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Two microphone acoustic feedback cancellation in digital hearing aids: A step size controlled frequency domain approach

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A R T I C L E I N F O

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A B S T R A C T

The feedback cancellation performance of behind the ear hearing aids can be improved by employing two microphones in the feedback cancellation process. A frequency domain implementation of feedback cancellation in a two microphone behind the ear hearing aid has been proposed in this paper. A frequency domain step size control scheme has been further introduced to improve the convergence behaviour. The implementation of a the frequency domain step size control method requires the microphone input signal as well as the loudspeaker output signal to be uncorrelated. In order to reduce the correlation effects and enable such an implementation, we have incorporated frequency shifting in the forward path. The proposed approach has been shown to provide improved convergence behaviour as well as offer enhanced speech quality. Further, the proposed scheme is expected to offer a reduced computational load for longer feedback paths.

1. Introduction

Hearing aid, which essentially amplifies the sound for an enhanced listening experience, is one of the most widely used assistive listening device [1,2]. A basic behind the ear (BTE) hearing aid consists of a microphone which picks up the sound to be amplified, an amplifier, a battery, and a loudspeaker which reproduces the amplified sound [3]. In persons suffering from severe-to-profound hearing loss, a hearing aid must offer a very high gain. However, the leakage from the loudspeaker to microphone due to its close proximity may lead to acoustic feedback. In order to avoid the feedback, an ear mould may be used, which may inturn lead to occlusion effect [4]. One of the frequently adopted solutions to this issue is an open fitting hearing aid, which may further lead to acoustic feedback. Acoustic feedback can result in a deteriorated listening experience [5]. Adaptive signal processing methods have been recently employed to improve the listening experience in the presence of acoustic feedback. An adaptive feedback cancellation (AFC) scheme essentially tries to model the acoustic feedback path using an adaptive finite impulse response (FIR) filter and thereby internally cancelling the feedback signal [6]. The weights of the adaptive FIR filter are traditionally updated using a normalized least mean square (NLMS) algorithm [7,8], with an objective to minimize the difference between the internally generated feedback cancellation signal and the actual acoustic feedback signal picked up by the microphone.

The closed loop nature of the AFC can lead to a biased estimation of the acoustic feedback path because of the finite correlation between the loudspeaker and microphone signals [9–11]. Several schemes have been proposed in the literature to overcome this limitation. One such method is the probe noise injection scheme, whereby an inaudible uncorrelated probe noise is injected before the loudspeaker to reduce the correlation effects [12–14]. A de-correlation approach has been presented in [4,8], in which the signals used in the weight update of adaptive feedback canceller are filtered through two extra adaptive FIR filters before using it in the update process. The presence of the three adaptive filters in the AFC increases the computational load. Another popular approach to reduce the correlation issue is the prediction error method [9,15]. A two microphone feedback cancellation scheme for BTE hearing aids has been recently proposed [12]. The two microphone scheme uses the signals from a microphone placed near the ear canal and another microphone present behind the ear to enhance the feedback cancellation performance. The microphone near the ear canal picks up the desired sound signal and a significant portion of the feedback signal. The BTE microphone receives only a significantly smaller magnitude feedback signal in addition to the desired signal. Such a placement of the two microphones may aid both natural hearing and higher gain. The two microphone feedback cancellation method

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has been shown in literature to provide improved feedback cancellation over single microphone BTE hearing aids as well the ones which uses prediction error method for feedback cancellation [12,16,17].

The computational complexity of the NLMS algorithm based feedback cancellation in a single microphone BTE hearing aid is proportional to the length of the impulse response of the acoustic feedback path. The computational complexity can be reduced by implementing the algorithm in the frequency domain, using the principles of fast Fourier transform (FFT) algorithm [18]. A frequency domain Kalman filter implementation of a feedback cancellation scheme, which uses a prediction error approach has been reported in [19]. A variable step size implementation of the prediction error method based frequency domain scheme has also been presented in [10]. In an endeavour to reduce the computational load in a two microphone feedback cancellation scheme, a frequency domain implementation has been presented in this paper. In addition to the computational advantages, the frequency domain implementation is expected to improve the convergence speed by reducing the eigenvalue spread. In order to further improve the convergence behaviour we have incorporated a bin dependent step size in the proposed frequency domain implementation of a two microphone feedback cancellation scheme [20]. The introduction of a bin dependent step size requires the input signal to the microphone and the output signal from the loudspeaker to be uncorrelated. This condition is satisfied in implementations of bin dependent step size scheme in echo cancellation. However, the finite correlation between the two signals makes the implementation of such a method in an acoustic feedback cancellation scheme in a hearing aid a difficult task. To reduce this issue, frequency shifting (FS) has been introduced in the forward path in the proposed approach [21–24].

The rest of the paper is organized as follows. A brief description of the dual microphone BTE hearing aid is made in Section 2. The frequency domain implementation of a two microphone BTE hearing aid, with bin dependent step size control approach is presented in Section 3. A comparison of computational complexity is also made in the section. The performance of the new schemes have been tested using a simulation study in Section 4 and the concluding remarks are made in Section 5.

2. Dual microphone BTE hearing aid

The hearing aid considered in this study is a BTE hearing aid, with two microphones. The first microphone is placed near the ear canal and the other is situated behind the ear. The locations of the microphones are selected in such a way that the two microphones receive the desired sound signal with almost equal magnitude and the feedback signal with significantly different magnitude. Fig. 1 shows the basic schematic diagram of a two microphone BTE hearing aid. Even though there are both continuous time and discrete time signals present in a real implementation of a BTE hearing aid, for the sake of simplicity, we have considered all the signals to be of discrete time nature. In the figure, the microphone-1 denotes the microphone near the ear canal and microphone-2 represents the microphone which is behind the ear. The two microphones are hereafter denoted by M_i for $i = 1, 2$. The microphone situated behind the ear is used to estimate the incoming desired signal and thereby reducing the effect of bias. The separation distance between the two microphones should be relatively small compared to the distance between the microphones and the signal source. The distance between microphone-2 and feedback source (loudspeaker) should be relatively large compared to the distance between microphone-1 and loudspeaker. The microphone near the ear canal is expected to pick up both the desired signal and a significant portion of the feedback signal. On the other hand, the microphone which is behind the ear is expected to pick up only a smaller portion of the feedback signal in addition to the desired sound signal [12,17].

Let $x_i(n)$ be the desired signal captured by M_i, $f_i(n)$ be the feedback signal received at M_i and $m_i(n) = x_i(n) + f_i(n)$ be the output of M_i. The feedback path from the output of the loudspeaker to M_i has a transfer function $G_i(q)$. In a two microphone method, an adaptive FIR filter with a transfer function $\widehat{G}_1(q)$ is employed to model the true acoustic feedback path $G_1(q)$ and thus compensate for the effect of the acoustic feedback. The forward path also contains an amplifier with a transfer function $F(q)$, which represents a delay together with a gain. The forward path delay before amplification can reduce the correlation between the loudspeaker input and the desired microphone input [25]. In addition, in the forward path, there is a fixed delay of d_m samples, which helps the system in maintaining causality. In [21–23], it has been reported that frequency shifting can be used to alleviate the bias issue. A frequency shift up to 5 Hz is inaudible both for speech and music signals [26]. The frequency shifting operation is carried out in the forward path to obtain $y(n)$, which is fed to the loudspeaker.

In a conventional two microphone BTE hearing aid, the adaptive weight vector (of length N) $\hat{\mathbf{g}}_1(n) = [\hat{g}_{1_0}(n), \hat{g}_{1_1}(n), \ldots, \hat{g}_{1_{N-1}}(n)]^T$ of a filter with transfer function $\widehat{G}_1(q)$ is updated in such a way as to minimize the cost function $\xi = \mathrm{E}[e^2(n)] \approx e^2(n)$, with

$$e(n) = \widetilde{x}_1(n) - \hat{x}_1(n), \tag{1}$$

and $\mathrm{E}[\cdot]$ representing the expectation operator. In (1), $\widetilde{x}_1(n)$ is the signal $m_1(n) - \widehat{f}_1(n)$ delayed by d_m samples, where

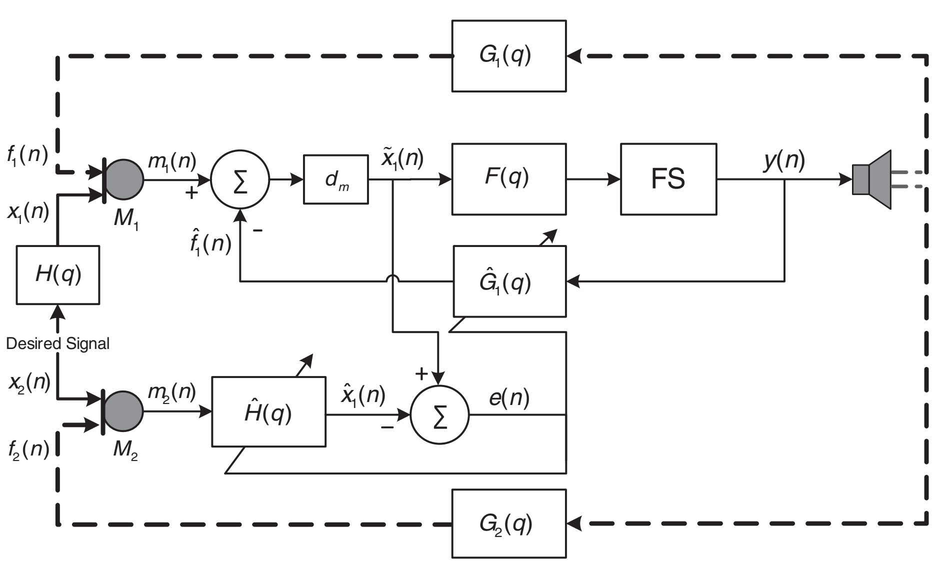

Fig. 1. Two-microphone method feedback cancellation in digital hearing aid with frequency shifting.