COMPARISON OF TWO BINAURAL BEAMFORMING APPROACHES FOR HEARING AIDS

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ABSTRACT

Beamforming algorithms in binaural hearing aids are crucial to improve speech understanding in background noise for hearing impaired persons. In this study, we compare and evaluate the performance of two recently proposed minimum variance (MV) beamforming approaches for binaural hearing aids. The binaural linearly constrained MV (BLCMV) beamformer applies linear constraints to maintain the target source and mitigate the interfering sources, taking into account the reverberant nature of sound propagation. The inequality constrained MV (ICMV) beamformer applies inequality constraints to maintain the target source and mitigate the interfering sources, utilizing estimates of the direction of arrivals (DOAs) of the target and interfering sources. The similarities and differences between these two approaches is discussed and the performance of both algorithms is evaluated using simulated data and using real-world recordings, particularly focusing on the robustness to estimation errors of the relative transfer functions (RTFs) and DOAs. The BLCMV achieves a good performance if the RTFs are accurately estimated while the ICMV shows a good robustness to DOA estimation errors.

Index Terms— Binaural signal processing, acoustic beamforming, binaural hearing aids, LCMV, noise reduction

1. INTRODUCTION

Although in the last decades hearing aids have evolved from simple sound amplifiers to modern digital devices with complex functionalities, speech understanding is still a challenging problem for the hearing aid user in the presence of background noise and reverberation. Hence, beamforming 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 but to use the microphones of both hearing aids simultaneously (binaural configuration) for improved noise reduction [1, 2, 3, 4, 5, 6, 7].

Binaural beamforming approaches usually require an estimate of the correlation matrices of the desired target and the undesired interference components and/or an estimate of the direction of arrivals (DOAs) of the target and interfering sources [7]. The estimated correlation matrices can be used to estimate the relative transfer functions (RTFs) or DOAs of the target and interfering sources in order to achieve noise reduction [8, 9]. However, the performance of these algorithms may significantly deteriorate in case of estimation errors [10, 11]. Hence, for binaural hearing aids, approaches have been proposed that aim to increase the robustness to estimation errors by using additional equality constraints [12] or inequality constraints [10, 11].

The aim of this study is to compare the performance and robustness of two minimum variance (MV) approaches for noise reduction in binaural hearing aids which aim to minimize the background noise component, while preserving the target source and suppressing interfering sources. On the one hand, the binaural linearly constrained MV (BLCMV) beamformer aims to achieve noise reduction by exploiting estimates of the RTFs of the target and interfering sources. On the other hand, the inequality constrained MV (ICMV) beamformer aims to achieve noise reduction by exploiting estimates of the DOAs of the target and interfering sources and utilizes additional robustness constraints for the target source in order to increase the robustness to DOA estimation errors [10, 11].

First, the BLCMV and the ICMV will be reviewed and similarities and differences between these two approaches will be discussed. Second, the performance of both algorithms will be evaluated in a simulated environment and using real-world recordings, particularly focusing on the robustness to estimations errors of the estimated RTFs and DOAs.

2. PROBLEM FORMULATION

We consider a binaural hearing aid system consisting of two hearing devices with a total of M microphones and an acoustic scenario comprising one target speech source and N_u multiple directional interfering sources in a noisy and reverberant environment. In the frequency-domain, the M-dimensional stacked vector of the received microphone signals $\mathbf{y}(\omega)$ can be written as

$$\mathbf{y}(\omega) = \mathbf{x}(\omega) + \mathbf{u}(\omega) + \mathbf{n}(\omega), \tag{1}$$

where $\mathbf{x}(\omega)$ is the target component, $\mathbf{u}(\omega)$ the interfering sources component, and $\mathbf{n}(\omega)$ the background noise component (e.g., diffuse noise). For brevity, the frequency variable ω is henceforth omitted.

The target and interfering sources' components can be written as $\mathbf{x} = s_{\mathrm{x}}\mathbf{h}_x$ and $\mathbf{u} = \sum_{r=1}^{N_u} s_{\mathrm{u,r}}\mathbf{h}_{u,r}$, where s_{x} and $s_{\mathrm{u,r}}$ denote the target and rth interfering signals and $\mathbf{h}_x$ and $\mathbf{h}_{u,r}$ denote the entire *reverberant* acoustic transfer function (ATF) vectors relating the target and the rth interfering source to the microphones, respectively.

Note that the entire reverberant ATF vector for the target source can be decomposed as

$$\mathbf{h}_x = \mathbf{h}_\theta + \mathbf{h}_{\text{reverb}}, \tag{2}$$

where θ is the DOA angle from the listener's head center to the source, $\mathbf{h}_\theta$ is the anechoic (direct path) ATF vector, and $\mathbf{h}_{\text{reverb}}$ is the residual binaural room reverberant (early reflections plus a late reverberation) ATF vector. Similar decomposition can be defined for the interfering sources. The background noise correlation matrix $\mathbf{R}_n = \mathcal{E}\left\{\mathbf{n}\mathbf{n}^H\right\}$ is assumed to be full-rank and $\mathcal{E}\{\cdot\}$ is the expectation operator.

In this paper, a binaural setup with monaural output is described. Without loss of generality, the reference microphone is selected as the first microphone at the left hearing device (e.g., closest to the left ear). The reference microphone signal is given by $y_L = \mathbf{e}_L^H \mathbf{y}$, where $\mathbf{e}_L$ is the M-dimensional selector vector with one element equal to one and all other elements equal to zero. The monaural output signal for the left hearing aid is obtained by applying the beamformer to all microphone signals from both hearing aids, i.e., $z = \mathbf{w}^H\mathbf{y}$, where $\mathbf{w}$ is the M-dimensional complex-valued weight vector.

3. TWO BINAURAL BEAMFORMING FORMULATIONS

Section 3.1 and Section 3.2 briefly review the considered BLCMV and ICMV beamformers. In Section 3.3, a comparison between the two considered beamformers is given.

3.1. Binaural LCMV (BLCMV)

The BLCMV beamformer is designed to reproduce the target component at the reference microphone, while reducing the directional interfering sources and minimizing the background noise power [13, 14]. The formulation of the BLCMV is

$$\begin{aligned} \min_{\mathbf{w}} \quad \mathbf{w}^H\mathbf{R}_n\mathbf{w} \quad \text{s.t. } & \mathbf{w}^H\bar{\mathbf{h}}_x = 1, \\ & \mathbf{w}^H\bar{\mathbf{h}}_{u,k} = \eta,\ k = 1...N_u, \end{aligned} \tag{3}$$

where the scaling parameter η, with $0 \le \eta \le 1$, sets the amount of interference reduction, and $\bar{\mathbf{h}}_x \triangleq \mathbf{h}_x/h_{x,r}$ and $\bar{\mathbf{h}}_{u,k} \triangleq \mathbf{h}_{u,r}/h_{u,k,r}$ are the reverberant RTF vectors defined as the normalized reverberant ATF vectors with respect to the rth microphone, which serves as a reference. The criterion can be written in a compact way, i.e.,

$$\min_{\mathbf{w}} \left\{\mathbf{w}^H\mathbf{R}_n\mathbf{w}\right\} \quad \text{s.t.} \quad \mathbf{C}_{\text{BLCMV}}^H\mathbf{w} = \mathbf{g}_{\text{BLCMV}}, \tag{4}$$

where the BLCMV constraint set is given by

$$\begin{aligned} \mathbf{C}_{\text{BLCMV}} &= \begin{bmatrix}\bar{\mathbf{h}}_x & \bar{\mathbf{h}}_{u,1} & \cdots & \bar{\mathbf{h}}_{u,N_u}\end{bmatrix}, \\ \mathbf{g}_{\text{BLCMV}} &= \begin{bmatrix} 1 \\ \eta\mathbf{1}_{N_u\times 1}\end{bmatrix}. \end{aligned} \tag{5}$$

The solution to the BLCMV problem is given by

$$\mathbf{w} = \mathbf{R}_n^{-1}\mathbf{C}_{\text{BLCMV}}\left[\mathbf{C}_{\text{BLCMV}}^H\mathbf{R}_n^{-1}\mathbf{C}_{\text{BLCMV}}\right]^{-1}\mathbf{g}_{\text{BLCMV}}. \tag{6}$$

3.2. ICMV

The ICMV is a binaural beamforming algorithm, which is designed with robustness to variations in the real-world by imposing DOA-based inequality constraints to protect the target speaker signal and reject interfering source signals. The ICMV formulation [11] is revisited briefly below.

Suppose that the anechoic ATF vector $\mathbf{h}_\theta$ from different incidence angle θ is available from a pre-existing database. The formulation of ICMV can be written as

$$\begin{aligned} \min_{\mathbf{w}} \ \mathbf{w}^H\mathbf{R}_n\mathbf{w} \quad \text{s.t. } & |\mathbf{w}^H\bar{\mathbf{h}}_\theta - 1|^2 \le \epsilon_\theta^2,\ \forall\, \theta \in \Theta \\ & |\mathbf{w}^H\bar{\mathbf{h}}_\phi|^2 \le \epsilon_\phi^2,\ \forall\, \phi \in \Phi, \end{aligned} \tag{7}$$

where $\bar{\mathbf{h}}_\theta \triangleq \mathbf{h}_\theta/h_{\theta,r}$ is the anechoic RTF vector defined as the normalized anechoic ATF vector with respect to the rth microphone, which serves as a reference. The angle set Θ includes a finite number of directions that are close to the estimated DoA τ of the target source, for instance, $\Theta = \{\tau - 10^\circ, \tau, \tau + 10^\circ\}$. The corresponding ϵ_θ specifies a tolerable speech distortion (SD). Similarly, the angle set Φ includes the estimated DOAs of the interfering speakers, of which the amplification strength should not exceed a pre-defined threshold ϵ_ϕ.

The optimization problem in (7) is a convex quadratically constrained quadratic program (QCQP), which does not have a closed-form solution. To design a low-complexity algorithm, we introduce auxiliary variables $\{\delta_\theta\}$, $\theta \in \Theta$ and $\{\delta_\phi\}$, $\phi \in \Phi$ and reformulate (7) as

$$\min_{\{\delta_\theta\},\{\delta_\phi\}} \min_{\mathbf{w}} \quad \mathbf{w}^H\mathbf{R}_n\mathbf{w}$$

$$\text{s.t.} \quad \mathbf{w}^H\bar{\mathbf{h}}_\theta = \delta_\theta,\ \forall\theta \in \Theta \tag{8a}$$

$$\mathbf{w}^H\bar{\mathbf{h}}_\phi = \delta_\phi,\ \forall\phi \in \Phi \tag{8b}$$

$$|\delta_\theta - 1|^2 \le \epsilon_\theta^2,\ \forall\theta \in \Theta \tag{8c}$$

$$|\delta_\phi|^2 \le \epsilon_\phi^2,\ \forall\phi \in \Phi. \tag{8d}$$

The structured optimization problem in (8) can be solved efficiently by the celebrated ADMM algorithm [15], where in each update step all optimization variables are obtained in closed-form with low computational effort. Detailed derivations can be found in [11].

3.3. Comparison of Algorithms

Both the BLCMV and ICMV beamformers aim to extract the target source while reducing the interfering sources and minimizing noise, i.e., both criteria are MV subject to constraints for both the target and the interfering sources. However, several major differences between the considered beamformers should be noted, which relate to 1) the type of steering vectors that construct the beamformers, 2) the type of constraints imposed on their cost function, and 3) their trade-off parameters.

The BLCMV utilizes the reverberant RTF steering vectors. The reverberant RTF vectors are estimated from the recorded data. Hence, the RTF vectors are data-dependent and vary for different recorded data. The beamformer performance strongly relies on the quality of the reverberant RTF estimation. The estimated RTF vectors take into account the specific listener head transfer function. Moreover, the estimated RTF vectors also take into account the specific room transfer function such that the spatial filtering is matched to the room acoustic. Reverberant RTF estimation procedures are described in [13, 16, 17].

As a result of the BLCMV equality constraints, a distortionless response towards the target source direction is imposed. The scaling parameter η controls the exact amount of interference reduction, i.e., the depth of the null.

	BLCMV	ICMV
Criterion	MV	MV
Steering vectors	Reverberant RTFs	Anechoic RTFs
	Data driven/Estimated	Fixed/From database
Constraints		
Target source	Equality	Inequality
Interfering source	Equality	Inequality
Estimation Requirement		
Directional sources	RTF steering vectors	DOA
Background noise	$\mathbf{R}_n$	$\mathbf{R}_n$

Table 1. Comparison of BLCMV and ICMV beamformers.

The ICMV utilizes anechoic RTF steering vectors with respect to an estimated DOA of the sources. The anechoic RTF vectors are fixed (data independent), and obtained from an existing database. The anechoic RTF vectors are typically measured on a head and torso simulator in an anechoic room, and hence, do not take into account the specific listener head related transfer function or the room acoustic.

The number of ICMV inequality constraints around the estimated DOA of the sources, and the trade-off parameters ϵ_θ and ϵ_ϕ control the robustness to head movements and steering errors.

In general, as the number of either BLCMV equality constraints or the ICMV inequality constraints increases, the degrees of freedom for the MV minimization decrease, which results in a lower noise reduction performance. In practice, the number of constraints in the BLCMV and the ICMV is a trade-off between robustness and noise reduction.

Relation between the ICMV and BLCMV: despite the use of different steering vectors (anechoic RTF for the ICMV versus estimated reverberant RTF for the LCMV) in the two beamforming approaches, the ICMV can be regarded as a generalization of the BLCMV, since the equality constraints are relaxed to inequality constraints. The ICMV problem in (8) can be solved sequentially in two stages: in the first stage, we minimize $\mathbf{w}^H\mathbf{R}_n\mathbf{w}$ subject to the linear constraints (8a) and (8b) with fixed right-hand side values δ_θ and δ_ϕ, which is exactly BLCMV, whereas in the second stage, we optimize δ_θ and δ_ϕ in the parameter space defined by (8c) and (8d). Thus, ICMV can be viewed as selecting a BLCMV beamformer with the optimal parameters (δ_θ and δ_ϕ) in the linear constraints.

The comparison of the considered beamformers is summarized in Table 1.

4. EXPERIMENTAL STUDY

In this section, we present simulation results for simulated data (Section 4.2) and real-world recordings (Section 4.3). All signals were sampled with a sampling frequency of 20 kHz. For the desired speaker, 10 groups of 2 sentences (each group has a length of at least 3 seconds) from the HINT database [18] were used with 2 seconds of silence between subsequent groups. For the simulated data, the original HINT recordings have been used while for the real-world recordings HINT sentences were spoken by the target speaker. The interfering sources are continuous speech signals taken from the HINT database, the rainbow passage [19], the ISMADHA test signal [20] and a male recording of Arizona Travelogue (Cosmos, Inc.) [21]. Each stimulus started with a 3 seconds long diffuse babble noise initialization phase to estimate the noise correlation matrix. After the initialization phase each target and interfering speaker talked individually for several seconds while diffuse background noise is continuously present. These segments have been used to estimate the target source plus noise correlation matrix and the interfering source plus noise correlation matrix. The estimated correlation matrices are then used to estimate the RTFs of each source (required in the BLCMV) using generalized eigenvalue decomposition [13, 16] and the DOAs for each source (required in the ICMV) are estimated during these segments using the generalized cross-correlation function with phase transform [22]. Based on the DOA estimate, the anechoic ATFs of the hearing aid microphones, which were measured on a KEMAR dummy head in an anechoic chamber, were used in the ICMV with a resolution of 5°. The two approaches are evaluated using the intelligibility-weighted signal-to-noise ratio improvement (IW-SNRI) [23] and the intelligibility-weighted speech distortion (IW-SD) [24].

4.1. Simulation Setup and Algorithm Parameters

For both the BLCMV and the ICMV the number of constraints depends on the number of interfering sources in the acoustic scenario. While for the BLCMV the number of linear constraints is equal to the number of interfering sources plus 1 (cf. Section 3.1), for the ICMV additional robustness constraints for the target source are imposed if a sufficient number of degrees of freedom is available. Since for the simulated data (cf. Section 4.2) altogether 6 hearing aid microphones are available, additional degrees of freedom are utilized to increase the robustness of the ICMV to DOA estimation errors. Since for the recorded data (cf. Section 4.3) only 4 hearing aid microphones are available, only one inequality constraint for each source is imposed due to an insufficient number of degrees of freedom. For the BLCMV the trade-off parameter η is set to zero for all scenarios and the scenario dependent parameter settings for the ICMV are presented in Tables 2 and 3. The signals are processed in a weighted overlap-add framework with a block-length of 1024 samples and 50 % overlap between successive blocks.

	Target Source	Interfering Sources
Est. DOA	$\tau = 10^\circ$	$\zeta_1 = 133^\circ$,$\zeta_2 = 327^\circ$
Angle Set	$\Theta = \{\tau - 10^\circ, \tau, \tau + 10^\circ\}$	$\Phi = \{\zeta_1\}, \{\zeta_1, \zeta_2\}$
Tolerance	$\epsilon_\theta = \{0.2, 0.1, 0.2\}$	$\epsilon_\phi = \{0.1\}, \{0.1, 0.1\}$

Table 2. Setup of ICMV for simulated data.

	Target Source	Interfering Sources
Est. DOA	$\tau = 0^\circ$	$\zeta_1 = 50^\circ$,$\zeta_2 = 315^\circ$
Angle Set	$\Theta = \{\tau\}$	$\Phi = \{\zeta_1, \zeta_2\}$
Tolerance	$\epsilon_\theta = \{0.1\}$	$\epsilon_\phi = \{0.5, 0.5\}$

Table 3. Setup of ICMV for recorded data.

4.2. Simulated Data

For the simulated data, the room impulse responses from the sources to the left and the right hearing aid microphones are generated using the image method [25], where the hearing aid microphones are simulated as being positioned on a rigid sphere [26, 27] in order to take the head shadow effect into account. For each hearing aid 3 microphones with a distance of 7.5 mm are available, i.e. altogether 6 microphones. The size of the room is similar to the room in the Starkey database [28] and the reflection coefficient of the surfaces are chosen such that the reverberation time is the same as the reverberation time of the room in the Starkey database which is 0.6 s. The target source is positioned at 10° with a distance of 1 m to the hearing aid user and the interfering sources are positioned at 140° and 330° with a distance of 1.5 m from the HA user. We used a scenario with one interfering source and two interfering sources with a signal-to-interference ratio (SIR) of 0 dB and SNRs of $(5, 20)$ dB (cf. Table 4). Diffuse babble noise is simulated by adding up the signals of 24 simulated speakers distributed over the simulated room [28]. The RTFs and DOAs are estimated as described in the beginning of Section 4. The impact of source position misalignment on the performance of the BLCMV and the ICMV is evaluated by estimating the RTFs and DOAs of all sources with a mismatch of 10° to the actual source position denoted as *BLCMV (mis.)* and *ICMV (mis.)*. For the ICMV this results in an DOA estimate of 0° for the target source and 147° and 320° for the interfering sources (cf. Table 2). In order to increase the robustness of the BLCMV, an average RTF for the target source has been calculated by averaging the signal statistics over 4 source positions between 0° and 10°, denoted as *BLCMV (avg.)*.

The results are depicted in Fig. 1 and Fig. 2. For the matched case (*BLCMV* and *ICMV*), the BLCMV shows the best performance in terms of IW-SNRI and IW-SD for both acoustic scenarios and input SNRs. In the mismatch case (*BLCMV (mis.)* and *ICMV (mis.)*), the performance of the BLCMV significantly drops while the ICMV shows a good robustness to estimation errors especially in terms of IW-SD. Using the average RTF estimate for the target source in the BLCMV (*BLCMV (avg.)*) increases the robustness of the BLCMV, while the IW-SNRI and IW-SD performance is comparable to the performance of the *ICMV* and the *ICMV (mis.)*.

Target [$^\circ$]	SNR [dB]	SIR [dB]	Interferers [$^\circ$]
10	5,20	0	140
10	5,20	0	140,330

Table 4. Simulation conditions for the simulated data

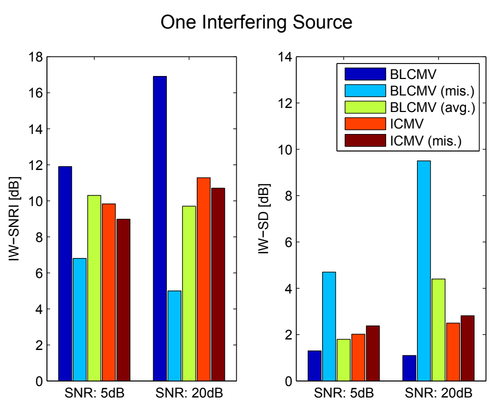

Fig. 1. IW-SNRI and IW-SD for the simulated scenario with one interfering source.

4.3. Recorded Data

For the second experiment, real-world recordings from the Starkey database [28] have been used. This database contains recordings with binaural hearing aids with 2 microphones in each hearing aid mounted on actual people. The target talker was a male talker who was sitting at a table in front of the hearing aid user. The two interfering talkers are two female talkers who were positioned at 45° and 315° at the same table. The diffuse background noise was generated by 56 talking people distributed over the room. The room had a size of $12.7 \times 10 \times 3.6$ m and a reverberation time of approximately 0.6 s. The SIR was equal to -5 dB and 5 dB, respectively and the SNR was equal to 5 dB and 20 dB, respectively (cf. Table 5).

The results are depicted in Fig. 3. For the real-world recordings the IW-SNRI of both the BLCMV and the ICMV are lower

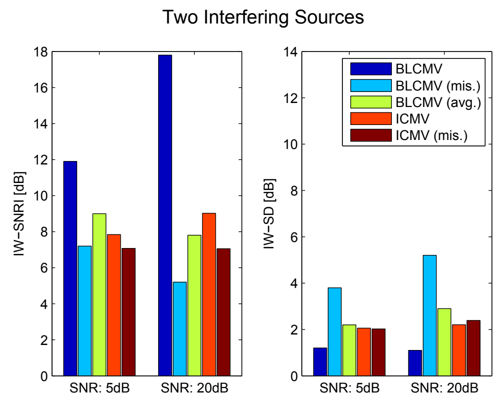

Fig. 2. IW-SNRI and IW-SD for the simulated scenario with two interfering sources.

compared to the results for the simulated data. The BLCMV generally shows a better performance in terms of IW-SNRI while the ICMV generally performs better in terms of IW-SD. While for an input (SIR,SNR) of $(-5, 5)$ dB the performance of the BLCMV and the ICMV are very similar, for all other conditions the BLCMV outperforms the ICMV by $1-2$ dB in terms of IW-SNRI and shows a similar performance in terms of IW-SD.
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Target [$^\circ$]	SNR [dB]	SIR [dB]	Interferers [$^\circ$]
0	5,20	-5,5	45,315

Table 5. Simulation conditions for the recorded data

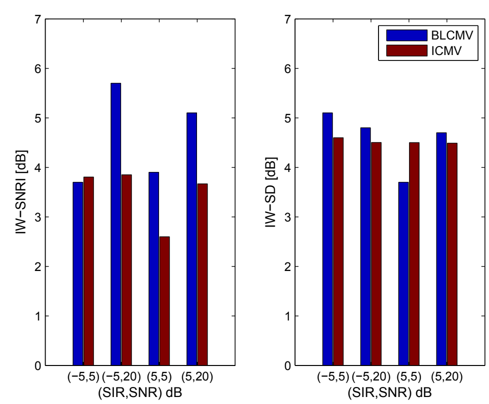

Fig. 3. IW-SNRI and IW-SD for the recorded scenario with two interfering sources.

5. CONCLUSIONS AND FURTHER RESEARCH

In this paper, two MV beamforming approaches for binaural hearing aids application, namely the BLCMV and the ICMV beamformers, were explored. The two approaches differ in their treatment of the constraint set. While the BLCMV uses equality constraints regarding the RTFs of the sources, the ICMV uses inequality constraints regarding the DOAs of the sources. The BLCMV beamformer achieves a good performance if the RTF vectors are accurately estimated while the ICMV beamformer shows a good robustness to DOA estimation errors. Integrating inequality reverberant RTF constraints into the MV cost function is topic for further research.

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