ADAPTIVE FEEDBACK CANCELLATION IN HEARING AIDS BASED ON THE AUTOREGRESSIVE MODELLING OF THE DESIRED SIGNAL

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Acoustic feedback limits the maximum amplification that can be used in a hearing aid. However, because of the presence of a closed signal loop, i.e., the so-called forward path $G(q)$, standard continuous adaptation feedback cancellers (CAF) fail to provide a reliable feedback path estimate if the desired signal $x[k]$ is spectrally colored [1, 2]. In this paper, we discuss adaptive feedback cancellation techniques that are based on direct closed-loop identification of the feedback path as well as the (autoregressive) modelling of the desired signal. Simulation results demonstrate that the proposed techniques outperform the standard CAF algorithm and the filtered-X algorithm of [2] that uses a fixed estimate of the average desired signal spectrum.

Notation. The symbol q^{-1} denotes the discrete-time delay operator, i.e., $q^{-1}u[k] = u[k-1]$. The filter $\mathbf{f}=[\, f_0 \quad \cdots \quad f_{L_F-1}\,]^T$ is represented as a polynomial $F(q)$ in q, i.e.,$F(q) = f_0 + f_1 q^{-1} + \ldots + f_{L_F-1}q^{-L_F+1}$. Filtering $u[k]$ with $F(q)$ is denoted as $F(q)u[k]$ or $\mathbf{f}^T\mathbf{u}[k]$ with $\mathbf{u}[k]=[\, u[k] \quad u[k-1] \quad \cdots \quad u[k-L_F+1]\,]^T$. The filter $F(q,k)$ refers to a time-varying filter with coefficient vector $\mathbf{f}[k]$.

Closed-loop system set-up

Figure 1(a) depicts the closed-loop system set-up of a hearing aid. The *open-loop system to be identified* is described by

$$y[k] = F(q)u[k] + x[k]; \quad x[k] = H(q)w[k] = \frac{1}{1+q^{-1}P(q)}w[k] \tag{1}$$

where $y[k]$ is the microphone signal, $u[k]$ the loudspeaker signal and $w[k]$ is white noise. This implies that the desired signal $x[k]$ is modelled as an autoregressive (AR) random process, which is a common assumption for speech[1] and other audio signals. The output signal $y[k]$ is fed-back to the input $u[k]$ according to $u[k] = G(q)\left(y[k] - \hat{F}_0(q)u[k]\right)$. The filter $\hat{F}_0(q)$ is an initial estimate of the feedback path $F(q)$. During adaptation, it is typically replaced with the feedback path estimate $\hat{F}(q)$. In the sequel, we assume that the forward path $G(q)$ contains a delay $d_f \geq 1$ sample.

Direct closed-loop identification of the feedback path

The direct method of closed-loop identification is only applicable with some specific open-loop identification methods, such as the prediction error method (PEM). The PEM produces an estimate $\hat{F}(q)$ and $A(q)$ of the feedback path $F(q)$ and the inverse desired signal model $H^{-1}(q)$, respectively, by minimizing the prediction error $e_p[k]$

$$e_p[k] = A(q)\left(y[k] - \hat{F}(q)u[k]\right). \tag{2}$$

To avoid the trivial solution $A(q) = 0$, the filter $A(q)$ is constrained to be monic, i.e., $A(q) = 1 + q^{-1}\bar{A}(q)$.

A standard optimization criterion for $\hat{F}(q)$ and $A(q)$ is to minimize the time-average of squared prediction errors, i.e., $\sum_{\bar{k}=0}^{k} \lambda^{k-\bar{k}} e_p^2[\bar{k}]$, where $\lambda \in (0,1]$ is a forgetting factor. In general though, the feedback path $F(q)$ and the inverse desired signal model $H^{-1}(q)$ are not both identifiable in the closed-loop system at hand [2]. In [3], we have shown that not only inserting non-linearities or a probe signal, but also adding a delay in the forward path or the cancellation path can render the system identifiable. If the total delay $d = d_f + d_c \geq L_{H^{-1}}$ with d_c the common delay in the feedback path $F(q)$ and the cancellation path $\hat{F}(q)$, $H(q)$ and $F(q)$ can be both identified in closed-loop. Assuming that $L_A \geq L_{H^{-1}}$, we set $d_f \geq L_A$.

Depending on the definition of $e_p[k]$, different update equations for $\hat{F}(q)$ and $A(q)$ are obtained.

[1]Note that the white noise assumption is not satisfied for periodic signals such as voiced speech segments.

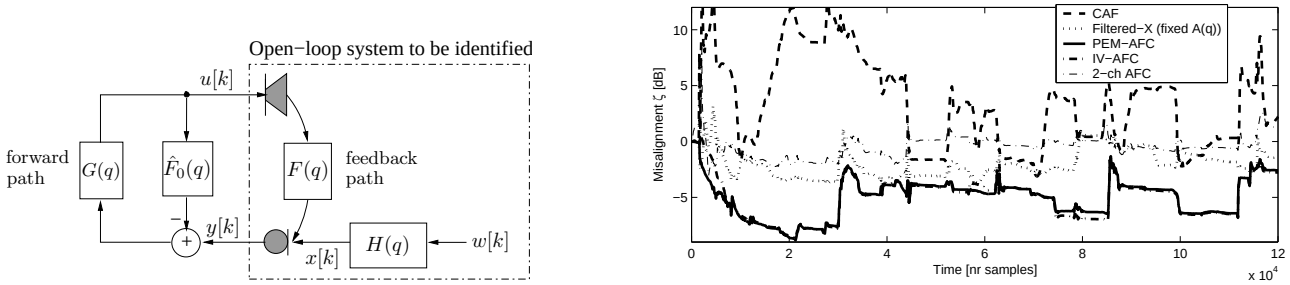


Figure 1: (a) Closed-loop system set-up of a hearing aid. (b) Misalignment ζ of the CAF, the 2-ch AFC, the filtered-X method with fixed $A(q)$, the PEM-AFC and the IV-AFC as a function of time.

- *Two-channel adaptive feedback canceller (2-ch AFC)[3]*
  In the 2-ch AFC, the prediction error $e_p[k]$ is formulated as $e_p[k]{=}A(q)y[k]{+}B(q)u[k]$, where $B(q)$ models $-H^{-1}(q)F(q)$. This results in an optimization criterion that is linear in $A(q)$ and $B(q)$, so that the ordinary LMS or RLS updating algorithms can be applied. The main drawback of this technique is that the same adaptation speed has to be used for estimating $H^{-1}(q)$ and $F(q)$. In practice, the desired signal model $H(q)$ is often highly time-varying, while the feedback path $F(q)$ varies more slowly in time, motivating the use of a faster adaptation speed for $H^{-1}(q)$ than for $F(q)$.
- *PEM-based adaptive feedback canceller (PEM-AFC)[3]*
  The PEM-AFC $\{A(q), \hat{F}(q)\}$ minimizes $\sum_{\bar{k}=0}^{k} \lambda^{k-\bar{k}} e_p^2[\bar{k}]$ with $e_p[k]$ defined in (2). Applying the stochastic Gauss-Newton update equation and forcing the off-diagonal blocks of the Hessian approximation to zero, results in a decoupled update equation for $A(q)$ and $\hat{F}(q)$. The update equation for the feedback path $\hat{\mathbf{f}}[k]$ corresponds to the filtered-X algorithm:
  $$\hat{\mathbf{f}}[k] = \hat{\mathbf{f}}[k-1] + \mathbf{R}_{\hat{\mathbf{f}}}^{-1}[k]\mathbf{u}_p[k]\epsilon_p[k] \text{ with } \epsilon_p[k] = A(q,k-1)\left(y[k] - F(q,k-1)u[k]\right)$$
  $$\mathbf{R}_{\hat{\mathbf{f}}}[k] = \lambda_{\hat{\mathbf{f}}}[k]\mathbf{R}_{\hat{\mathbf{f}}}[k-1] + \mathbf{u}_p[k]\mathbf{u}_p^T[k]$$
  with $\mathbf{u}_p[k] = \left[\begin{array}{ccc} u_p[k] & \cdots & u_p[k-L_{\hat{F}}+1] \end{array}\right]^T$ and $u_p[k] = A(q,k-1)u[k]$. If the estimate $A(q)$ of $H^{-1}(q)$ contains a group delay, the correction term in the adaptation of $\hat{\mathbf{f}}[k]$ is delayed. As a result, instability may occur when the adaptive filter $\hat{\mathbf{f}}[k]$ is fast time-varying, e.g., in highly time-varying environments.
- *IV-based adaptive feedback canceller (IV-AFC)[4]*
  Formulating the prediction error $e_p[k]$ as $e_p[k] = y_p[k] - F(q)u_p[k]$ with $y_p[k] = A(q)y[k]$ and $u_p[k] = A(q)u[k]$, results in an alternative update equation for $\hat{\mathbf{f}}$ that can be interpreted as an (optimal) instrumental variable (IV) method with pre-filtering of the data:
  $$\hat{\mathbf{f}}[k] = \hat{\mathbf{f}}[k-1] + \mathbf{R}_{\hat{\mathbf{f}}}^{-1}[k]\mathbf{u}_p[k]\tilde{\epsilon}_p[k] \text{ with } \tilde{\epsilon}_p = y_p[k] - \hat{\mathbf{f}}^T[k-1]\mathbf{u}_p[k].$$
  In contrast to the PEM-AFC, any change to the adaptive filter $\hat{\mathbf{f}}$ has an immediate effect on the adaptation error $\tilde{\epsilon}_p[k]$, irrespective of the group delay in $A(q)$. Hence, the IV-AFC does not suffer from stability problems.

Simulation results

Figure 1(b) depicts the misalignment $\zeta(\mathbf{f}[k], \hat{\mathbf{f}}[k]) = \left\|\mathbf{f}[k] - \hat{\mathbf{f}}[k]\right\|_2^2 / \left\|\mathbf{f}[k]\right\|_2^2$ of the CAF, the 2-ch AFC, the PEM-AFC, the IV-AFC as a function of time for a real speech signal $x[k]$ consisting of sentences spoken by a male speaker. The sampling frequency equals 16 kHz. For comparison, the filtered-X method of [2] with an AR model $A(q)$ of the (unknown) long-term average spectrum of the speech signal is depicted too. The filter length L_F of the feedback path $F(q)$ equals 50. The gain $|G(q)|$ has been set to 5. The delay d_f in $G(q)$ equals 30 samples, i.e., 1.9 msec.The forgetting factor $\lambda_{\hat{\mathbf{f}}}$ in all algorithms was set to 0.99985. In the IV-AFC and the PEM-AFC, the filter $A(q)$ was updated by means of the Burg lattice algorithm with $\lambda_{\bar{\mathbf{a}}} = 1 - \frac{1}{160}$. The filter length $L_A = 21$, $L_B = 70$, $L_{\hat{F}} = 50$. The 2-ch AFC outperforms the CAF. The IV-AFC and the PEM-AFC outperform the CAF, the 2-ch AFC and the filtered-X method of [2].

References

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