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Improved prediction error filters for adaptive feedback cancellation in hearing aids

Kim Ngo [a,1], Toon van Waterschoot [a,*], Mads Græsbøll Christensen [b], Marc Moonen [a], Søren Holdt Jensen [c]

[a] KU Leuven, Department of Electrical Engineering (ESAT-SCD)/IBBT Future Health Department, Kasteelpark Arenberg 10, 3001 Leuven, Belgium
[b] Aalborg University, Department of Architecture, Design & Media Technology (AD:MT), Audio Analysis Lab, Sofiendalsvej 11, 9200 Aalborg, Denmark
[c] Aalborg University, Department of Electronic Systems, Fredrik Bajers Vej 7, 9220 Aalborg, Denmark

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ABSTRACT

Acoustic feedback is a well-known problem in hearing aids, caused by the undesired acoustic coupling between the hearing aid loudspeaker and microphone. Acoustic feedback produces annoying howling sounds and limits the maximum achievable hearing aid amplification. This paper is focused on adaptive feedback cancellation (AFC) where the goal is to adaptively model the acoustic feedback path and estimate the feedback signal, which is then subtracted from the microphone signal. The main problem in identifying the acoustic feedback path model is the correlation between the near-end signal and the loudspeaker signal caused by the closed signal loop, in particular when the near-end signal is spectrally colored as is the case for a speech signal. This paper adopts a prediction-error method (PEM)-based approach to AFC, which is based on the use of decorrelating prediction error filters (PEFs). We propose a number of improved PEF designs that are inspired by harmonic sinusoidal modeling and pitch prediction of speech signals. The resulting PEM-based AFC algorithms are evaluated in terms of the maximum stable gain (MSG), filter misadjustment, and computational complexity. Simulation results for a hearing aid scenario indicate an improvement up to 5–7 dB in MSG and up to 6–8 dB in terms of filter misadjustment.

1. Introduction

Acoustic feedback is a well-known problem in hearing aids. It is caused by the undesired acoustic coupling between the hearing aid loudspeaker and microphone. It has become an even more important problem due to two recent trends in hearing aids design, both of which further increase the loudspeaker–microphone coupling: (1) the use of open fittings, in which the ear canal is intentionally left open to avoid the occlusion effect and hence improve the user comfort, (2) the use of smaller form factors which implicitly reduce the hearing aid dimensions, including the loudspeaker–microphone distance. The acoustic loudspeaker–microphone coupling results in a closed signal loop which may become unstable, resulting in acoustic oscillations known as howling. Therefore, as a consequence of acoustic feedback, the speech intelligibility and listening comfort for hearing aid users is compromised in two ways: acoustic feedback may result in

* Corresponding author. Tel.: +32 16 321788; fax: +32 16 321970.
E-mail addresses: kim.ngo@citrix.com (K. Ngo), toon.vanwaterschoot@esat.kuleuven.be (T. van Waterschoot), mgc@create.aau.dk (M. Græsbøll Christensen), marc.moonen@esat.kuleuven.be (M. Moonen), shj@es.aau.dk (S. Holdt Jensen).
[1] Present address: Citrix Online GmbH, Theaterstrasse 6, 01067 Dresden, Germany.

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howling artifacts that interfere with desired speech components, and it may severely constrain the maximum hearing aid amplification that can be used if howling, due to instability, is to be avoided. In many cases this maximum amplification is too small to compensate for the hearing loss, i.e., the auditory loss in the user, and therefore feedback cancellation is considered a crucial component in present-day hearing aids [1–3].

The goal of adaptive feedback cancellation (AFC) is to adaptively model the acoustic feedback path and estimate the feedback signal, which is then subtracted from the microphone signal. The main problem in identifying the acoustic feedback path model is the correlation between the near-end signal and the loudspeaker signal caused by the closed signal loop, in particular when the near-end signal is spectrally colored, as it is the case for a speech signal. This correlation problem causes standard adaptive filtering algorithms to converge to a biased solution. A major challenge is therefore to reduce the correlation between the near-end signal and the loudspeaker signal. Typically, there exist two approaches to achieve this decorrelation [4], i.e., decorrelation in the closed signal loop and decorrelation in the adaptive filtering circuit. Recently proposed methods for decorrelation in the closed signal loop consist of inserting all-pass filters in the forward path of the hearing aid [5], applying a clipping operation to the feedback signal arriving at the microphone [6], or inserting a probe noise signal into the closed signal loop [7]. However, decorrelation in the closed signal loop implicitly affects the desired (near-end) speech component, hence a trade-off between signal decorrelation and perceptual degradation is unavoidable [4].

Alternatively, an unbiased identification of the feedback path model can be achieved by applying decorrelation in the adaptive filtering circuit, i.e., by first prefiltering the loudspeaker and microphone signals with the inverse near-end signal model before feeding these signals to the adaptive filtering algorithm [8–10]. In this way, the desired (near-end) speech component remains unaffected and so the signal decorrelation does not induce any perceptual degradation. The near-end signal model and the feedback path model can then be jointly estimated using the prediction error method (PEM) [11]. For PEM-based AFC with near-end speech signals, a linear prediction (LP) model is commonly used [8,12]. Other near-end speech signal models have been based on a pole-zero LP (PZLP), a warped LP, or a pitch prediction model, cascaded with an LP model [9].

Recently, the use of a harmonic sinusoidal near-end signal model for PEM-based AFC has been proposed by the authors [13,14] and was shown to improve the AFC performance compared to using a PZLP near-end signal model. The main difference with the PZLP model of [9] is that the near-end signal model in [13,14] and the corresponding pitch estimation [15,16] rely on harmonicity, i.e., the sinusoidal frequencies are assumed to be integer multiples of a fundamental frequency, which in the near-end speech case follows naturally from voiced speech being quasi-periodic, whereas the sinusoidal frequencies in a PZLP model are estimated independently [17]. In [13] it has been shown how different pitch estimation techniques based on subspace shift-invariance, subspace orthogonality, and optimal filtering [15] can be employed to improve the resulting AFC performance. In [14] it has further been shown how the harmonic sinusoidal near-end signal model and the corresponding design of the prediction error filter (PEF) can be improved by including a variable model order (corresponding to the number of near-end signal harmonics) and a variable amplitude, next to a variable pitch.

In this paper, different designs for the PEF are analyzed and it is shown that a more accurate modeling of the near-end signal generally results in a significant performance improvement in PEM-based AFC in terms of the achieved maximum stable gain (MSG) and filter misadjustment. As compared to our previous work in [13,14], two improved PEF designs are presented, which are inspired by harmonic sinusoidal speech models, as used in speech applications other than AFC, for the extraction, separation, and enhancement of periodic signals [15,16]. The first improvement is based on a refinement of the harmonic sinusoidal model such as to incorporate a number of typical speech features into the PEF design. Since speech is highly non-stationary, the PEF should be able to adapt quickly both in terms of tracking the pitch, number of harmonics, and amplitude changes as well as in terms of characterizing voiced versus unvoiced frames. To this end, the PEM-AFC algorithm proposed in [14] is extended by including a non-intrusive voiced–unvoiced detection algorithm in the PEF design, and the different impact of voiced and unvoiced speech frames on the resulting PEF design is investigated. The second improvement is based on the use of so-called amplitude and phase estimation (APES) filters in the PEF design, which are specifically suited for periodic signals and are optimal and signal-adaptive given the observed signals [18]. Both improvements are then evaluated in terms of the resulting PEM-based AFC performance. Simulation results for a hearing aid scenario indicate an improvement up to 5–7 dB in MSG and up to 6–8 dB in terms of filter misadjustment. Finally, a computational complexity analysis of the competing PEF designs is conducted, which in particular supports the computational benefit of including a voiced–unvoiced detector.

The paper is organized as follows. Section 2 defines the notation and reviews the PEM-based AFC concept, while Section 3 introduces the harmonic sinusoidal near-end signal model. Section 4 elaborates on the different existing and proposed approaches to PEF design. Section 5 shows the effect of voiced–unvoiced detection on the PEF performance, and Section 6 contains a computational complexity analysis. In Section 7 experimental results are presented. The work is summarized in Section 8.

2. Adaptive feedback cancellation

The typical AFC set-up is shown in Fig. 1. The microphone signal is given by

$$y(t) = v(t) + x(t) = v(t) + F(q,t)u(t) \tag{1}$$

where q denotes the time shift operator, e.g., $q^{-k}u(t) = u(t-k)$, and t is the discrete time variable. $F(q,t) = f_0(t) + f_1(t)q^{-1} + \cdots + f_{n_F}(t)q^{-n_F}$ represents a linear finite impulse