SVD-Based Optimal Filtering for Noise Reduction in Dual Microphone Hearing Aids: A Real Time Implementation and Perceptual Evaluation

Jean-Baptiste Maj*, Liesbeth Royackers, Marc Moonen, *Member, IEEE*, and Jan Wouters

***Abstract*—In this paper, the first real-time implementation and perceptual evaluation of a singular value decomposition (SVD)-based optimal filtering technique for noise reduction in a dual microphone behind-the-ear (BTE) hearing aid is presented. This evaluation was carried out for a speech weighted noise and multitalker babble, for single and multiple jammer sound source scenarios. Two basic microphone configurations in the hearing aid were used. The SVD-based optimal filtering technique was compared against an adaptive beamformer, which is known to give significant improvements in speech intelligibility in noisy environment. The optimal filtering technique works without assumptions about a speaker position, unlike the two-stage adaptive beamformer. However this strategy needs a robust voice activity detector (VAD). A method to improve the performance of the VAD was presented and evaluated physically. By connecting the VAD to the output of the noise reduction algorithms, a good discrimination between the speech-and-noise periods and the noise-only periods of the signals was obtained. The perceptual experiments demonstrated that the SVD-based optimal filtering technique could perform as well as the adaptive beamformer in a single noise source scenario, i.e., the ideal scenario for the latter technique, and could outperform the adaptive beamformer in multiple noise source scenarios.**

***Index Terms*—Adaptive beamformer, hearing aids, noise reduction algorithms, optimal filtering technique, singular value decomposition, voice activity detector.**

I. Introduction

HEARING-IMPAIRED listeners have great difficulty to understand speech in a noisy background. They require a signal-to-noise ratio (SNR) of about 5–10 dB higher than normal hearing listeners to achieve the same level of speech understanding [1]. To compensate for this difference, noise reduction algorithms are being developed and implemented in hearing aids.

Based on a single microphone signal the separation of desired and interfering sounds is impossible when the spectral and temporal characteristics are similar [2], [3]. By using multiple microphones spatial information can be used. The latter approach has already been applied for some time in directional microphones by combining the outputs of two omnidirectional microphones [4], [5]. More recently, adaptive signal processing has been added to two microphone configurations, e.g., adaptive directional microphones in commercial hearing aids [6]–[8] or adaptive beamformers in the research area [9]–[11]. In principle, this allows the spatially-selective suppression of jammer sources, even with changing positions.

As a possible adaptive beamformer approach, an extension of the generalized sidelobe canceller (GSC) [12] was developed for dual microphone BTE hearing aids. This extension of the GSC was called a *two-stage adaptive beamformer* and an improvement of speech understanding in noise of 5 dB was obtained, relative to the effect of one hardware directional microphone [11]. Other evaluations of this technique were carried out with normal hearing listeners and hearing impaired listeners [13], [14] and with cochlear implant patients [15]. The two-stage adaptive beamformer was also implemented in a commercial hearing aid to make a comparison with an adaptive directional microphone [7]. Also, in these studies, significant improvements of the speech intelligibility in noise were obtained.

Recently, an *optimal filtering* technique for multimicrophone noise reduction was presented [16]. The optimal filtering technique works without assumptions about a position of the speaker, unlike the two-stage adaptive beamformer, and needs a robust voice activity detector (VAD) to discriminate the *speech-and-noise periods* from the *noise-only periods* in noisy speech signals. In the present study, the computation of the optimal filter is based on a singular value decomposition (SVD). Theoretical and physical evaluations for dual microphone BTE hearing aid applications have already been carried out [17], [18].

In this paper, the first *perceptual evaluation* of the SVD-based optimal filtering technique is presented. The strategy was implemented in real time on a PC-platform, to which the microphone signals of BTE hearing aids were connected. For the real time implementation, a method to improve the performance of the VAD was presented and evaluated physically. In speech-in-noise enhancement strategies for hearing aids, several signals are available to the VAD, such as the signal of an omnidirectional microphone, a directional microphone and even the output of the noise reduction technique. The behavior of the VAD was evaluated when the VAD was connected to these different signals. The assessment was performed by

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measuring the percentage of well-detected speech-and-noise periods and noise periods in a noisy signal. This evaluation was performed off line, with signals recorded under the same conditions as for the perceptual evaluation. For the perceptual assessment, the optimal filtering technique was compared to the two-stage adaptive beamformer for two types of microphone configurations in hearing aids. A first had one hardware directional microphone and one omnidirectional microphone, and a second had two omnidirectional microphones allowing the implementation of a so-called software directional microphone. The perceptual evaluations were performed in a stationary speech weighted noise and in multitalker babble for a single noise source and a multiple jammer sound source scenario, all in a moderately reverberant room. The perceptual experiments consisted in measuring the speech reception threshold (SRT: defined as the sound-pressure level of speech at which 50% of the speech is correctly understood by the listener) as obtained with the noise reduction strategies. The SRT-improvements of the noise reduction technique were relative to the case no noise reduction is applied, i.e., where the omnidirectional microphone signal is used.

In this paper, the microphone configuration in the hearing aids and the two noise reduction strategies are reviewed. A real time implementation of the SVD-based optimal filtering technique is described. Then, the experimental setup for the physical and the perceptual evaluations are examined. Finally, the results of the physical and the perceptual experiments are discussed.

II. Methods

A. Hearing Aids

To evaluate the performance of the noise reduction algorithms, two types of BTE hearing aids were used and referred to as *DO* and *OO* (Fig. 1). They had two different microphone configurations and were prototypes based on the Nucleus-Esprit behind-the-ear headset housing of Cochlear Ltd. The first type *DO* had one hardware directional microphone (Microtronic 6001), as front microphone, and one omnidirectional microphone (Knowles FG-3452), as rear microphone. The hardware directional microphone had a cardioid spatial characteristic (null at 180°) in anechoic conditions. The distance between the front entry port and the back entry port of the hardware directional microphone was 1 cm. The distance between the front entry port of the hardware directional microphone and the omnidirectional microphone was 2.5 cm. The second type *OO* had two omnidirectional microphones (Knowles FG-3452) mounted in an endfire configuration, and spaced 1 cm apart. A so-called software directional microphone was implemented, where a directional microphone is computed as the difference between the signal from the front microphone and the delayed-weighted signal of the rear microphone [5]. The microphone parameters were the interport distance d, internal delay τ and the weight factor for the back port was $w(f) = a \cdot e^{-j2\pi f\tau}$. The delay τ and the weight a were chosen to give a hypercardioid spatial characteristic in anechoic conditions for a speech weighted noise (null at 110°). The software directional microphone and the rear omnidirectional microphone signal were then used as the inputs to the noise reduction algorithms.

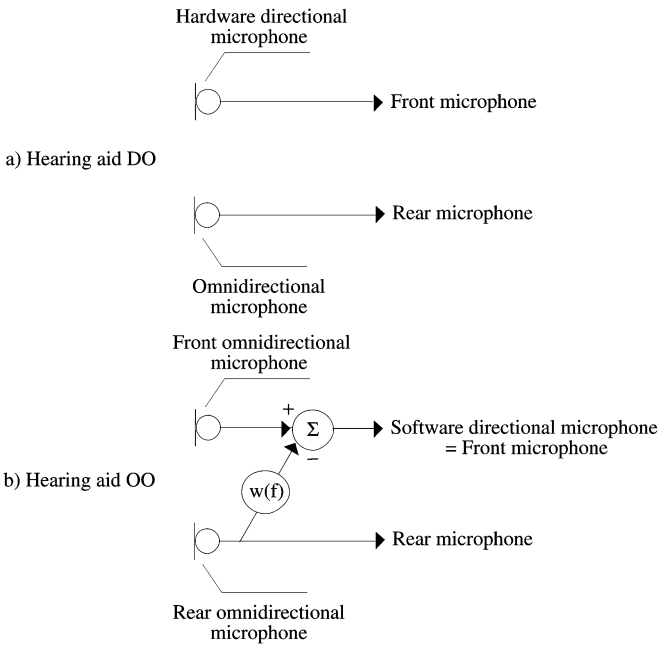


Fig. 1. Representation of the microphone configuration of hearing aids *DO* and *OO*.

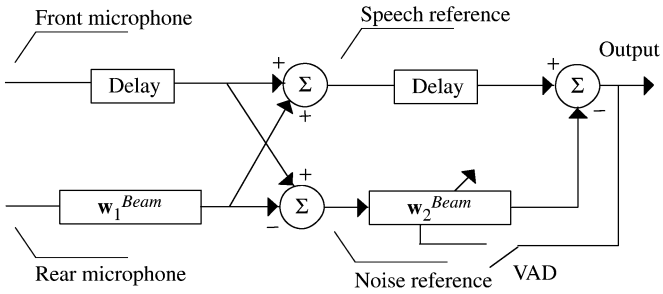


Fig. 2. Representation of the two stage adaptive beamformer.

B. SVD-Based Optimal Filtering Technique

The SVD-based optimal filtering technique has already been evaluated theoretically and physically in previous studies [16], [17], [18], [19]. In the latter works, the optimal filtering technique was introduced, and theoretical and physical evaluations were performed. These evaluations were carried out in unrealistic conditions, unlike in the present paper. Indeed, the reverberant speech signal was created by using speech material recorded in anechoic conditions and filtered by a simulated acoustic room impulse response. To create the noisy speech signal, an uncorrelated white noise signal was added to the speech signal. Furthermore, a perfect VAD was considered and the GSVD was computed with a batch method [16].

This paper presents, for the first time, the complete strategy of the real time implementation of the optimal filtering technique and a perceptual evaluation performed in a moderately reverberant condition. The optimal filtering strategy is also compared to a two-stage adaptive beamformer. The scheme of the two-stage adaptive beamformer is presented in Fig. 2 [11]. This algorithm is an extension of the GSC [12] and was already implemented in real time and evaluated with normal hearing listeners and hearing impaired subjects [13], [14] and cochlear implant users [15].

1) Method: The SVD-based optimal filtering technique considered here, in general reconstructs the speech signal $\mathbf{s}_k$ from noisy data $\mathbf{u}_k = \mathbf{s}_k + \mathbf{n}_k$ by means of a linear filter $\mathbf{W} \in \mathbb{R}^{N\times N}$, i.e., $\hat{\mathbf{s}}_k = \mathbf{W}^T\mathbf{u}_k$ at time k. The $\hat{\mathbf{s}}_k$ is then an estimate for the speech constituent $\mathbf{s}_k$ in $\mathbf{u}_k$. Using a minimum mean square error-criterion (MMSE), the optimal filter $\mathbf{W}_{WF}$ is equal to

$$\mathbf{W}_{WF} = \mathcal{E}\left\{\mathbf{u}_k \cdot \mathbf{u}_k^T\right\}^{-1} \cdot \left(\mathcal{E}\left\{\mathbf{u}_k \cdot \mathbf{u}_k^T\right\} - \mathcal{E}\left\{\mathbf{n}_k \cdot \mathbf{n}_k^T\right\}\right). \tag{1}$$

Doclo [16] used an interesting and useful simplification in (1) where $\mathbf{W}_{WF}$ is derived from the *generalized singular value decomposition* (GSVD) of the data matrices $\mathbf{U}_k \in \mathbb{R}^{p\times N}$ and $\mathbf{N}_k \in \mathbb{R}^{q\times N}$ (with p and q typically larger than N), such that $\mathcal{E}\left\{\mathbf{u}_k \cdot \mathbf{u}_k^T\right\} \Rightarrow \left(\mathbf{U}_k^T \cdot \mathbf{U}_k\right)/p$ and $\mathcal{E}\left\{\mathbf{n}_{k'} \cdot \mathbf{n}_{k'}^T\right\} \Rightarrow \left(\mathbf{N}_k^T \cdot \mathbf{N}_k\right)/q$. $\mathbf{u}_k$ is collected during speech-and-noise periods while $\mathbf{n}_k$ is collected during noise-only periods. In a two microphone application, the vector $\mathbf{u}_k \in \mathbb{R}^L$ took the form

$$\mathbf{u}_k = \left[\mathbf{u}_{1k}^T \quad \mathbf{u}_{2k}^T\right]^T \tag{2}$$

with

$$\mathbf{u}_{jk} = \left[u_j(k) \quad u_j(k-1) \quad \ldots \quad u_j(k-N+1)\right]^T \tag{3}$$

where the j refers to the j-th microphone and then $L = 2N$. The vector $\mathbf{n}_k$ is similarly defined. And the matrices $\mathbf{U}_k$ and $\mathbf{N}_k$ are given as

$$\mathbf{U}_k = \begin{bmatrix} \mathbf{u}_k^T \\ \mathbf{u}_{k+1}^T \\ \vdots \\ \mathbf{u}_{k+p-1}^T \end{bmatrix} \quad \mathbf{N}_k = \begin{bmatrix} \mathbf{n}_k^T \\ \mathbf{n}_{k+1}^T \\ \vdots \\ \mathbf{n}_{k+p-1}^T \end{bmatrix}. \tag{4}$$

The data in $\mathbf{u}_k$ $(\mathbf{n}_k)$ are collected during speech-and-noise (noise-only) periods. The generalized singular value decomposition of the matrices $\mathbf{U}_k$ and $\mathbf{N}_k$ is defined as

$$\begin{cases} \mathbf{U}_k = \mathbf{Y}\ \mathrm{diag}\{\sigma_i\}\mathbf{X}^T \\ \mathbf{N}_k = \mathbf{V}\ \mathrm{diag}\{\eta_i\}\mathbf{X}^T \end{cases} \tag{5}$$

where $\mathbf{Y} \in \mathbb{R}^{p\times L}$ and $\mathbf{V} \in \mathbb{R}^{q\times L}$ are orthogonal matrices, $\mathbf{X} \in \mathbb{R}^{L\times L}$ is an invertible matrix, and, σ_i and η_i are the generalized singular values. By substituting the above formulas in (1), an alternative expression for $\mathbf{W}_{WF}$ is obtained

$$\mathbf{W}_{WF} = \mathbf{X}^{-T}\mathrm{diag}\left\{\frac{q\sigma_i^2 - p\eta_i^2}{q\sigma_i^2}\right\}\mathbf{X}^T. \tag{6}$$

Ephraim and Trees [20] introduced a time constraint where the energy of the signal distortion ϵ_s^2 is minimized under the constraint that the residual noise energy ϵ_n^2 stays under a threshold α.

$$\min_{\mathbf{W}} \epsilon_s^2 \text{ subject to } \epsilon_n^2 \leq \alpha \text{ where } 0 \leq \alpha \leq 1. \tag{7}$$

Thus, the filter $\mathbf{W}_{WF}$ becomes

$$\mathbf{W}_{WF} = \mathbf{X}^{-T}\ \mathrm{diag}\left\{\frac{q\sigma_i^2 - p\eta_i^2}{q\sigma_i^2 + (\mu-1)p\eta_i^2}\right\}\mathbf{X}^T \tag{8}$$

where the speech distortion parameter $\mu \in [0, \infty[$ is a tradeoff between signal distortion and noise reduction. If $\mu = 1$ the original MMSE solution is obtained. More emphasis is put on the signal distortion when $\mu < 1$ at the expense of decreasing the noise reduction performance. The residual noise level is reduced when $\mu > 1$ at the expense of increasing the speech distortion. With $\mu \to \infty$, all the emphasis was put on the noise reduction without taking into account of the signal distortion.

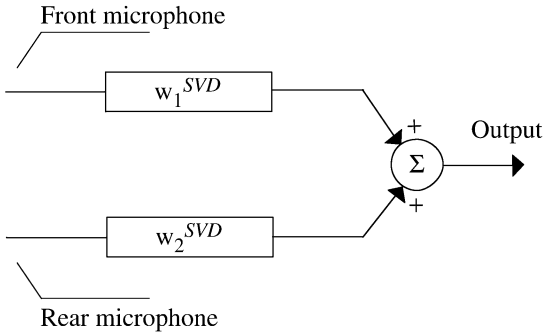


Fig. 3. Representation of the SVD-based optimal filtering technique.

The computation of the optimal filter $\mathbf{W}_{WF}$ provides L estimators $\mathbf{w}_{WF}$, where $\mathbf{w}_{WF}$ is a column of $\mathbf{W}_{WF}$, for an optimal filtered estimate $\hat{s}_k$ at time k. Maj *et al.* [21] showed that, for a BTE configuration with a directional microphone and an omnidirectional microphone, using the middle column of $\mathbf{W}_{WF}$ in the directional microphone part, a good speech estimate is obtained. This filter $\mathbf{w}_{WF}$ (see Fig. 3) is as a two-channel filter, where each microphone was filtered with a N-taps filter $\mathbf{w}_j^{\mathrm{SVD}}$

$$\mathbf{w}_{WF} = \begin{bmatrix} \mathbf{w}_1^{\mathrm{SVD}} \\ \mathbf{w}_2^{\mathrm{SVD}} \end{bmatrix}. \tag{9}$$

The number of taps used in $\mathbf{w}_1^{\mathrm{SVD}}$ and $\mathbf{w}_2^{\mathrm{SVD}}$ was 15.

2) Real Time Implementation: The real time implementation of the SVD-based technique is illustrated in Fig. 4. Five steps are necessary to compute the filter coefficients in real time.

- *Step 1)* The VAD decides if the new input sample vector is a speech-and-noise period or a noise-only period. The VAD used in this study is an extension of the VAD developed by Van Gerven and Xie [22] which is based on the log-energy of the signal. The VAD induces only a delay of 2.7 ms and did not affect lip-reading performance [23]. The sensitivity of the VAD was fitted with the signal of the software directional microphone for having a good discrimination between the speech-and-noise periods and the noise-only periods at a SNR of 5 dB in stationary and non stationary conditions [7], [8], [14].
- *Step 2)* Classification errors between the speech-and-noise periods and the noise-only periods occur with the VAD. If the speech-and-noise periods are wrongly classified, speech-and-noise vectors are added to the noise matrix $(\mathbf{N}_k)$. In this case, the factor $F = \sum_{i=1}^{L} 1 - \eta_i^2/\sigma_i^2$ of the filter $\mathbf{W}_{WF}$ tends to be small $\left(\sigma_i^2 \to \eta_i^2\right)$, resulting in signal cancellation at the output of the optimal filtering technique. Since F varies in time, the derivative D_F of this factor can be measured during the processing

$$D_F = \frac{\delta\left(\frac{1}{L}\cdot\sum_{i=1}^{L}\left(\frac{1-\eta_i^2}{\sigma_i^2}\right)\right)}{\delta t}. \tag{10}$$

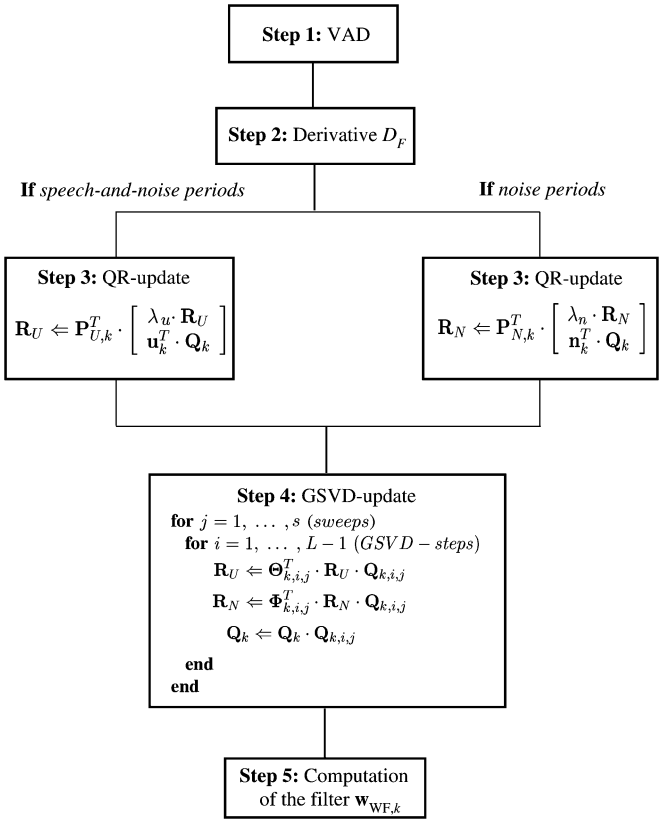


Fig. 4. Real time implementation of the SVD-based optimal filtering technique.

If the derivative D_F is below a given (negative) threshold T_D, this means that the VAD misclassified speech-and-noise periods. Then, a correction is brought to the VAD and the decision made in *Step 1)* is modified. Otherwise, when $D_F > T_D$, the decision made in *Step 1)* is kept valid.

The function D_F is effective ($D_F < T_D$) with VAD errors, but can also be effective with temporal variation of a non stationary noise. For instance, in a multitalker babble (perfect VAD is assumed) the level of the generalized singular value η_i varies while the level of the generalized singular value σ_i stays roughly the same. When the generalized singular values η_i increase, the factor F becomes smaller resulting in the decrease of the derivative D_F. If the latter function becomes effective ($D_F < T_D$), the noise signal is detected as a speech-and-noise signal and the optimal filtering technique does not perform an optimal cancellation of the noise source. In this study, the threshold T_D was chosen low enough to avoid that the function D_F was effective ($D_F < T_D$) with a perfect VAD in a stationary and a nonstationary (e.g., multitalker babble) noise at a SNR of -10 dB. The threshold T_D was also chosen high enough that no temporal distortions were observed when the VAD could only detect correctly 50% of the speech-and-noise periods.

- *Step 3) and Step 4)* A recursive technique is used to approximate the SVD-based optimal filtering technique. Since the updating scheme of the SVD-based technique has already been published, a brief overview is presented in this paper. For details on the updating scheme, see [24] and [25].

  This technique is based on a Jacobi-type GSVD-updating algorithm [24]. Recursive GSVD-updating algorithms use the decomposition of the GSVD at time $k-1$ to compute the decomposition at time k. The (5) at time $k-1$ can be rewritten as

$$\begin{cases} \mathbf{U}_{k-1} = \mathbf{Y}_{k-1} \cdot \mathbf{R}_{U,k-1} \cdot \mathbf{Q}_{k-1}^T \\ \mathbf{N}_{k-1} = \mathbf{V}_{k-1} \cdot \mathbf{R}_{N,k-1} \cdot \mathbf{Q}_{k-1}^T. \end{cases} \tag{11}$$

  where $\mathbf{R}_{U,k-1} \in \mathbb{R}^{L \times L}$ and $\mathbf{R}_{N,k-1} \in \mathbb{R}^{L \times L}$ are upper triangular matrices having parallel rows and $\mathbf{Q}_{k-1} \in \mathbb{R}^{L \times L}$ is an orthogonal matrix. For the computation, only $\mathbf{R}_{U,k-1}$, $\mathbf{R}_{N,k-1}$ and $\mathbf{Q}_{k-1}$ are stored. When a new data vector $\mathbf{u}_k$ (speech-and-noise) or $\mathbf{n}_k$ (noise) is present at time k, the GSVD of $\mathbf{U}_k$ and $\mathbf{N}_k$ need to be recomputed as

$$\mathbf{U}_k = \begin{bmatrix} \lambda_u \cdot \mathbf{U}_{k-1} \\ \mathbf{u}_k \end{bmatrix} \text{ or } \mathbf{N}_k = \begin{bmatrix} \lambda_n \cdot \mathbf{N}_{k-1} \\ \mathbf{n}_k \end{bmatrix} \tag{12}$$

  where λ_u and λ_n are exponential weighting factors for speech and noise matrix, respectively.
- *Step 5)* This step consists of computing the optimal filter $\mathbf{w}_{WF,k}$ after the update of the recursive GSVD-updating algorithm. Substituting (12) into (1), the equation can be rewritten as

$$\mathbf{W}_{WF,k} = \mathbf{Q}_k . \mathbf{R}_{U,k}^{-1} \cdot \mathrm{diag}\left\{ \frac{\left(1-\lambda_n^2\right)\left(\mathbf{R}_{U,k}^{ii}\right)^2 - \left(1-\lambda_u^2\right)\cdot\left(\mathbf{R}_{N,k}^{ii}\right)^2}{\left(1-\lambda_n^2\right)\left(\mathbf{R}_{U,k}^{ii}\right)^2 + (\mu-1)\left(1-\lambda_u^2\right)\cdot\left(\mathbf{R}_{N,k}^{ii}\right)^2} \right\} \cdot \mathbf{R}_{U,k}\mathbf{Q}_k^T. \tag{13}$$

  The factor p/q is replaced by $\left(1-\lambda_n^2\right) / \left(1-\lambda_u^2\right)$. Since only one column (the i-th column, $\mathbf{w}_{WF,k}^i$ of $\mathbf{W}_{WF,k}$) needs to be computed, this column can be computed as the solution of the linear set of equations, shown in (14) at the bottom of the page. The calculation of the $\mathbf{w}_{WF,k}^i$ consists by computing $\tilde{\mathbf{x}}_k$, solving the equation $\mathbf{R}_{U,k} \cdot \tilde{\mathbf{w}}_k = \tilde{\mathbf{x}}_k$ by back-substitution and computing $\mathbf{w}_{WF,k}^i$ as

$$\mathbf{w}_{WF,k}^i = \mathbf{Q}_k \cdot \tilde{\mathbf{w}}_k. \tag{15}$$

$$\mathbf{R}_{U,k} \underbrace{\mathbf{Q}_k^T \mathbf{w}_{WF,k}^i}_{\tilde{\mathbf{w}}_k} = \underbrace{\mathrm{diag}\left\{ \frac{\left(1-\lambda_n^2\right)\left(\mathbf{R}_{U,k}^{i,i}\right)^2 - \left(1-\lambda_u^2\right)\left(\mathbf{R}_{N,k}^{i,i}\right)^2}{\left(1-\lambda_n^2\right)\left(\mathbf{R}_{U,k}^{i,i}\right)^2 + (\mu-1)\left(1-\lambda_u^2\right)\left(\mathbf{R}_{N,k}^{i,i}\right)^2} \right\} \mathbf{R}_{U,k}\mathbf{Q}_k^{i\,T}}_{\tilde{\mathbf{x}}_k} \tag{14}$$

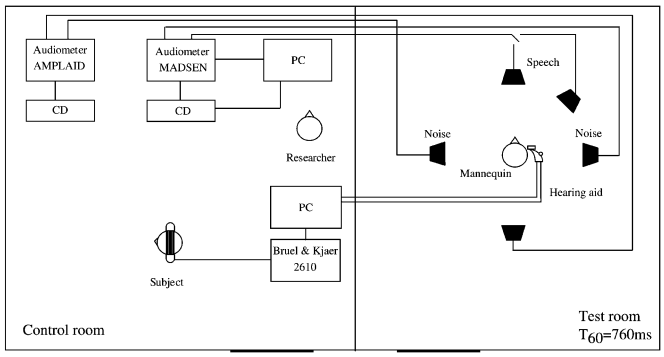


Fig. 5. The experimental setup for the perceptual evaluations of the SVD-based optimal filtering technique.

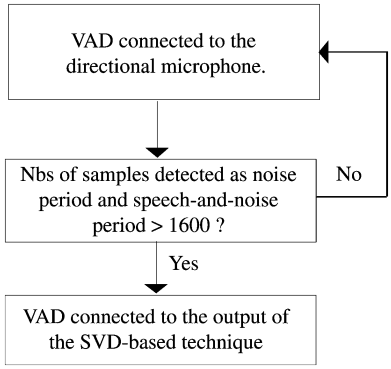


Fig. 6. Initialization strategy when the VAD is connected to the output of the SVD-based noise reduction scheme.

III. Experimental Setup

Two neighboring rooms were used for the experiments, a test room and a control room (Fig. 5). The BTE-devices were positioned on the right ear of a mannequin in the test room and subjects evaluated the processed speech in the control room.

The *test room* simulates the acoustics of a living room situation with a volume of 70 m^3 and a reverberation time T_{60} of 0.76 s. Five loudspeakers (Yamaha CBX-S3) were used for the speech and discrete noise sources: one for the speech signal and three for the noise signals. The loudspeaker for the speech signal was situated in front of the mannequin at 0° or at 45° (side of the hearing aid) depending on the test scenario. The loudspeakers for the noise signals were located at 90° (side of the hearing aid), at 270° (the opposite side of the hearing aid) and behind the mannequin at 180°. The loudspeakers were calibrated separately to obtain the same sound level for each loudspeaker in the middle of the five loudspeaker positions. The center of the loudspeakers was at the same height as the hearing aid of the mannequin (140 cm) and the distance between each loudspeaker and the center of the mannequin's head was one meter. For noise scenarios with more than one spectrally identical jammer noise, the presented sources were uncorrelated.

In the *control room*, the speech signals were output by a computer sound card (Sound blaster 16 bits) and the noise signals by a SONY CD991 CD player. For amplification purposes and intensity control, the speech signal and the noise at 90° were sent through a MADSEN OB822 audiometer, and the noises at 180° and 270° through an AMPLAID 309 audiometer. The signals of the microphones of the hearing aids were amplified with a two-channel Larson Davis 2200C amplifier and were digitized, at a sampling frequency of 12 kHz using a LynxONE PC-sound card having two input channels with 16-bit analog-to-digital conversion (ADC). For the perceptual evaluation, C-code was developed to run the algorithms in real time. The output signal of the noise reduction algorithms were sent to an analog output of the LynxOne card. This analog signal was connected to a Bruel & Kjaer (2610) amplifier. Since the BTE-devices were positioned on the right ear of the mannequin during the tests, the output signal was presented monaurally to the right ear of the subjects through headphones TDH39. This Bruel & Kjaer amplifier was calibrated to get an overall level of 65 dBSPL on the right channel of the headphone when a level of 65 dBPSL was present at the center of the loudspeakers with the head absent.

A. Physical Evaluation

In noise reduction strategies, several signals are available to the VAD, such as the signal of the omnidirectional microphone, the directional microphone or even the output of the noise reduction technique. In this study, the behavior of the VAD was evaluated when the VAD was connected to these different signals. The signals of the omnidirectional or the directional microphones (software or hardware), are directly available. However, when the VAD is connected to the output of the noise reduction algorithms, the signals are only available after a first update of the adaptive filters. The optimal filtering technique needs at least a noise-only period and a speech-and-noise period. To solve this problem of initialization (Fig. 6), the VAD is connected to the directional microphone (software or hardware) and when several samples (e.g., 1600) are classified as speech-and-noise periods or noise-only periods, the optimal filters are updated. Only then, the VAD algorithm is connected to the output of the SVD-based optimal filtering strategy.

1) Signals: In the test room, the hearing aid was positioned on a dummy head and two loudspeakers were used. One loudspeaker was situated in front of the dummy head (at 0°) and presented the speech material (sentences), and a second at 90° (on the side of the hearing aid) presented the noise material. The VAD was evaluated for stationary and nonstationary noise. As stationary noise, a long term average speech-weighted noise of sentences was used, and as nonstationary noise, a multitalker noise was used. The signals of the microphones were recorded during 90 s. The speech and the noise signals were recorded separately. In this way, it was feasible to know perfectly the speech-and-noise periods and the noise-only periods. This perfect VAD was used as a reference for the evaluation of the VAD algorithm.

2) Performance Measure: The performance of the VAD was evaluated by calculating the percentage correctly detected samples by the VAD algorithm for speech-and-noise periods and noise-only periods of the signals. The percentage (Per) was calculated by

$$\mathrm{Per} = \frac{\mathrm{SN}_{\mathrm{RealTime}} \times 100}{\mathrm{SN}_{\mathrm{Perfect}}}$$

or

$$\mathrm{Per} = \frac{N_{\mathrm{RealTime}} \times 100}{N_{\mathrm{Perfect}}} \tag{16}$$

where N_{Perfect} and $\mathrm{SN}_{\mathrm{Perfect}}$ are the number of samples, which are known to be belong to noise-only periods (N) or speech-and-noise periods (SN), based on the "perfect" VAD. N_{Realtime} and $\mathrm{SN}_{\mathrm{RealTime}}$ are the number of samples which are correctly classified as noise-only periods or speech-and-noise periods by the real time VAD. In the calculation, the first 20 s of the signals were not taken into account, which gave time to the noise reduction algorithms to converge.

B. Perceptual Evaluations

The main perceptual research questions of this study are related to the optimization of the speech distortion parameter μ of the SVD-based optimal filtering technique, and, a comparison between the latter technique and the two-stage adaptive beamformer.

1) Subjects: To avoid eventual fatigue and/or training effects in the data, two different groups of normal hearing listeners participated in this study. The first group consisted of ten normal hearing listeners and their ages were ranged from 20 to 27 (mean of 22 years). This group performed the tests to optimize the speech distortion parameter μ of the SVD-optimal filtering technique and the tests with the hearing aid *DO* to evaluate the performance of the two noise reduction techniques. The second group consisted of five normal hearing persons and their ages varied from 20 to 42 years (mean of 30 two years). This group performed the tests with the hearing aid *OO* to evaluate the two noise reduction techniques. In both group, the pure tone threshold was not less than or equal to 15 dBHL at the octave frequencies from 125 Hz to 8 kHz.

2) Method: The speech distortion parameter μ of the SVD-based technique was optimized for one jammer sound scene (speech at 0° and noise at 90°), one spectro-temporal character of the jammer sound (stationary speech weighted noise of the sentences) and one basic microphone configuration in the hearing aid (*DO*). The first group of subjects (ten persons) performed twice the SRT-measurements for the omnidirectional microphone and the optimal filtering technique for different values of μ. The SRT-improvements of the noise reduction strategies were relative to the omnidirectional microphone. For the comparison of the two noise reduction algorithms in a single and in a multiple jammer sound scenario, the tests were performed for different spectro-temporal characters of the jammer sounds, and two basic microphone configurations in the hearing aid. SRT-measurements were carried out with two groups of subject consisting of ten and five normal hearing persons. The group of ten subjects (the same as for the optimization of the speech distortion parameter μ) performed the evaluation with the hearing aid *DO* and the group of five subjects, performed the evaluation with the hearing aid *OO*. The first group (ten persons) did not perform the tests with the hearing aid *OO* because of the limited availability of lists of speech material. The tests of the omnidirectional microphone, the two-stage adaptive beamformer and the SVD-based optimal filtering technique with hearing aids *OO* and *DO* were carried out in two spectro-temporally different jammer sounds (SW: unmodulated speech weighted noise and MB: multitalker babble) and in two different jammer sound scenarios (speech source at 0° and a single noise at 90°, versus speech source at 45° and three uncorrelated noise sources at 90°/180°/270°).

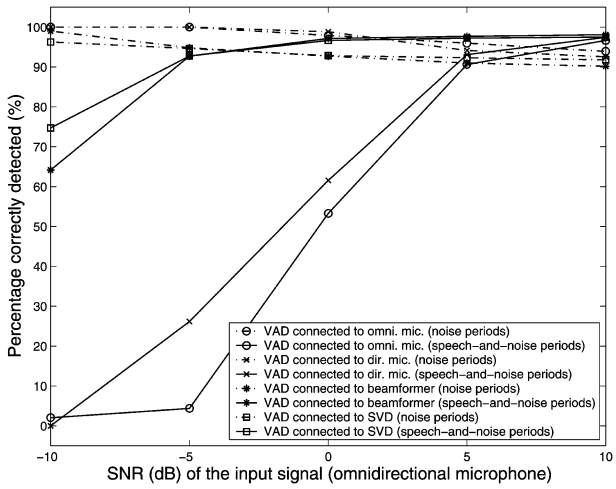


Fig. 7. Performance of the VAD when it is connected to the front omnidirectional microphone, the software directional microphone, the output of the adaptive beamformer and the output of the SVD-based technique in stationary conditions (hearing aid *OO*).

3) Test Materials: All the tests were carried out with the same speech material (sentences from a male speaker). To measure the SRT, an adaptive procedure was used [26]. The adaptive procedure adjusts the level of the speech material to extract the 50% speech recognition level, i.e., the SRT. The first sentence of a speech list was repeated with increasing level until the subject correctly identifies it. Once this level was determined, every other sentence of the list was subsequently presented only once at a level lower or higher, depending on the former item being identified correctly or not. The level step size for each sentence was taken as 2 dB. In order to determine the SRT, the levels of the last six responses of a list of 13 sentences were averaged. The three noise sources were calibrated separately, to obtain a constant sound level at the reference point. Thus, a sound level was obtained of 65 dBSPL and 69.8 dBSPL with one noise and three noise sources, respectively. Sentences spoken by a male voice were used as speech materials, and, unmodulated speech weighted noise (with spectrum identical to the corresponding speech material) and multitalker babble were used as noise materials. The sentence speech materials were the Dutch sentences developed by Versfeld *et al.* [27]. These sentences are an extension of the materials of Plomp and Mimpen [26] to measure speech reception thresholds. Thirty nine lists were available, where each list contained 13 sentences. One of the two test noises was unmodulated noise speech weighted according to the spectrum of the specific speech materials used. The other noise was multitalker babble, taken from the compact disk "Auditory Tests" (Revised) edited by Auditec of St. Louis.

IV. Results and Discussion

A. Physical Evaluation

1) Stationary Conditions: Figs. 7 and 8 show the results of the percentage (Per) of samples correctly detected by the VAD algorithm in a stationary speech weighted noise by the VAD algorithm for speech-and-noise periods and noise-only periods. For both types of the hearing aids, *OO* and *DO*, the VAD algorithm detected almost correctly $(\mathrm{Per} > 90\%)$ the noise-only

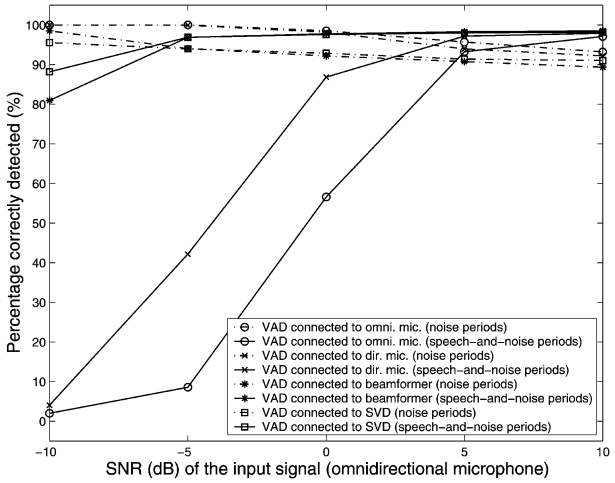


Fig. 8. Performance of the VAD when it is connected to the front omnidirectional microphone, the hardware directional microphone, the output of the adaptive beamformer and the output of the SVD-based technique in stationary conditions (hearing aid *DO*).

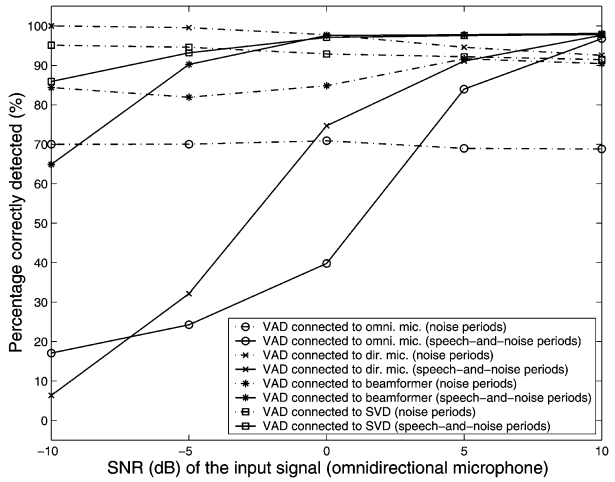


Fig. 9. Performance of the VAD when it is connected to the front omnidirectional microphone, the software directional microphone, the output of the adaptive beamformer and the output of the SVD-based technique in nonstationary conditions (hearing aid *OO*).

periods when it was connected to the omnidirectional microphone, the directional microphone (software and hardware) or the output of the noise reduction strategies (two-stage adaptive beamformer or optimal filtering technique). The detection performance for the speech-and-noise periods was clearly dependent on where the VAD was connected. The performance of the VAD dropped significantly when it was connected to the omnidirectional microphone or the directional microphones for a SNR below 5 dB. When the VAD was connected to the output signal of the noise reduction algorithm, the percentage well-detected samples stayed above 90% for a SNR above -5 dB. With the hearing aid *OO* at $\mathrm{SNR} = -10$ dB, the percentage obtained with the optimal filtering technique was 75% and with the two-stage adaptive beamformer the percentage was 65%. With the hearing aid *DO* at $\mathrm{SNR} = -10$ dB, the scores were about 90% with the optimal filtering technique and about 80% with the adaptive beamformer. Per was roughly similar with the two omnidirectional microphones of the hearing aids and the hardware directional microphone gave a better performance than the software directional microphone for a SNR below 0 dB.

2) Nonstationary Conditions: In nonstationary conditions, the performance of the VAD is illustrated in Figs. 9 and 10. The omnidirectional microphone of the device *DO* had a higher Per than the omnidirectional microphone of the device *OO* for the speech-and-noise periods below 5 dB SNR. The percentages for the noise-only periods were similar. This may be due to the fact that the omnidirectional microphones were not at the same position in the hearing aids.

Connecting the VAD to the software and the hardware directional microphones did not have the same impact on the detection performance of the VAD. With the software directional microphone, the noise-only periods were well-detected ($\mathrm{Per} > 90\%$) independent of the SNR, while, detection of the speech-and-noise periods was dependent on the SNR. The lower the SNR, the lower the Per. Relative to the software directional microphone, the VAD connected to the hardware directional microphone detected better the speech-and-noise periods but performed worse for the noise-only periods ($\mathrm{Per} \approx 45\%$) relative to 85%. The VAD-output is a function of the statistical

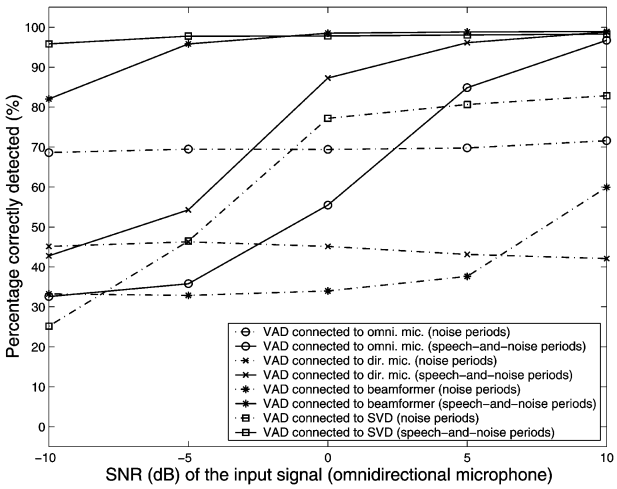


Fig. 10. Performance of the VAD when it is connected to the front omnidirectional microphone, the software directional microphone, the output of the adaptive beamformer and the output of the SVD-based technique in nonstationary conditions (hearing aid *DO*).

variations of the signals and a speech-and-noise period is detected if the statistics of the new frame exceed the estimated statistics of the previous frames. The more the statistics change, the more the speech-and-noise periods are detected. Fig. 11 shows the frequency responses of the hardware directional microphone and the software directional microphone. The directional processes attenuate the sensitivity at the low frequencies ($f < 1$ kHz), where both directional microphones have roughly the same frequency response (intermicrophone or interport distance in both cases is 10 mm). Between the frequencies 1 kHz and 4 kHz, the frequency response of the hardware directional microphone exceeds the frequency response of the software directional microphone. Having more high frequency component, the signal of the hardware directional microphone has more temporal changes than the signal of the software directional microphone. This can explain the difference in the VAD-output between both directional microphones.

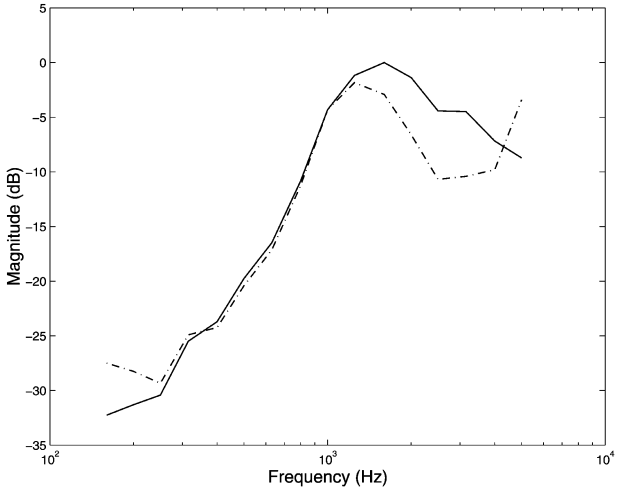


Fig. 11. Frequency response of the hardware directional microphone $(-)$, the software directional microphone $(-\cdot)$.

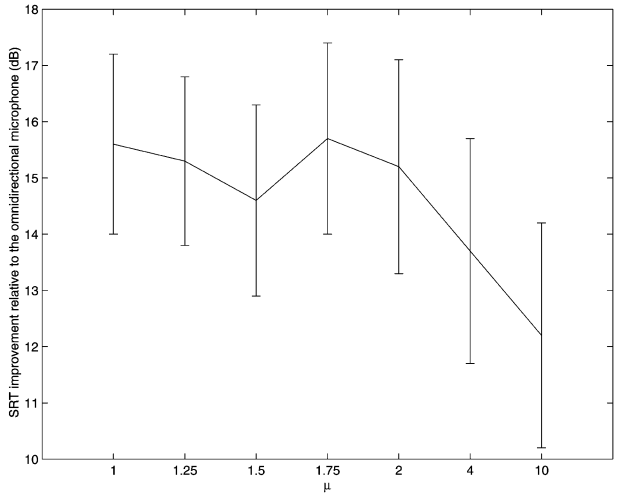


Fig. 12. Influence of the speech distortion parameter μ on the SRT-improvements of the SVD-based optimal filtering technique.

Similar to the results under stationary conditions, the best performance was revealed by connecting the VAD to the output of the noise reduction strategy. For a $\mathrm{SNR} \geq -5$ dB, the speech-and-noise periods were well-detected. $\mathrm{Per} > 90\%$ was observed for both microphone configurations when the VAD used the output signals of the two-stage adaptive beamformer and the optimal filtering technique. It seems that the frequency responses of the two directional microphones also have an influence on the VAD-output. Indeed with the hearing aid *DO*, the VAD was very sensitive to the temporal variations of the multitalker babble and tended to detect only the speech-and-noise periods $(\mathrm{Per} > 80\%)$ whereas the noise-only periods were practically not detected, $\mathrm{Per} \approx 30\%$ at a SNR of -10 dB. With the hearing aid *OO*, Per was above 80% for the detection of the noise-only periods and Per was above 65% for the detection of the speech-and-noise periods. This means for having a similar performance of the VAD, the sensitivity of the algorithm needs to be fitted for every signal.

The performance of the VAD in a multiple jammer sound scenario was also evaluated. A similar VAD-output was obtained for the different signals. Results similar to the above results for stationary conditions were obtained with the multitalker babble because the sum of the nonstationary signals at the omnidirectional microphones resulted in a stationary-like signal.

The VAD works on the log-energy of the signal, hence, its performance depends on the SNR. The better the SNR, the better the discrimination between the speech-and-noise period and the noise-only period and connecting the VAD to the output of the noise reduction algorithm revealed the best performance. In the sequel, all the experiments will be performed with the VAD connected to the output of the noise reduction strategy.

B. Perceptual Evaluations

In this section, the results for the optimization of the speech distortion parameter μ of the SVD-based optimal filtering technique and the gain in speech intelligibility of both noise reduction algorithms for the two jammer sound scenarios are presented.

1) Speech Distortion Parameter: Fig. 12 shows the influence of the speech distortion parameter μ on the SRT-improvements of the SVD-based optimal filtering strategy. The best improvement of the speech intelligibility was obtained for $\mu = 1.75$. For $1 \leq \mu \leq 2$, the obtained SRT-improvements were similar (within 1 dB), but the SRT-improvement decreased quite rapidly for $\mu = 4$ and $\mu = 10$. For these two values of μ, too much distortion was brought to the speech signal and affected the speech intelligibility. Hence, the speech distortion parameter μ was fixed to 1.75.

TABLE I
SRT-IMPROVEMENT (DB) AVERAGED MEAN(STD)) PER GROUP OF SUBJECTS OBTAINED WITH THE TWO-STAGE ADAPTIVE BEAMFORMER AND THE SVD-BASED OPTIMAL FILTERING TECHNIQUE, RELATIVE TO THE OMNIDIRECTIONAL MICROPHONE FOR DIFFERENT CONDITIONS. THE GROUP OF TEN NORMAL HEARING LISTENERS WERE TESTED WITH THE HEARING AID *DO* AND THE SECOND GROUP OF FIVE NORMAL HEARING LISTENERS WERE TESTED WITH THE HEARING AID *OO*

		Adaptive beamformer		Optimal filtering technique	
Noise scenario	Noise material	DO	OO	DO	OO
$S_{0^\circ}N_{90^\circ}$	SW	15.8 (1.8)	14.3 (2.2)	15.1 (1.3)	14.0 (2.4)
	MB	17.0 (2.7)	14.3 (1.3)	13.3 (3.0)***	12.6 (1.7)
$S_{45^\circ}N_{90^\circ/180^\circ/270^\circ}$	SW	7.5 (0.9)	6.5 (1.8)	9.0 (1.5)**	8.4 (1.5)*
	MB	5.8 (2.8)	4.7 (2.4)	6.3 (2.1)	5.2 (1.9)

2) Noise Scenarios: Table I shows the improvements (in decibels) of the SRT relative to the omnidirectional microphone for the two-stage adaptive beamformer and for the SVD-based optimal filtering technique and for both types of hearing aids (*OO* and *DO*). Every data cell corresponds to the mean (and the standard deviation) of all subjects in one group in each test condition. Columns three to four show the results of the SRT-improvement between the omnidirectional microphone and the two-stage adaptive beamformer, and, columns five to six show the results of the SRT-improvement between the omnidirectional microphone and the optimal filtering technique. This is presented for the two noise materials (SW and MB) and the two jammer sound scenes ($S_{0^\circ}N_{90^\circ}$ and $S_{0^\circ}N_{90^\circ/180^\circ/270^\circ}$). Statistic analysis were performed on the data using SPSS 10.0 software where detailed aspects were studied with paired comparisons analysis.

In the single noise source scenario, the behavior of both noise reduction strategies differed in multitalker babble with the hearing aid *DO*. The paired comparisons analysis gave a significant difference between the beamformer and the optimal

filtering technique ($p < 0.001$). Moreover, the two-stage adaptive beamformer was dependent on the microphone configuration in the hearing aids ($p = 0.023$). In the multiple noise sources scenario, the SVD-based optimal filtering technique was significantly better than the two-stage adaptive beamformer when a stationary speech weighted noise was present with the hearing aid *OO* ($p = 0.011$) and *DO* ($p = 0.005$).

3) Discussion:

a) Single noise source scenario: The single noise scenario was optimal for the *two-stage adaptive beamformer* because the desired target was in the look direction given to the beamformer. Although a leakage of the speech signal occurred in the noise reference of the adaptive beamformer (the first filter was calibrated in anechoic conditions and fell short in reverberant conditions), the presence of the speech signal in the noise reference was limited. The tests were carried out at SNRs about -10 dB and 0 dB in stationary and non stationary conditions, respectively, and the detection performance of the vad for the speech-and-noise periods was high enough to avoid that the anc stage was corrupted by the statistics of the speech signal.

With the *optimal filtering technique*, the VAD discriminated well the speech-and-noise periods ($\mathrm{Per} > 75\%$) from the noise-only periods ($\mathrm{Per} > 90\%$) in stationary conditions for both microphone configuration and in nonstationary conditions with the hearing aid *OO*. The perceptual tests were performed at a SNR about -10 dB in stationary condition and about -5 dB in non stationary conditions. With the hearing aid *DO* in the multitalker babble a significant difference was found between the two noise reductions strategies. In this configuration, the VAD was sensitive to the temporal variation of the noise and tended to detect only speech-and-noise periods even when the noise-only periods were present. At a SNR of -10 dB, similar to the SNR to which the perceptual tests were performed, the VAD detected correctly about 95% the speech-and-noise periods and only 25% of the noise-only periods (Fig. 10). On the one hand, this did not result in a cancellation of the desired signal but on the other hand the optimal filtering technique did not have a good estimate of the noise statistics and hence could not set up an optimal noise cancellation. The function based on the derivative D_F had a moderate influence on the performance of the optimal filtering technique because the speech-and-noise periods were well detected by the VAD and the threshold T_D was fitted low enough so that the spectral changes of the multitalker babble did not make the derivative D_F effective ($D_F > T_D$).

b) Multiple noise sources scenario: In stationary conditions, the optimal filtering technique gave a better SRT-improvement than the two-stage adaptive beamformer. The noise scenario was not optimal for the beamformer technique because the desired signal was not in the look direction given to the strategy. The level of the speech signal leakage into the noise reference increased [10], [18], [28] and the ANC stage was corrupted by the statistics of the speech signal. Hence, the two-stage adaptive beamformer became more sensitive to the errors of the VAD and a partial cancellation of the speech signal was performed at the output of the algorithm [18], [19]. Unlike the adaptive beamformer, the performance of the optimal filtering technique is not a function of the position of the speaker because this technique works without assumptions on the desired target. The differences were small between the two strategies (1.5 dB and 1.9 dB with the hearing aids *DO* and *OO*, respectively) but very important for hearing-aid users. In critical listening conditions (close to 50% of speech understood by the listener) an improvement of 1 dB in SNR can correspond to an increase of speech understanding of about 15% in every day speech communication [26].

Significant differences were found between the stationary and nonstationary conditions. This is may be due to the fact that the asymptotic maximum noise reduction performance of the noise reduction strategies is reached if the time constants of the techniques are small enough to track the temporal and spatial changes of the noises during the tests. Under stationary conditions, the spectral and the spatial variations are minimal and in nonstationary conditions with a single noise source only the spectral changes need to be tracked. The time constant was small enough (< 1 s) to follow the changes in these noise scenarios. In a multiple noise scenario with a nonstationary noise, the time constant of the noise reduction strategies was certainly not small enough to track the variations in the spectral and the spatial domains. The technique could not perform an optimal noise reduction and the asymptotic maximum noise reduction performance was not reached.

Relatively to the single noise source scenario, the SRT-improvements of the noise reduction strategies decreased by between 6 and 10 dB. Peterson [9] analyzed the effects of the number of noise sources and microphones on the performance of the noise reduction algorithms: the more independent sensor inputs the better noise sources can be nulled out with appropriate signal processing. Thus, two microphones configurations are only most optimal for the cancellation of just one noise source, separately from the desired speech signal.

c) Microphone configuration: With the two-stage adaptive beamformer, the microphone configuration of the hearing aid had an influence on the performance of the strategy. The hearing aid *DO* gave an additional improvement relative to the hearing aid *OO*. These results agree with a previous study, but a larger difference between both types of microphone configurations was observed [14]. The tests were carried out in the same conditions as in this study (same speech material, speech weighted noise, noise scenario and room configuration), and a difference of 3.6 dB was obtained between the devices *OO* and *DO* for the adaptive beamformer. In this study, the difference was only 1.5 dB between both devices. The positions of the microphones were different in the hearing aid for both studies and this can explain the dissimilarity in the results.

d) Computational complexity: The computational complexity of the two-stage adaptive beamformer equals $(2.N_1 + 7.N_2) \times fs$ operations (multiplications or additions) per second (ops/s). N_1 represents the size of the fixed filter, N_2 the size of the adaptive filter and fs the sampling frequency. In this study, the two-stage adaptive beamformer had a computational complexity of 2Mops/s, with $N_1 = 8$ taps and $N_2 = 22$ taps and a sampling frequency of 12 kHz. The computational cost of the SVD-based optimal filtering equals $(17.5.N^2 + 4.N^2) \times fs$ ops/s. Here N represents the size of the filters per channel and was set to $N = 15$, so that the computational cost equals 232 Mops/s at 12 kHz. SVD procedures have a high computational complexity, but, recent studies showed that this complexity can be further reduced, making this approach attractive for practical systems. A subband approach or an alternative algorithms based on a QR Decomposition were proposed in [30] and [29]. More

recently, a LMS approach was proposed and had approximately the same computational cost as the two-stage adaptive beamformer, be it at the expense of a larger memory requirement [29].

V. Conclusion

In this paper, a first real-time implementation and perceptual evaluation of a SVD-based optimal filtering technique was performed in the context of noise reduction in two microphone hearing aids. The perceptual experiments were carried out in two different noise scenarios with two microphone configurations. The SVD-based optimal filtering technique was compared to an adaptive beamformer. A method to improve the performance of the VAD was presented. The behavior of the VAD was evaluated physically when the VAD was connected to different signals, such as omnidirectional microphone, directional microphone signals and the output of the noise reduction strategy. The VAD could discriminate well the speech-and-noise periods from the noise-only periods, when it was connected to the output of the algorithms. The perceptual evaluation showed that, in an optimal noise scenario for the adaptive beamformer, the optimal filtering technique performs as well as the beamformer under the condition that the VAD performs a good discrimination between the speech-and-noise periods and the noise-only periods. Important SRT-improvements were obtained for both noise reduction strategies because single jammer sound scenarios are appropriate for dual microphone hearing aids. In a multiple jammer sound scenario, the optimal filtering technique performed better than the adaptive beamformer in stationary conditions.

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