BINAURAL LINEARLY CONSTRAINED MINIMUM VARIANCE BEAMFORMER FOR HEARING AID APPLICATIONS

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ABSTRACT

In many cases hearing impaired persons suffer from hearing loss in both ears, necessitating two hearing apparatuses. In such cases, the applied speech enhancement algorithms should be capable of preserving the, so called, binaural cues. In this paper, a binaural extension of the linearly constrained minimum variance (LCMV) beamformer is proposed. The proposed algorithm, denoted binaural linearly constrained minimum variance (BLCMV) beamformer, is capable of extracting desired speakers while suppressing interference speakers. The BLCMV maintains the binaural cues of both the desired and the interference sources in the constrained space. The ability of preserving the binaural cues makes the BLCMV beamformer particularly suitable for hearing aid applications. It is further proposed to obtain a reduced complexity implementation by sharing common blocks in both sides of the hearing aid device. The performance of the proposed method, in terms of imposed distortion, interference cancellation and cue preservation, is verified by an extensive experimental study using signals recorded by a dummy head in an actual room.

Index Terms— Binaural cues, array signal processing, LCMV beamforming, hearing aids, noise reduction.

1. INTRODUCTION

Modern hearing aid devices may consist of two multi-microphone apparatuses mounted on both ears of the hearing impaired person. A wireless link makes the sharing of information (e.g. all microphone signals) feasible. Binaural hearing devices outperform monaural hearing aid devices in their interference cancellation ability and in their capability to preserve the binaural cues and spatial impression.

It is well known that the preservation of the sources' spatial information plays an important role in speech intelligibility improvement. Interaural time difference (ITD) and interaural level difference (ILD) are widely used binaural cues in the context of hearing aids.

Two main families of methods are commonly used in binaural hearing aids design. The first family utilizes two microphone signals (one at each side of the head). As an identical and real-valued gain is applied in each side of the hearing aid, both ILD and ITD cues are preserved. In [1] this method was proposed based on the concept of computational auditory scene analysis. Alternative procedure, consisting of a binaural super-directive beamformer, was proposed in [2]. In the second family of methods, all microphones from both sides are centrally processed yielding two different beamformer outputs that achieve noise reduction and maintain the binaural cues. In [3] a binaural noise reduction algorithm based on the multichannel

Wiener filter (MWF), was presented. This method inherently preserves the interaural cues of the desired speech components while the noise cues are not preserved. Several extensions of the MWF, that incorporate a simplified quadratic cost function for the preservation of the noise interaural cues, were proposed in [4]. Similar extensions can also be applied to the minimum variance distortionless response (MVDR) criterion[1].

In [5] the spatial cues of a single desired source are preserved by applying multiple constraints and implementing a closed-form LCMV beamformer with broad beam. In the current contribution, a member of the second family of methods, we propose an extension of the eigenspace LCMV beamformer [6] that is applicable to binaural hearing aid devices. We show that the proposed BLCMV is capable of extracting the desired speakers, suppressing the competing speakers and preserving the binaural cues of all constrained signals. The paper is organized as follows. In Sec. 2, the binaural problem is formulated. The BLCMV is introduced in Sec. 3. Sec. 4 outlines the analysis of the binaural cue preservation capability of the BLCMV. The BLCMV beamformer implemented in generalized sidelobe canceler (GSC) structure is reviewed in Sec. 5. Experimental results of the proposed algorithm in actual environments are given in Sec. 6.

2. PROBLEM FORMULATION

Consider a scenario with multiple simultaneous conversations. N_d desired speakers $s_1^d(n), \ldots, s_{N_d}^d(n)$, N_i interference speakers $s_1^i(n), \ldots, s_{N_i}^i(n)$ are received in noisy and reverberant environment by a bilateral, fully connected, hearing device consists of M microphones. The received signals can be formulated in a vector notation, in the short-time Fourier transform (STFT) domain:

$$\boldsymbol{z}(\ell, k) = \boldsymbol{H}_d(\ell, k)\boldsymbol{s}_d(\ell, k) + \boldsymbol{H}_i(\ell, k)\boldsymbol{s}_i(\ell, k) + \boldsymbol{v}(\ell, k) \quad (1)$$

where $\boldsymbol{s}_d(\ell, k) = \begin{bmatrix} s_1^d(\ell, k) & \cdots & s_{N_d}^d(\ell, k) \end{bmatrix}^T$ and $\boldsymbol{s}_i(\ell, k) = \begin{bmatrix} s_1^i(\ell, k) & \cdots & s_{N_i}^i(\ell, k) \end{bmatrix}^T$ are vectors comprising the desired and interfering speech signals, respectively, k denotes the frequency index and ℓ the frame index. $\boldsymbol{H}_d(\ell, k) = \begin{bmatrix} \boldsymbol{h}_1^d(\ell, k) & \cdots & \boldsymbol{h}_{N_d}^d(\ell, k) \end{bmatrix}$ and $\boldsymbol{H}_i(\ell, k) = \begin{bmatrix} \boldsymbol{h}_1^i(\ell, k) & \cdots & \boldsymbol{h}_{N_i}^i(\ell, k) \end{bmatrix}$ are $M \times N_d$ and $M \times N_i$ matrices depicting the time-varying acoustic transfer function (ATF)s relating the desired and interfering sources and the microphone array. $\boldsymbol{v}(\ell, k)$ denotes a stationary noise component of the received signals, consisting of directional as well as diffused signals in the unconstrained subspace. Henceforth, ℓ and k will be omitted for brevity.

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[1]S. Doclo, unpublished.

Assuming the sources and the various interference signals are uncorrelated, the spatial correlation matrix of the received signals can be written as:

$$\mathbf{\Phi}_{zz} = \mathbf{H}_d(\ell, k)\mathbf{\Lambda}_d\big(\mathbf{H}_d\big)^\dagger + \mathbf{H}_i\mathbf{\Lambda}_i\big(\mathbf{H}_i\big)^\dagger + \mathbf{\Phi}_{vv} \qquad (2)$$

where $\mathbf{\Lambda}_d \triangleq \mathrm{diag}\left(\begin{bmatrix} (\sigma_1^d)^2 & \ldots & (\sigma_{N_d}^d)^2 \end{bmatrix}\right)$ and $\mathbf{\Lambda}_i \triangleq \mathrm{diag}\left(\begin{bmatrix} (\sigma_1^i)^2 & \ldots & (\sigma_{N_i}^i)^2 \end{bmatrix}\right)$ are diagonal matrices with the spectral variances of the desired and interfering sources on their main diagonal, respectively. $\mathbf{\Phi}_{vv}$ is the stationary noise spatial correlation matrix.

Let m_ℓ and m_r be the indexes of the left and right reference microphones, respectively (usually, selected as the microphones closest to the ears). The respective reference microphone signals at the left and the right devices are given by $z_\ell = \mathbf{e}_\ell^\dagger \mathbf{z}$ and $z_r = \mathbf{e}_r^\dagger \mathbf{z}$, where $\mathbf{e}_\ell$ and $\mathbf{e}_r$ are M dimensional vectors with 1 in the m_ℓ and m_r position, respectively, and 0 elsewhere. Two spatial M-dimensional filters $\mathbf{w}_\ell$ and $\mathbf{w}_r$ (one for each side), utilize all M microphones, constitute the binaural beamformer

$$y_\ell = \mathbf{w}_\ell^\dagger \mathbf{z}; \quad y_r = \mathbf{w}_r^\dagger \mathbf{z}. \qquad (3)$$

3. BINAURAL LCMV

The BLCMV algorithm consists of two beamformers designed to reproduce the desired signal components as received by the reference microphones in each hearing aid apparatus, while canceling the interference signal component and minimizing the overall noise power. Define the LCMV criterion with multiple constraints on both the desired and interference sources for the left beamformer:

$$\mathbf{w}_\ell = \underset{\mathbf{w}_\ell}{\mathrm{argmin}}\{\mathbf{w}_\ell^\dagger \mathbf{\Phi}_{zz} \mathbf{w}_\ell\}; \text{ s.t. } \mathbf{C}^\dagger \mathbf{w}_\ell = \mathbf{g}_\ell \qquad (4)$$

where $\mathbf{g}_\ell$ is the left response vector, $\mathbf{C}$ is the constrains matrix, and $\mathbf{C}^\dagger \mathbf{w}_\ell = \mathbf{g}_\ell$ is the left constraints set. The criterion for the right beamformer, $\mathbf{w}_r$, is similarly defined by changing the left response vector with its right counterpart $\mathbf{g}_r$.

Denote by $\mathbf{Q}_d \triangleq \begin{bmatrix} \mathbf{q}_1^d & \ldots & \mathbf{q}_{N_d}^d \end{bmatrix}$ a basis spanning the desired subspace and by $\mathbf{Q}_i \triangleq \begin{bmatrix} \mathbf{q}_1^i & \ldots & \mathbf{q}_{N_i}^i \end{bmatrix}$ a basis spanning the interference subspace. Then $\mathbf{H}_d = \mathbf{Q}_d\mathbf{\Theta}_d$ and $\mathbf{H}_i = \mathbf{Q}_i\mathbf{\Theta}_i$ where $\mathbf{\Theta}_d \triangleq \begin{bmatrix} \boldsymbol{\theta}_1^d & \ldots & \boldsymbol{\theta}_{N_d}^d \end{bmatrix}$ and $\mathbf{\Theta}_i \triangleq \begin{bmatrix} \boldsymbol{\theta}_1^i & \ldots & \boldsymbol{\theta}_{N_i}^i \end{bmatrix}$ are the projection coefficients matrices. Let $\mathbf{C} = \begin{bmatrix} \mathbf{Q}_d & \mathbf{Q}_i \end{bmatrix}$ be the constraint matrix, and define the left desired response vector

$$\mathbf{g}_\ell = \begin{bmatrix} \underbrace{\eta(q_{1\ell}^d)^* \ldots \eta(q_{N_d\ell}^d)^*}_{N_d} & \underbrace{\mu(q_{1\ell}^i)^* \ldots \mu(q_{N_i\ell}^i)^*}_{N_i} \end{bmatrix}^T$$
$$= \begin{bmatrix} \eta(\mathbf{g}_\ell^d)^\dagger & \mu(\mathbf{g}_\ell^i)^\dagger \end{bmatrix}^\dagger \qquad (5)$$

where $0 \leq \eta \leq 1$ and $0 \leq \mu \leq 1$ are real-valued scalars defined as the *cue gain factors* for the desired and the interference signals, respectively. $(\mathbf{g}_\ell^d)^\dagger$ and $(\mathbf{g}_\ell^i)^\dagger$ are the m_ℓth and m_rth rows of the respective basis matrices given by:

$$\mathbf{g}_\ell^{d\dagger} = \mathbf{e}_\ell^\dagger \mathbf{Q}_d; \quad \mathbf{g}_\ell^{i\dagger} = \mathbf{e}_\ell^\dagger \mathbf{Q}_i. \qquad (6)$$

Following [6], the ATF is taken into account rather than just the direct arrival. The desired and interference speaker subspaces can be estimated by selecting the major generalized-eigenvectors of the generalized eigenvalue decomposition (GEVD) of either the desired or the interference correlation matrixes, respectively, and the stationary noise correlation matrix.

4. THEORETICAL ANALYSIS OF CUE PRESERVATION

The interaural transfer function (ITF) is defined as the ratio of the acoustical transfer functions relating a single source position and the two ears. The ILD and ITD are defined as the amplitude and the phase of the complex-valued ITF, respectively. For a single source the ITF captures the necessary interaural information. When multiple speakers, both desired and interfering sources, are concurrently active it is important to preserve the entire acoustical scene. Preservation of the binaural cues of only the desired sources is known to be insufficient in practical scenarios, when the interference sources are not entirely suppressed. We will therefore extend now the ITF definition to the multiple concurrent speakers scenario. We will then show that the proposed BLCMV algorithm is capable of preserving the cues of all constrained sources.

Define the interaural signal ratio (ISR) as the ratio of the signal at the left hearing aid to the signal at the right hearing aid. This definition is also applicable to the multiple sources scenario. The ISR at the reference microphones is defined as[2]:

$$\mathrm{ISR}^{in} = \frac{z_\ell}{z_r} = \frac{\mathbf{e}_\ell^\dagger \mathbf{H}_d \mathbf{s}_d + \mathbf{e}_\ell^\dagger \mathbf{H}_i \mathbf{s}_i}{\mathbf{e}_r^\dagger \mathbf{H}_d \mathbf{s}_d + \mathbf{e}_r^\dagger \mathbf{H}_i \mathbf{s}_i}. \qquad (7)$$

The ISR at the output of the beamformers is given by:

$$\mathrm{ISR}^{out} = \frac{\mathbf{w}_\ell^\dagger \mathbf{z}}{\mathbf{w}_r^\dagger \mathbf{z}} = \frac{\mathbf{w}_\ell^\dagger(\mathbf{H}_d\mathbf{s}_d + \mathbf{H}_i\mathbf{s}_i)}{\mathbf{w}_r^\dagger(\mathbf{H}_d\mathbf{s}_d + \mathbf{H}_i\mathbf{s}_i)}$$
$$= \frac{\mathbf{w}_\ell^\dagger \mathbf{Q}_d\mathbf{\Theta}_d\mathbf{s}_d + \mathbf{w}_\ell^\dagger \mathbf{Q}_i\mathbf{\Theta}_i\mathbf{s}_i}{\mathbf{w}_r^\dagger \mathbf{Q}_d\mathbf{\Theta}_d\mathbf{s}_d + \mathbf{w}_r^\dagger \mathbf{Q}_i\mathbf{\Theta}_i\mathbf{s}_i}. \qquad (8)$$

If the constraints are satisfied then:

$$\mathbf{w}_\ell^\dagger \mathbf{Q}_d = \eta(\mathbf{g}_\ell^d)^\dagger; \quad \mathbf{w}_\ell^\dagger \mathbf{Q}_i = \mu(\mathbf{g}_\ell^i)^\dagger$$
$$\mathbf{w}_r^\dagger \mathbf{Q}_d = \eta(\mathbf{g}_r^d)^\dagger; \quad \mathbf{w}_r^\dagger \mathbf{Q}_i = \mu(\mathbf{g}_r^i)^\dagger. \qquad (9)$$

Hence,

$$\mathrm{ISR}^{out} = \frac{\eta\mathbf{g}_\ell^{d\dagger}\mathbf{\Theta}_d\mathbf{s}_d + \mu\mathbf{g}_\ell^{i\dagger}\mathbf{\Theta}_i\mathbf{s}_i}{\eta\mathbf{g}_r^{d\dagger}\mathbf{\Theta}_d\mathbf{s}_d + \mu\mathbf{g}_r^{i\dagger}\mathbf{\Theta}_i\mathbf{s}_i}$$
$$\overset{(6)}{=} \frac{\mathbf{e}_\ell^\dagger(\eta\mathbf{Q}_d\mathbf{\Theta}_d\mathbf{s}_d + \mu\mathbf{Q}_i\mathbf{\Theta}_i\mathbf{s}_i)}{\mathbf{e}_r^\dagger(\eta\mathbf{Q}_d\mathbf{\Theta}_d\mathbf{s}_d + \mu\mathbf{Q}_i\mathbf{\Theta}_i\mathbf{s}_i)}$$
$$= \frac{\mathbf{e}_\ell^\dagger(\eta\mathbf{H}_d\mathbf{s}_d + \mu\mathbf{H}_i\mathbf{s}_i)}{\mathbf{e}_r^\dagger(\eta\mathbf{H}_d\mathbf{s}_d + \mu\mathbf{H}_i\mathbf{s}_i)}. \qquad (10)$$

It can be easily verified that the ISR identifies with the ITF whenever only single source (either desired or interference) is active and that $\mathrm{ISR}^{out} = \mathrm{ISR}^{in}$, i.e. the spatial cues of this source are preserved. It is also evident that the ISR is indeed a generalization of the ITF to the multiple sources case. Moreover, the ISR is also preserved for concurrent sources, provided that they belong to the same group of sources. Hence, even if a time segment consists of several simultaneously active desired sources, the BLCMV beamformer will preserve the entire spatial scene, if the interference sources are inactive. The same applies to the complementary scenario, in which the some of the interference sources are active and all desired sources are inactive. Since speech signals in the STFT domain are only sparsely active, the previous discussion can also be extended to segments with arbitrary activity pattern. Although, in strict sense, setting $\eta \approx 1$

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[2]We assume $v(\ell, k)$ is negligible and all directional interferences are in the constraint subspace.

and $0 < \mu \ll 1$ changes the ISR, as will be shown in Sec. 6, the spatial impression of the user of the hearing aid device will still be preserved. Binaural cue preservation is only guaranteed for the constrained sources and the unconstrained stationary noise source will "inherit" the input cues of the dominant source. The same applies to residual (constrained) interference sources that leak to the output of the beamformer due to estimation inaccuracies. To mitigate the consequences of this phenomenon we set μ to a small value that is *higher* than 0. This will add interference component with the correct cue to the output of the beamformer, and will hence mask any artifacts due to the leakage.

5. BLCMV IMPLEMENTATION

We turn now to the implementation of the BLCMV beamformer that solves criterion (4) and its equivalent criterion for the right hearing aid. An efficient implementation of the BLCMV beamformer can be obtained by sharing common blocks between the beamformers at both sides.

Consider the GSC implementation of the LCMV [6] $\boldsymbol{w} = \boldsymbol{w}_0 - \boldsymbol{B}^\dagger \boldsymbol{q}$ where $\boldsymbol{B}$ is the projection matrix to the "null" subspace, denoted blocking matrix (BM), $\boldsymbol{w}_0$ is the fixed beamformer (FBF) satisfying the constraints set and $\boldsymbol{q}$ is a set of adaptive noise canceler (ANC) filters adjusted to obtain the (unconstrained) minimization. The FBF in both sides are given by:

$$\boldsymbol{w}_{0\ell} = \boldsymbol{F}\boldsymbol{g}_\ell; \quad \boldsymbol{w}_{0r} = \boldsymbol{F}\boldsymbol{g}_r \qquad (11)$$

where $\boldsymbol{F} = \boldsymbol{C}(\boldsymbol{C}^\dagger \boldsymbol{C})^{-1}$. The BM is identical for both the left and right beamformers and is given by $\boldsymbol{B} = \boldsymbol{I}_{M \times M} - \boldsymbol{C}(\boldsymbol{C}^\dagger \boldsymbol{C})^{-1}\boldsymbol{C}^\dagger$. The ANC filters $\boldsymbol{q}_\ell$ and $\boldsymbol{q}_r$ can be implemented by the normalized least mean squares (NLMS) algorithm. Step-size control algorithms can be utilized to guarantee the algorithm convergence. The block diagram for the BLCMV algorithm sharing common blocks is depicted in Fig. 1. Detailed description of the GSC implementation the LCMV can be found in [6].
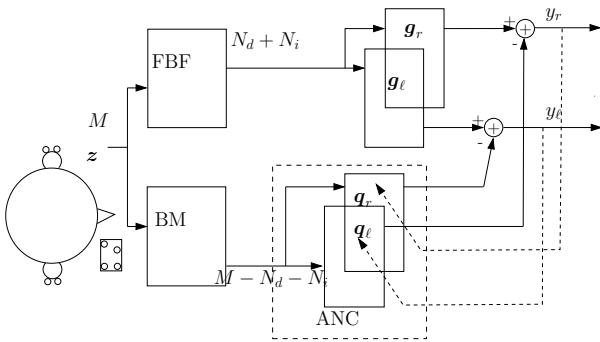

**Fig. 1**. Block diagram of the proposed BLCMV algorithm.

6. EXPERIMENTAL STUDY

The proposed algorithm was evaluated using signals recorded in the acoustic lab at Bar-Ilan University. The experimental setup consisted of two hearing aid devices, each with two omni-directional microphones mounted on a B&K head and torso simulator (HATS)

with 2cm inter-distance and a 9×5 utility device with four omni-directional microphones mounted close to the corners of the device with average distance 3.5cm. The device was placed on a table in front of the HATS at a distance of 0.5m. All microphone signals are utilized by the two beamformers. The test scenario comprised one desired speaker at $\theta_x = 30^o$ 1m from the HATS, one interference speaker at $\theta_v = 290^o$ 1m from the HATS, and one directional stationary noise at $\theta_n = 320^o$ 2.5m from the HATS. The acoustic lab is a $6 \times 6 \times 2.4$ room with reverberation time controlled by 60 panels covering the room facets. In our test T_{60} was set to 0.3s. The signal to interference ratio (SIR) with respect to the non-stationary interference speaker and the signal to noise ratio (SNR) with respect to the stationary noise were set to 0dB and 14dB, respectively. The sampling frequency was 8kHz. The signals were transformed to the STFT domain with 4096 points and 75% overlap. Only the desired and interference speakers were constrained by the BLCMV. The ANC was responsible for suppressing the stationary noise source. The correlation matrices of the desired and interference speakers were estimated during non-concurrent activity of the desired and interference speakers. The sources are static, resulting in time invariant spatial filter. Hence, no smoothing artifacts are encountered. The cue gain factor of the desired speech was set to $\eta = 1$, and the respective interference gain was set to $\mu = 0.1$, equivalent to 20dB attenuation. Fig. 2 depicts the sonograms at the right device as well
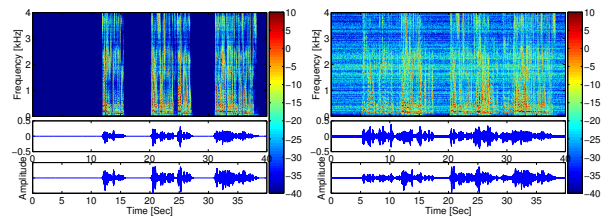
(a) Desired speaker at the reference microphones

(b) Received reference microphones
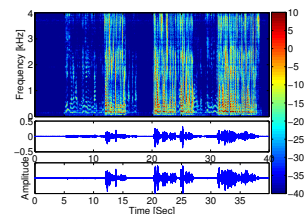
(c) BLCMV outputs

Fig. 2. Sonograms of the right signal and stereo waveforms (left and right signals).

as both the left and right waveforms at various points in the signal flow. It is evident that the BLCMV algorithm significantly attenuates the interference speaker and the stationary noise. The SIR improvements using the proposed algorithm are 19.7dB and 18.1dB in the left and right sides, respectively, in correspondence with the values of η and μ. The SNR improvements are 27.2dB and 19.8dB in the left and right sides, respectively. The analysis of cue preservation for the multiple concurrent speaker scenario can be conducted using an auditory-based processing [7]. The frequency analysis of the basilar membrane within the cochlea of the inner ear can be modeled by analyzing the left and right ear signals with a gammatone filterbank. This auditory filterbank decomposes each input signal
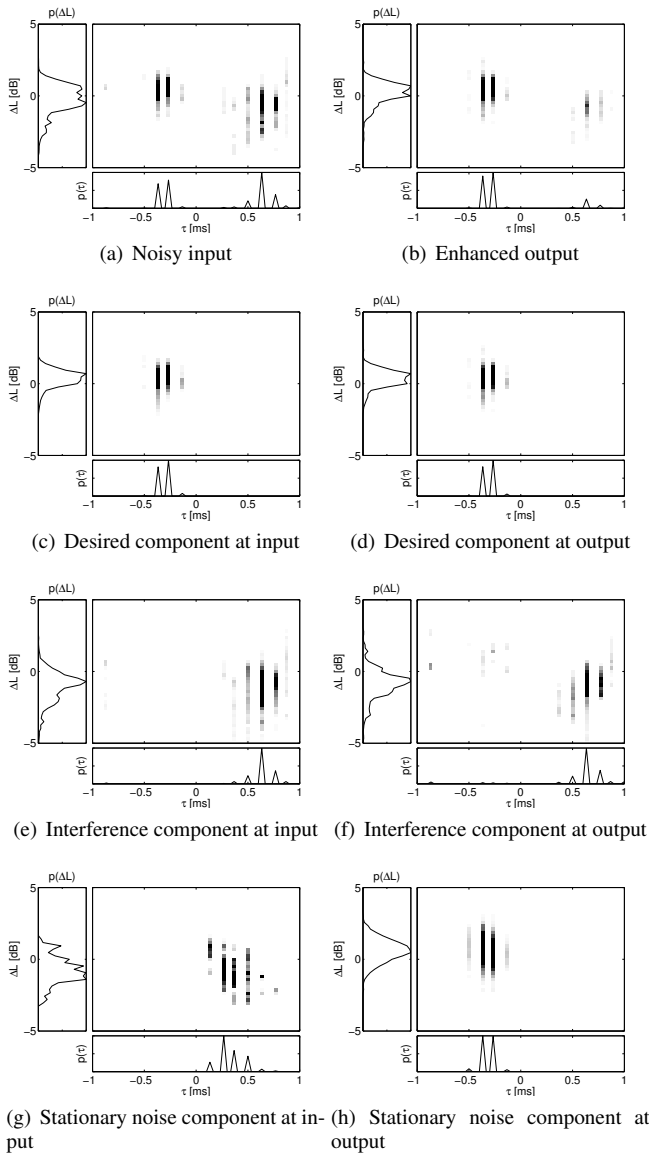
(a) Noisy input

(b) Enhanced output

(c) Desired component at input

(d) Desired component at output

(e) Interference component at input

(f) Interference component at output

(g) Stationary noise component at input

(h) Stationary noise component at output

Fig. 3. p.d.f.'s of ITD and ILD for a scenario with concurrent desired and interference speech signals and directional stationary noise. $T_{60} = 300ms$. Cue selection threshold 0.98. Data are shown for critical band at 500Hz. Graphs produced by the Binaural Cue Selection Matlab Toolbox described in [7].

into 24 sub-bands, named critical bands. Following Faller and Merimaa [7], we evaluate the cue preservation of the proposed algorithm, in a critical band centered at 500Hz. In [7] a model based on the interaural coherence (IC) value is introduced. The model determines the time segments that best predicts the localization cues (ITD and ILD) of a reverberated signal, by applying a predefined threshold to the IC. Fig. 3 depicts the p.d.f.'s of the selected ITD and ILD cues[3]. The p.d.f.'s are estimated by computing histograms of the

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[3]The graphs are produced by the Matlab$^{©}$ toolbox presented by C. Faller and J. Merimaa and available online at http://www.acoustics.hut.fi/software/cueselection.

ITD and ILD cues for IC higher than 0.98. The height of the maximum peak is normalized to 1. The cues of the two dominant sources, namely the concurrent desired and interference sources, are depicted in Fig. 3(a). In Fig. 3(b) the cues at output of the BLCMV beamformer are shown. Although the cues of both signals are shown to be preserved, those related to the interference source are weaker, due to the inherent attenuation of the BLCMV beamformer. To further analyze the cue preservation capabilities of the proposed algorithm, each source is separately analyzed as well. In Fig. 3(c) and Fig. 3(d) the desired signal component at the input and at the output of the algorithm are shown to be preserved. Similar conclusions regarding the interference signal are drawn from Fig. 3(e) and Fig. 3(f). On the contrary, by inspecting Fig. 3(g) and Fig. 3(h) it is evident that the cues of the (unconstrained) noise signal are not preserved but rather replaced by the cues of the dominant (desired speech) signal. This result is verified by our informal listening tests[4].

7. CONCLUSIONS

A binaural extension of the LCMV beamformer was introduced. The proposed method extracts multiple desired speakers in a noisy and reverberant environment with multiple interference speakers. The algorithm is capable of preserving the the ITD and ILD cues of all constrained sources (desired as well as interference). The experimental study verifies the theoretical study. An efficient implementation of the BLCMV beamformer can be obtained by sharing common blocks between the beamformers at both sides. Improving input-output latency, that will enable the implementation of the proposed algorithm in hearing aid devices, is a subject of current research effort.

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