Katholieke Universiteit Leuven

Departement Elektrotechniek ESAT-SISTA/TR 2004-226

Adaptive feedback cancellation in hearing aids based on the autoregressive modelling of the desired signal [1]

Ann Spriet[2], Ian Proudler[3] ,Marc Moonen[4] ,Jan Wouters[5]

Submitted for publication in the first annual IEEE BENELUX/DSP Valley Signal Processing Symposium, Antwerp, Belgium, April 2005.

[1]This report is available by anonymous ftp from *ftp.esat.kuleuven.ac.be* in the directory *pub/sista/spriet/reports/04-226.pdf*

[2]K.U.Leuven, Dept. of Electrical Engineering (ESAT), SISTA, Kasteelpark Arenberg 10, 3001 Leuven-Heverlee, Belgium, Tel. 32/16/32 18 99, Fax 32/16/32 19 70, WWW: *http://www.esat.kuleuven.ac.be/sista*. E-mail: *ann.spriet@esat.kuleuven.ac.be*. K.U.Leuven, Lab. Exp. ORL/ ENT-Dept., Kapucijnenvoer 33, 3000 Leuven, Belgium, Tel. 32/16/33 24 15, Fax 32/16/33 23 35, WWW: *http://www.kuleuven.ac.be/exporl/Lab/Default.htm*. This research work was carried out at the ESAT laboratory and Lab. Exp. ORL of the Katholieke Universiteit Leuven, in the the frame of IUAP P5/22 ('Dynamical Systems and Control: Computation, Identification and Modelling'), the Concerted Research Action GOA-MEFISTO-666 (Mathematical Engineering for Information and Communication Systems Technology)of the Flemish Government, Research Project FWO nr.G.0233.01 ('Signal processing and automatic patient fitting for advanced auditory prostheses'), IWT project 020540 ('Innovative Speech Processing Algorithms for Improved Performance of Cochlear Implants') and was partially sponsored by QinetiQ. The scientific responsibility is assumed by its authors.

[3]QinetiQ Ltd., Malvern Technology Centre, St Andrews Road, Malvern, Worcestershire, WR14 3PS, UK. Tel: +44 (0) 1684 894228, Fax: +44 (0) 1684 896502, E-mail: *i.proudler@signal.qinetiq.com*.

[4]K.U.Leuven, Dept. of Electrical Engineering (ESAT), SISTA, Kasteelpark Arenberg 10, 3001 Heverlee, Belgium, Tel. 32/16/32 17 09, Fax 32/16/32 19 70, WWW: *http://www.esat.kuleuven.ac.be/sista*. E-mail: *marc.moonen@esat.kuleuven.ac.be*. Marc Moonen is a professor at the Katholieke Universiteit Leuven.

[5]K.U.Leuven, Lab. Exp. ORL, Dept. Neurowetenschappen, Kapucijnenvoer 33, 3000 Leuven, Belgium, Tel. 32/16/33 23 42, Fax 32/16/33 23 35, WWW: *http://www.kuleuven.ac.be/exporl/Lab/Default.htm* E-mail: *jan.wouters@uz.kuleuven.ac.be*. Jan Wouters is a professor at the Katholieke Universiteit Leuven.

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

[1,2]*Ann Spriet,* [3]*Ian Proudler,*[1]*Marc Moonen,*[2]*Jan Wouters*

[1] ann.spriet@esat.kuleuven.ac.be

[1]Katholieke Universiteit Leuven - ESAT, Kasteelpark Arenberg 10, B-3001 Leuven, Belgium

[2] Katholieke Universiteit Leuven - Lab. Exp. ORL, Kapucijnenvoer 33, B-3000 Leuven, Belgium

[3]QinetiQ Ltd., Malvern Techn. Centre, St Andrews Road, Malvern, Worcestershire, WR14 3PS, UK

ABSTRACT

In this paper, we propose 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, with the aim of providing an unbiased feedback path estimate for unknown and time-varying desired signals. Simulation results demonstrate that the proposed techniques outperform the standard continuous-adaptation algorithm and the filtered-X algorithm that uses a fixed estimate of the average desired signal spectrum.

1. INTRODUCTION

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 [2, 3], an adaptive feedback canceller based on the direct method of closed-loop identification and a *fixed* estimate of the desired signal model has been proposed. It has been shown that an unbiased feedback path estimate can be obtained by means of a filtered-X algorithm if the desired signal $x[k]$ can be modelled as $H(q)w[k]$, with $w[k]$ white noise, and the desired signal model $H(q)$ is known. In practice, $H(q)$ is unknown and highly time-varying so that it is desirable to also estimate this model adaptively.
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.

This research was carried out at ESAT and Lab. Exp. ORL of K.U. Leuven, in the frame of IUAP P5/22, the Concerted Research Action GOA-MEFISTO-666, FWO Project nr. G.0233.01, *Signal Processing and automative patient fitting of auditory prostheses*, IWT project 020540, *Innovative speech processing algorithms for improved performance in cochlear implants* and was partially sponsored by QinetiQ. The scientific responsibility is assumed by its authors.

Notation

The symbol q^{-1} denotes the discrete-time delay operator, i.e., $q^{-1}u[k] = u[k-1]$. A discrete-time filter with coefficient vector $\mathbf{f} = \begin{bmatrix} f_0 & f_1 & \cdots & f_{L_F-1} \end{bmatrix}^T$ and filter length L_F is represented as a polynomial $F(q)$ in q, i.e.,

$$F(q) \quad = \quad f_0 + f_1 q^{-1} + \ldots + f_{L_F-1} q^{-L_F+1}. \quad (1)$$

Filtering $u[k]$ with $F(q)$ is denoted as $F(q)u[k]$ or $\mathbf{f}^T\mathbf{u}[k]$ with $\mathbf{u}[k] = \begin{bmatrix} u[k] & u[k-1] & \cdots & u[k-L_F+1] \end{bmatrix}^T$. The filter $F(q,k)$ refers to a time-varying filter with coefficient vector $\mathbf{f}[k]$.

2. CLOSED-LOOP SYSTEM SET-UP

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

$$y[k] = F(q)u[k] + x[k], \quad (2)$$

where $y[k]$ is the microphone signal and $u[k]$ the loudspeaker signal. In general, the desired signal $x[k]$ is an audio signal (e.g., a speech signal). Many audio signals can be closely approximated by a low-order autoregressive (AR) random process

$$x[k] = H(q)w[k] = \frac{1}{1 + q^{-1}P(q)} w[k], \quad (3)$$

with $w[k]$ white noise[1] and $P(q)$ an FIR filter.
The output signal $y[k]$ is fed-back to the input $u[k]$ according to

$$u[k] = G(q)\left(y[k] - \hat{F}_0(q)u[k]\right). \quad (4)$$

Using (2) and (4), the input $u[k]$ can be written as

[1]Note that the white noise assumption is not satisfied for periodic signals such as voiced speech segments. E.g., the excitation $w[k]$ of a voiced speech segment is an impulse train [4].

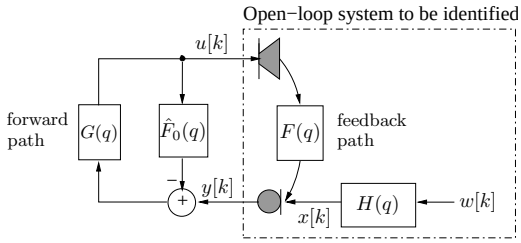


Figure 1: Closed-loop system set-up of a hearing aid.

$$u[k] = \underbrace{\frac{G(q)}{1 - G(q)\left(F(q) - \hat{F}_0(q)\right)}}_{C(q)} x[k]. \qquad (5)$$

The filter $\hat{F}_0(q)$ in the feedback cancellation path is an initial estimate of the feedback path $F(q)$ chosen such that the closed-loop system $C(q)$ is stable. 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.

3. DIRECT CLOSED-LOOP IDENTIFICATION OF THE FEEDBACK PATH

In this section, we describe 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.

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 $e_p[k]$

$$e_p[k] = A(q)\left(y[k] - \hat{F}(q)u[k]\right). \qquad (6)$$

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 [5], 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.

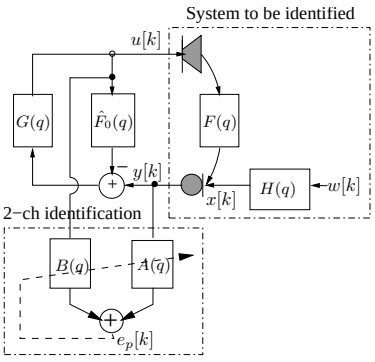


Figure 2: Two-channel adaptive feedback canceller.

3.1. *Two-channel adaptive feedback canceller [5]*

Figure 2 depicts the two-channel adaptive feedback canceller (2-ch AFC). In the 2-ch AFC, the prediction error $e_p[k]$ is formulated as

$$e_p[k] = A(q)y[k] + B(q)u[k], \qquad (7)$$

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. For RLS updating, we obtain:

$$\begin{bmatrix} \bar{\mathbf{a}}[k] \\ \mathbf{b}[k] \end{bmatrix} = \begin{bmatrix} \bar{\mathbf{a}}[k-1] \\ \mathbf{b}[k-1] \end{bmatrix} - \mathbf{R}^{-1}[k] \begin{bmatrix} \bar{\mathbf{y}}[k] \\ \mathbf{u}[k] \end{bmatrix} \epsilon_p[k], \quad (8)$$

with

$$\mathbf{u}[k] = \begin{bmatrix} u[k] & u[k-1] & \cdots & u[k-L_B+1] \end{bmatrix}^T \qquad (9)$$

$$\bar{\mathbf{y}}[k] = \begin{bmatrix} y[k-1] & y[k-2] & \cdots & y[k-L_A+1] \end{bmatrix}^T \qquad (10)$$

$$\mathbf{R}[k] = \lambda \mathbf{R}[k-1] + \begin{bmatrix} \bar{\mathbf{y}}[k] \\ \mathbf{u}[k] \end{bmatrix} \begin{bmatrix} \bar{\mathbf{y}}^T[k] & \mathbf{u}^T[k] \end{bmatrix} \qquad (11)$$

and $\epsilon_p[k]$ the a priori residual, defined as

$$\epsilon_p[k] \quad = \quad A(q, k-1)y[k] + B(q, k-1)u[k]. \qquad (12)$$

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 $F(q)$.

3.2. *PEM-based adaptive feedback canceller [5]*

The PEM-based adaptive feedback canceller (PEM-AFC) $\{A(q), \hat{F}(q)\}$, depicted in Figure 3, minimizes

$$\sum_{\bar{k}=0}^{k} \lambda^{k-\bar{k}} e_p^2[\bar{k}] \qquad (13)$$

with $e_p[k]$ defined in (6). Applying the stochastic Gauss-Newton update equation and forcing the off-diagonal blocks

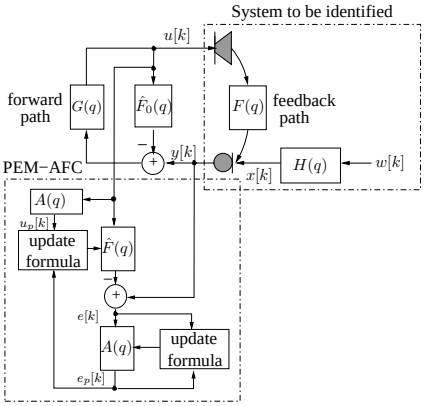


Figure 3: PEM-based adaptive feedback canceller.

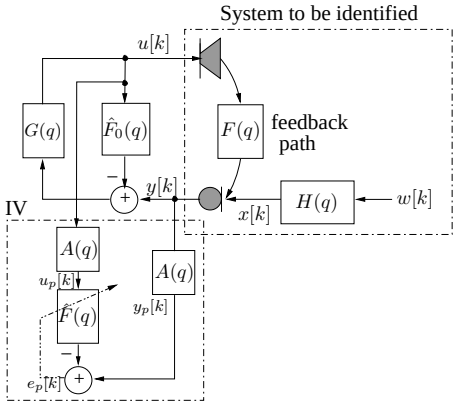


Figure 4: IV-based adaptive feedback canceller.

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:

$$\hat{\mathbf{f}}[k] = \hat{\mathbf{f}}[k-1] + \mathbf{R}_{\hat{\mathbf{f}}}^{-1}[k]\mathbf{u}_p[k]\epsilon_p[k] \quad (14)$$

$$\mathbf{R}_{\hat{\mathbf{f}}}[k] = \lambda_{\hat{\mathbf{f}}}[k]\mathbf{R}_{\hat{\mathbf{f}}}[k-1] + \mathbf{u}_p[k]\mathbf{u}_p^T[k] \quad (15)$$

with

$$\epsilon_p[k] = A(q,k-1)\left(y[k] - F(q,k-1)u[k]\right) \quad (16)$$

$$\mathbf{u}_p[k] = \left[\begin{array}{ccc} u_p[k] & \cdots & u_p[k-L_{\hat{F}}+1] \end{array}\right]^T \quad (17)$$

and $u_p[k] = A(q,k-1)u[k]$.

The update equation of $\mathbf{a}[k] = [1\ \bar{\mathbf{a}}[k]]^T$ equals

$$\bar{\mathbf{a}}[k] = \bar{\mathbf{a}}[k-1] - \mathbf{R}_{\bar{\mathbf{a}}}^{-1}[k]\frac{\partial \epsilon_p[k]}{\partial \bar{\mathbf{a}}[k-1]}\epsilon_p[k] \quad (18)$$

$$\mathbf{R}_{\bar{\mathbf{a}}}[k] = \lambda_{\bar{\mathbf{a}}}[k]\mathbf{R}_{\bar{\mathbf{a}}}[k-1] + \frac{\partial \epsilon_p[k]}{\partial \bar{\mathbf{a}}[k-1]}\left(\frac{\partial \epsilon_p[k]}{\partial \bar{\mathbf{a}}[k-1]}\right)^T \quad (19)$$

with

$$\frac{\partial \epsilon_p[k]}{\partial \bar{\mathbf{a}}[k-1]} = \left[\begin{array}{ccc} \epsilon[k-1] & \cdots & \epsilon[k-L_A+1] \end{array}\right]^T \quad (20)$$

and $\epsilon[k] = y[k] - F(q,k-1)u[k]$. To guarantee a stable speech model $A^{-1}(q)$, the AR coefficients $\bar{\mathbf{a}}[k]$ are updated with the Burg lattice algorithm [6].

In the filtered-X algorithm (14), the feedback compensated signal $e[k]$ is filtered with the estimate $A(q)$ before using it to update the adaptive feedback canceller $\hat{\mathbf{f}}[k]$. If the estimate $A(q)$ 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.

3.3. *IV-based adaptive feedback canceller (IV-AFC)[7]*

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:

$$\hat{\mathbf{f}}[k] = \hat{\mathbf{f}}[k-1] + \mathbf{R}_{\hat{\mathbf{f}}}^{-1}[k]\mathbf{u}_p[k]\tilde{\epsilon}_p[k]$$

with

$$\tilde{\epsilon}_p = y_p[k] - \hat{\mathbf{f}}^T[k-1]\mathbf{u}_p[k]. \quad (21)$$

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.

4. SIMULATION RESULTS

In this section, we compare the performance of the 2-ch AFC, the PEM-AFC and the IV-AFC with the CAF and the filtered-X method of [2]. To assess the performance, we use the misalignment $\zeta(\mathbf{f}[k], \hat{\mathbf{f}}[k])$ between the true and estimated feedback path $\hat{F}(q)$, defined as

$$\zeta(\mathbf{f}[k], \hat{\mathbf{f}}[k]) = \frac{\left\|\mathbf{f}[k] - \hat{\mathbf{f}}[k]\right\|_2^2}{\|\mathbf{f}[k]\|_2^2}. \quad (22)$$

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 sampling frequency equals 16 kHz.

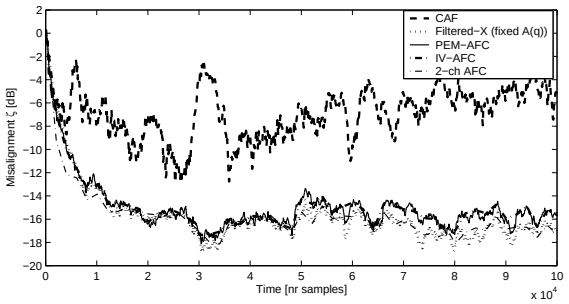


Figure 5: Misalignment ζ of the CAF, the 2-ch AFC, the filtered-X method with fixed $A(q)$, the PEM-AFC and the IV-AFC as a function of time for a stationary desired signal $x[k]$.

4.1. Stationary desired signal

Figure 5 depicts the misalignment of the CAF, the PEM-AFC and the IV-AFC as a function of time for a stationary speech-weighted noise signal $x[k]$ created by passing Gaussian white noise through a 20-th order all-pole filter $H(q)$. For comparison, the filtered-X method of [2] using the true desired signal model $H(q)$, which we consider in some sense an idealized solution, is also shown. The 2-ch AFC, the PEM-AFC and the IV-AFC clearly outperform the CAF and achieve the same performance as the idealized filtered-X method with $A(q) = H^{-1}(q)$.

4.2. Real speech signal

In a second example, $x[k]$ is a real speech signal consisting of sentences spoken by a male speaker. Figure 6 depicts the misalignment $\zeta(\mathbf{f}[k], \hat{\mathbf{f}}[k])$ of the CAF, the 2-ch AFC, the PEM-AFC, the IV-AFC and the filtered-X algorithm of [2] with an AR model $A(q)$ of the (unknown) long-term average spectrum of the speech signal as a function of time. 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]. Experiments in [7] have shown that, in contrast to the PEM-AFC, the IV-AFC does not suffer from stability problems when the adaptive feedback canceller is highly time-varying.

5. CONCLUSIONS

In this paper, we discussed 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 demonstrated 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.

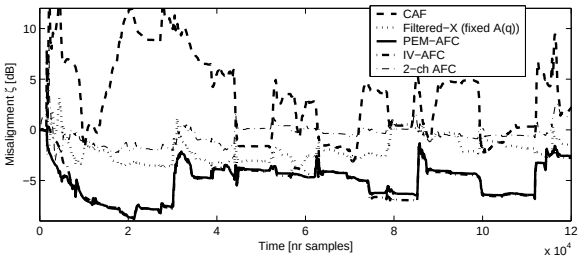


Figure 6: Misalignment ζ of the CAF, the 2-ch AFC, the filtered-X method with fixed $A(q)$, the PEM-AFC and the IV-AFC as a function of time for a real speech signal $x[k]$.

6. REFERENCES

[1] M. G. Siqueira and A. Alwan, "Steady-state analysis of continuous adaptation in acoustic feedback reduction systems for hearing-aids," *IEEE Trans. Speech, Audio Processing*, vol. 8, no. 4, pp. 443–453, July 2000.

[2] J. Hellgren and U. Forssell, "Bias of feedback cancellation algorithms in hearing aids based on direct closed loop identification," *IEEE Trans. Speech, Audio Processing*, vol. 9, no. 7, pp. 906–913, Nov. 2001.

[3] J. Hellgren, "Analysis of feedback cancellation in hearing aids with filtered-X LMS and the direct method of closed loop identification," *IEEE Trans. Speech, Audio Processing*, vol. 10, no. 2, pp. 119–131, Feb. 2002.

[4] J. R. Deller, J. G. Proakis, and J. H. L. Hansen, *Discrete-Time Processing of Speech Signals*, Macmillan Publishing Company, Englewood Cliffs, New Jersey, 1993.

[5] A. Spriet, I. Proudler, M. Moonen, and J. Wouters, "Adaptive feedback cancellation in hearing aids with linear prediction of the desired signal," Accepted for publication in *IEEE Transactions on Signal Processing*, 2004.

[6] J. Makhoul, "Stable and efficient lattice methods for linear prediction," *IEEE Trans. Acoust., Speech, and Signal Processing*, vol. 25, no. 5, pp. 423–428, Oct. 1977.

[7] A. Spriet, I. Proudler, M. Moonen, and J. Wouters, "An instrumental variable method for adaptive feedback cancellation in hearing aids." of Accepted for publication in *Proc. ICASSP 2005*, Philadelphia, PA, USA, March 2005.