DUAL MICROPHONE SOLUTION FOR ACOUSTIC FEEDBACK CANCELLATION FOR ASSISTIVE LISTENING

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ABSTRACT

The method proposed in this paper improves the identification and cancellation of the feedback path by making the adaptive canceller robust against the impact of the desired speech signal. The proposed method allows for the canceller's coefficients to be continuously adapted allowing it to track variations in the feedback path even in the presence of the desired signal. It suggests the use of dual microphones and dual adaptive filters arranged in such a way that allows the speech signal to be identified and removed from the adaptation process. This results in a more robust solution which was verified by our experiments and evaluations. The perceptual evaluation of speech quality (PESQ) measure was also used to show that the proposed method results in better signal quality.

***Index Terms*—hearing aids, assistive listening, acoustic feedback, adaptive filter, feedback canceller**

I. INTRODUCTION

With the advance of technology, such as, advances in digital signal processing, hearing aids are becoming smaller and smaller in size. Many hearing devices today can be fitted completely inside the ear canal of the user [1]. This reduction in size leads to a decreasing distance between the loudspeaker and the microphone. As a result, acoustic feedback occurs due to the acoustic coupling between the loudspeaker and the microphone.

Acoustic feedback poses a problem in the normal operation of hearing aids. The feedback limits the maximum achievable amplification possible by the hearing device, deteriorates the sound quality by producing a distortion of the desired signal, and is a cause of instability in hearing aids [2, 3]. The feedback path possesses some general characteristics. One characteristic is that the feedback varies under different conditions and environments. There has been some study in the literature of the variability of the feedback path [4, 5]. Causes of the feedback path and it's variations are mentioned in [1, 2, 4, 5, 6]. The general observation is that the feedback path tend to show less attenuation at high frequencies than at low. Thus, oscillations due to feedback often occurs at higher frequencies [2].

Acoustic feedback control techniques tries to minimise the effect of the feedback on the performance of hearing aids. [7] defines acoustic feedback control as to the process of attempting to solve the acoustic feedback problem either completely (i.e., to remove the acoustic coupling) or partially (e.g., to remove the howling artefacts from the loudspeaker signal). Many feedback control methods have been proposed in the literature, however, there is still

*This research is partly funded by the Australian Research Council Grant No. DP1096348.

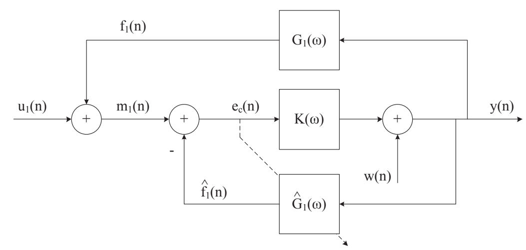

Fig. 1. Adaptive Canceller

a lack of reliability in the available automatic acoustic feedback control solutions [7]. Thus, there is still a need and demand for improved feedback suppression and/or cancellation techniques [3]. Proposed techniques in the literature can be generally classified into feedforward suppression and feedback cancellation techniques [3].

The use of feedback cancellation techniques in the acoustic feedback control is a preferred option as it is able to be made adaptive to track the variations in the feedback path [3, 8]. Fig. 1 illustrates a classic feedback canceller. One main challenge with adaptive feedback cancellers is that the unobservable desired input signal $u_1(n)$ acts as a disturbance to the adaptation to the canceller. If the feedback estimate $\hat{f}_1(n) = f_1(n)$ then the error signal $e_c(n) = u_1(n)$. Therefore, if this error signal is used to adapt the filter's coefficients it will result in the cancellation of the desired signal leading to degraded signal quality.

This paper proposes a method to identify the desired input signal $u_1(n)$ and remove it from the error signal prior to adapting the feedback canceller's coefficients. Thus, making the adaptation more robust against the disturbance. The proposed idea also allows for the filter to be continuously adapted even in the presence of the desired input signal. This method results in better signal quality than the classic approach based on the PESQ method.

II. BACKGROUND

One main challenge with adaptive feedback cancellers is that the desired input signal $u_1(n)$ acts as a disturbance to the canceller's adaptation. The presence of the closed-signal loop gain $K(\omega)$ introduces signal correlation when the desired signal is spectrally coloured (e.g. speech or music signal) [3]. If the loudspeaker signal $y(n)$ and the speech signal $u_1(n)$ are not correlated, then the feedback path estimate is said to be unbiased. As a result of the bias term, the adaptive feedback canceller fails to provide a reliable feedback estimate and even cancels the desired signal instead.

The adaptive filter continuously adapts the coefficients $\hat{\mathbf{g}}_1 = [\ \hat{g}_0 \quad \hat{g}_1 \quad \dots \quad \hat{g}_{L-1}\ ]^T$ of the feedback canceller based on standard adaptive filtering procedures (Wiener filtering) where L is the length of the impulse response of the feedback path.

The adaptive filter tries to minimise the error signal $e(n)$ using the cost function

$$J(\hat{\mathbf{g}}_1) = E\{|m_1(n) - \hat{\mathbf{g}}_1^T \mathbf{y}(n)|^2\}. \tag{1}$$

With $\mathbf{y}(n) = [\ y(n) \quad y(n-1) \quad \dots \quad y(n-L+1)\ ]^T$ then (1) results in the Wiener filter

$$\hat{\mathbf{g}}_1 = \mathbf{R}_{yy}^{-1}(n)\mathbf{R}_{ym_1}(n). \tag{2}$$

where $\mathbf{R}_{\alpha\beta}(n)$ is the cross-correlation (autocorrelation when $\alpha = \beta$).

Assuming a sufficient-order L, and using $m_1(n) = \mathbf{g}^T\mathbf{y}(n) + u_1(n)$ then, (2) can be written as

$$\hat{\mathbf{g}}_1 = \mathbf{g}_1 + \underbrace{\mathbf{R}_{yy}^{-1}(n)\mathbf{R}_{yu_1}(n)}_{bias}. \tag{3}$$

Ideally, $\hat{\mathbf{g}}_1 = \mathbf{g}_1$, however, from (3) it can be seen that the desired signal $u_1(n)$ acts as a disturbance to the adaptation of the feedback canceller.

In the literature, several solutions have been proposed to reduce the bias problem. One solution is to incorporate signal decorrelating operations in the signal processing path of the hearing aid, such as introducing delays, probe signals, and non-linearities [3, 9, 10]. However, decorrelation tends to degrade the sound quality, making full decorrelation impossible [3]. Another attempt to minimise the bias is to reduce the adaptation speed of the adaptive feedback canceller [11] or constrain its adaptation based on prior knowledge of the feedback path [3]. Yet another approach, is to do a closed-loop system identification [8, 11, 12]. A more recent approach is to use dual microphones for feedback cancellation where the coefficients of feedback canceller are updated after subtracting the speech signal from the input signal by dual microphones [13].

The proposed method here differentiates itself by using dual microphones and dual adaptive filters on each ear plug, it also allows for continuous adaptation of the feedback canceller, and more flexibility with the desired input source location.

III. PROPOSED DUAL MICROPHONE METHOD

This paper proposes an alternative way to improve the identification and cancellation of the feedback path $G_1(\omega)$ in the presence of the desired speech signal $u_1(n)$ by reducing the impact of the desired signal on the adaptation of the feedback canceller $\hat{G}_1(\omega)$. This method also allows for the canceller's coefficients to continuously adapt allowing it to track variations in the feedback path.

The proposed idea is to use a second microphone in the assistive listening device. The location of such microphone is important. On one hand, the two microphones should be located as close as possible to each other so that the desired signal picked up by the microphones be as similar as possible. On the other hand, the two microphones should be as far as possible from each other, so that the feedback picked up by the second microphone be more attenuated than the first microphone. By this, the second microphone is able to capture the desired speech signal, with minimum presence of the feedback signal. This new signal is then removed from the error signal prior to adapting the feedback canceller, thus, removing the bias term from the adaptation.

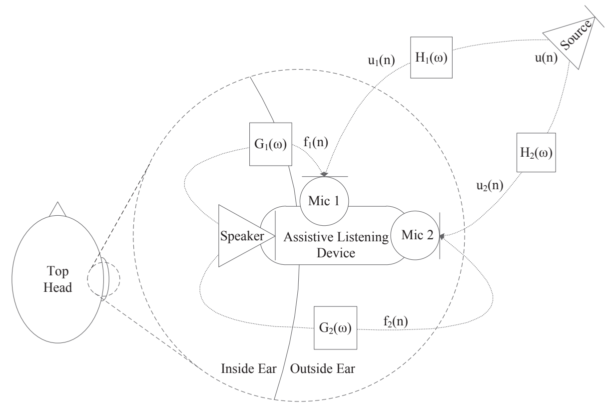

Fig. 2. Microphone Placement

Fig. 2 illustrates an assistive listening device with two microphones. The device shown is plugged into the user's right ear. Microphone 1 faces forward from the head and microphone 2 faces outward. The desired input wave $u(t)$ travels through two separate channels, $H_1(\omega)$ and $H_2(\omega)$ to reach each of the microphones. The signal picked up from microphone 1 is amplified and played out through the device's loudspeaker. The amplified signal is fed back into the microphones through two separate channels $G_1(\omega)$ and $G_2(\omega)$.

It is desired that channel $H_1(\omega)$ be as similar as possible to $H_2(\omega)$ and that the feedback channel $G_2(\omega)$ have high attenuation. To achieve this, the placement of the microphones is crucial. The distance between the microphones compared to the distance from the microphones to the desired signal source should be relatively small. Also, the distance between microphone 2 and the feedback source should be relatively large.

Fig. 3 illustrates the block diagram of the proposed dual microphones method with two adaptive filters, $\hat{G}_1(\omega)$ and $\hat{H}(\omega)$. The first filter, $\hat{G}_1(\omega)$ is adapted to match the feedback channel $G_1(\omega)$. The second filter $\hat{H}(\omega)$ is adapted to match the channel $H(\omega)$ which is the transfer function from $H_2(\omega)$ to $H_1(\omega)$ in Fig. 2. $K(\omega)$ is the signal processing path of the assistive listening device, which is generally some selective frequency gain.

The error equation $e_p(n)$ is given as

$$\begin{aligned} e_p(n) = \mathbf{g}_1^T\mathbf{y}(n) - \hat{\mathbf{g}}_1^T\mathbf{y}(n) \\ + \mathbf{h}^T\mathbf{u}_2(n) - \hat{\mathbf{h}}^T\mathbf{u}_2(n) - \hat{\mathbf{h}}^T\mathbf{f}_2(n) \end{aligned} \tag{4}$$

where $u_1(n) = \mathbf{h}^T\mathbf{u}_2(n)$, $\mathbf{g}_1 = [\ \hat{g}_0 \quad \hat{g}_1 \quad \dots \quad \hat{g}_{L-1}\ ]^T$ is the coefficients of the feedback channel $G_1(\omega)$, $\hat{\mathbf{h}} = [h_0\, h_1\, \dots\, h_{L-1}]^T$ is the coefficients of the estimate $\hat{H}(\omega)$, $\mathbf{y}(n) = [y(n)\, y(n-1)\, \dots\, y(n-L+1)]^T$ is the loudspeaker signal, and $\mathbf{f}_2(n) = [f_2(n)\, f_2(n-1)\, \dots\, f_2(n-L+1)]^T$ is the feedback signal picked up by the second microphone.

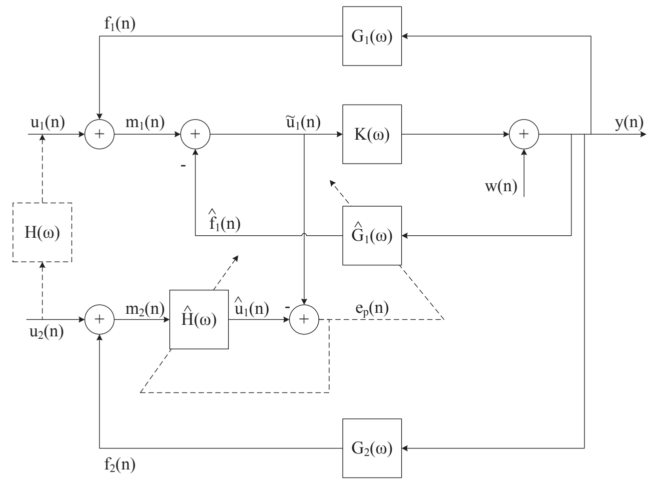

Fig. 3. Proposed Block Diagram

Let $\mathbf{a}(n) = [\mathbf{g}_1\ \mathbf{h}]^T$, $\mathbf{z}(n) = [\mathbf{y}(n)\ \mathbf{u}_2(n)]^T$, and $\xi(n) = [\mathbf{0}\ \mathbf{f}_2(n)]^T$, then (4) becomes

$$e_p(n) = [\mathbf{a} - \hat{\mathbf{a}}]^T \mathbf{z}(n) - \hat{\mathbf{a}}^T \xi(n). \tag{5}$$

We wish to minimise the cost function

$$J\left\{|e_p(n)|^2\right\} = E\left\{\left|[\mathbf{a} - \hat{\mathbf{a}}]^T \mathbf{z}(n) - \hat{\mathbf{a}}^T \xi(n)\right|^2\right\}. \tag{6}$$

By differentiating (6) with respect to $\hat{\mathbf{a}}^T$ leads to

$$\hat{\mathbf{a}} = [\mathbf{R}_{zz}(n) + \mathbf{R}_{z\xi}(n) + \mathbf{R}_{\xi z}(n) - \mathbf{R}_{\xi\xi}(n)]^{-1} \cdot [\mathbf{R}_{zz}(n) + \mathbf{R}_{\xi z}(n)]\,\mathbf{a}. \tag{7}$$

If $\|\mathbf{R}_{zz}(n)\| \gg \|\mathbf{R}_{\xi\xi}(n)\|$ and the correlation between $u_2(n)$ and $f_2(n)$ is small, then the terms $\mathbf{R}_{\xi\xi}(n)$, $\mathbf{R}_{z\xi}(n)$, and $\mathbf{R}_{\xi z}(n)$ in (7) can be ignored resulting in

$$\hat{\mathbf{a}} = \mathbf{a}. \tag{8}$$

Such assumptions can be made due to the microphone arrangement proposed. The greater the attenuation in the feedback channel $G_2(\omega)$, the weaker the signal $f_2(n)$ becomes and the less impact it will have in the system.

IV. EXPERIMENT RESULTS

Experiments were conducted in order to verify the performance of the proposed method and to validate our assumptions. The assistive listening device used was Sensear's ear plug SP1x with 16 kHz sampling rate and with modified firmware to suit our real time experiment requirements. The layout of the microphone placement is illustrated in Fig. 2 where one microphone faces forward from the head and the second microphone faces outward and is further away from the feedback source.

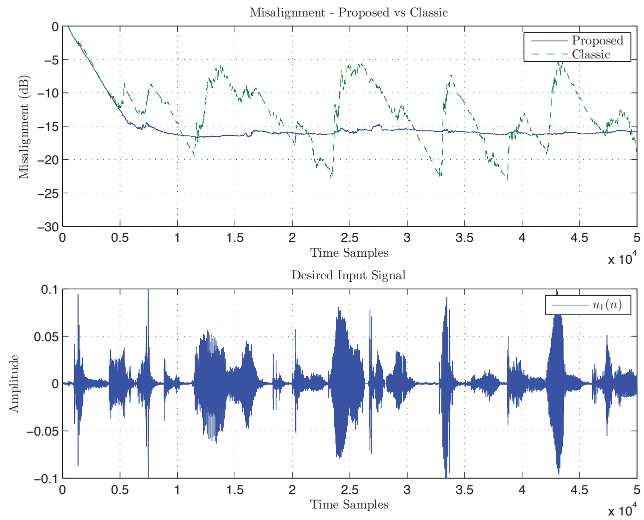

Fig. 4. Misalignment proposed method vs classic adaptation

To measure the feedback path, a Gaussian white noise signal $w(n)$ was injected into the loudspeaker and both microphones were set to record. When no speech signal is present, the feedback path signals $f_1(n)$ and $f_2(n)$ can be measured and used to estimate the channels $G_1(\omega)$ and $G_2(\omega)$. When the ear plug is properly fitted into the user's ears, we found that the $\|G_1(\omega)\| \gg \|G_2(\omega)\|$ for most frequencies. At some frequency locations $\|G_1(\omega)\|$ is over 32 dB higher than $\|G_2(\omega)\|$. The further away the second microphone is from the feedback source the more attenuation there will be in $G_2(\omega)$.

Included in the feedback paths $G_1(\omega)$ and $G_2(\omega)$ is the characteristic of the loudspeaker, the microphone, the analogue-to-digital converter (ADC), the digital-to-analogue converter (DAC), and low-pass filters [3, 14].

Speech signals were also recorded during the experiments using dual microphones. Three locations for the speech signal source was used: speaker placed in front, side, and back of the head. These signals were recorded when no feedback was present and were used in our evaluations.

V. EVALUATION BASED ON EXPERIMENTAL DATA

This section presents some of our evaluations based on experimental data. Fig. 4 compares the misalignment, defined as $\Delta = \frac{|\mathbf{g_1} - \hat{\mathbf{g}}_1|^2}{|\mathbf{g_1}|^2}$, of the proposed method verses the classic adaptive filter illustrated in Fig. 1. These plots were obtained using 256 tap filters, with a modified LMS (MLMS) algorithm - where the step size is normalised with respect to both the filter's input and error signal. The value of the step size was set to 0.1. The speaker was located facing the side of the head and the injected noise variance was set to a value of 0.1. The forward path gain $K(\omega)$ was set to 0 dB. All adaptive filters were set to the same parameter values and algorithm.

Fig. 4 and Fig. 5 shows that the proposed method is more robust in the presence of disturbance caused by the desired input signal. Fig. 4 shows that the misalignment does not diverge as wildly as the classic approach does and Fig. 5 presents the errors signals for the

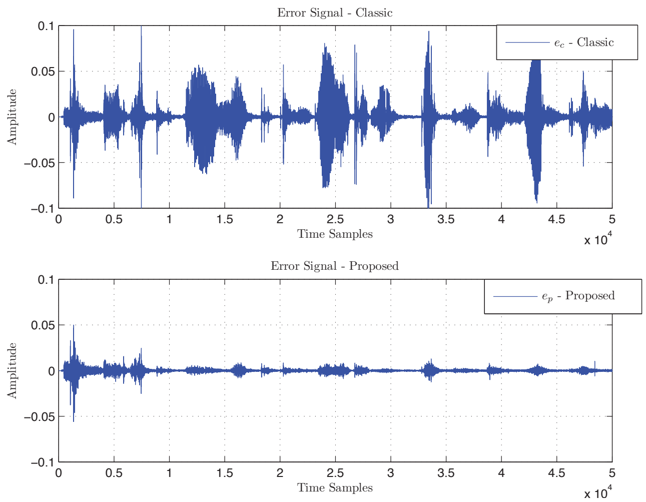

Fig. 5. Error signals - proposed vs classic

Table I. PESQ measure - proposed vs classic

Step Size	Proposed	Classic
0.01	**4.1890**	4.0944
0.05	**4.2543**	3.5212
0.10	**4.1925**	3.3001
0.50	**4.2507**	3.2807

classic and proposed approach. With the classic approach, $e_c(n)$ is very similar to $u_1(n)$ whereas in $e_p(n)$ there is less impact from $u_1(n)$.

Another objective measure used is the PESQ measure. The classic adaptive canceller tends to degrade the desired signal quality as it cancels the speech due to the bias term. Therefore, the use of the PESQ measure quantifies the speech quality and is an appropriate measure to compare the proposed method against the classic approach. PESQ provides a score in the range of 1 to 5 where 1 is unacceptable and 5 is excellent. Table I presents the results. The reference signal used is $u_1(n)$ and the degraded signal used is the loudspeaker signal $y(n)$ without the injected noise $w(n)$.

VI. CONCLUSION

This paper proposed an approach that improves the identification and cancellation of the feedback path by reducing the impact of the desired signal on the adaptation of the feedback canceller. This method allows for the canceller's coefficients to continuously adapt allowing it to track variations in the feedback path. The suggested microphone layout assumes that the speech signal received by both microphones are similar, but the feedback received by the second microphone has greater attenuation than the first. Two adaptive filters were used, the first was used as the feedback canceller and the second was used to match the desired speech signal recorded by the dual microphones. With such arrangement, the speech signal from the second microphone is subtracted from the error signal before adapting the canceller. This results in a more robust solution which was verified by our experiments and evaluations. The perceptual evaluation of speech quality (PESQ) measure was also used to show that the proposed method results in better signal quality.

VII. REFERENCES

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