Acoustic Feedback Cancellation in Hearing Aids Using Spectral Subtraction

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Abstract: - Acoustic feedback is a common complaint of hearing aids users and reduces the maximum gain of hearing aids. In this paper, an acoustic feedback cancellation scheme using spectral subtraction is proposed. Acoustic feedback signals are continuously estimated by approximated Wiener filter and are subtracted from microphone input signal of hearing aids. Using cross-correlation between microphone input signal and receiver output signal, approximated Wiener filter estimates the path of acoustic feedback of hearing aids. Acoustic feedback signals are canceled by the spectrum of output of approximated Wiener filter. Feedback signals are hardly occurred because they are continuously estimated and canceled by the proposed scheme. As results of computer simulation, maximum gain of proposed scheme is better than that of conventional ones.


1 Introduction

Hearing aids does to hear voice by amplifying input audible signal for a person that is hardness of hearing. Signal of acoustic feedback in hearing aids, which, if large enough, cause the hearing aids to emit a high-intensity oscillation, such as the howling, whistling, is annoying problem. It is the acoustic feedback cancellation that remove acoustic feedback as increasing gains of amplifier inserted to hearing aids[1].

Many previous works, including notch filtering, frequency shifting and adaptive feedback cancellation(AFC)[1] to remove feedback have been proceeded in the recent years. Adaptive cancellation scheme using adaptive filter theory is widely utilized. In AFC methods, the feedback signal is cancelled by the output signal of adaptive filter which models the acoustic feedback path. The algorithms for estimation and adaptation vary among implementations. The methods are generally dichotomized by continuous and non-continuous adaptation.

The continuous adaptation method amplify and pass the input signal to user while simultaneously adapts adaptation filter coefficients continuously. The non-continuous filter adaptation is that it detects acoustic feedback occurrence availability and adapts adaptation filter coefficients.

Bustamante et al[2], who were the first to propose a continuous adaptation system that used the least mean squares(LMS)-based adaptive filter[Fig. 1], indicate that a delay in the cancellation path is sufficient for decorrelating input signals. However, they do not present a mathematical justification for their choice of delay[3].

Kates[4] proposes a non-continuous adaptation system that used LMS-based adaptive filter and Wiener filer when howling is detected. Maxwell[1] also propose a non-continuous adaptation system[Fig. 2] using the normalized-LMS(NLMS) algorithm that adjusts the adaptive filter coefficients only when the input signal level is low. Both systems are able to active when howling or silent interval are detected. The continuous adaptation systems are better than the non-continuous ones, under assumption of well-estimated acoustic feedback, it does not need a kind of training sequence and continuously adapts filter coefficients using input signal.
In this paper, a method continuously estimates the feedback signal is proposed to cancel feedback signal. The feedback path is modeled by Wiener filter which is updated by the cross-correlation between input signals of adaptive filter and microphone input signals. It is not need acoustic feedback detector and the proposed method has simple structure. The proposed method can remove robustly acoustic feedback signal regardless of change of feedback path within extent that do not pass over maximum stable gains of amplifier. Through simulations, the proposed method has better performance than conventional ones.

2 Feedback Cancellation Methods

2.1 Continuous adaptation method

Fig. 1 shows the continuous adaptation method that estimate feedback path continuously adapting adaptive filter. It can estimate change of feedback path continuously as estimate that a continuation feedback path compares to non-continuous adaptation method. However, because IIR structure, it has disadvantage of the limitation of amplifier’s gain to guarantee the stability of system. In particular, IIR filter structure has two disadvantages. First, they become unstable if their poles move outside of a unit circle during the adaptation process. Second, their performance surfaces are generally non-quadratic and may have local minima. In experiment of computer simulation and real time implementation, even if use variants of several LMS algorithm, it cause that whole system is unstable[1].

2.2 Non-continuous adaptation method

Fig. 2 shows the non-continuous adaptation method that estimate feedback path to reduce feedback when resonance occurred in hearing aids. Kates[4] estimate the feedback path during howling is detected. Maxwell[1] proposed system estimates it during silent as well as howling is detect. Acoustic feedback cancellation method that takes advantage of the non-continuous adaptation method has short duration comparing with continuous adaptation method, and if alarm or warning signals without attenuation is detected, feedback loop is interrupt. It is that changes in the feedback path that do not immediately cause instability but that may nonetheless be disturbing because of their effects on the overall transfer function are undetected. And the signal is interrupted for a short period and that a high-level noise is presented to the user.

2.3 Adaptation algorithm

The NLMS algorithm is used the adaptation algorithm as computation is low, structure is simple. It compares required response with output of error signal. It adapts coefficients so that mean-square value of error signal may be minimized. This algorithm is described by

\[ e(k) = d(k) - W^T(k)X(k) \]  
\[ W(k+1) = W(k) + 2\mu(k)e(k)X(k) \]  
\[ \mu(k) = \frac{\alpha}{LP_s(k)} \]  
\[ P_s(k+1) = \beta P_s(k) + (1-\beta)x^2(k) \]
where $W(k)$ is the L-tap coefficients vector, $e(k)$ is error, $\alpha$ is the adaptation constant(0 to 1), $P_e(k)$ is the running power estimate of error signal. Maxwell[1] uses NLMS algorithm that was modified to what is known as the sum method

$$\mu(k) = \frac{\alpha}{L(P_x(k) + P_e(k))} \tag{5}$$

where $P_x(k)$ is the running power estimate of input signal. Computation is not much high and coefficients error adjustment of target signal is low. However it has the disadvantage that convergence time is quite long.

### 3 Proposed feedback reduction method

#### 3.1 Spectral subtraction(SS)

The SS algorithm processes frame-based input signal. Hanning or Hamming window are used as weight function. There is no correlation between speech signal $s(k)$ and noise signal $f(k)$. And they are locally stationary short duration. This algorithm is described by

\[ f_u(k,m) = f_s(k,m) + f_f(k,m) \tag{6} \]
\[ U(\omega;m) = S(\omega;m) + F(\omega;m) \tag{7} \]
\[ P_u(\omega;m) = P_s(\omega;m) + P_f(\omega;m) \tag{8} \]
\[ P(\omega;m) = \frac{1}{N}|U(\omega;m)|^2 \tag{9} \]

where $u(k)$ is input signal, $w(k)$ is window, $k$ is time index, $m$ is frame index, and $N$ is frame length.

\[ f_u(k,m) = u(k)w(m-k), \]
\[ f_s(k,m) = s(k)w(m-k), \]
\[ f_f(k,m) = f(k)w(m-k). \]

$U(\omega;m)$, $S(\omega;m)$, and $F(\omega;m)$ are short-time discrete time Fourier transform of the $u(k)$, $s(k)$, and $f(k)$. $P_u(\omega;m)$, $P_s(\omega;m)$, and $P_f(\omega;m)$ are power spectrum of them.

If the spectrum of estimated speech signal is same to that of estimated input signal.

\[ \hat{S}(\omega;m) = [U(\omega;m)^a - |F(\omega;m)|^a] e^{i \varphi_u(\omega;m)} \tag{10} \]
\[ \hat{S}(\omega;m) = [U(\omega;m)^a - |F(\omega;m)|^a]^\frac{1}{2} e^{i \varphi_u(\omega;m)} \tag{11} \]

Following is its general expression.

\[ \hat{S}(\omega;m) = [U(\omega;m)^a - |F(\omega;m)|^a]^\frac{1}{2} e^{i \varphi_u(\omega;m)} \tag{12} \]

At equation (12), a is 1 at magnitude SS algorithm, a is 2 at power SS algorithm. When the signal-to-noise ratio(SNR) is low, however, even if it uses SS method, fidelity drops more or less.

#### 3.2 Feedback Cancellation using SS

In this paper, the feedback signal is cancelled by adaptive cancellation method using spectral subtraction with approximated Wiener filter. Following is equation of approximated Wiener fieler.
\[ H^2(e^{jo}) = \frac{|U(e^{jo})|^2 - |\mathcal{F}(e^{jo})|^2}{|U(e^{jo})|^2} \]  

(13)

where \(|U(e^{jo})|^2\) is power spectrum of input signal in hearing aids. \(|\mathcal{F}(e^{jo})|^2\) is power spectrum of feedback path.

Fig. 4 is block diagram of overall system in hearing aids. Input acoustic signal amplifies through pass processing stage of hearing aids into microphone input. Therefore, the acoustic signal feedback through unknown path. Feedback signal causes howling that make unstable hearing aids. Stable hearing aids needs acoustic feedback reduction system. At fig. 4 wiener filter is used to reduce acoustic feedback signal. The Wiener filter passes the input signal when feedback signal does not exist. When feedback signal is occurred, however, it reduces the feedback signal according to cross-correlation between input signal of microphone and input signal of Wiener filter.

In this paper, the Wiener filter supposes that feedback path is stationary, it is most suitable in least square viewpoint. Of course, LMS algorithm gets Wiener solution through adapting. However, in steady-state LMS algorithm has more error than the Wiener filter.

\[ \hat{F}(e^{jo}) = \left( \frac{1}{X(e^{jo})} \left( \frac{|Y(e^{jo})|^2}{|X(e^{jo})|^2} \right) \right)^{1/2} X(e^{jo}) \]  

(14)

The output \( \hat{F}(e^{jo}) \) of approximated Wiener filter is value of correlation between input signal and target signal. When \(|X(e^{jo})| << |Y(e^{jo})|\) in transfer function, transfer function of this Wiener filter is closed to 1. Therefore, feedback path is dominant in overall spectrum domain. So, this Wiener filter reduces acoustic feedback path that estimate feedback path. When \(|X(e^{jo})| >> |Y(e^{jo})|\), transfer function of this Wiener filter is closed to 0. Therefore, input signal is dominant in overall spectrum domain. So, input signal according to that by out send because output of this Wiener filter is closed to 0.

4 Simulation
Simulation assumes that resonance frequency is 7kHz in hearing aids. Input signal uses white noise signal normally distributed. Feedback path is fixed and its modeling is 128-order FIR filter. See fig. 5. In NLMS algorithm, adaptation constant \(\alpha\) is 0.0002, adaptation processing is sample by sample. In SS, the block processing is by 256 samples, window length is 256, using Hanning window.

![Fig. 5 Transfer function of the feedback path](image)

Simulation compares proposed method to Maxwell [1]'s non-continuous adaptation method. It is important problem that maximum stable gain of the amplifier. The gain of the amplifier uses linear gain(eq. 15). Equation 16 is the SNR.

\[ g_{linear} = 10^{-\frac{e_m}{2}} \]  

(15)
SNR = 10\log_{10} \frac{E_i}{E_c} = 10\log_{10} \frac{\sum_{k} k^2}{\sum_{k} k (s(n) - s(n))^2} \quad (16)

Where \( g_{\text{linear}} \) is value of the linear gain, \( g_{\text{dB}} \) is value of gain in decibel, and \( k \) is 10 during power of input signal and 20 during magnitude of input signal.

Fig. 6 ~ Fig. 8 show simulation results. In Fig. 6 ~ Fig. 8, (a) is white Gaussian noise input, (b) is output of non-processing, (c) is output of non-continuous adaptation, (d) is the output of proposed method respectively. In simulation, Acoustic feedback is occurred 11dB of amplifier’s gain at first.

In Fig. 6, the (c) is same to the (b), because The non-continuous method(Fig. 6(c)) does not adapt until howling is detect, the proposed method(Fig. 6(d)) adapts continuously. So there is no acoustic feedback signal. Fig. 7 shows result that gain of the amplifier is 14dB. In Fig. 7(c), as shows that acoustic feedback is detected at 6,300 sample. After 6,300 sample, adaptation is beginning. So it reduces feedback signal. In Fig. 7 (d), there is no acoustic feedback signal. If gain of the amplifier were more than 14.1dB, non-continuous adaptation method goes divergence. Fig. 8 show results that gain of the amplifier is 16.8dB. If gain of the amplifier were more than 14.8dB, proposed continuous adaptation method goes to divergence. 16.8dB is critical point that does not converge. As above shows, non-continuous adaptation method procedure takes detection and adaptation. However, proposed method adapts continuously. From the viewpoint of maximum stable gain of amplifier, proposed method is 2.8dB better than previous method.

<table>
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<tr>
<th>Gain of amp (dB)</th>
<th>NLMS (dB)</th>
<th>Proposed (dB)</th>
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<tbody>
<tr>
<td>11</td>
<td>-3.1</td>
<td>-2.7</td>
</tr>
<tr>
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<td>-87.8</td>
<td>-11.9</td>
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</table>
5 Conclusion

Hearing aids is amplifier of microphone input signal and the vent in hearing aids occurs acoustic feedback. In this paper, continuous adaptation method proposed in order to reducing acoustic feedback signal. The proposed method uses approximated Wiener filter using spectral subtraction. The advantages of the proposed method are that does not need feedback signal detector and are able to reduce effectively before occur acoustic feedback signal.

As results of simulation, it is shown that the proposed method has better maximum stable gain than conventional ones.

References: