Speech enhancement with multichannel Wiener filter techniques in multimicrophone binaural hearing aids

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This paper evaluates speech enhancement in binaural multimicrophone hearing aids by noise reduction algorithms based on the multichannel Wiener filter (MWF) and the MWF with partial noise estimate (MWF-N). Both algorithms are specifically developed to combine noise reduction with the preservation of binaural cues. Objective and perceptual evaluations were performed with different speech-in-multitalker-babble configurations in two different acoustic environments. The main conclusions are as follows: (a) A bilateral MWF with perfect voice activity detection equals or outperforms a bilateral adaptive directional microphone in terms of speech enhancement while preserving the binaural cues of the speech component. (b) A significant gain in speech enhancement is found when transmitting one contralateral microphone signal to the MWF active at the ipsilateral hearing aid. Adding a second contralateral microphone showed a significant improvement during the objective evaluations but not in the subset of scenarios tested during the perceptual evaluations. (c) Adding the partial noise estimate to the MWF, done to improve the spatial awareness of the hearing aid user, reduces the amount of speech enhancement in a limited way. In some conditions the MWF-N even outperformed the MWF possibly due to an improved spatial release from masking.
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I. INTRODUCTION

Hearing aid users often have great difficulty understanding speech in a noisy background. They typically 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. Therefore, several single- and multimicrophone noise reduction strategies have been developed for modern hearing aids. Multimicrophone noise reduction systems are able to exploit spatial in addition to spectral information and are hence preferred to single-microphone systems (Welker *et al.*, 1997; Lotter, 2004). However, the noise reduction systems currently implemented in modern hearing aids, typically adaptive directional noise reduction systems, are designed to optimize speech in noise monaurally (Wouters and Vanden Berghe, 2001; Luo *et al.*, 2002; Maj *et al.*, 2004). In a bilateral hearing aid configuration, these systems do not take the contralateral ear into account and hence may incorrectly represent the binaural cues (Keidser *et al.*, 2006; Van den Bogaert *et al.*, 2006, 2008). The main binaural cues are interaural time differences (ITDs) and interaural level differences (ILDs). These cues play a major role in directional hearing in the horizontal plane and in spatial awareness and also contribute to an improved speech understanding in a noisy environment due to spatial release from masking also known as "the cocktail party effect" (Plomp and Mimpen, 1981; Bronkhorst and Plomp, 1988; Bronkhorst, 2000). By combining the microphones of the left and right hearing aids into one binaural hearing aid configuration, adaptive algorithms may be controlled more easily to preserve binaural cues, thereby enhancing directional hearing and speech perception in noisy environments. Moreover, an additional improvement in speech perception may be obtained by an increased noise reduction performance due to the advanced signal processing on an increased number of available microphone signals. Spectral cues, more related to resolving front-back confusions and elevation, are not discussed in this manuscript.

Several algorithms have been studied in the past decennium to combine noise reduction with the preservation of binaural localization cues. First, Wittkop and Hohmann (2003) proposed a method based on computational auditory scene analysis in which the input signal is split into different frequency bands. By comparing the estimated binaural properties, such as the coherence, of each frequency band with the expected properties of the signal component (typically it is assumed that the signal component arrives from the frontal area with ITD and ILD values close to 0 μs and 0 dB), these frequencies are either enhanced or attenuated. By applying identical gains to the left and the right hearing aid, binaural cues should be preserved. However, spectral enhancement artifacts such as "musical noise" will typically occur. Moreover, localization performance when using this technique was never evaluated.

A second class of systems is based on fixed or adaptive beamforming. Desloge *et al.* (1997) introduced two methods to combine fixed beamforming strategies with the preservation of localizing abilities in a binaural hearing aid. In the first method, the amount of ITD distortion introduced by the fixed beamformer, averaged over all directions, was constrained to 40 μs. In the second method, the fixed directional beamformer was limited to frequencies higher than 800 Hz. The monaural output of the beamformer was then combined with the unprocessed low ($f < 800$ Hz) frequencies of the omnidirectional microphone at each hearing aid. These frequencies could then be used to localize sound sources. Both of these methods are inspired by observations that the ITD information, present at the lower frequencies, is a dominant localization cue compared to the ILD information, present at the higher frequencies (Wightman and Kistler, 1992). With both systems, a reasonable localization performance was obtained with a root mean square error smaller than 20°. The high pass–low pass method was expanded by Welker *et al.* (1997). An adaptive beamformer was used to process the high-frequency part of the signal. When using this approach with hearing impaired subjects, Zurek and Greenberg (2000) obtained a noise reduction performance of 2.0 dB. However, these systems usually rely on the assumption that the speech signal is arriving from the frontal hemisphere and that the noise signal is arriving from the back hemisphere. Therefore, a good noise reduction performance is only obtained in these specific scenarios. Moreover, localization cues are typically preserved for the targeted speech component but not for the noise component, and this only when speech is arriving from the forward field of view (Van den Bogaert *et al.*, 2008).

A third class of systems is based on the multichannel Wiener filter (MWF). In general, the goal of the Wiener filter is to filter out noise corrupting a desired signal. By using the second-order statistical properties of the desired speech signal and the noise, the optimal filter or Wiener filter can be calculated. It generates an output signal that estimates the desired signal in a minimum mean square error sense. In contrast with a standard beamformer, it can do so without any prior assumption on the angle of arrival of the signal. In Doclo and Moonen (2002), it was shown that a MWF can be used for monaural hearing aid applications. Later on, this approach was extended to a binaural hearing aid configuration in which one or more contralateral microphone signals can be added. One of the main benefits of a MWF is that it inherently preserves the interaural cues of the estimated speech component. This was mathematically proven in the work of Doclo *et al.* (2006). However, it was also proven that the interaural cues of the noise component are distorted into those of the speech component. To preserve binaural information for both the speech and the noise component, an extension, the MWF with partial noise estimation (MWF-N), was proposed by Klasen *et al.* (2007). The rationale of the MWF-N is to remove only part of the noise component. The remaining unprocessed part of the noise signal then restores the spatial cues of the noise component in the signal at the output of the algorithm. Obviously this may come at the cost of a reduced noise reduction. In a way, this is similar to the work of Noble *et al.* (1998) and Byrne *et al.* (1998), in which improvements in localization performance were found when using open instead of closed earmolds. The open earmolds enables the use of the direct unprocessed sound at frequencies with low hearing loss to improve localization performance.

In Van den Bogaert *et al.* (2008) it was shown perceptually that in a binaural hearing aid configuration, the MWF and the MWF-N, have advantages in terms of spatial awareness for the hearing aid user in comparison with an adaptive directional microphone (ADM), which is the most frequently implemented adaptive multimicrophone noise reduction system in modern digital hearing aids. This was done by using a localization experiment in the frontal horizontal hemisphere with a realistic environment (T_{60}=0.61 s). In contrast with the ADM, the MWF preserves the location of the target speech sound, independently of its angle of arrival. However, in some conditions subjects located the noise source at the place of the speech source as mathematically predicted by the work of Doclo *et al.* (2006). When using the MWF-N, however, subjects correctly localized both the speech and the noise source.

Until now, noise reduction and speech enhancement performance of the MWF and MWF-N have not been evaluated thoroughly. The study of Klasen *et al.* (2007) focused on the concept of partial noise estimation and how it can decrease ITD and ILD errors. Only a limited set of objective measurements of monaural SNR improvements, done in anechoic conditions with a single noise source fixed at 90°, was reported. The study of Van den Bogaert *et al.* (2008) mainly focused on localization performance. A limited set of speech perception data of three-microphone MWF and MWF-N was also presented. This was done for two single noise source scenarios in a realistic environment. In both scenarios—S_0N_{60} and $S_{90}N_{270}$ with S_xN_y defining the spatial scenario with speech arriving from angle x and a noise signal arriving from angle y—the MWF and MWF-N outperformed a two-microphone ADM. The MWF and MWF-N can increase noise reduction performance by using microphone signals from both the ipsilateral and the contralateral hearing aid. However, transmitting microphone signals between hearing aids comes at the large cost of power consumption and bandwidth, especially since commercial manufacturers prefer a wireless connection between both devices. Therefore, a thorough evaluation in realistic listening conditions is needed on the obtained gain in speech understanding when transmitting no (a bilateral configuration), one, or all contralateral microphone signals. A commonly used ADM is used as a reference noise reduction system. The algorithms discussed in this manuscript are evaluated using monaural and binaural presentations.

This paper presents objective and perceptual evaluations of the noise reduction and speech enhancement performance of the MWF and MWF-N approaches using different microphone combinations under several spatial sound scenarios in different acoustical environments. The main research questions answered in this manuscript are the following: (a) What is the speech enhancement performance of a MWF in comparison with a standard bilateral ADM in a monaural and a binaural hearing aid configuration? (b) What is the gain in speech enhancement when evolving from a monaural hearing aid design to a binaural hearing aid design, i.e., adding a third and/or a fourth microphone, positioned at the contralateral hearing aid, to a MWF already using two microphones of the ipsilateral hearing aid? (c) What is the cost in speech enhancement performance when adding a partial noise estimate into the MWF-scheme, i.e., the MWF-N, which enables a correct sound localization of both the speech and the noise component? (Van den Bogaert *et al.*, 2008). All three questions will be evaluated using both objective performance measures, using a semianechoic and a realistic reverberant environment, and perceptual performance measures, only for the realistic environment, in different single and multiple noise source scenarios. The correlation between both performance measures is also discussed.

II. HEARING AID CONFIGURATION

The hearing aid configuration used in this study is identical to the one used in Van den Bogaert *et al.* (2008). The microphone array of the left and right behind-the-ear hearing aids consists of two omnidirectional microphones with an intermicrophone distance of approximately 1 cm. In a general binaural configuration, microphone signals from the ipsilateral (M_I) and contralateral (M_C) hearing aids can be used to generate an output signal for each ear. Three different noise reduction algorithms were evaluated with these hearing aids: the MWF, the MWF-N, and the ADM. For all algorithms a sampling frequency of f_s=20 480 Hz was used.

A. MWF and MWF-N

Different microphone combinations were evaluated to measure the benefit of adding one or two contralateral microphone signals to the MWF or MWF-N algorithm active at the ipsilateral hearing aid. A monaural system with each hearing aid using only its own two microphone signals was first evaluated. The MWF-based systems were then extended by transmitting one or two contralateral microphone signals to the ipsilateral hearing aid. The three different implementations of the MWF algorithm used in this study are denoted as MWF_{2+M_C}, with $0 \leqslant M_C \leqslant 2$. The three different implementations of the MWF-N algorithm are denoted similarly as MWF_{2+M_C}-N. A list of algorithms evaluated during this study is given in the left column of Table I. A description of the algorithmic aspects of the MWF and MWF-N algorithms is already presented in Van den Bogaert *et al.* (2008). A brief summary is given here. The algorithms are described in the frequency domain.

Transmitting M_C contralateral microphone signals to the ipsilateral hearing aid results in an M-dimensional ($M = M_I + M_C$) input vector $\mathbf{Y}_L(\omega)$ and $\mathbf{Y}_R(\omega)$ for the left and right hearing aid, respectively. Each signal vector $\mathbf{Y}(\omega)$ can be written as a sum of a speech component $\mathbf{X}(\omega)$ and a noise component $\mathbf{V}(\omega)$, which are equal to the speech and noise source signals convolved with the impulse responses of the room. The output signal of the noise reduction algorithm at the left and the right hearing aid can be described by the filtered input vectors, i.e.,

$$Z_L(\omega) = \mathbf{W}_L^H(\omega)\mathbf{Y}_L(\omega), \quad Z_R(\omega) = \mathbf{W}_R^H(\omega)\mathbf{Y}_R(\omega), \tag{1}$$

where $\mathbf{W}_L(\omega)$ and $\mathbf{W}_R(\omega)$ are M-dimensional complex vectors representing the calculated Wiener filters for each hearing aid. The MWF uses the M available microphone signals at each hearing aid to produce the filters $\mathbf{W}_L(\omega)$ and $\mathbf{W}_R(\omega)$. These filters create a minimum mean square error estimate of the speech component at the reference microphone, usually the front omnidirectional microphone for the left [for $\mathbf{W}_L(\omega)$] and for the right [for $\mathbf{W}_R(\omega)$] hearing aid, respectively. By doing so, an MWF inherently preserves the binaural cues of the speech component. Through the remainder of the paper, the frequency domain variable ω is omitted for conciseness.

The filter $\mathbf{W} = [\mathbf{W}_L^T \mathbf{W}_R^T]^T$ with T the transpose operator, is calculated by minimizing the cost function

$$J_{\mathrm{MWF}}(\mathbf{W}) = \mathcal{E}\left\{ \left\| \begin{bmatrix} X_{L,1} - \mathbf{W}_L^H \mathbf{X}_L \\ X_{R,1} - \mathbf{W}_R^H \mathbf{X}_R \end{bmatrix} \right\|^2 + \mu \left\| \begin{bmatrix} \mathbf{W}_L^H \mathbf{V}_L \\ \mathbf{W}_R^H \mathbf{V}_R \end{bmatrix} \right\|^2 \right\}, \tag{2}$$

with H the Hermitian transpose operator and $\mathcal{E}$ the expected value operator. μ is a parameter which trade offs noise reduction performance and speech distortion (Spriet *et al.*, 2004). The rationale of the MWF-N is to remove not the full noise component from the reference microphone signal but to remove only a part $(1-\eta)$ of it. The other part (η) remains unprocessed. This changes the original cost function to

$$J_{\mathrm{MWF\text{-}N}}(\mathbf{W}) = \mathcal{E}\left\{ \left\| \begin{bmatrix} X_{L,1} - \mathbf{W}_L^H \mathbf{X}_L \\ X_{R,1} - \mathbf{W}_R^H \mathbf{X}_R \end{bmatrix} \right\|^2 + \mu \left\| \begin{bmatrix} \eta V_{L,1} - \mathbf{W}_L^H \mathbf{V}_L \\ \eta V_{R,1} - \mathbf{W}_R^H \mathbf{V}_R \end{bmatrix} \right\|^2 \right\}. \tag{3}$$

TABLE I. The list of algorithms, microphone combinations, and spatial scenarios evaluated in this paper. All algorithms use two microphone signals of the left/right hearing aid and M_C (0,1 or 2) microphone signals of the contralateral hearing aid to generate a signal for the left/right ear. This is depicted as $2+M_C$. The second column represents whether a bilateral/binaural (b) and/or a monaural presentation (m) was used during the perceptual evaluation of the corresponding algorithm. The third and fourth columns represent the list of spatial scenarios, S_xN_y, evaluated during the objective evaluations. x represents the location of the speech source; y represents the location of the noise source(s). The conditions S_0N_{60}, $S_{90}N_{270}$, and S_0N_3 were also evaluated perceptually.

Evaluated algorithms		Spatial scenarios	
MWF_{2+0}	$b+m$	S_0N_x	x between 0° and 330°
MWF_{2+1}	b	$S_{90}N_{180}$	Single noise source N at 180°
MWF_{2+2}	b	$S_{90}N_{270}$	Single noise source N at 270° (=−90°)
MWF_{2+0}-$\mathrm{N}_{0.2}$	b	$S_{45}N_{315}$	Single noise source N at 315° (=−45°)
MWF_{2+1}-$\mathrm{N}_{0.2}$	b	S_0N_{2a}	Noise sources at −60° and +60°
MWF_{2+2}-$\mathrm{N}_{0.2}$	b	S_0N_{2b}	Noise sources at −120° and +120°
ADM	$b+m$	S_0N_{2c}	Noise sources at 120° and 210°
Unproc	$b+m$	S_0N_3	Noise sources at 90°, 180° and 270°
		S_0N_{4a}	Noise sources at 60°, 120°, 180° and 210°
		S_0N_{4b}	Noise sources at 60°, 120°, 180° and 270°

Note that Eq. (2) is a special case of Eq. (3) with $\eta=0$. Both cost functions are minimized by using estimates of the speech and noise correlation matrices (Klasen *et al.*, 2007; Van den Bogaert *et al.*, 2008). The Wiener solution minimizing $J_{\text{MWF-N}}(\mathbf{W})$ equals

$$\mathbf{W} = \begin{bmatrix} \mathbf{R}_{x,L} + \mu\mathbf{R}_{v,L} & 0_M \\ 0_M & \mathbf{R}_{x,R} + \mu\mathbf{R}_{v,R} \end{bmatrix}^{-1} \begin{bmatrix} \mathbf{R}_{x,L}\mathbf{e}_L \\ \mathbf{R}_{x,R}\mathbf{e}_R \end{bmatrix}, \tag{4}$$

where $\mathbf{e}_L$ and $\mathbf{e}_R$ are all zero vectors, except for a "1" in the position corresponding to the selected reference microphone, i.e., $\mathbf{e}_L(1)=1$ and $\mathbf{e}_R(1)=1$. $\mathbf{R}_x$, and $\mathbf{R}_v$, are defined as the $(M\times M)$-dimensional speech and noise correlation matrices, e.g., for the left hearing aid $\mathbf{R}_{x,L}=\mathcal{E}\{\mathbf{X}_L\mathbf{X}_L^H\}$ and $\mathbf{R}_{v,L}=\mathcal{E}\{\mathbf{V}_L\mathbf{V}_L^H\}$. A voice activity detector (VAD) is used to discriminate between "speech and noise periods" and "noise only periods." The noise correlation matrix $\mathbf{R}_v$ was calculated during the noise only periods. The speech correlation matrix $\mathbf{R}_x$ was estimated by subtracting $\mathbf{R}_v$ from the correlation matrix $\mathbf{R}_y$ of the "speech and noise" signal vector $\mathbf{Y}$. For both the MWF and MWF-N algorithms, speech and noise and noise only correlation matrices were calculated using a perfect VAD. A filter length of 96 taps was used per microphone channel. When using block processing, an overlap of 48 samples was used, leading to a total delay of approximately 4.7 ms for the MWF and MWF-N algorithms. Pilot experiments showed that $\mu=5$ provides a good trade-off between noise reduction and speech distortion. In the work of Van den Bogaert *et al.* (2008), it was shown that $\eta=0.2$ resulted in a good localization performance. Therefore these parameter settings were used throughout this study.

B. Adaptive directional microphone

An ADM was used as a reference multimicrophone noise reduction algorithm. This algorithm is commonly used in modern digital hearing aids (Luo *et al.*, 2002; Maj *et al.*, 2004). Unlike the MWF-based algorithms, the ADM relies on the assumption that the target signal arrives from the frontal field of view and that jammer signals arrive from the back hemisphere. The ADM exploits the time of arrival differences between the microphones on a hearing aid to improve the SNR by steering a null in the direction of the jammer signals. The ADM used the two omnidirectional microphones of the ipsilateral hearing aid. A first stage generated two software directional microphone signals corresponding to, respectively, a front and a back oriented cardioid pattern. These signals were then combined by an adaptive scalar β to minimize the energy arriving from the back hemisphere at the output of the algorithm (Maj *et al.*, 2006). The parameter β was constrained between 0 and 0.5 to avoid noise reduction in the frontal hemisphere.

III. METHODS

A. General

First, different sets of impulse responses were measured between a loudspeaker and the microphones in two behind-the-ear hearing aids worn by a CORTEX MK2 manikin. Loudspeakers were placed at 1 m distance of the center of the head, and impulse responses were measured in the horizontal plane in steps of 30°. Measurements were done in a room with dimensions $5.50\times4.50\times3.10$ m^3 (length $\times$width$\times$height), and acoustical curtains were used to change its acoustical properties. Two different acoustical environments were studied with a reverberation time, linearly averaged over all one-third octave bands between 100 and 8000 Hz, of, respectively, $T_{60}=0.21$ s and $T_{60}=0.61$ s, with the latter value corresponding to a realistic living room condition.

The measured impulse responses were convolved with the appropriate speech and noise material to generate the four microphone signals for the different spatial scenarios used in the perceptual and the objective evaluations. A spatial scenario, with a target signal (S) arriving from angle x and one or multiple noise sources (N) arriving from angle(s) y, is denoted as S_xN_y. The angles were defined clockwise with 0° being in front of the subject. The generated microphone signals were used as input for the different algorithms. Besides the different algorithms, an unprocessed condition, using the front microphones of each hearing aid, was used as a reference condition. In each spatial scenario, the input SNR was calibrated to 0 dBA, measured in absence of the head. A full list of tested conditions is given in Table I. Evaluations were done after convergence of the filters for all algorithms.

B. Objective evaluation

The improvement in speech intelligibility weighted SNR ($\Delta\text{SNR}_{\text{SI}}$), defined by Greenberg *et al.* (1993), was used to evaluate the noise reduction performance of the algorithms. This is defined as the difference between the output SNR_{SI} and the input SNR_{SI}. The input SNR_{SI} was calculated between the front omnidirectional microphone of the left and the right hearing aid. For the left hearing aid, this gives

$$\Delta\text{SNR}_{\text{SI},L} = \sum_i I(\omega_i)\text{SNR}_{\text{out},L}(\omega_i) - I(\omega_i)\text{SNR}_{\text{in},L}(\omega_i), \tag{5}$$

with $\text{SNR}(\omega_i)$ as the SNR measured in the ith third-octave band and $I(\omega_i)$ as the importance of the ith frequency band for speech intelligibility, as defined by ANSI-SII (1997).

Noise reduction performance was evaluated using an average speech spectrum of a Dutch male speaker from the VU test material (Versfeld *et al.*, 2000) as target sound (S) and multitalker babble (Auditec of St. Louis) as jammer sound (N). The long term average spectrum of both the speech and the noise material is given in Van den Bogaert *et al.* (2008). For multiple noise source scenarios, time-shifted versions of the same noise source signal were generated to obtain "uncorrelated" noise sources. Since simulations are a time-efficient way to assess the performance of noise reduction algorithms, a large number of spatial conditions were examined using one target signal and one to four noise sources. A full list of studied spatial scenarios is given in the right column of Table I. Simulations were done for both $T_{60}=0.21$ s and $T_{60}=0.61$ s to evaluate the influence of reverberation on the algorithms.

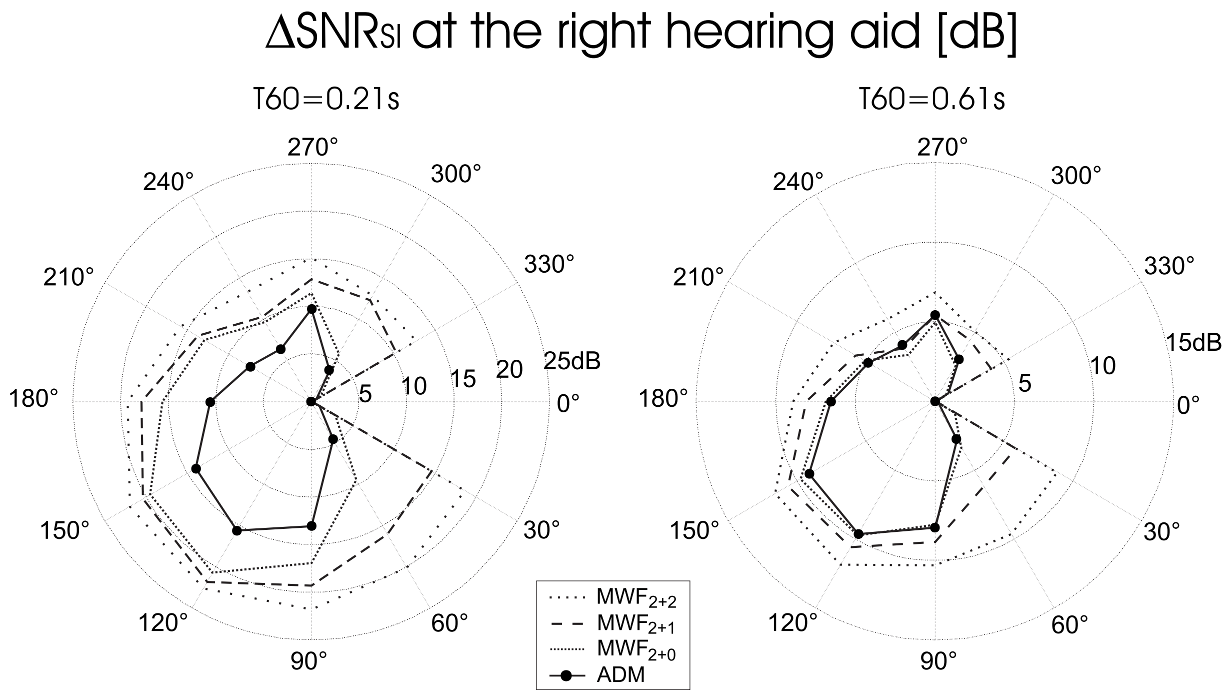

FIG. 1. $\Delta\mathrm{SNR}_{\mathrm{SI}}$ of the ADM and the MWF with different microphone combinations (denoted as MWF_{2+M_C}) for single noise source scenarios with speech arriving from 0° and noise arriving from x° (S_0N_x). The data of the right hearing aid are presented in two reverberant environments, with x being varied per 30°.

C. Perceptual evaluation

Speech reception thresholds (SRTs) were measured with ten normal hearing subjects using an adaptive test procedure (Plomp and Mimpen, 1979). The procedure adjusts the level of the speech signal in steps of 2 dB to extract the 50% SRT. The level of the noise signal was calibrated with the sound pressure level, averaged over the left and the right ear, equal to 65 dBA. The male sentences of the VU test material (Versfeld *et al.*, 2000) were used as speech material, and a multitalker babble (Auditec of St. Louis) was used as a noise source.

The algorithms were perceptually evaluated using a binaural presentation with signals presented to both the left and the right ear. The MWF_{2+0} and the ADM were also tested for one ear only with a monaural presentation of the stimuli (for the full list of conditions, see Table I). In the monaural evaluation, signals were presented to the right ear of the subjects. In both the binaural and the monaural presentation, an unprocessed condition was used as a reference, bringing the total of tested conditions to 11. The speech enhancement achieved by each algorithm was calculated by subtracting the SRT score (in dB SNR) of the algorithm from the unprocessed SRT score, i.e.,

$$\Delta\mathrm{SRT}_{\mathrm{algo}} = \mathrm{SRT}_{\mathrm{unproc}} - \mathrm{SRT}_{\mathrm{algo}}. \qquad (6)$$

Tests were performed in a double walled sound booth under headphones (TDH-39) using an RME Hamerfall Multiface II soundcard and a Tucker Davis HB7 headphone driver. The perceptual evaluations were carried out using the impulse responses of the acoustical environment with T_{60} =0.61 s, i.e., a realistic living room condition. Because of practical considerations, three spatial scenarios, selected from the list of scenarios tested in the objective evaluation, were perceptually evaluated, i.e., S_0N_{60}, $S_{90}N_{270}$ and a triple noise source condition $S_0N_{90/180/270}$.

IV. RESULTS AND ANALYSIS

A. Objective evaluation

First, the noise reduction performance of the MWF is discussed and compared with the ADM. Second, the MWF-N is evaluated.

1. MWF

Figure 1 shows the measured speech intelligibility weighted gain in SNR, $\Delta\mathrm{SNR}_{\mathrm{SI}}$, for a target speech source arriving from 0° and a single noise source arriving from x° (S_0N_x) for the ADM and the three different MWF algorithms. This is done for a room with a low (T_{60}=0.21 s) and a living room (T_{60}=0.61 s) reverberation time, respectively. The data are given only for the right hearing aid as, for a single noise source scenario, the directivities of the left and the right hearing aid are almost identical (if one changes positive angles into negative angles). The noise reduction data of more challenging scenarios, with multiple noise sources or a nonzero speech source angle, are shown in Fig. 2.

For both the single noise source data and the more complex spatial scenarios, it was observed that the acoustical parameters have a very large effect on the noise reduction performance of the algorithms. Due to the presence of reflections, the performance of all algorithms decreased significantly, which is a well known effect from literature. In case of a low reverberant condition, gains of up to 23 dB were obtained. In a more realistic environment, this performance

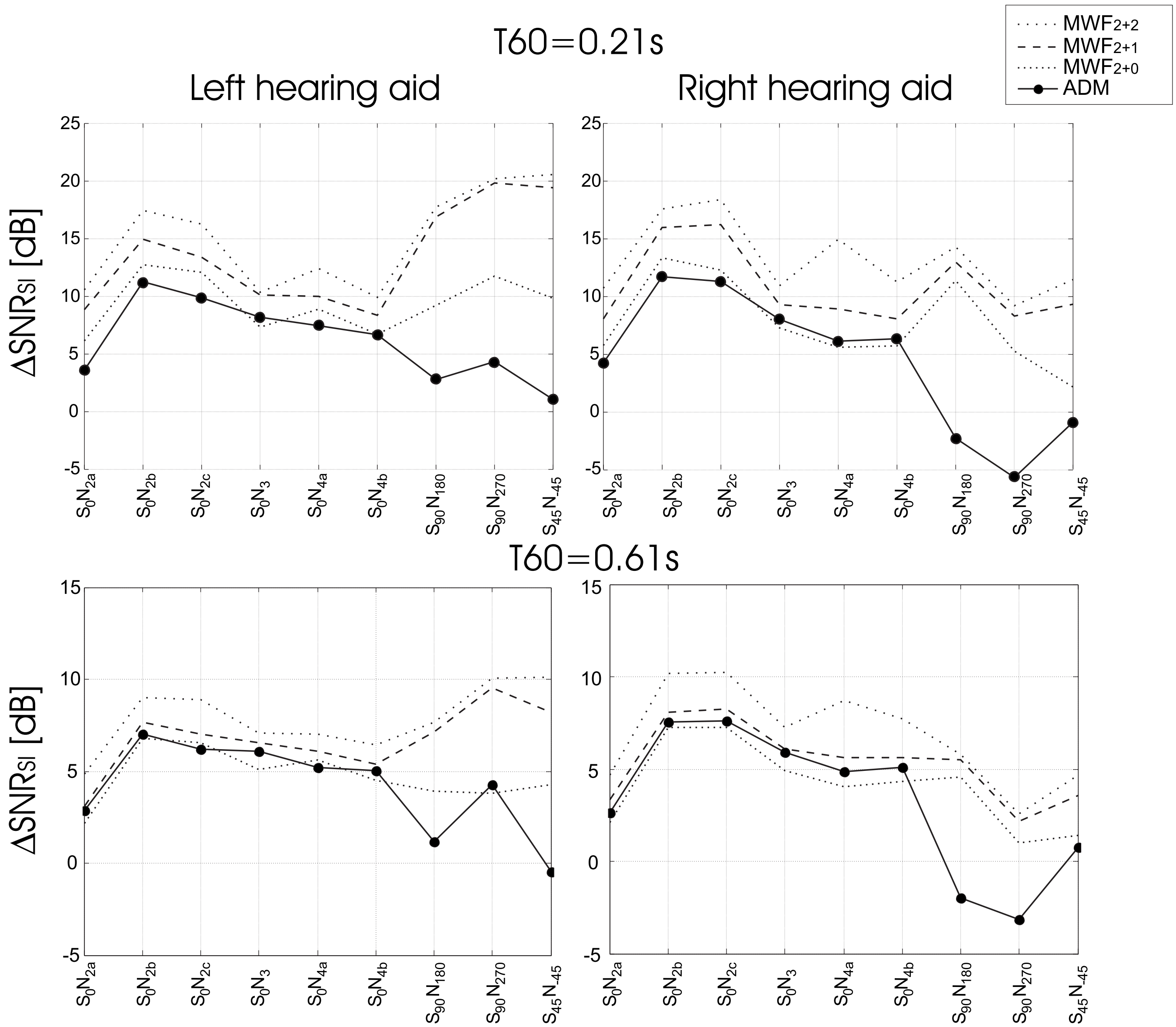

FIG. 2. ΔSNR$_{SI}$ of the MWF with different microphone combinations and the ADM for multiple noise sources. The abbreviations of the spatial scenarios are explained in Table I. Two different acoustical environments are evaluated.

dropped to 12 dB for the same spatial scenario and the same hearing aid, i.e., the scenario S_0N_{120} at the right hearing aid.

In single noise source scenarios (Fig. 1), extending the MWF$_{2+0}$ with contralateral microphone signals substantially increased noise reduction performance, especially if the speech and the noise source were positioned within 60° of each other. In these spatial scenarios, an additional gain of 7.5–14 dB in T_{60}=0.21 s and of 3.1–7.6 dB in T_{60}=0.61 s was obtained for the right hearing aid when going from the MWF$_{2+0}$ to the MWF$_{2+2}$. In the other single noise source scenarios, the benefit was much more modest. An average difference (and standard deviation) between the MWF$_{2+0}$ and, respectively, the MWF$_{2+1}$ and the MWF$_{2+2}$ of 1.4 ± 0.7 and 3.3 ± 1.0 dB for T_{60}=0.21 s and of 0.8 ± 0.3 and 2.2 ± 0.3 dB for T_{60}=0.61 s was measured over these spatial scenarios. Interestingly the MWF$_{2+0}$ outperformed the ADM in low reverberant conditions. However, in a realistic environment both bilateral algorithms had a similar performance.

For the multiple noise source scenarios, as shown in Fig. 2, the same trends were observed, with the MWF$_{2+2}$ outperforming the MWF$_{2+1}$, which in turn performed better than the MWF$_{2+0}$ and ADM. For both acoustic environments, both two-microphone algorithms, i.e., the ADM and the MWF$_{2+0}$, tend to have a similar performance. However, for the spatial scenarios with the target signal not arriving from 0°, all MWF-based algorithms easily outperformed the ADM. In these scenarios, the ADM only showed very small improvements or even a decrease in ΔSNR$_{SI}$ (up to −5 and −2.5 dB for T_{60}=0.21 s and T_{60}=0.61 s, respectively). For the more complex spatial scenarios shown in Fig. 2, it is observed that the gain in noise reduction achieved by extending the MWF$_{2+0}$ with contralateral microphone signals was highly dependent on the spatial scenario and the ear of interest. For instance, a large gain in ΔSNR$_{SI}$ for the left hearing aid is observed in $S_{90}N_{270}$, while a more modest gain is present at the right hearing aid. For the right hearing aid, a large gain is observed for, e.g., condition S_0N_{4a}, while a more modest gain is observed in, e.g., condition S_0N_3.

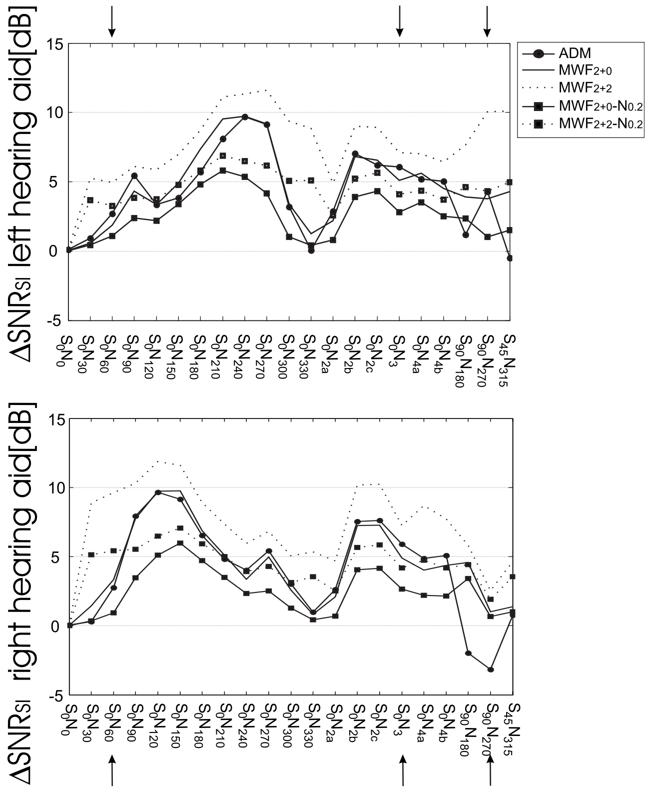

FIG. 3. The influence of η=0.2 on $\Delta\mathrm{SNR}_{\mathrm{SI}}$ of the MWF, the MWF-$\mathrm{N}_{0.2}$, and the ADM for T_{60}=0.61 s. A four- and two-microphone MWF-based system have been tested. The abbreviations of the spatial scenarios are explained in Table I. The arrows highlight the spatial scenarios that have been evaluated perceptually.

2. MWF-N

As discussed in the Introduction, the MWF-N enables the user to correctly localize the speech and the noise component when used in a binaural hearing aid configuration. This is in contrast with other signal processing schemes for hearing aids, e.g., the ADM and partly (only for the noise component) the MWF (Van den Bogaert *et al.*, 2008). The parameter η controls the amount of noise that remains unprocessed by the algorithm.

Figure 3 illustrates the influence of the parameter η =0.2 on the estimated noise reduction performance of the MWF_{2+2} and the MWF_{2+0}. The performance of the ADM is also shown as a reference noise reduction system. This figure illustrates that when adding a partial noise estimate to the MWF algorithm (MWF-Nη), the loss in noise reduction is not only dependent on the parameter η, but also on the amount of noise reduction originally obtained by the MWF. Larger losses are observed if a high noise reduction performance was already obtained by the MWF algorithm. As a consequence, the influence of the parameter η is more pronounced on the MWF_{2+2} than on the MWF_{2+0} algorithm. The figure shows that when using η=0.2, the estimated noise reduction performance of the MWF_{2+2}-$\mathrm{N}_{0.2}$ drops, in most conditions, below the performance of the ADM and the MWF_{2+0}. Other simulations have shown that when using η =0.1, the MWF-N still outperforms the ADM. If the speech source is located outside the forward field of view, all MWF- and MWF-N-based algorithms outperform the ADM.

B. Perceptual evaluation

To further validate the performance of the MWF and MWF-N, a number of perceptual evaluations were performed. Three spatial scenarios were selected (see Table I or the arrows in Fig. 3). Table II shows the improvement in SRT relative to an unprocessed condition averaged over ten normal hearing subjects obtained when using the different algorithms. The bottom two rows show the SRT levels of the unprocessed reference condition. The gains in $\Delta\mathrm{SNR}_{\mathrm{SI}}$ measured during the objective evaluation were added for both the left and the right hearing aid. All statistical analyses were done using SPSS 15.0. For conciseness, the term "factorial repeated measures analysis of variance (ANOVA)" is abbre-

TABLE II. The gain in SRT, $\Delta\mathrm{SRT}_{\mathrm{algo}}$, averaged over ten normal hearing subjects. The bottom rows show the SNRs at which the unprocessed reference SRTs have been measured for the monaural and the binaural presentations. A "*" depicts a significant noise reduction performance ($p<0.05$) compared to the unprocessed condition. $\Delta\mathrm{SNR}_{\mathrm{SI}}$, calculated for the left and right hearing aids in the objective evaluation, is also added to the table.

	S_0N_{60}			$S_{90}N_{270}$			$S_0N_{90/180/270}$		
Bilat/bin ΔSRT (dB)	Perceptual	Left	Right	Perceptual	Left	Right	Perceptual	Left	Right
ADM	2.1 ± 1.9	2.7	2.8	−4.3 ± 1.3*	4.3	−3.2	1.3 ± 1.4	6.0	5.9
MWF_{2+2}	4.3 ± 1.5*	4.9	9.6	0.7 ± 1.4	10.0	2.5	4.6 ± 0.8*	7.1	7.2
MWF_{2+1}	3.8 ± 1.6*	4.0	6.2	0.3 ± 2.0	9.6	2.1	4.0 ± 1.5*	6.6	6.0
MWF_{2+0}	1.0 ± 0.7*	1.9	3.3	−1.2 ± 1.6	3.8	1.0	2.8 ± 1.3*	5.1	4.9
MWF_{2+2}-$\mathrm{N}_{0.2}$	3.6 ± 1.4*	3.3	5.4	2.0 ± 1.4*	4.3	1.9	3.2 ± 0.8*	4.1	4.2
MWF_{2+1}-$\mathrm{N}_{0.2}$	2.7 ± 1.3*	2.6	3.0	1.5 ± 1.6	3.9	1.6	3.4 ± 0.8*	3.7	3.3
MWF_{2+0}-$\mathrm{N}_{0.2}$	1.0 ± 2.1	1.1	0.9	0.0 ± 1.5	1.0	0.7	2.3 ± 1.4*	2.8	2.6
Monaural ΔSRT (dB)									
ADM	5.4 ± 2.0*		2.8	−5.4 ± 1.2*		−3.2	3.4 ± 2.3*		5.9
MWF_{2+0}	3.4 ± 1.3*		3.3	−0.7 ± 1.4		1.0	5.0 ± 1.6*		4.9
SNR-unproc (dB)									
Binaural	−6.2 ± 1.8			−9.1 ± 1.7			−7.2 ± 1.6		
Monaural	2.8 ± 2.0			−8.0 ± 1.7			−3.0 ± 2.1		

viated as ANOVA, and pairwise comparisons discussed throughout the document were always Bonferroni corrected for multiple comparisons. The reported p-values of the pairwise comparisons are lower bound values. A p-value of $p = 0.05$ was used as a threshold for significance.

1. Bilateral/binaural presentation

To compare the different algorithms, an ANOVA is carried out on the SRT data. These data were also used to calculate the average gains shown in Table II [see Eq. (6)]. The ANOVA was carried out using the factor algorithm (seven algorithms and an unprocessed condition) and spatial scenario (three spatial scenarios). An interaction was found between both factors ($p = 0.005$). This was expected since the performance of the algorithms was clearly dependent on the location of the speech and the noise source(s). Therefore an ANOVA and pairwise comparisons were carried out for each spatial scenario. For all three spatial scenarios, a main effect for the factor algorithm was found ($p = 0.002$, $p < 0.001$, and $p < 0.001$ for respectively, S_0N_{60}, $S_{90}N_{270}$, and $S_0N_{90/180/270}$).

First, an overview is given of the comparisons made between the algorithms and the unprocessed condition. An "*" was added in Table II if the algorithm generated a significant gain in SRT compared to the unprocessed condition. For the scenario S_0N_{60}, a significant gain in noise reduction was achieved by all algorithms except for the ADM ($p = 0.155$) and the MWF_{2+0}-$\mathrm{N}_{0.2}$ ($p = 1.000$). The highest significant gain was obtained by the MWF_{2+2} algorithm (4.3 dB, $p < 0.001$). The lowest significant gain was obtained when using the MWF_{2+0} (1.0 dB, $p = 0.036$). For the scenario $S_{90}N_{270}$, a significant gain was achieved only by the MWF_{2+2}-$\mathrm{N}_{0.2}$ algorithm (2.0 dB, $p = 0.047$). When using the ADM, a significant decrease in speech understanding was observed (−4.3 dB, $p < 0.001$). For the triple noise source scenario, all MWF algorithms showed a significant gain in speech understanding ranging from 2.3 dB for the MWF_{2+0}-$\mathrm{N}_{0.2}$ ($p = 0.019$) to 4.6 dB for the MWF_{2+2} ($p < 0.001$). The ADM showed no significant improvement compared to the unprocessed condition ($p = 0.435$).

Second, an overview is given of the pairwise comparisons between the ADM and all MWF and MWF-N approaches. For the spatial scenario S_0N_{60}, only the MWF_{2+2} showed a significant gain in speech enhancement compared to the ADM (2.2 dB, $p = 0.013$); the MWF_{2+1} showed a nonsignificant gain of 1.6 dB ($p = 0.061$). The performance of the MWF_{2+0} showed no significant difference with the ADM (which is also a two microphone algorithm). For the scenario $S_{90}N_{270}$ all MWF and MWF-N algorithms showed a clear significant benefit (all p-values $p \leqslant 0.001$) compared to the ADM. This benefit is in the range of 3.1 dB for the MWF_{2+0} to 6.3 dB for the MWF_{2+2}-$\mathrm{N}_{0.2}$. For the triple noise source scenario, a significant benefit is found for the MWF_{2+2} (3.3 dB, $p < 0.001$), the MWF_{2+1} (2.7 dB, $p < 0.001$), and the MWF_{2+1}-$\mathrm{N}_{0.2}$ (2.1 dB, $p = 0.002$). Since the MWF_{2+1}-$\mathrm{N}_{0.2}$ showed a significant gain compared to the ADM, it was expected that also the MWF_{2+2}-$\mathrm{N}_{0.2}$, which has an extra microphone input, would show this benefit. However, no statistically significant difference is found between this algorithm and the ADM (1.9 dB, $p = 0.164$).

Third, the influence of adding contralateral microphones to the original two-microphone MWF-scheme (MWF_{2+0}) can be observed. For S_0N_{60} both the MWF_{2+1} and MWF_{2+2} showed a significant increase in performance of, respectively, 2.8 dB ($p = 0.022$) and 3.3 dB ($p = 0.001$) compared to the MWF_{2+0}. The MWF_{2+2} and MWF_{2+1} were statistically not significantly different. For the MWF-$\mathrm{N}_{0.2}$ algorithms the same trends were observed, but these differences were not statistically significant [MWF_{2+1}-$\mathrm{N}_{0.2}$ and MWF_{2+2}-$\mathrm{N}_{0.2}$ show an average improvement of, respectively, 1.7 dB ($p = 0.341$) and 2.5 dB ($p = 0.125$) compared to the MWF_{2+0}-$\mathrm{N}_{0.2}$]. For the spatial scenario $S_{90}N_{270}$, the same observations are made, with MWF_{2+1} and MWF_{2+2} performing statistically better than MWF_{2+0} (respectively, 1.5 dB, $p = 0.033$ and 1.8 dB, $p = 0.001$) and with no significant difference between MWF_{2+2} and MWF_{2+1}. Again both the MWF_{2+1}-$\mathrm{N}_{0.2}$ and MWF_{2+2}-$\mathrm{N}_{0.2}$ show the same nonsignificant trend compared to the MWF_{2+0}-$\mathrm{N}_{0.2}$ (with, respectively, a gain of 1.5 dB, $p = 0.454$ and 1.9 dB, $p = 0.215$). For the triple noise source scenario, only the MWF_{2+2} performed significantly better than the MWF_{2+0} (1.7 dB, $p = 0.004$). Again both the MWF_{2+1}-$\mathrm{N}_{0.2}$ and MWF_{2+2}-$\mathrm{N}_{0.2}$ show a nonsignificant improvement compared to the MWF_{2+0}-$\mathrm{N}_{0.2}$.

Finally the last comparisons examine the impact of introducing the partial noise estimate using $\eta = 0.2$ to the original MWF algorithm (MWF versus MWF-$\mathrm{N}_{0.2}$). In the three different ANOVAs, one for each spatial scenario, only one significant difference was found when comparing the performance of the MWF_{2+M_C} with the MWF_{2+M_C}-$\mathrm{N}_{0.2}$, with M_C ranging from 0 to 2. A significant decrease in performance of −1.4 dB is observed ($p = 0.016$) when comparing the MWF_{2+2}-$\mathrm{N}_{0.2}$ with the MWF_{2+2} in the triple noise source scenario. Some other nonsignificant trends were also observed. In the triple noise source scenario and in scenario S_0N_{60}, the MWF_{2+M_C}-$\mathrm{N}_{0.2}$ tends to have a decreased performance compared to the MWF_{2+M_C} condition, which was expected since the parameter $\eta = 0.2$ introduces an unprocessed noise component at the output of the noise reduction algorithm. Interestingly this trend is not observed in the scenario $S_{90}N_{270}$. In this scenario the MWF-N algorithms typically outperformed the MWF algorithms.

These trends were verified by a different ANOVA. In this refined analysis, the factor algorithm (three different MWF algorithms: MWF_{2+0}, MWF_{2+1}, and MWF_{2+2}) and eta ($\eta = 0$ and $\eta = 0.2$) were used per spatial condition. For all three ANOVAs, no interactions were found between both factors. For the scenario S_0N_{60}, no significant effect is observed. For the condition $S_{90}N_{270}$, a significant increase in performance of 1.3 dB ($p = 0.002$) is observed when comparing $\mathrm{MWF}_{2+M_C} - \mathrm{N}_{0.2}$ with MWF_{2+M_C}. For the triple noise source scenario, a significant decrease in performance of 0.8 dB ($p = 0.001$) is observed when introducing $\eta = 0.2$. In all three of these ANOVAs, a significant increase in performance is found when introducing one or two contralateral microphones, but no significant difference is observed between the three and four-microphone algorithms, confirming the observations made in the paragraph on contralateral mirophones.

2. Monaural presentation

The monaural SRT data, used to calculate the gains shown in Table II, were used in an ANOVA. Again the factor algorithm (two algorithms and an unprocessed condition) and spatial scenario were used. Similar to the analysis of the bilateral/binaural presentation, an interaction is found between both factors ($p<0.001$). This leads to a separate ANOVA and separate pairwise comparisons for each spatial scenario.

In the scenario S_0N_{60}, both algorithms perform significantly better than the unprocessed condition with an average gain of 3.4 dB by the MWF_{2+0}, $p<0.001$ and an average gain of 5.4 dB by the ADM, $p<0.001$. Both algorithms are significantly different from each other, with the performance of the ADM being 2.0 dB better than the MWF_{2+0} ($p=0.007$). For the scenario $S_{90}N_{270}$, the MWF_{2+0} is not significantly different from the unprocessed condition. The ADM shows a significant decrease in performance compared to both the MWF_{2+0} and the unprocessed condition (respectively, 5.4 and 4.7 dB, both $p<0.001$). In the triple noise source scenario, both the MWF_{2+0} and the ADM show a significant improvement compared to the unprocessed condition (respectively, 5.0 dB, $p<0.001$ and 3.4 dB, $p=0.004$).

3. Comparison with the objective data

In Table II, the noise reduction gains ($\Delta\mathrm{SNR}_{\mathrm{SI}}$) calculated during the objective evaluations are shown together with the speech enhancement data of the perceptual evaluations. Large correlations are present between the data of both evaluations. In the bilateral/binaural configuration, perceptual results correlated best with $\Delta\mathrm{SNR}_{\mathrm{SI}}$ of the hearing aid that had the best input SNR (e.g., the left ear for S_0N_{60}, the right ear for $S_{90}N_{270}$, and both ears for $S_0N_{90/180/270}$). It was observed that this hearing aid is typically the device with the lowest gain in noise reduction, $\Delta\mathrm{SNR}_{\mathrm{SI}}$. Although large correlations between both performance measures were observed, Table II illustrates that the performance of the ADM and the MWF seems to be overestimated by approximately 2 dB in $S_{90}N_{270}$ and the triple noise source scenario.

V. DISCUSSION

This paper evaluates two recently introduced MWF-based noise reduction algorithms for multimicrophone hearing aids, which offer the ability to preserve the spatial awareness of hearing aid users. A verification of the speech enhancement and the noise reduction performance of the algorithms is presented in this study. A bilateral ADM was used as a reference noise reduction algorithm as this is commonly implemented in current bilateral hearing aids. Three research questions on combining noise reduction with preserving sound source localization in multimicrophone noise reduction algorithms were raised in the Introduction. The results and analysis from the previous sections will be used to answer these questions.

A. Noise reduction performance of the MWF

In Sec. IV A the performance of the MWF was evaluated objectively in two different acoustical environments, i.e., $T_{60}=0.21$ s and $T_{60}=0.61$ s. In the low reverberant condition, the two-microphone MWF, i.e., the MWF_{2+0}, outperformed the ADM, especially in single noise source scenarios (Fig. 1) and in conditions in which the target signal was not arriving from the forward field of view (the three rightmost data-points of Fig. 2). The performance of all the adaptive algorithms dropped significantly in a more realistic acoustic environment. This phenomenon is well known and commonly found in literature [e.g., see Kompis and Dillier (2001) and Greenberg and Zurek (1992)]. In this more realistic acoustic environment, the MWF_{2+0} outperformed the ADM only if the speech source is not arriving from the forward field of view. In all other spatial scenarios, both two-microphone algorithms had approximately the same performance. The perceptual evaluation, also carried out with $T_{60}=0.61$ s, supported these conclusions. When using a bilateral configuration that consists of two independent monaural systems, no significant differences were apparent between the ADM and the MWF_{2+0} if the speech source arrives from 0° (Sec. IV B). Still, unlike the ADM, the MWF preserves the binaural cues of the speech component independent of the angle of arrival of the signal (Doclo *et al.*, 2006; Van den Bogaert *et al.*, 2008).

Why the ADM caught up with the performance of the MWF in more reverberant conditions can be explained by the MWF, unlike an ADM, not performing any dereverberation. The MWF is designed to estimate the speech component, X, present at a reference microphone, which is the convolution of the target signal S with the room impulse response. Hence, no dereverberation is performed. The ADM, on the other hand, is designed to preserve signals arriving from the frontal hemisphere. In other words, reflections arriving from the back hemisphere are reduced in amplitude. However, this also implies that the ADM will reduce speech perception if the target signal arrives from the side or the back of the head. Therefore the ADM was significantly outperformed by the MWF in these spatial scenarios. This was also validated by the perceptual evaluation in which all MWF-based algorithms outperformed the ADM in the condition $S_{90}N_{270}$.

The two bottom rows of Table II show the SNRs at which the unprocessed reference SRTs were measured. It is observed that if a bilateral/binaural configuration was used, subjects always benefited from the best ear advantage. This means that if both ears are available, one of the ears has a better SNR than the other ear due to the headshadow effect and the positioning of the sound sources. This enables the human auditory system to focus on the ear with the best SNR. In condition S_0N_{60}, the noise source was close to the right ear, i.e., the ear used in the monaural evaluation. Therefore, the SRT level was much higher in the monaural presentation compared to the binaural presentation. Overall, it is observed that a binaural presentation, i.e., accessing the signals from both ears, always resulted in lower SRT values compared to the monaural presentations. This has motivated the standard use of bilateral hearing aids in case of a bilateral hearing deficit (Libby, 2007).

During this study a perfect VAD was used to demonstrate the potential of the noise reduction performance of the MWF-based algorithms. It is clear that VAD performance will have an impact on the noise reduction performance of the algorithms. Simulations of Doclo *et al.* (2007) with a monaural spatially preprocessed MWF show that no large degradations (<1 dB) in performance should be expected when using an energy-based VAD at input SNRs higher than -2 dB. In the work of Wouters *et al.* (2008), hearing impaired subjects were evaluated with an adaptive version of this algorithm using a monaural energy-based VAD. Also in their experiments, a clear and robust gain in speech perception of several decibels was observed in multisource setups. Binaural algorithms also offer the possibility of integrating contralateral information into the VAD, which could lead to an improved VAD performance.

B. Adding contralateral microphones

Adding contralateral microphone signals to the ipsilateral hearing aid clearly comes at the cost of transmitting and processing those signals. To evaluate this trade-off, different microphone combinations were evaluated.

The objective evaluations showed that in single noise source scenarios with speech arriving from 0°, adding contralateral microphones introduced a large gain in noise reduction performance if the speech and noise sources were relatively close to each other (Fig. 1), i.e., within 60°. In other words the directional pattern generated by the MWF became more narrow when more microphones were used. This effect is well known in sensor array processing. Typically a large impact is obtained if additional sensors, in our case the contralateral front microphone, are placed sufficiently far away from the original sensors, thereby enhancing the size of the array. Extreme examples of this phenomenon are, in the specific case of hearing aids, often referred to as tunnel-hearing (Stadler and Rabinowitz, 1993). Soede *et al.* (1993) proved that very narrow beams in the horizontal hemisphere could be created when using several (4–17) microphones positioned on eyeglasses. If the speech and the single noise source were more spatially separated, adding more microphones did not result in large improvements in noise reduction performance (Fig. 1). This is due to the fact that in single noise source scenarios, the MWF only has to create a single null pointed toward the location of the noise source. As a consequence, adding more degrees of freedom, i.e., more microphones, to a two-microphone system does not significantly improve noise reduction performance.

Significant gains in noise reduction performance were also obtained during the objective evaluations for some asymmetric single noise source scenarios. In these scenarios, i.e., $S_{90}N_{270}$, $S_{45}N_{315}$, and $S_{90}N_{180}$, a significant improvement in performance were observed at the ear with the worst input SNR, i.e., the left ear. This is due to the asymmetrical setting of the speech source. Since the microphone inputs of the left hearing aid had a low input SNR, due to the headshadow effect, the noise reduction algorithm on this hearing aid produced a nonoptimal estimate of the speech component. However, if a contralateral microphone signal, which has a higher SNR, was added to the system, a better estimate of the speech component could be generated and noise reduction performance increased. One may interpret this as introducing the best ear advantage, used by our own auditory system, into the noise reduction algorithm.

One should be aware that this increased performance at the hearing aid with the worst SNR may be limited in daily life. The human auditory system focuses on the ear with the best SNR to listen to speech. The hearing aid with the large gain in SNR, obtained at the ear with the worst input SNR, will typically produce a similar output SNR as the hearing aid on the other side of the head. Therefore, perceptual SRT measurements with a bilateral/binaural hearing aid configuration will not show the large predicted gain in SNR. This was confirmed when comparing the objective and the perceptual data. It was shown that the actual gain in SRT correlates best with the predicted ΔSNR_{SI} performance obtained at the ear with the best input SNR. The more spectacular improvements found during the simulations, obtained at the ear with the worst input SNR, were not realistic predictions of the SRT gains. This illustrates that input as well as output SNRs or the best ear advantage should be taken into account when interpreting measurements of noise reduction gains for binaural or bilateral hearing aid configurations.

For the multiple noise source scenarios (Fig. 2), objective evaluations demonstrated that adding more microphones or more degrees of freedom does result in a significant gain in noise reduction. For the very asymmetrical condition, i.e., $S_0N_{60/120/180/210}$ (S_0N_{4a}), it was again observed that a larger benefit was obtained at the ear with the worst input SNR, i.e., the right ear.

In the perceptual evaluations, it was observed that in the scenarios S_0N_{60} and $S_{90}N_{270}$ the MWF_{2+1} and MWF_{2+2} outperformed the MWF_{2+0}. These observations confirm the objective evaluation, discussed earlier. In the triple noise source scenario only the MWF_{2+2} significantly outperformed the MWF_{2+0}, which can be explained by taking into account the degrees of freedom needed to reduce three noise sources. The grouped analysis of the perceptual data of the MWF and MWF-N showed that, in general, a three-microphone system, consisting of two ipsilateral and one contralateral microphone outperformed the two-microphone system. Adding a fourth microphone did not, in general, add a significant improvement over the three-microphone system. Intuitively this can be explained by the fact that adding a third microphone placed at the other side of the head will introduce a significant amount of "new information" to the noise reduction system. The fourth microphone will increase the degrees of freedom of the system, but its impact will be much smaller since it is located very close to the third microphone.

C. Noise reduction performance of the MWF-N

Van den Bogaert *et al.* (2008) showed that adding a partial noise estimate with $\eta=0.2$ to the MWF algorithm not only preserves the capability to localize the targeted speech component but also restores the capability to localize the noise component. This is important for hearing aid users in terms of spatial awareness and release from masking. However, this clearly comes at the cost of some noise reduction. Figure 3 demonstrates the influence of the parameter $\eta=0.2$ on $\Delta\mathrm{SNR}_{\mathrm{SI}}$ of the MWF_{2+0} and MWF_{2+2} in an environment with a realistic reverberation. It showed that the loss in noise reduction due to the partial noise estimate was dependent on its original noise reduction performance. This can be explained by using the relation between the output of the MWF and MWF-N. The output of the MWF-N [Eq. (1)] can be written as the sum of a scaled proportion of the input signal added to the output of the MWF (Van den Bogaert *et al.*, 2008), i.e.,

$$Z_{\mathrm{MWF\text{-}N}_{\eta}L}(\eta) = \eta Y_{L,1} + (1-\eta) Z_{\mathrm{MWF},L}, \tag{7}$$

$$Z_{\mathrm{MWF\text{-}N}_{\eta}R}(\eta) = \eta Y_{R,1} + (1-\eta) Z_{\mathrm{MWF},R}. \tag{8}$$

It was also observed that when adding a partial noise estimate with $\eta=0.2$, the predicted performance, $\Delta\mathrm{SNR}_{\mathrm{SI}}$, could drop below the performance of the ADM for some spatial scenarios (Fig. 3). This may be interpreted as a cost to sufficiently preserve the binaural cues of the speech and the noise component. However, during the perceptual evaluations, no significant difference was found between the ADM and the MWF_{2+0}-$\mathrm{N}_{0.2}$ in scenarios S_0N_{60} and $S_0N_{90/180/270}$. Moreover, the ADM showed a significant loss in performance compared to all MWF-$\mathrm{N}_{0.2}$ algorithms in the scenario $S_{90}N_{270}$ for reasons already discussed in the previous section. The lack of a significant SRT difference ($\Delta\mathrm{SRT}_{\mathrm{algo}}$) between the ADM and the MWF_{2+0}-$\mathrm{N}_{0.2}$, which was in contrast with the objective evaluation, may be explained by spatial release from masking. Since the MWF-$\mathrm{N}_{0.2}$ preserved the localization of both the speech and noise component, a slightly better speech perception in noise compared to the performance predicted by $\Delta\mathrm{SNR}_{\mathrm{SI}}$ could be expected. The same spatial release from masking may also explain why the MWF_{2+M_C}-$\mathrm{N}_{0.2}$ outperformed the MWF_{2+M_C} in the condition with the largest spatial separation between speech and noise sources, i.e., $S_{90}N_{270}$. In this condition, the MWF_{2+M_C}-$\mathrm{N}_{0.2}$ produced a worse $\Delta\mathrm{SNR}_{\mathrm{SI}}$, but since it preserves the user's ability to localize both the speech and the noise component correctly, a significantly better SRT could be obtained. This also illustrates that although $\Delta\mathrm{SNR}_{\mathrm{SI}}$ is a useful tool for predicting noise reduction performance, other factors such as binaural cues should be taken into account when evaluating speech enhancement by noise reduction algorithms in hearing aids.

VI. CONCLUSION

In Van den Bogaert *et al.* (2008), it was shown that MWF-based noise reduction approaches have interesting features in terms of preserving binaural cues and hence spatial awareness for hearing aid users. Unlike other noise reduction approaches, the MWF and MWF-N approaches are capable of using multimicrophone information; they can easily integrate contralateral microphone signals, and they inherently preserve the binaural cues of the speech component, independent of the angle of arrival of the signal. By preserving part of the noise component (MWF-N), the ability to localize both the speech and the noise component can be preserved. This paper presented a thorough evaluation of the noise reduction performance of the MWF and MWF-N algorithms in comparison with an unprocessed condition and an ADM, which is a commonly used noise reduction system in commercial digital hearing aids. This was done by evaluating noise reduction and speech perception performance in different speech-in-multitalker-babble scenarios. Three different research questions have been addressed.

First, it was shown that a two-microphone MWF (MWF_{2+0}) has approximately the same performance as an ADM. It does so while preserving the binaural cues of the speech component. Since the MWF operates independently of the angle of arrival of the signal, it easily outperformed the ADM if the speech signal was not arriving from the forward field of view. Moreover, in these scenarios the ADM may even reduce the speech perception of the hearing aid user compared to the unprocessed condition. This was observed during the perceptual evaluation of both the monaural (−5.4 dB) and the bilateral (−4.3 dB) ADM configuration in the spatial scenario $S_{90}N_{270}$. Large differences were observed when comparing the monaural with the bilateral data. It was observed that a bilateral presentation leads to an improved speech perception in noisy environments due to the best ear benefit. This confirms, although tests were performed with normal hearing subjects, the common practice of using bilateral/binaural hearing aids for a bilateral hearing impaired subject.

Second, different microphone combinations were evaluated. A significant gain in performance was found if one contralateral microphone signal was added to the ipsilateral hearing aid. This shows that transmitting microphone signals can result in a significant gain in noise reduction, especially in multiple noise source scenarios or if the speech and the noise source(s) are placed asymmetrically around the head. Adding a second contralateral microphone signal to the ipsilateral hearing aid did not, in general, show a significant SRT improvement in the perceptual evaluations.

Finally, it was demonstrated that adding a partial noise estimate to the MWF, large enough to sufficiently preserve the binaural cues to restore the directional hearing and spatial awareness (MWF-$\mathrm{N}_{0.2}$), only slightly affects noise reduction performance. Moreover, perceptual evaluations showed that in some conditions ($S_{90}N_{270}$) the MWF-$\mathrm{N}_{0.2}$ could even outperform the MWF, which may be due to improved spatial release from masking. The parameter η controls the amount of noise reduction. Therefore it may also be used as a control mechanism to maximize or to limit the amount of noise reduction if necessary. This can be done adaptively using sound classification algorithms, which are often available in present-day high-end digital hearing aids.

The study also demonstrated that carefully selected objective performance measures can be very useful in predicting the performance of noise reduction algorithms. However, one has to take into account psychophysical properties of the auditory system for a correct interpretation of these objective measures, e.g., the best ear benefit and spatial release from masking effects.

The chosen experimental setup, used to investigate and demonstrate the previously mentioned effects, does not represent all of the many conditions and noise sources encountered by hearing impaired subjects. The effect of head movements, which may interfere with the adaptation of the filters, other noise source scenarios, and a real-time implementation with a realistic, perhaps binaural, multimicrophone VAD were not discussed in this manuscript. Therefore more validation is preferred before implementing these algorithms into hearing aids. However, recent research with hearing impaired subjects indicates that a robust gain in speech perception is found when using a monaural real time MWF algorithm together with an energy-based VAD (Wouters *et al.*, 2008).

In conclusion, it seems that the binaural MWF-based algorithms offer a valid alternative for standard adaptive directional algorithms. Unlike these algorithms, the MWF does not rely on the direction of arrival of the speech signal nor on assumptions of the microphone characteristics of the hearing aids. In this paper, it was shown that the bilateral and the binaural MWF are capable of offering a good noise reduction performance in an environment with realistic acoustical parameters. Since it is often assumed that localization performance is mainly dominated by low-frequency ITD cues, future research may also include the investigation of a frequency dependent parameter η.

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