Available online at www.sciencedirect.com
ScienceDirect
Signal Processing 88 (2008) 571–581
SIGNAL PROCESSING
www.elsevier.com/locate/sigpro

Generalized sidelobe canceller based combined acoustic feedback- and noise cancellation

Geert Rombouts*, Ann Spriet, Marc Moonen

Katholieke Universiteit Leuven/ESAT-SCD, Kasteelpark Arenberg 10, 3001 Heverlee, Belgium

Received 16 March 2007; received in revised form 14 August 2007; accepted 31 August 2007
Available online 11 September 2007

Abstract

We propose a combination of the well-known generalized sidelobe canceller (GSC) or Griffiths–Jim beamformer, and the so-called PEM-AFROW algorithm for joint estimation under closed loop conditions of a room impulse response and a desired speech signal model, resulting in a system for multimicrophone combined acoustic feedback and noise cancellation. For specific applications (e.g. public address systems), the computational complexity may be reduced dramatically compared to state-of-the-art proactive acoustic feedback cancellers, while feedback cancellation performance is only marginally degraded.

Keywords: Generalized sidelobe canceller; Acoustic feedback cancellation; Whitening

1. Introduction

Acoustic feedback is a well-known phenomenon that appears in systems that have an electro-acoustic closed loop. Such systems (Fig. 1) consist of a microphone, a forward path gain g and delay q^{-D} (remark: the delay operator q^{-1} is used to denote a unit delay in the time domain, as is often done in system identification literature, while z^{-1} is a unit delay in the z-domain), a loudspeaker and a room impulse response (RIR) $\mathbf{f}_r$. Hearing aids (HA) and public address (PA) systems are typical examples. If the loop gain exceeds unity for frequencies ω_i where the loop phase is $2n_i\pi$ radians (with n_i integer), system instability becomes audible as a loud 'howling' sound (the Larsen-effect). In such applications, background *noise* is also picked up by the microphones, and degrades the signal quality. Examples are babble noise in the vicinity of the speaker, or cabin noise in a train.

Many approaches can be found in literature for acoustic feedback cancellation as well as for noise cancellation.

Most of the *acoustic feedback cancellation* techniques that have been derived up till now are single channel techniques [1–5] (although some multichannel examples exist [6,7]). Traditional approaches are mostly 'reactive', as they allow the system to become unstable in order to then identify the frequencies where acoustic feedback occurs, and introduce notch filters for these frequencies into the signal path. More recent approaches [8–10] are 'proactive' and do not introduce signal distortion, as they are based on an adaptive filter that models

*Corresponding author.
E-mail address: geert.rombouts@esat.kuleuven.be (G. Rombouts).

0165-1684/$ - see front matter
doi:10.1016/j.sigpro.2007.08.018

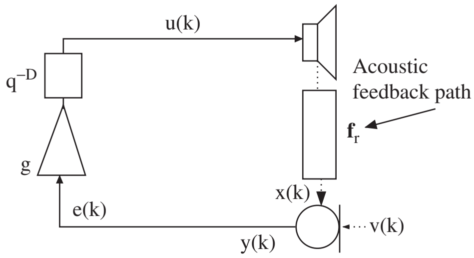

Fig. 1. An electro-acoustic loop.

the RIR, and insert a so-called controller into the scheme, which effectively removes the component from the microphone signal that stems from the loudspeaker. Such approaches are based on acoustic echo cancellation procedures, where an additional signal modelling is required to avoid a biased RIR estimate due to the correlation between the desired speech signal ("near-end signal") and the ("far-end") loudspeaker signal (indeed meant to be a processed (amplified) version of the speech signal). The PEM-AFROW algorithm of [8], applies signal model based prewhitening and effectively achieves an unbiased estimate. It is a prediction error method, where the whitening process performs row-operations on a least-squares system of equations, hence the name. Modelling the RIR may, however, be computationally expensive, especially in the PA context, and therefore we will concentrate on a simpler approach.

Good *noise cancellation methods* typically rely on multiple microphones (microphone arrays) providing spatial information on the location of the desired speech source and the noise source. Single channel approaches can only use spectral information to discriminate between noise and speech, and their performance is inferior to multichannel methods (e.g. they are often prone to a phenomenon called 'musical noise', although remedies exist for this [11]).

In several applications (e.g. HA and PA again), it is desirable to combine feedback- and noise cancellation. A simple cascading of both systems, even for the single channel case, does not lead to an optimal solution (see Section 4.1). It is also expected that PEM-AFROW based feedback cancellation schemes could benefit from additional multichannel processing. In this paper, we first derive a PEM-AFROW based proactive multichannel feedback cancellation algorithm which does not introduce signal distortion, and which especially for PA applications exhibits a dramatically reduced computational complexity compared to existing adaptive filtering based proactive feedback cancellation schemes. We show that this scheme is robust to noise, and in certain circumstances even provides some noise reduction. Then we extend this setup by adding multiple noise references, such that it exhibits both feedback- and noise cancellation behaviour in low-reverberant environments.

In Section 2, the Griffiths–Jim (or GSC) based multichannel noise reduction scheme is reviewed. In Section 3, it is shown that for white input signals, a GSC scheme can already be used for feedback cancellation. When signal model based whitening filters are added, it can also provide feedback cancellation for (non-white) speech input signals. In Section 4, we first discuss the problems inherent to a simple cascading of a noise-reduction and a feedback-reduction scheme, and then proceed by stating and analyzing a minimization problem corresponding to the GSC-based combined noise- and feedback cancellation scheme, and describing an implementation (the PEM-AFROW algorithm). In Section 5, a number of simulations are shown for different scenarios, which prove the effectiveness of the new scheme. Conclusions are given in Section 6.

2. Noise reduction

A traditional approach to multichannel noise reduction is the so-called Griffiths–Jim beamformer [12] or generalized sidelobe canceller (GSC), see Fig. 2, which we briefly review in this section. For the time being, we assume a simple scenario with one desired speech source $v(k)$ and one noise source $x(k)$. The input vector of the M-microphone array can be written as

$$\mathbf{y}(k) = \mathbf{v}(k) + \mathbf{x}(k),$$

$$\mathbf{y}(k) = \begin{pmatrix} \mathbf{y}^{(1)}(k) \\ \mathbf{y}^{(2)}(k) \\ \vdots \\ \mathbf{y}^{(M)}(k) \end{pmatrix},$$

i.e. as a stacked vector of the input vectors for microphones $1 \ldots M$,

$$\mathbf{y}^{(i)}(k) = \begin{pmatrix} y^{(i)}(k) \\ y^{(i)}(k-1) \\ \vdots \\ y^{(i)}(k-N_m+1) \end{pmatrix}.$$