ACOUSTIC FEEDBACK CANCELLATION FOR A MULTI-MICROPHONE EARPIECE BASED ON A NULL-STEERING BEAMFORMER

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ABSTRACT

In order to reduce acoustic feedback in hearing aids, adaptive filters are commonly used to estimate the feedback contribution in the microphone(s). While theoretically allowing for perfect feedback cancellation, in practice the solution is typically biased due to the closed-loop acoustical system. In this paper, we propose to use a fixed beamformer to cancel the acoustic feedback for an earpiece with multiple integrated microphones and loudspeakers. By steering a spatial null in the direction of the hearing aid loudspeaker we show that theoretically perfect feedback cancellation can be achieved. Experimental results using measured acoustic feedback paths from an earpiece with two microphones in the vent and a third microphone in the concha show that the proposed fixed beamformer provides a reduction of the acoustic feedback and substantially increases the added stable gain while maintaining a high perceptual speech quality even for unknown acoustic feedback paths, e.g., after repositioning of the earpiece or with a telephone receiver close to the ear.

***Index Terms*—** acoustic feedback cancellation, null-steering, beamforming, single-loudspeaker multiple-microphones, hearing aid

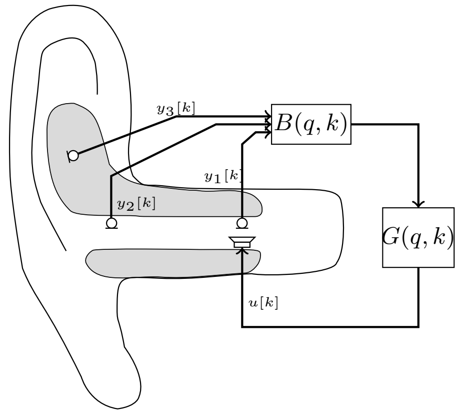

Fig. 1. Considered hearing aid setup with a single-loudspeaker three-microphone earpiece.

1. INTRODUCTION

Due to the acoustic coupling between the hearing aid loudspeaker and microphones(s), acoustic feedback is a common problem limiting the maximum applicable gain in a hearing aid. Most often acoustic feedback is perceived as whistling or howling. In order to increase the maximum gain that can be applied in a hearing aid, robust feedback cancellation strategies are required.

To cancel acoustic feedback an adaptive feedback cancellation is frequently used, where an adaptive filter models the acoustic feedback path between the hearing aid loudspeaker and the microphone(s) [1, 2, 3, 4, 5, 6]. Theoretically this allows for perfect cancellation of the acoustic feedback. However, due to the closed-loop acoustical system, the filter adaptation is usually biased, e.g., [2, 7]. Different approaches have been proposed to reduce the bias, e.g., by using the prediction-error-method [2, 8], by using probe noise [3] or by using phase modulation [9]. In addition, it has been shown that an improved performance can be achieved by using multiple microphones, e.g., by adaptively removing the contribution of the incoming signal in the filter adaptation [5] or by using a combined multi-microphone feedback cancellation and noise reduction scheme [10, 11].

In this paper we propose the use of a fixed beamformer, see e.g., [12], to cancel the contribution of the loudspeaker signal in the microphone signals. In particular we apply this approach to a newly developed earpiece [13] (see Figure 1) with two closely spaced microphones and a loudspeaker in the vent and a third microphone located in the concha. In contrast to conventional behind-the-ear hearing aids, this earpiece design allows to design a fixed beamformer with a spatial null in the direction of the hearing aid loudspeaker located in the vent. Thus the beamformer ideally cancels all signals originating from inside of the ear canal and does not impact the incoming (external) signal. Experimental results using measured acoustic feedback paths show that the proposed null-steering beamformer enables to substantially reduce the acoustic feedback, while preserving a high perceptual speech quality. Furthermore, the fixed beamformer also enables to increase the added stable gain for changing acoustic conditions, i.e., after repositioning of the earpiece and with a telephone receiver close to the ear.

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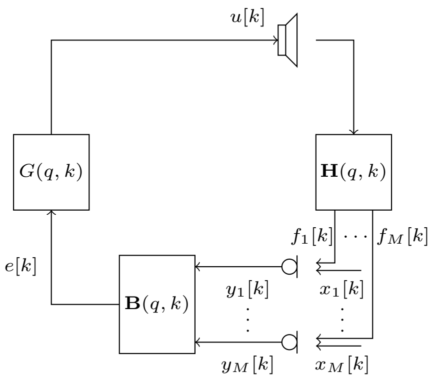

Fig. 2. Generic single-loudspeaker multi-microphone hearing aid system.

2. ACOUSTIC SCENARIO

Consider a single-loudspeaker multi-microphone hearing aid system with M microphones as depicted in Figure 2. The mth microphone signal $y_m[k]$, $m = 1, \dots, M$, at discrete time k is the sum of the incoming signal $x_m[k]$ and the loudspeaker contribution in the mth microphone $f_m[k]$, i.e.,

$$y_m[k] = x_m[k] + f_m[k] \tag{1}$$

$$= x_m[k] + \sum_{m=1}^{M} H_m(q,k)u[k], \tag{2}$$

where $u[k]$ denotes the loudspeaker signal and $H_m(q,k)$ denotes the acoustic feedback path between the mth microphone and the loudspeaker. We assume that the acoustic feedback path can be modelled as an L_H-dimensional polynomial in q, i.e.,

$$H_m(q,k) = h_{m,0}[k] + h_{m,1}[k]q^{-1} + \dots + h_{m,L_H-1}[k]q^{-L_H+1} \tag{3}$$

$$= \mathbf{h}_m^T[k]\mathbf{q}, \tag{4}$$

where $[\cdot]^T$ denotes transpose operation, $\mathbf{q}$ is the vector containing the delay-elements of q of appropriate length and $\mathbf{h}_m[k]$ denotes the impulse response of the mth acoustic feedback path, i.e.,

$$\mathbf{h}_m[k] = \begin{bmatrix} h_{m,0}[k] & \dots & h_{m,L_H-1}[k] \end{bmatrix}^T, \tag{5}$$

By stacking all microphone signals into an M-dimensional vector, (2) can be rewritten as

$$\mathbf{y}[k] = \mathbf{x}[k] + \mathbf{H}(q,k)u[k] \tag{6}$$

with

$$\mathbf{y}[k] = \begin{bmatrix} y_1[k] & \dots & y_M[k] \end{bmatrix}^T, \tag{7}$$

$$\mathbf{x}[k] = \begin{bmatrix} x_1[k] & \dots & x_M[k] \end{bmatrix}^T, \tag{8}$$

$$\mathbf{H}(q,k) = \begin{bmatrix} H_1(q,k) & \dots & H_M(q,k) \end{bmatrix}^T. \tag{9}$$

After applying a (possibly time-varying) filter-and-sum beamformer to the microphone signals the beamformer output signal $e[k]$ is obtained, i.e.,

$$e[k] = \mathbf{B}^T(q,k)\mathbf{y}[k], \tag{10}$$

where the $\mathbf{B}(q,k)$ denotes the weighting vector of the beamformer, i.e.,

$$\mathbf{B}(q,k) = \begin{bmatrix} B_1(q,k) & \dots & B_M(q,k) \end{bmatrix}^T. \tag{11}$$

The L_B-dimensional beamformer coefficient vector for the mth microphone is defined as

$$\mathbf{b}_m[k] = \begin{bmatrix} b_{m,0}[k] & \dots & b_{m,L_B-1}[k] \end{bmatrix}^T, \tag{12}$$

and the ML_B-dimensional stacked vector is defined as

$$\mathbf{b}[k] = \begin{bmatrix} \mathbf{b}_1^T[k] & \dots & \mathbf{b}_M^T[k] \end{bmatrix}^T. \tag{13}$$

The signal $e[k]$ is then processed using the hearing aid forward path $G(q,k)$, yielding in the loudspeaker signal $u[k]$, i.e.,

$$u[k] = G(q,k)e[k]. \tag{14}$$

3. SYSTEM ANALYSIS

In the following we analyse transfer function of the hearing aid system depicted in Figure 2. By combining (6),(10), and (14) we can rewrite the loudspeaker signal as

$$u[k] = G(q,k)\mathbf{B}^T(q,k)\mathbf{y}[k] \tag{15}$$

$$= G(q,k)\mathbf{B}^T(q,k)\mathbf{x}[k] + G(q,k)\mathbf{B}^T(q,k)\mathbf{H}(q,k)u[k], \tag{16}$$

such that

$$u[k] = \underbrace{\frac{G(q,k)\mathbf{B}^T(q,k)}{(1 - G(q,k)\mathbf{B}^T(q,k)\mathbf{H}(q,k))}}_{\mathbf{C}^T(q,k)}\mathbf{x}[k], \tag{17}$$

with $\mathbf{C}(q,k)$ the closed-loop transfer function. From this expression it can observed that perfect feedback cancellation can be achieved when the beamformer $\mathbf{B}(q,k)$ cancels the feedback contribution in the microphones, i.e.,

$$\mathbf{B}^T(q,k)\mathbf{H}(q,k) = 0, \tag{18}$$

with $B_m(q,k) \neq 0$ for at least one $m \in [1, \dots, M]$ to avoid the trivial solution. If (18) holds, then from (17) we obtain

$$u[k] = G(q,k)\mathbf{B}^T(q,k)\mathbf{x}[k]. \tag{19}$$

Note that although (18) perfectly solves the feedback cancellation problem, applying the beamformer coefficients will hence also modify the incoming signals $\mathbf{x}[k]$, possibly leading to sound quality degradation.

4. DESIGN OF A NULL-STEERING BEAMFORMER

In this section we design a fixed beamformer aiming to cancel the feedback contribution in a reference microphone m_0 as depicted in Figure 3. Assuming time-invariance of the acoustic feedback paths, i.e., $\mathbf{H}(q,k) = \mathbf{H}(q)$, and assuming knowledge of these acoustic feedback paths, e.g., by prior measurement, the goal is to compute the fixed beamformer coefficient vector $\mathbf{b}$ that minimizes the following least-squares (LS) optimization problem

$$\boxed{\begin{array}{ll} \min\limits_{\mathbf{b}} & \|\mathbf{H}^T\mathbf{b}\|_2^2 \\ \text{subject to} & \mathbf{b}_{m_0} = [\underbrace{\,0 \;\; \dots \;\; 0\,}_{L_d} \;\; 1 \;\; 0 \;\; \dots \;\; 0\,]^T \end{array}} \tag{20}$$

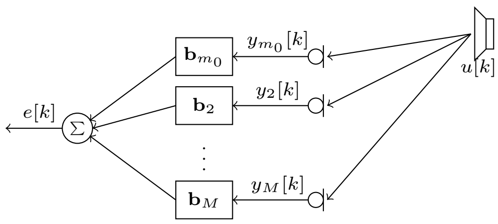

Fig. 3. Schematic of the null-steering beamformer.

where $\mathbf{b}_{m_0}$ is the beamformer coefficients in the reference microphone in order to prevent the trivial solution of $\mathbf{b} = \mathbf{0}$, L_d is a delay and $\mathbf{H}$ is the $ML_B \times (L_B + L_H - 1)$ dimensional matrix of concatenated convolution matrices of the acoustic feedback paths, i.e.,

$$\mathbf{H}^T = \begin{bmatrix} \mathbf{H}_1^T & \dots & \mathbf{H}_M^T \end{bmatrix} \tag{21}$$

where $\mathbf{H}_m$ is the $L_B \times (L_B + L_H - 1)$-dimensional convolution matrix of $\mathbf{h}_m$. The LS problem in (20) can be reformulated as

$$\min_{\mathbf{b}} \|\mathbf{h}_{m_0} + \sum_{\substack{m=1 \\ m \neq m_0}}^{M} \mathbf{H}_m^T \mathbf{b}_m\|_2^2 \tag{22}$$

with the closed-form solution

$$\mathbf{b}_{LS} = (\tilde{\mathbf{H}}\tilde{\mathbf{H}}^T)^{-1}\tilde{\mathbf{H}}\mathbf{h}_{m_0}, \tag{23}$$

where $\tilde{\mathbf{H}}$ is the matrix of concatenated convolution matrices $\mathbf{H}_m$, $m = 1, \dots, M, m \neq m_0$. Note that the solution in (23) only holds for $L_B < \frac{L_H - 1}{M - 1}$ [12].

5. EXPERIMENTAL EVALUATION

In this section the performance of the proposed null-steering beamformer is evaluated when using 2 or 3 microphones. In particular we consider the ability to cancel the acoustic feedback in different acoustic scenarios as well as the distortion of the incoming signal.

5.1. Setup and Performance Measures

Acoustic feedback paths were measured for the three-microphone earpiece as depicted in Figure 1 on a dummy head with adjustable ear canals [14]. The impulse responses were sampled at $f_s = 16\,\text{kHz}$ and truncated to length $L_H = 100$. Measurements were performed in an acoustically treated chamber and the distance between the external source and the dummy head was 1.2 m. Figure 4 shows the amplitude response of the measured acoustic feedback paths for the three different microphones and for different acoustic conditions. Note that due to the location of the loudspeaker and microphones inside the earpiece, these are no pure delays. The forward path of the hearing aid was of the set to $G(q, k) = q^{-96}10^{45/20}$, corresponding to a delay of 6 ms and a broadband amplification of 45 dB. For all experiments the reference microphone $m_0 = 2$, i.e., the microphone located at the outer phase of the vent, was chosen and $L_d = L_B/2$ was used.

We evaluated the feedback cancellation performance of the null-steering beamformer using the added stable gain (ASG) [8] and the perceptual quality of the signal after applying the null-steering beamformer using the perceptual quality of speech (PESQ) measure [15].

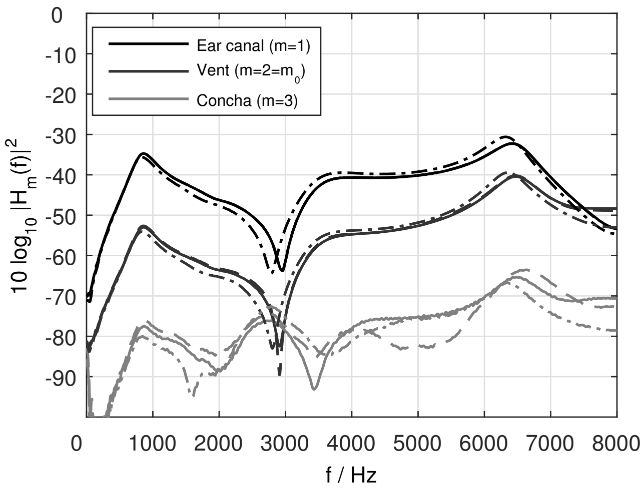

Fig. 4. Amplitude response of the measured acoustic feedback paths. Continuous lines show feedback paths in free-field, i.e., without any obstruction (used for computing the beamformer coefficients), dashed dotted lines show an exemplary responses after repositioning of the earpiece, and dashed lines show the acoustic feedback paths in the presence of a telephone receiver.

The ASG for the considered hearing aid setup is computed as [8]

$$ASG = 20\log_{10}\frac{1}{\max_\Omega |\sum_{m=1}^{M} H_m(e^{j\Omega})B_m e^{j\Omega})|} - MSG_{m_0}, \tag{24}$$

where MSG_{m_0} is the maximum stable gain in the reference microphone without applying the beamformer, i.e.,

$$MSG_{m_0} = 20\log_{10}\frac{1}{\max_\Omega |H_{m_0}(e^{j\Omega})|}. \tag{25}$$

The reference signal for the PESQ measure was the incoming signal $x_{m_0}[k]$ in the reference microphone, while the test signal was the error signal $e[k]$ after applying the beamformer. As speech source we concatenated 26 sentences spoken by 4 different speakers from the TIMIT database [16] resulting in an 80 s long signal.

5.2. Experiment 1: Optimal performance

Based on the measured acoustic feedback paths in free-field, i.e., without obstruction, the beamformer coefficient vector $\mathbf{b}$ was computed using (23). Figure 5 shows the results for the ASG for different numbers of microphones ($M = 2, M = 3$) and filter length L_B. As expected, by increasing the number of microphones from $M = 2$ to $M = 3$ the performance can be increased by approximately 25 dB. Table 1 depicts the obtained PESQ scores. Results show that for the condition in free-field a high perceptual speech quality is maintained for the incoming speech source with PESQ scores larger than 4.33.

5.3. Experiment 2: Influence of External Sound Field Variations

In the second experiment, we investigate the influence of an external sound field variation on the performance, when using the beamformer coefficients optimized for the free-field condition (see Experiment 1). More in particular we consider placing a telephone receiver

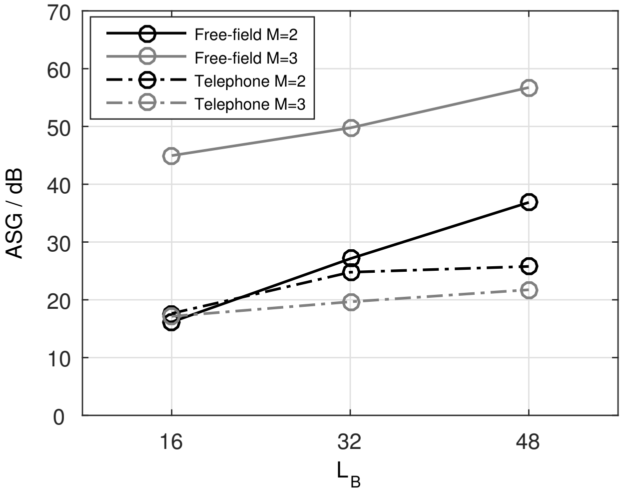

Fig. 5. ASG for different numbers of microphones and external sound fields as a function of the beamformer filter length L_B.

Table 1. PESQ score for different numbers of microphones and external sound fields as a function of the beamformer filter length L_B.

		$L_B = 16$	$L_B = 32$	$L_B = 48$
$M = 2$	Free-field	4.44	4.43	4.40
	Telephone	4.42	4.42	4.39
$M = 3$	Free-field	4.34	4.35	4.33
	Telephone	4.35	4.37	4.35

close to the ear of the dummy head. As can be observed from Figure 5, as expected the ASG is in general reduced. While for $M = 2$ the ASG is reduced by about 10 dB for $L_B = 48$, for $M = 3$ the ASG is drastically reduced by more than 30 dB, nevertheless, resulting in ASGs of about 20 dB. These results show that for $M = 2$ microphones the null-steering beamformer is rather robust to drastic changes in the sound field while for using $M = 3$ the beamformer is not so robust to these changes. The PESQ scores shown in Table 1 show that while the ASG can be drastically reduced the perceptual quality of the incoming speech source is not changed.

5.4. Experiment 3: Influence of Hearing Aid Repositioning

While placing a telephone receiver close to the ear leads to large changes of the acoustic feedback paths, repositioning of the earpiece may introduce small changes in the acoustic feedback paths [17]. Hence, in the third experiment we investigate the influence of refitting the earpiece to the dummy head. Under free-field conditions the earpiece was removed and repositioned 10 times. The null-steering beamformer was computed for the first position (see Experiment 1) of the earpiece and kept fix for the remaining 9 positions.

Figure 6 shows the ASG for different number of microphones as a function of the beamformer filter length. The solid lines show the results for the position for which the beamformer was optimized, while dashed lines show average results for all other positions. Errorbars denote minimum and maximum ASG. The results show that the performance is generally reduced after respositioning of the earpiece. This reduction is largest when using $M = 3$ microphones, where, e.g., for $L_B = 48$, the difference in performance is 35 dB. In contrast for $M = 2$ the proposed approach is more robust to repositioning of the earpiece across different values of L_B. In conclusion, even for different positions of the earpiece an ASG of approximately 20 dB can be obtained. Table 2 shows the PESQ scores, where in all cases a high perceptual quality is achieved as indicated by PESQ scores greater than 4.31.

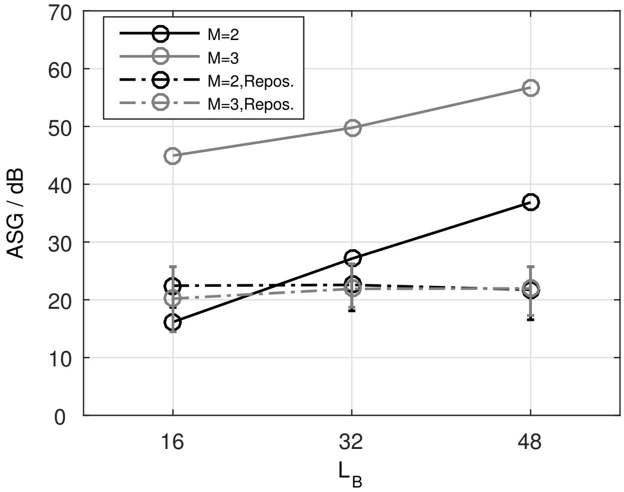

Fig. 6. ASG as a function of beamformer filter length L_B for different numbers of microphones for the optimized position and average ASG after repositioning the earpiece in the ear (Repos.), where errorbars show minimum and maximum ASG.

Table 2. PESQ scores as a function of beamformer filter length L_B for different numbers of microphones for the optimized position and average as well as minimum and maximum PESQ scores after repositioning the earpiece in the ear (Repos.).

			$L_B = 16$	$L_B = 32$	$L_B = 48$
$M = 2$	Free-field		4.44	4.43	4.40
	Repos.	mean	4.44	4.44	4.42
		min	4.44	4.44	4.40
		max	4.44	4.45	4.43
$M = 3$	Free-field		4.34	4.35	4.33
	Repos.	mean	4.34	4.33	4.33
		min	4.32	4.32	4.31
		max	4.36	4.36	4.36

6. CONCLUSION

In this paper we proposed a fixed beamformer method to perform acoustic feedback cancellation in an earpiece with multiple integrated microphones by steering a spatial null in the direction of the hearing aid loudspeaker. We theoretically showed that perfect feedback cancellation can be achieved when using a null-steering beamformer. We formulated the estimation of the beamformer coefficients as a least-squares optimization problem, where we constrain the beamformer coefficients in a reference microphone to a delay. Experimental results using measured acoustic feedback paths show that the proposed approach leads to a large ASG, while maintaining a good perceptual quality, even when placing a telephone close to the ear or repositioning the earpiece. In conclusion, by using the proposed fixed beamformer method the ASG is robustly improved by more than 20 dB while not compromising the speech quality.

7. REFERENCES

[1] T. van Waterschoot and M. Moonen, "Fifty Years of Acoustic Feedback Control: State of the Art and Future Challenges," *Proc. IEEE*, vol. 99, no. 2, pp. 288-327, Feb. 2011.

[2] A. Spriet, S. Doclo, M. Moonen, and J. Wouters, "Feedback Control in Hearing Aids," in *Springer Handbook of Speech Processing*, pp. 979-999. Springer-Verlag, Berlin, Germany, 2008.

[3] M. Guo, S. H. Jensen, and J. Jensen, "Novel Acoustic Feedback Cancellation Approaches in Hearing Aid Applications Using Probe Noise and Probe Noise Enhancement," *IEEE Trans. Audio, Speech, Language Process.*, vol. 20, no. 9, pp. 2549–2563, Nov. 2012.

[4] F. Strasser and H. Puder, "Sub-band feedback cancellation with variable step sizes for music signals in hearing aids," in *Proc. Int. Conf. Acoust. Speech Signal Process. (ICASSP)*, Florence, Italy, May 2014, pp. 8207–8211.

[5] C. R. C. Nakagawa, S. Nordholm, and W.-Y. Yan, "Analysis of Two Microphone Method for Feedback Cancellation," *IEEE Signal Process. Lett.*, vol. 22, no. 1, pp. 35–39, Jan. 2015.

[6] H. Schepker, L. T. T. Tran, S. Nordholm, S. Doclo, "Improving adaptive feedback cancellation in hearing aids using an affine combination of filters," in *Proc. Int. Conf. Acoust. Speech Signal Process. (ICASSP)*, Shanghai, China, Mar. 2016, pp. 231–235.

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

[8] A. Spriet, I. Proudler, M. Moonen, and J. Wouters, "Adaptive feedback cancellation in hearing aids with linear prediction of the desired signal," *IEEE Trans. Signal Process.*, vol. 53, no. 10, pp. 3749–3763, Oct. 2005.

[9] M. Guo, S. H. Jensen, J. Jensen, and S. L. Grant, "On the use of a phase modulation method for decorrelation in acoustic feedback cancellation," in *Proc. Europ. Signal Process. Conf. (EUSIPCO)*, Bucharest, Romania, Aug. 2012, pp. 2000–2004.

[10] A. Spriet, G. Rombouts, M. Moonen, and J. Wouters, "Combined feedback and noise suppression in hearing aids," *IEEE Trans. Audio, Speech, Language Process.*, vol. 15, no. 6, pp. 1777–1790, Aug. 2007.

[11] G. Rombouts, A. Spriet, and M. Moonen, "Generalized sidelobe canceller based combined acoustic feedback- and noise cancellation," *Signal Process.*, vol. 88, no. 3, pp. 571–581, Mar. 2008.

[12] J. Benesty, J. Chen, and Y. Huang, *Microphone Array Signal Processing*. Berlin, Germany: Springer-Verlag, 2008.

[13] F. Denk, M. Hiipakka, B. Kollmeier, and S. M. A. Ernst, "An individualized acoustically transparent earpiece for hearing devices," submitted to Trends in Hearing.

[14] M. Hiipakka, M. Tikander, and M. Karjalainen, "Modeling the External Ear Acoustics for Insert Headphone Usage," *J. Audio Eng. Soc.*, vol. 58, no. 4, pp. 269–281, Apr. 2010.

[15] ITU-T, Perceptual evaluation of speech quality (PESQ), An objective method for end-to-end speech quality assessment of narrowband tele-phone networks and speech codecs P.862 Int. Telecomm. Union (ITU-T) Rec., 2001.

[16] J. S. Garofolo, "Getting started with the darpa timit cd-rom: An acoustic phonetic con- tinuous speech database," Nat. Inst. Standards Technol. (NIST), Gaithersburg, MD, Dec. 1988.

[17] T. Sankowsky-Rothe, M. Blau, H. Schepker, S. Doclo, "Reciprocal measurement of acoustic feedback paths in hearing aids," *J. Acoust. Soc. Am.*, vol. 138, no. 4, pp. EL399–EL404, Oct. 2015.