Performance Comparison of Bilateral and Binaural MVDR-based Noise Reduction Algorithms in the Presence of DOA Estimation Errors

Daniel Marquardt, Simon Doclo

University of Oldenburg, Department of Medical Physics and Acoustics and Cluster of Excellence Hearing4All, Oldenburg, Germany
Email: {daniel.marquardt, simon.doclo}@uni-oldenburg.de
Web: www.sigproc.uni-oldenburg.de

Abstract

With the advent of wireless technology in hearing aids, it is currently possible to not only use the microphones of the left or the right hearing aid (bilateral configuration) but to use the microphones of both hearing aids (binaural configuration) to improve speech intelligibility in noisy environments. On the one hand, since a larger number of microphones is available, a better noise reduction performance can be achieved for the binaural configuration compared to the bilateral configuration. On the other hand, the sensitivity to DOA estimation errors is increased. In this paper, we objectively evaluate the performance of a bilateral and a binaural MVDR beamformer, which is steered towards the direction of the desired speech source using a DOA estimator based on a discriminative classifier. Simulation results show that in general the binaural MVDR beamformer shows a better performance compared to the bilateral MVDR beamformer even for very low input SNRs.

1 Introduction

In the last decades hearing aids have evolved from simple sound amplifiers to modern digital devices with complex functionalities, enabling to significantly improve speech intelligibility for hearing impaired especially in quiet acoustic environments. However, in the presence of background noise and reverberation speech understanding is still a challenging problem for the hearing aid user. Hence, noise reduction algorithms in hearing aids are crucial to improve speech understanding in background noise for hearing impaired persons. With the advent of wireless technology, it is currently possible to not only use the microphones of the left or the right hearing aid separately (bilateral configuration) but to use the microphones of both hearing aids simultaneously (binaural configuration) in order to achieve noise reduction [1–8].

For bilateral hearing aids (cf. Figure 1a), differential microphones (DM) [9] are widely used due to their simplicity and robust noise reduction performance if the speech source is located in the frontal hemisphere [10]. An extension of the DM is the adaptive differential microphone (ADM) [11], which consists of two directional microphones with a forward-facing and a backward-facing directivity pattern. These two directional microphones are then combined in an adaptive stage aiming to preserve the signals coming from the frontal direction and steering a spatial null into the direction of the strongest interferer in the rear hemisphere. Another frequently used algorithm in bilateral hearing aids is the fixed Minimum Variance Distortionless Response (MVDR) beamformer, also known as superdirective beamformer [12, 13], which, similarly as the DM, assumes the desired speech source to be located in the frontal direction. Hence, in the case of a diffuse noise field the bilateral MVDR beamformer shows a very similar performance as the DM. On the on hand, since the number of microphones available for noise reduction in bilateral hearing aids is rather low, the noise reduction performance is limited especially in diffuse noise fields. On the other hand, the noise reduction performance is quite robust if the desired speech source is not located in the frontal direction, since the directivity index is rather low.

[1]This work was supported in part by the joint Lower Saxony-Israeli Project ATHENA financially supported by the State of Lower Saxony and the Cluster of Excellence 1077 "Hearing4All", funded by the German Research Foundation (DFG).

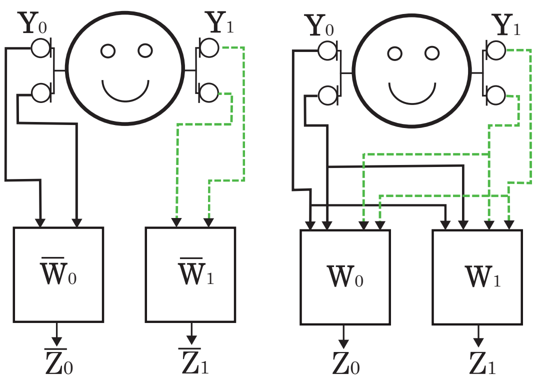
For binaural hearing aids (cf. Figure 1b), using a binaural MVDR beamformer, which exploits the microphones signals from both the left and the right hearing aid, is a straightforward extension [14]. On the one hand, since a larger number of microphones is available, a better noise reduction performance can be achieved compared to the bilateral MVDR beamformer. On the other hand, due to the increased directivity, the noise reduction performance may significantly decrease if the desired speech source is not located at the steering direction [4, 15, 16]. Hence, while for a bilateral configuration a fixed bilateral MVDR beamformer steered towards the frontal direction is a reasonable choice, for the binaural configuration a steerable binaural MVDR beamformer should be considered, which however requires an estimate of the DOA of the desired speech source.

The aim of this paper is to evaluate the noise reduction performance of bilateral and binaural MVDR beamformers for two reverberant, diffuse noise scenarios. In order to estimate the DOA of the desired speech source, we use a robust binaural DOA estimation algorithm proposed in [17]. The DOA is estimated using a binaural classification-based approach where a set of discriminative support vector machine (SVM) classifiers are used, which were trained using anechoic data. The performance of the steerable binaural MVDR beamformer is compared to a fixed bilateral and a fixed binaural MVDR beamformer, which were steered towards the frontal direction. The simulation results show that the steerable binaural MVDR beamformer in general shows a better noise reduction performance compared to the fixed MVDR beamformers even for very low input SNRs.

2 Configuration and Notation

Consider the bilateral and the binaural hearing aid configurations in Figure 1, consisting of two microphone arrays with M microphones on the left and the right hearing aid. The m-th microphone signal in the left hearing aid $Y_{0,m}(\omega)$ can be written in the frequency-domain as

$$Y_{0,m}(\omega) = X_{0,m}(\omega) + V_{0,m}(\omega),\ m = 1, \ldots, M,$$



(a) bilateral configuration (b) binaural configuration

Figure 1: Bilateral and binaural hearing aid configuration. For the bilateral configuration the microphone signals of the left and the right hearing aid are processed separately and for the binaural configuration the microphone signals of the left and the right hearing aid are processed simultaneously.

ISBN 978-3-8007-4275-2

with $X_{0,m}(\omega)$ and $V_{0,m}(\omega)$ representing the speech and the noise component and ω denoting the normalized radian frequency. The m-th microphone signal in the right hearing aid $Y_{1,m}(\omega)$ is defined similarly. For conciseness we will omit the frequency variable ω in the remainder of the paper. We define the two M-dimensional signal vectors $\mathbf{Y}_0$ and $\mathbf{Y}_1$ as

$$\mathbf{Y}_0 = \left[Y_{0,1}, \ldots, Y_{0,M}\right]^T, \quad \mathbf{Y}_1 = \left[Y_{1,1}, \ldots, Y_{1,M}\right]^T, \tag{1}$$

and the $2M$-dimensional stacked signal vector $\mathbf{Y}$ as

$$\mathbf{Y} = \left[\mathbf{Y}_0^T, \mathbf{Y}_1^T\right]^T, \tag{2}$$

which can be written as $\mathbf{Y} = \mathbf{X} + \mathbf{V}$, where $\mathbf{X}$ and $\mathbf{V}$ are defined similarly as $\mathbf{Y}$. For the bilateral configuration, which uses the microphone signals of the left and the right hearing aid separately, we define the M-dimensional weight vectors $\overline{\mathbf{W}}_0$ and $\overline{\mathbf{W}}_1$, such that the output signal at the left hearing aid $\overline{Z_0}$ is equal to

$$\overline{Z_0} = \overline{\mathbf{W}}_0^H \mathbf{Y}_0 = \overline{\mathbf{W}}_0^H \mathbf{X}_0 + \overline{\mathbf{W}}_0^H \mathbf{V}_0 = \overline{Z}_{\mathrm{x},0} + \overline{Z}_{\mathrm{v},0}, \tag{3}$$

where $\overline{Z}_{\mathrm{x},0}$ represents the output speech component and $\overline{Z}_{\mathrm{v},0}$ represents the output noise component. For the binaural configuration, which uses the microphone signals of the left and the right hearing aid simultaneously, we define the $2M$-dimensional weight vectors $\mathbf{W}_0$ and $\mathbf{W}_1$, such that the output signal at the left hearing aid Z_0 is equal to

$$Z_0 = \mathbf{W}_0^H \mathbf{Y} = \mathbf{W}_0^H \mathbf{X} + \mathbf{W}_0^H \mathbf{V} = Z_{\mathrm{x},0} + Z_{\mathrm{v},0}. \tag{4}$$

The output signals at the right hearing aid $\overline{Z}_1$ and Z_1 can be defined similarly as $\overline{Z}_0$ in (3) and Z_0 in (4).
For the special case of a single desired speech source S, the received speech component can be written as $\mathbf{X} = S\mathbf{A}$, with $\mathbf{A}$ the acoustic transfer function (ATF) vector between the speech source and the microphones. Furthermore, in the case of a diffuse noise field, the noise correlation matrix can be written as

$$\mathbf{R}_\mathrm{v} = \mathscr{E}\left\{\mathbf{V}\mathbf{V}^H\right\} = P_\mathrm{n}\boldsymbol{\Gamma}, \tag{5}$$

with $P_\mathrm{n} = \mathscr{E}\left\{|V|^2\right\}$ the power spectral density (PSD) of the noise component in all microphone signals and $\boldsymbol{\Gamma}$ the spatial coherence matrix of the diffuse noise field.

3 MVDR beamformer

The binaural MVDR beamformer [12, 14] aims to minimize the output PSD of the noise component in the left and the right hearing aid while preserving the speech component in the so-called reference microphone signals, which are typically chosen as the frontal microphones. The constrained optimization problem for the left and the right hearing aid can hence be formulated as

$$\min_{\mathbf{W}_0} \mathbf{W}_0^H \mathbf{R}_\mathrm{v} \mathbf{W}_0 \quad \text{subject to} \quad \mathbf{W}_0^H \mathbf{A} = A_{0,1}, \tag{6}$$

$$\min_{\mathbf{W}_1} \mathbf{W}_1^H \mathbf{R}_\mathrm{v} \mathbf{W}_1 \quad \text{subject to} \quad \mathbf{W}_1^H \mathbf{A} = A_{1,1}, \tag{7}$$

with $A_{0,1}$ and $A_{1,1}$ the ATFs between the speech source and the reference microphones of the left and the right hearing aid, respectively. In the case of a diffuse noise field, the solution to the optimization problem in (6) and (7) is equal to [12, 13]

$$\mathbf{W}_0 = \frac{\boldsymbol{\Gamma}^{-1}\mathbf{A}A_{0,1}^*}{\mathbf{A}^H\boldsymbol{\Gamma}^{-1}\mathbf{A}}, \qquad \mathbf{W}_1 = \frac{\boldsymbol{\Gamma}^{-1}\mathbf{A}A_{1,1}^*}{\mathbf{A}^H\boldsymbol{\Gamma}^{-1}\mathbf{A}}. \tag{8}$$

The binaural MVDR beamformer in (8) requires an estimate of the relative transfer function vectors $\mathbf{A}/A_{0,1}$ and $\mathbf{A}/A_{1,1}$, which may become difficult to estimate, particularly when using short filter lengths in noisy and reverberant rooms and if the desired speech source is moving. Alternatively, the anechoic ATF vectors of the binaural hearing aid setup $\mathbf{d}(\theta_\mathrm{s})$, with θ_s denoting the steering angle, can be used, which can be obtained based on anechoic measurements or head models. The filter vectors for the binaural and the bilateral MVDR beamformer are then equal to

$$\mathbf{W}_0 = \frac{\boldsymbol{\Gamma}^{-1}\mathbf{d}(\theta_\mathrm{s})d_{0,1}^*(\theta_\mathrm{s})}{\mathbf{d}^H(\theta_\mathrm{s})\boldsymbol{\Gamma}^{-1}\mathbf{d}(\theta_\mathrm{s})}, \qquad \mathbf{W}_1 = \frac{\boldsymbol{\Gamma}^{-1}\mathbf{d}(\theta_\mathrm{s})d_{1,1}^*(\theta_\mathrm{s})}{\mathbf{d}^H(\theta_\mathrm{s})\boldsymbol{\Gamma}^{-1}\mathbf{d}(\theta_\mathrm{s})}, \tag{9}$$

$$\overline{\mathbf{W}}_0 = \frac{\boldsymbol{\Gamma}^{-1}\mathbf{d}_0(\theta_\mathrm{s})d_{0,1}^*(\theta_\mathrm{s})}{\mathbf{d}_0^H(\theta_\mathrm{s})\boldsymbol{\Gamma}^{-1}\mathbf{d}_0(\theta_\mathrm{s})}, \quad \overline{\mathbf{W}}_1 = \frac{\boldsymbol{\Gamma}^{-1}\mathbf{d}_1(\theta_\mathrm{s})d_{1,1}^*(\theta_\mathrm{s})}{\mathbf{d}_1^H(\theta_\mathrm{s})\boldsymbol{\Gamma}^{-1}\mathbf{d}_1(\theta_\mathrm{s})}, \tag{10}$$

with $d_{0,1}(\theta_\mathrm{s})$ and $d_{1,1}(\theta_\mathrm{s})$ the anechoic ATFs in the reference microphones of the left and the right hearing aid, respectively.
A quite common assumption for noise reduction algorithms in hearing aids is that the desired speech source is located in the frontal direction. Since the noise reduction performance of the bilateral MVDR beamformer is very robust to speech sources coming from different directions in the frontal hemisphere (cf. Section 5), it is reasonable to steer the bilateral MVDR beamformer towards the 0° direction in order to avoid the usage of a DOA estimator which might provide erroneous results, especially in very noisy conditions. This leads to a quite robust performance in terms of noise reduction for different speech source positions in the frontal hemisphere but also to a limited amount of noise reduction. Contrary, for a binaural MVDR beamformer a DOA estimator is definitely required since the robustness to different positions of the desired speech source in the frontal hemisphere is strongly decreased for the ipsilateral hearing aid (cf. Section 5). In order to obtain a reliable estimate of the DOA of the desired speech source in reverberant and noisy conditions, in the next section we briefly review a classification-based binaural DOA estimator, which we have used for the experiments in Section 5.

4 DOA estimation

For estimating the DOA in binaural hearing aids several approaches have been proposed, e.g. based on computational auditory scene analysis [18], using head-models [19] or measured anechoic ATFs [20]. We use a binaural classification-based approach proposed in [17], where a set of discriminative support vector machine (SVM) classifiers is used, which are trained to distinguish between the presence of the desired speech source for a certain direction and the absence of the desired speech source for all other directions. Hence, for each considered angle θ a SVM classifier is trained. The obtained decision value for each SVM classifier is mapped to a source presence probability for the given direction using a generalized linear model. As feature vectors instantaneous, i.e. non-smoothed, short-term generalized cross-correlation functions with phase transform (GCC-PHAT) [21] have been used, since they have been shown to be relatively robust against noise and reverberation [22]. The direction-dependent SVM models were trained using binaural Behind-The-Ear impulse responses (BTE-IRs) downsampled to a sampling frequency of 16 kHz, which were measured in an anechoic environment [23]. Each BTE hearing aid was equipped with 2 microphones with a distance of about 7 mm and was mounted on an artificial head. The BTE-IRs were measured in an anechoic environment (angles ranging from $-180°$ to $180°$ in steps of 5°, with the source at 3 m from the artificial head). The speech material was taken from the TIMIT training data set and a diffuse noise field was generated by convolving a speech-shaped noise signal with the BTE-IRs from all directions and subsequent superposition of all resulting signals. The diffuse noise signal has been added to the speech signals at an SNR of -20 dB to 20 dB in steps of 10 dB in order to take the impact of the diffuse noise on the GCC-PHAT features into account. The GCC-PHAT features are calculated using segment lengths of 10 ms with an overlap of 5 ms. Since instantaneous GCC-PHAT features were used, the source presence probabilities are smoothed across time using recursive averaging with a time constant of 160 ms. The estimated DOA is then determined by selecting the direction with the highest source probability for angles between $-90°$ and 90°. Figure 2

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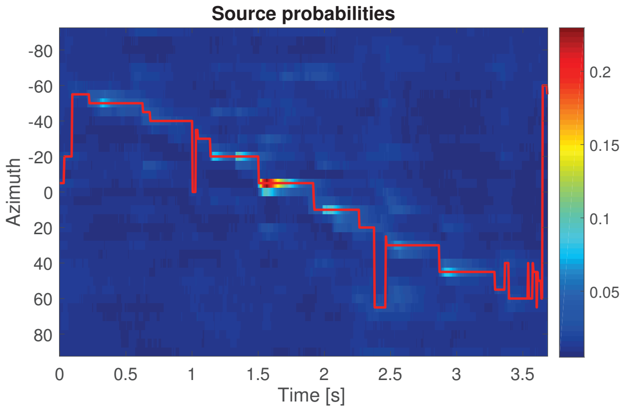

Figure 2: Source presence probabilities for a speech source in a reverberant office environment moving from $-60°$ to $60°$ in a diffuse noise field at an average input iSNR of -5 dB.

exemplary depicts the source presence probabilities for a source moving from $-60°$ to $60°$ in a reverberant office environment and an average intelligibility-weighted SNR (iSNR) of -5 dB.

5 EXPERIMENTAL RESULTS

In this section, we first compare the performance of a fixed bilateral and binaural MVDR beamformer by calculating the SNR improvement for several speech source position in a diffuse noise field. Secondly, we present simulation results for a speech source in a cafeteria and an office scenario in order to evaluate the performance of the fixed binaural and bilateral MVDR beamformers and the steerable MVDR beamformer in realistic scenarios.

5.1 Setup

The anechoic ATFs which were used in the MVDR beamformers were calculated from the same BTE-IRs that have been used for the training of the SVM classifiers (cf. Section 4). The (i, j)-th element of the spatial coherence matrix $\mathbf{\Gamma}_{i,j}$ has been calculated using these anechoic ATFs, i.e.,

$$\mathbf{\Gamma}_{i,j} = \frac{\sum_{k=1}^{K} d_i(\theta_k) d_j^*(\theta_k)}{\sqrt{\sum_{k=1}^{K} |d_i(\theta_k)|^2 \sum_{k=1}^{K} |d_j(\theta_k)|^2}}, \tag{11}$$

with K denoting the total number of angles, i.e. $K = 72$. For the binaural MVDR beamformer $i, j \in \{1, \ldots, 2M\}$ and for the bilateral MVDR beamformer $i, j \in \{1, \ldots, M\}$.

The hearing aid microphone signals for the second experiment have been generated using measured BTE-IRs for a binaural hearing aid setup mounted on a dummy head in an office room ($T_{60} = 300$ ms, DRR = 4 dB) and a cafeteria ($T_{60} = 1250$ ms, DRR = 16 dB) [23]. Each hearing aid was equipped with 2 microphones with a distance of 7 mm. For the office scenario a moving speech source was simulated which moved from $-60°$ to $0°$ in steps of $5°$ and for the cafeteria scenario the speech source was located at fixed positions of $0°$ and $-35°$. The speech signals were taken from the OLSA sentence test material [24]. Two different noise types have been used for the experiments:

- *Babble noise:* For the office scenario, a spatially stationary noise field was generated using the method described in [25], where the time-varying PSD of the noise component was calculated from a babble noise signal and the time-invariant spatial coherence matrix was calculated according to (11), i.e. perfectly matching the covariance matrix used in the MVDR beamformers. This noise field reflects the so-called cocktail party scenario where a large number of competing talkers is simultaneously active in a reverberant room.
- *Ambient noise:* For the cafeteria scenario, realistic ambient noise including babble noise, clacking plates and interfering speakers, recorded in the same cafeteria, has been used as the noise component [23].

For the office scenario the speech-and-noise signal had a length of 3.7 s and for the cafeteria scenario the speech-and-noise signals had a length of 20.6 s. The signals were sampled at a sampling frequency of 16 kHz. For both scenarios the iSNR at the left hearing aid was set to 0 dB, -5 dB and -10 dB. The corresponding iSNRs for the right hearing aid are shown in Figures 5 - 7. The signals were processed using a weighted overlap-add framework with a block size of 10 ms and an overlap of 5 ms.

5.2 Experiment 1

In order to compare the directivity of the binaural and the bilateral MVDR beamformer we calculate the narrowband SNR improvement for the left and the right hearing aid for a source coming from a direction θ, i.e.,

$$gSNR_0(\theta) = \frac{|\mathbf{W}_0^H \mathbf{d}(\theta)|^2}{\mathbf{W}_0^H \mathbf{\Gamma} \mathbf{W}_0} \frac{1}{|d_{0,1}(\theta)|^2}, \tag{12}$$

$$gSNR_1(\theta) = \frac{|\mathbf{W}_1^H \mathbf{d}(\theta)|^2}{\mathbf{W}_1^H \mathbf{\Gamma} \mathbf{W}_1} \frac{1}{|d_{1,1}(\theta)|^2}. \tag{13}$$

The narrowband SNR improvement for the bilateral MVDR beamformer can be calculated by replacing $\mathbf{W}_0$ and $\mathbf{W}_1$ with $\overline{\mathbf{W}}_0$ and $\overline{\mathbf{W}}_1$ and $\mathbf{d}(\theta)$ with $\mathbf{d}_0(\theta)$ and $\mathbf{d}_1(\theta)$, respectively.

Figure 3 depicts the narrowband SNR improvement for the binaural and the bilateral MVDR beamformer (steered towards $0°$) for different speech source positions. Figure 4 depicts the global intelligibility-weighted SNR (iSNR) improvement [26].

For the binaural MVDR beamformer the global iSNR improvement decreases for the ipsilateral hearing aid and is rather stable for the contralateral hearing aid if the speech source is not located at the steering direction of $0°$ and the DOA error is less than $80°$. On the one hand, for the ipsilateral hearing aid the global iSNR improvement becomes negative if the DOA estimation error is larger or equal to $10°$ due to the sidelobes caused by the distance of 0.17 m between the left and the right hearing aid (cf. Figure 3a and 3b). On the other hand, for the contralateral hearing aid the global iSNR improvement is always larger than 0 dB for DOA estimation errors up to $80°$.

For the bilateral MVDR beamformer, the global iSNR improvement in the ipsilateral hearing aid is much less sensitive compared to the binaural MVDR beamformer. Up to an DOA estimation error of $60°$ the global iSNR improvement is larger than 0 dB and hence very similar to the performance of a DM [10]. However, for a speech source position of $0°$, i.e. no DOA estimation error, the global iSNR improvement is 3 dB lower than for the binaural MVDR beamformer. Similarly as for the binaural MVDR beamformer, for the contralateral hearing aid the global iSNR improvement is always larger than 0 dB for DOA estimation errors up to $80°$ and is generally lower than for the binaural MVDR beamformer for speech source positions in the frontal hemisphere.

5.3 Experiment 2

In the second experiment we compare the performance of the fixed bilateral MVDR beamformer, the fixed binaural MVDR beamformer and the steerable binaural MVDR beamformer in a cafeteria and an office scenario. The results for a speech source position at $0°$ and $-35°$ in the cafeteria are depicted in Figure 5 and 6. For the speech source position at $0°$ all beamformers improve the output SNR, where the iSNR improvement of the fixed bilateral and binaural MVDR beamformers are independent of the input iSNR since the beamformers are always steered towards $0°$. The noise reduction performance of the steerable binaural MVDR beamformer decreases with decreasing input iSNR from 7.1 dB (input iSNR of 0 dB) to 6.3 dB (input iSNR of -10 dB) in the left hearing and from 6.9 dB to 4.7 dB in the right hearing aid. For both hearing aids the binaural MVDR beamformers show a better noise reduction performance compared to the bilateral MVDR beamformer at the considered SNRs.

For the speech source position at $-35°$, again all beamformers improve the output SNR for all input SNRs. Compared to the fixed binaural MVDR beamformer, the fixed bilateral MVDR beamformer shows a better performance in the left hearing aid and a worse performance in the right hearing, as expected from

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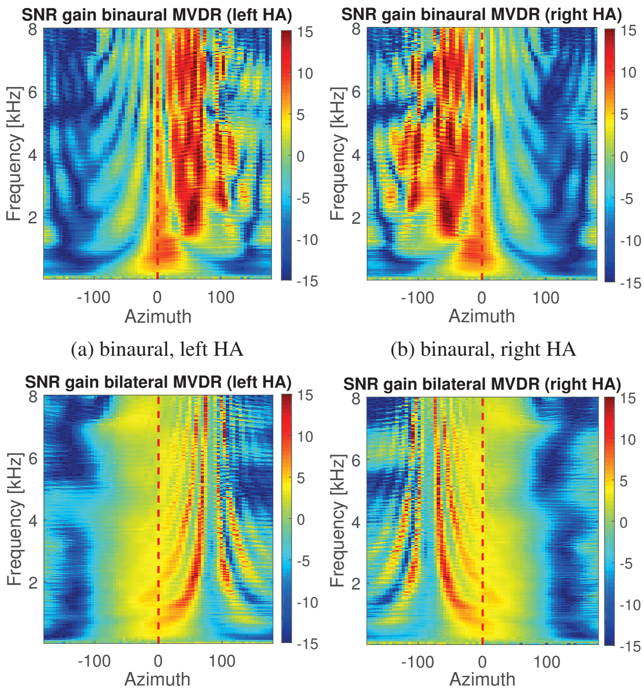

(a) binaural, left HA (b) binaural, right HA

(c) bilateral, left HA (d) bilateral, right HA

Figure 3: Narrowband SNR improvement for a binaural and a bilateral MVDR beamformer steered towards 0° for different speech source positions in a diffuse noise field. The vertical dashed line indicates the steering direction of the MVDR beamformers.

(a) left HA (b) right HA

Figure 4: Global intelligibility-weighted SNR improvement for a binaural and a bilateral MVDR beamformer steered towards 0° for different speech source positions in a diffuse noise field. The vertical dashed line indicates the steering direction of the MVDR beamformers.

Figure 4. The noise reduction performance of the steerable binaural MVDR beamformer decreases with decreasing input iSNR from 4.8 dB (input iSNR of 0 dB) to 2.9 dB (input iSNR of -10 dB) in the left hearing and from 10.5 dB to 5.9 dB in the right hearing aid. The performance of the steerable binaural MVDR beamformer is generally better compared to the bilateral MVDR beamformer except for an input iSNR of -10 dB in the left hearing aid.

The results for the speech source in the office scenario, moving from $-60°$ to $0°$, are depicted in Figure 7. For all beamformers the noise reduction performance is generally lower than for the cafeteria scenario, while the steerable binaural MVDR beamformer shows the best performance. Again, compared to the fixed binaural MVDR beamformer, the fixed bilateral MVDR beamformer shows a better noise reduction performance in the left hearing aid and a worse performance in the right hearing aid. For the steerable binaural MVDR beamformer the noise reduction performance in the left hearing aid is rather independent of the input iSNR while for the right hearing aid the performance decreases with decreasing input iSNR.

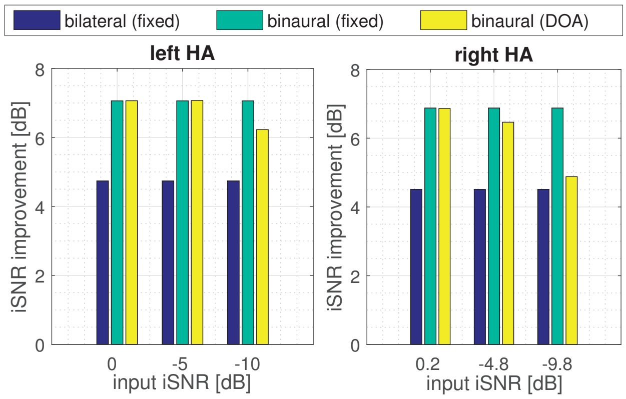

Figure 5: iSNR improvement in the left and the right hearing aid for the binaural and bilateral MVDR beamformers. The speech source is positioned at 0° in a cafeteria and recorded ambient noise is added at different iSNRs.

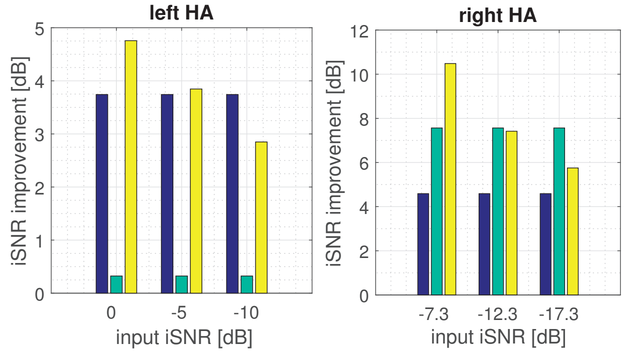

Figure 6: iSNR improvement in the left and the right hearing aid for the binaural and bilateral MVDR beamformers. The speech source is positioned at $-35°$ in a cafeteria and recorded ambient noise is added at different iSNRs.

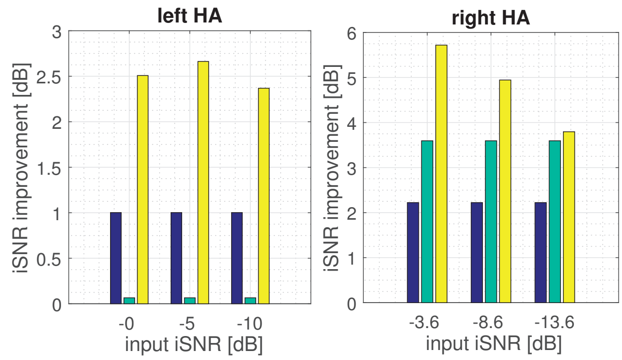

Figure 7: iSNR improvement in the left and the right hearing aid for the binaural and bilateral MVDR beamformers. The speech source moves from $-60°$ to $0°$ in an office room and diffuse babble noise is added at different iSNRs.

6 Conclusions

In this paper we have shown that for a fixed binaural MVDR beamformer the noise reduction performance may significantly decrease in the ipsilateral hearing aid if the desired speech source is not located in the frontal direction. For a fixed bilateral MVDR beamformer the noise reduction performance is lower compared to the binaural MVDR beamformer but also much more robust for target positions apart from the frontal direction. Simulation results in two reverberant, diffuse noise scenarios show that using a steerable binaural MVDR beamformer with a classification based DOA estimator in general provides a better noise reduction performance compared to a fixed bilateral MVDR beamformer.

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