cm} (18)

where $f_o$ denotes the center frequency of the oscillation, $f_i$ denotes the sampling frequency, $z_i$ denotes the zero crossing rates of the autocorrelation signal, $M$ denotes the number of points of the autocorrelation signal.

4. Simulation results

In this section, the simulation of frequency domain normalized LMS algorithm (FNLMS), band-limited frequency domain normalized LMS algorithm (BL-FNLMS) and band-limited frequency domain normalized LMS algorithm with an oscillation detector (BL-FNLMS-OSD) are performed to compare the performance of band-limited adaptive filter and oscillation detector in feedback cancellation and noise reduction. The two feedback paths $W_{o1} = [1 \ldots 1] \times \sum_{k=1}^{N} R_{o1} M$ and $W_{o2} = [1 \ldots 1] \times \sum_{k=1}^{N} R_{o2} M$ are shown in Figure 3, the feedback path applied in the simulation is changed from $W_{o1}$ to $W_{o2}$ in the 3000th frame.

Figure 3: The amplitude responses of feedback paths $W_{o1}$ and $W_{o2}$

The sampling frequency is chosen to be 16kHz, the length of the frame is set to be 64, and the gains of sub-band $k_1$ to $k_8$ which are depicted in Figure 2 are set to be equal constant $G$.

The performance of the three aforementioned algorithms are compared in Figure 4 by the curves of error metric of $W$ and $C$ which are defined as follows,

$$\xi_W = 10 \log \left( \frac{1}{N} \sum_{k=1}^{N} \frac{|W(k) - W_{o1}(k)|^2}{\sum_{k=1}^{N} |W_{o1}(k)|^2} \right)$$  \hspace{1cm} (19)

Where $N_G$ represents the number of sub-bands, the parameter $\eta$ is an empirical value which is set to 3 in the simulation of section 4. When the oscillation is detected, the center frequency of the oscillation can be calculated by the following equation.

\[ f_i = \frac{f_o \cdot z_i}{2 \cdot M} \]
\[ \xi_C = 10 \log \left( \frac{\sum_{m=1}^{M} \left| C_m(k) - C_m(k) \right|^2}{\sum_{m=1}^{M} \left| C_m(k) \right|^2} \right) \quad (20) \]

Where \( W \) represents the optimum \( W \), and \( C \) represents the optimum \( C \).

As shown in Table 2, the informal subjective evaluation is performed by letting subjects listen and grade the residual feedback component according to the principles.

Table 2. Description of subjective ratings.

<table>
<thead>
<tr>
<th>Score</th>
<th>Feedback</th>
<th>Noise</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>Loud howling</td>
<td>Speech totally covered</td>
</tr>
<tr>
<td>1</td>
<td>Loud continuous whistling</td>
<td>Extremely loud noise</td>
</tr>
<tr>
<td>2</td>
<td>Soft continuous whistling</td>
<td>Somewhat loud noise</td>
</tr>
<tr>
<td>3</td>
<td>Soft intermittent whistling</td>
<td>Soft noise</td>
</tr>
<tr>
<td>4</td>
<td>Slight audible feedback</td>
<td>Slight audible noise</td>
</tr>
<tr>
<td>5</td>
<td>No audible feedback</td>
<td>No audible noise</td>
</tr>
</tbody>
</table>

As shown in Figure 4, simulations are carried out to compare the convergence speed and steady state error of adaptive filters \( W \) and \( C \). Nearly 15 dB steady-state error of \( W \) is reduced by the band-limited adaptation when comparing the FNLMS and BL-FNLMS before the 3000th frame, and more than 10 dB steady-state error is reduced with the advantage of oscillation detector when comparing BL-FNLMS and BL-FNLMS-OSD after the feedback path is changed. The diverse convergence speeds of the three aforementioned algorithms of \( C \) show that the band-stop pre-filters contribute significantly to increase the convergence speed of \( C \) with the help of oscillation detector especially when the feedback path is changed.

Additionally, the subjective rating is carried out to evaluate the feedback and noise reduction performance with an input SNR of 0dB. Five sentences which were picked randomly from the TIMIT speech database are taken as input with five person to rate at three gain levels: Maximum Stable Gain(MSG), and 5 and 10 dB below MSG.

From Table 3, the experiment shows that in a considerably noisy environment, the feedback cancellation and noise reduction can be improved in a large scale with different forward path gain when the oscillation detector is applied.

Table 3. Subjective ratings of five sentences.

<table>
<thead>
<tr>
<th></th>
<th>MSG-10</th>
<th>MSG-5</th>
<th>MSG</th>
</tr>
</thead>
<tbody>
<tr>
<td>Feedback</td>
<td>4.6</td>
<td>4.0</td>
<td>1.8</td>
</tr>
<tr>
<td>Noise</td>
<td>4.0</td>
<td>3.8</td>
<td>3.0</td>
</tr>
<tr>
<td>BL-FNLMS-OSD</td>
<td>4.2</td>
<td>3.0</td>
<td>1.2</td>
</tr>
<tr>
<td>Noise</td>
<td>3.8</td>
<td>4.2</td>
<td>1.6</td>
</tr>
</tbody>
</table>

5. Conclusions

This paper proposes a novel approach to improve the performance of feedback cancellation and noise reduction as a whole. The band-pass adaptive strategy with fixed band-width of pass band is applied to the adaptive feedback cancellation, and similarly, the band-stop adaptive strategy with fixed band-width of stop band is applied to the adaptive noise reduction to constrain the adaptive filters to operate in limited frequencies. Besides, an oscillation detector is designed to deliver a more robust performance when facing with the varying environments. Simulation results show that the steady-state performance of \( W \) and the convergence behavior of \( C \) which aim to reduce the feedback and noise have a significant improvement.

6. References