

THEORETICAL ANALYSIS OF BINAURAL CUE PRESERVATION USING MULTI-CHANNEL WIENER FILTERING AND INTERAURAL TRANSFER FUNCTIONS

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ABSTRACT

In this paper a theoretical analysis of the binaural cue preservation of the multi-channel Wiener filter (MWF) is performed. We will prove that in the case of a single speech source the MWF perfectly preserves the binaural cues of the speech component, but changes the binaural cues of the noise component to the cues of the speech component. In addition, we show that by extending the MWF cost function with terms related to the interaural transfer function it is possible to preserve the binaural cues of both the speech and the noise component, without considerably reducing the noise reduction performance.

1. INTRODUCTION

Noise reduction algorithms in hearing aids are crucial for hearing impaired persons to improve speech intelligibility in background noise. Multi-microphone systems are able to exploit spatial information in addition to spectral information and are hence preferred to single-microphone systems [1]-[6]. In addition to reducing noise and limiting speech distortion, another important objective of a binaural noise reduction algorithm is to preserve the listener's impression of the auditory environment in order to exploit the binaural hearing advantage [7]. This can be achieved by preserving the binaural cues, e.g. the interaural time and level difference (ITD, ILD), of the speech and the noise sources. ITD is the time-delay of arrival of the sound signal between the left and right ear, whereas ILD is the intensity difference between the two ears.

A binaural multi-channel Wiener filtering (MWF) technique has been presented in [3], where it has been shown experimentally that this technique preserves the binaural cues of the speech component, but does not preserve the binaural cues of the noise component. This observation will be theoretically proved in Section 4 in the case of a single speech source. In order to also preserve the binaural cues of the noise component, the MWF cost function has been extended either with terms related to the ITD and ILD of the noise component [4] or with terms related to the interaural transfer function (ITF) [5]. In Section 5 we will perform simulations for the SDW-MWF and the ITF extension for a simple scenario consisting of one speech source and one noise source and we will investigate the noise reduction performance and the binaural cue preservation.

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2. CONFIGURATION AND NOTATION

Consider the binaural hearing aid configuration in Figure 1, where the left and the right hearing aid have a microphone array consisting of M_0 and M_1 microphones. In the frequency-domain, the m th microphone signal in the left hearing aid $Y_{0,m}(\omega)$ can be written as

$$Y_{0,m}(\omega) = X_{0,m}(\omega) + V_{0,m}(\omega), \quad m = 0 \dots M_0 - 1, \quad (1)$$

where $X_{0,m}(\omega)$ and $V_{0,m}(\omega)$ represent the speech and the noise component. Similarly, the m th microphone signal in the right hearing aid is $Y_{1,m}(\omega) = X_{1,m}(\omega) + V_{1,m}(\omega)$.

Assuming some sort of communication (e.g. wireless link) between both hearing aids, all microphone inputs can be used to generate an output for the left and the right ear. We define the M -dimensional signal vector $\mathbf{Y}(\omega)$, with $M = M_0 + M_1$, as

$$\mathbf{Y}(\omega) = [Y_{0,0}(\omega) \dots Y_{0,M_0-1}(\omega) \quad Y_{1,0}(\omega) \dots Y_{1,M_1-1}(\omega)]^T.$$

The signal vector can be written as $\mathbf{Y}(\omega) = \mathbf{X}(\omega) + \mathbf{V}(\omega)$, where $\mathbf{X}(\omega)$ and $\mathbf{V}(\omega)$ are defined similarly as $\mathbf{Y}(\omega)$. The output signals for the left and the right ear are equal to

$$Z_0(\omega) = \mathbf{W}_0^H(\omega)\mathbf{Y}(\omega), \quad Z_1(\omega) = \mathbf{W}_1^H(\omega)\mathbf{Y}(\omega), \quad (2)$$

with $\mathbf{W}_0(\omega)$ and $\mathbf{W}_1(\omega)$ M -dimensional complex vectors. We define the $2M$ -dimensional stacked weight vector $\mathbf{W}(\omega)$ as

$$\mathbf{W}(\omega) = \begin{bmatrix} \mathbf{W}_0(\omega) \\ \mathbf{W}_1(\omega) \end{bmatrix}. \quad (3)$$

The output signal for the left ear can be written as

$$Z_0(\omega) = Z_{x0}(\omega) + Z_{v0}(\omega) = \mathbf{W}_0^H(\omega)\mathbf{X}(\omega) + \mathbf{W}_0^H(\omega)\mathbf{V}(\omega),$$

where $Z_{x0}(\omega)$ and $Z_{v0}(\omega)$ represent the speech and the noise component. Similarly, the output signal for the right ear $Z_1(\omega) = Z_{x1}(\omega) + Z_{v1}(\omega)$. For conciseness, we will omit the frequency-domain variable ω in the remainder of the paper.

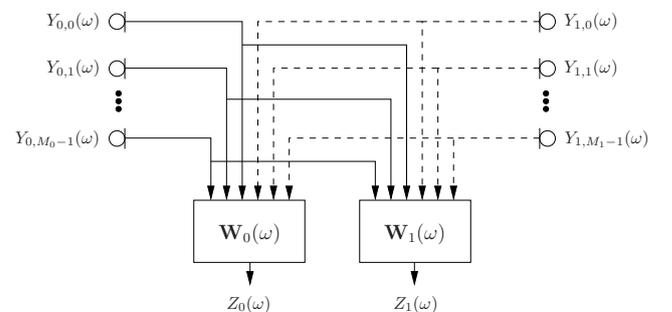


Figure 1: Binaural hearing aid configuration

3. BINAURAL NOISE REDUCTION TECHNIQUES

In this section we briefly discuss the cost functions for the binaural MWF and the extension with the interaural transfer function.

3.1. Binaural multi-channel Wiener filter (MWF)

The binaural MWF produces an MMSE (minimum-mean-square-error) estimate of the speech component in both hearing aids, hence simultaneously reducing noise and limiting speech distortion [3]. The MSE cost function for the filter \mathbf{W}_0 estimating the speech component X_{0,r_0} in the r_0 th microphone of the left hearing aid and the filter \mathbf{W}_1 estimating the speech component X_{1,r_1} in the r_1 th microphone of the right hearing aid is equal to¹

$$J_{MSE}(\mathbf{W}) = \mathcal{E} \left\{ \left\| \begin{bmatrix} X_{0,r_0} - \mathbf{W}_0^H \mathbf{Y} \\ X_{1,r_1} - \mathbf{W}_1^H \mathbf{Y} \end{bmatrix} \right\|^2 \right\}. \quad (4)$$

In order to provide a trade-off between speech distortion and noise reduction, the speech distortion weighted multi-channel Wiener filter (SDW-MWF) minimizes the weighted sum of the residual noise energy and the speech distortion energy [6]. The binaural SDW-MWF cost function is equal to

$$J_{SDW}(\mathbf{W}) = \mathcal{E} \left\{ \left\| \begin{bmatrix} X_{0,r_0} - \mathbf{W}_0^H \mathbf{X} \\ X_{1,r_1} - \mathbf{W}_1^H \mathbf{X} \end{bmatrix} \right\|^2 + \mu \left\| \begin{bmatrix} \mathbf{W}_0^H \mathbf{V} \\ \mathbf{W}_1^H \mathbf{V} \end{bmatrix} \right\|^2 \right\} \quad (5)$$

where μ provides a trade-off between noise reduction and speech distortion. The filter minimizing $J_{SDW}(\mathbf{W})$ is equal to

$$\mathbf{W}_{SDW} = \mathbf{R}^{-1} \mathbf{r}, \quad (6)$$

with

$$\mathbf{R} = \begin{bmatrix} \mathbf{R}_x + \mu \mathbf{R}_v & \mathbf{0}_M \\ \mathbf{0}_M & \mathbf{R}_x + \mu \mathbf{R}_v \end{bmatrix}, \quad \mathbf{r} = \begin{bmatrix} \mathbf{R}_x \mathbf{e}_0 \\ \mathbf{R}_x \mathbf{e}_1 \end{bmatrix}.$$

\mathbf{R}_x and \mathbf{R}_v are the speech and the noise correlation matrix, i.e. $\mathbf{R}_x = \mathcal{E}\{\mathbf{X}\mathbf{X}^H\}$ and $\mathbf{R}_v = \mathcal{E}\{\mathbf{V}\mathbf{V}^H\}$, and \mathbf{e}_0 and \mathbf{e}_1 are vectors of which only one element is equal to 1 and the other elements are equal to 0, i.e. $\mathbf{e}_0(r_0) = 1$ and $\mathbf{e}_1(r_1) = 1$.

3.2. Extension with the interaural transfer function

In order to control the binaural cues of the speech and the noise component, the cost function in (5) has been extended with terms related to the interaural transfer function (ITF) in [5]. The ITFs of the input speech and noise component are defined as

$$ITF_{in}^x = \frac{X_{0,r_0}}{X_{1,r_1}}, \quad ITF_{in}^v = \frac{V_{0,r_0}}{V_{1,r_1}}. \quad (7)$$

Similarly, the ITFs of the output speech and noise component are defined as

$$ITF_{out}^x(\mathbf{W}) = \frac{\mathbf{W}_0^H \mathbf{X}}{\mathbf{W}_1^H \mathbf{X}}, \quad ITF_{out}^v(\mathbf{W}) = \frac{\mathbf{W}_0^H \mathbf{V}}{\mathbf{W}_1^H \mathbf{V}}. \quad (8)$$

When the binaural cues are to be preserved, the desired output ITFs are equal to the input ITFs in (7). We assume the input ITFs to be constant², such that they can be estimated in a least-squares sense using the correlation matrices as

¹Typically, the first microphone is used, i.e. $r_0 = r_1 = 0$.

²In the case of a single localized source, the input ITF is equal to the ratio of the acoustic transfer functions between the source and the reference microphone signals. In this case, it can also be shown that preserving the ITF is equivalent to preserving the phase of the cross-correlation, i.e. the ITD, and preserving the power ratio, i.e. the ILD.

$$ITF_{des}^x = \frac{\mathcal{E}\{X_{0,r_0} X_{1,r_1}^*\}}{\mathcal{E}\{X_{1,r_1} X_{1,r_1}^*\}}, \quad ITF_{des}^v = \frac{\mathcal{E}\{V_{0,r_0} V_{1,r_1}^*\}}{\mathcal{E}\{V_{1,r_1} V_{1,r_1}^*\}}.$$

The ITF cost function for preserving the binaural cues of the noise component then is defined as

$$J_{ITF}^v(\mathbf{W}) = \mathcal{E} \left\{ \left| \frac{\mathbf{W}_0^H \mathbf{V}}{\mathbf{W}_1^H \mathbf{V}} - ITF_{des}^v \right|^2 \right\}, \quad (9)$$

which, in the case of a single localized source, is equal to

$$J_{ITF}^v(\mathbf{W}) = \frac{\mathcal{E}\{|\mathbf{W}_0^H \mathbf{V} - ITF_{des}^v \mathbf{W}_1^H \mathbf{V}|^2\}}{\mathcal{E}\{|\mathbf{W}_1^H \mathbf{V}|^2\}} = \frac{\mathbf{W}^H \mathbf{R}_{vt} \mathbf{W}}{\mathbf{W}^H \mathbf{R}_{v1} \mathbf{W}} \quad (10)$$

with

$$\mathbf{R}_{vt} = \begin{bmatrix} \mathbf{R}_v & -ITF_{des}^{v,*} \mathbf{R}_v \\ -ITF_{des}^v \mathbf{R}_v & |ITF_{des}^v|^2 \mathbf{R}_v \end{bmatrix} \quad (11)$$

$$\mathbf{R}_{v1} = \begin{bmatrix} \mathbf{0}_M & \mathbf{0}_M \\ \mathbf{0}_M & \mathbf{R}_v \end{bmatrix}. \quad (12)$$

The ITF cost function $J_{ITF}^x(\mathbf{W})$ for the speech component is defined similarly as $J_{ITF}^v(\mathbf{W})$, by replacing the noise correlation matrix with the speech correlation matrix and the desired noise ITF with the desired speech ITF. The total cost function trading off noise reduction, speech distortion and binaural cue preservation is defined as

$$J_{tot}(\mathbf{W}) = J_{SDW}(\mathbf{W}) + \alpha J_{ITF}^x(\mathbf{W}) + \beta J_{ITF}^v(\mathbf{W}) \quad (13)$$

where the parameters α and β enable to put more emphasis on binaural cue preservation for the speech and the noise component. Since no closed-form expression is available for the filter minimizing $J_{tot}(\mathbf{W})$, we will use iterative optimization techniques. Many of these techniques (e.g. quasi-Newton method) are able to exploit the analytical expressions for the gradient and the Hessian, which can be derived using (5) and (10).

4. THEORETICAL ANALYSIS OF BINAURAL MWF

In this section we assume that a single speech source is present, but we do not make any assumptions about the noise sources. We will prove that the binaural SDW-MWF preserves the binaural speech cues, but changes the binaural noise cues to the binaural speech cues. In Section 5 we will show using simulations that by extending the SDW-MWF with ITF terms it is possible to preserve the binaural cues of both the speech and the noise component, without considerably reducing the noise reduction performance.

4.1. Performance measures

The *SNR improvement* is defined as the difference between the output and the input SNR, i.e. for the left hearing aid

$$\Delta SNR_0 = 10 \log_{10} \frac{\mathcal{E}\{|Z_{x0}|^2\}}{\mathcal{E}\{|Z_{v0}|^2\}} - 10 \log_{10} \frac{\mathcal{E}\{|X_{0,r_0}|^2\}}{\mathcal{E}\{|V_{0,r_0}|^2\}}. \quad (14)$$

The SNR improvement ΔSNR_1 for the right hearing aid is defined similarly as ΔSNR_0 . The ITD is defined as the phase of the cross-correlation, i.e. for the noise component

$$c_v^{in} = \mathcal{E}\{V_{0,r_0} V_{1,r_1}^*\}, \quad c_v^{out} = \mathcal{E}\{Z_{v0} Z_{v1}^*\}, \quad (15)$$

such that the *ITD error* can be computed as

$$\Delta ITD_v = \frac{|\angle c_v^{out} - \angle c_v^{in}|}{\pi}. \quad (16)$$

The ITD error for the speech component is defined similarly. Note that ΔITD always lies between 0 and 1. The ILD is defined as the power ratio, i.e. for the noise component

$$P_v^{in} = \frac{\mathcal{E}\{|V_{0,r_0}|^2\}}{\mathcal{E}\{|V_{1,r_1}|^2\}}, \quad P_v^{out} = \frac{\mathcal{E}\{|Z_{v0}|^2\}}{\mathcal{E}\{|Z_{v1}|^2\}}, \quad (17)$$

such that the *ILD error* can be computed as

$$\Delta ILD_v = |10 \log_{10} P_v^{out} - 10 \log_{10} P_v^{in}|. \quad (18)$$

The ILD error for the speech component is defined similarly.

4.2. Single speech source

Assuming that a single speech source is present, the speech signal vector $\mathbf{X} = \mathbf{A}S$, with the vector \mathbf{A} containing the acoustic transfer functions between the speech source and the M microphones on the left and the right hearing aid (including head shadow effect, microphone characteristics and room acoustics) and S the speech signal. Hence, the speech correlation matrix

$$\mathbf{R}_x = P_s \mathbf{A} \mathbf{A}^H, \quad (19)$$

is a rank-1 matrix with $P_s = \mathcal{E}\{|S|^2\}$ the power of the speech signal, such that the filter \mathbf{W}_{SDW} in (6) can be written using the matrix inversion lemma as

$$\mathbf{W}_{SDW,0} = \frac{\mathbf{R}_v^{-1} \mathbf{A}}{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s}} A_{0,r_0}^*, \quad (20)$$

$$\mathbf{W}_{SDW,1} = \frac{\mathbf{R}_v^{-1} \mathbf{A}}{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s}} A_{1,r_1}^*. \quad (21)$$

Hence, the speech and the noise components of the output signal at the left and the right hearing aid are equal to

$$Z_{x0} = \frac{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A}}{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s}} X_{0,r_0}, \quad Z_{x1} = \frac{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A}}{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s}} X_{1,r_1},$$

$$Z_{v0} = \frac{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{V}}{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s}} A_{r_0}, \quad Z_{v1} = \frac{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{V}}{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s}} A_{r_1}.$$

The input cross-correlation and the power ratio for the speech component are equal to

$$c_x^{in} = P_s A_{0,r_0} A_{1,r_1}^*, \quad P_x^{in} = \frac{|A_{0,r_0}|^2}{|A_{1,r_1}|^2}. \quad (22)$$

Since the output cross-correlation and the power ratio for the speech component are equal to

$$c_x^{out} = \frac{(\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A})^2 P_s}{(\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s})^2} A_{0,r_0} A_{1,r_1}^*, \quad P_x^{out} = \frac{|A_{0,r_0}|^2}{|A_{1,r_1}|^2},$$

the SDW-MWF perfectly preserves the ITD and the ILD of the speech component. However, since the output cross-correlation and the power ratio for the noise component are equal to

$$c_v^{out} = \frac{\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A}}{(\mathbf{A}^H \mathbf{R}_v^{-1} \mathbf{A} + \frac{\mu}{P_s})^2} A_{0,r_0} A_{1,r_1}^*, \quad P_v^{out} = \frac{|A_{0,r_0}|^2}{|A_{1,r_1}|^2},$$

the ITD and ILD of the output noise component are equal to the ITD and ILD of the output speech component (and hence also the input speech component), which is obviously not desired.

5. SIMULATION RESULTS

In this section we perform simulations for the SDW-MWF and the ITF extension for a simple scenario consisting of one speech source and one noise source. We will investigate the effect of the parameters α and β in (13) on the noise reduction performance and the preservation of the binaural speech and noise cues.

5.1. Data model

We will assume that the sources are located in the far-field of the microphone arrays in a non-reverberant environment and that all microphones are omni-directional. The speech and the noise source are located at an angle θ_x and θ_v from the head³ ($\theta = 0^\circ$: front, $\theta = 90^\circ$: right). Hence, the speech and the noise components of the microphone signals can be written as

$$\mathbf{X}(\omega) = \mathbf{g}(\omega, \theta_x) S(\omega), \quad \mathbf{V}(\omega) = \mathbf{g}(\omega, \theta_v) V(\omega), \quad (23)$$

with the steering vector $\mathbf{g}(\omega, \theta)$ equal to

$$\mathbf{g}(\omega, \theta) = [g_{0,0}(\omega, \theta) \quad \dots \quad g_{0,M_0-1}(\omega, \theta) \quad (24)$$

$$g_{1,0}(\omega, \theta) \quad \dots \quad g_{1,M_1-1}(\omega, \theta)]^T. \quad (25)$$

Since the microphones are located on a head, the head shadow effect needs to be taken into account, which can be achieved by incorporating the head related transfer functions (HRTF) [8] in the steering vector. We will assume that the same HRTF can be used for all microphones at the left (right) hearing aid, i.e.

$$g_{0,m}(\omega, \theta) = HRTF_0(\omega, \theta) e^{-j\omega\tau_{0,m}(\theta)}, \quad (26)$$

$$g_{1,m}(\omega, \theta) = HRTF_1(\omega, \theta) e^{-j\omega\tau_{1,m}(\theta)}, \quad (27)$$

where $HRTF_0(\omega, \theta)$ and $HRTF_1(\omega, \theta)$ represent the HRTFs for the left and the right ear, and $\tau_{0,m}(\theta)$ and $\tau_{1,m}(\theta)$ represent the delay between the m th microphone at the left/right hearing aid and the reference point at the left/right hearing aid.

In practice sensor noise will always be present. We will assume that sensor noise can be modeled as spatially uncorrelated noise, such that using (23) the noise correlation matrix is equal to

$$\mathbf{R}_v(\omega) = P_v(\omega) [\mathbf{g}(\omega, \theta_v) \mathbf{g}^H(\omega, \theta_v) + \delta \mathbf{I}_M], \quad (28)$$

with $P_v(\omega) = \mathcal{E}\{|V(\omega)|^2\}$ the noise power, and δ the power of the (internal) sensor noise relative to the (external) noise power.

5.2. Noise reduction and binaural cue preservation

We have performed experiments using a speech source at -5° and a noise source at 40° . We have used a 2-microphone array both on the left and the right hearing aid. The microphone distance on the left hearing aid is 2 cm, whereas the microphone distance on the right hearing aid is 1.5 cm. The design frequency $\omega = 2\pi 2000$ rad/s and the sampling frequency $f_s = 16$ kHz. The signal-to-noise ratio $P_s/P_v = 0$ dB and the relative sensor noise power is -20 dB, corresponding to $\delta = 0.01$. The parameter μ in the SDW-MWF cost function in (5) is equal to 1.

Figure 2 depicts the ITD error (16) and the ILD error (18) of the speech and the noise component, and the average SNR improvement $(\Delta SNR_0 + \Delta SNR_1)/2$ for different values of the parameters α and β in (13). When $\alpha = 0$ and $\beta = 0$ (SDW-MWF),

³We will only consider the azimuthal plane, i.e. the elevation $\phi = 0$.

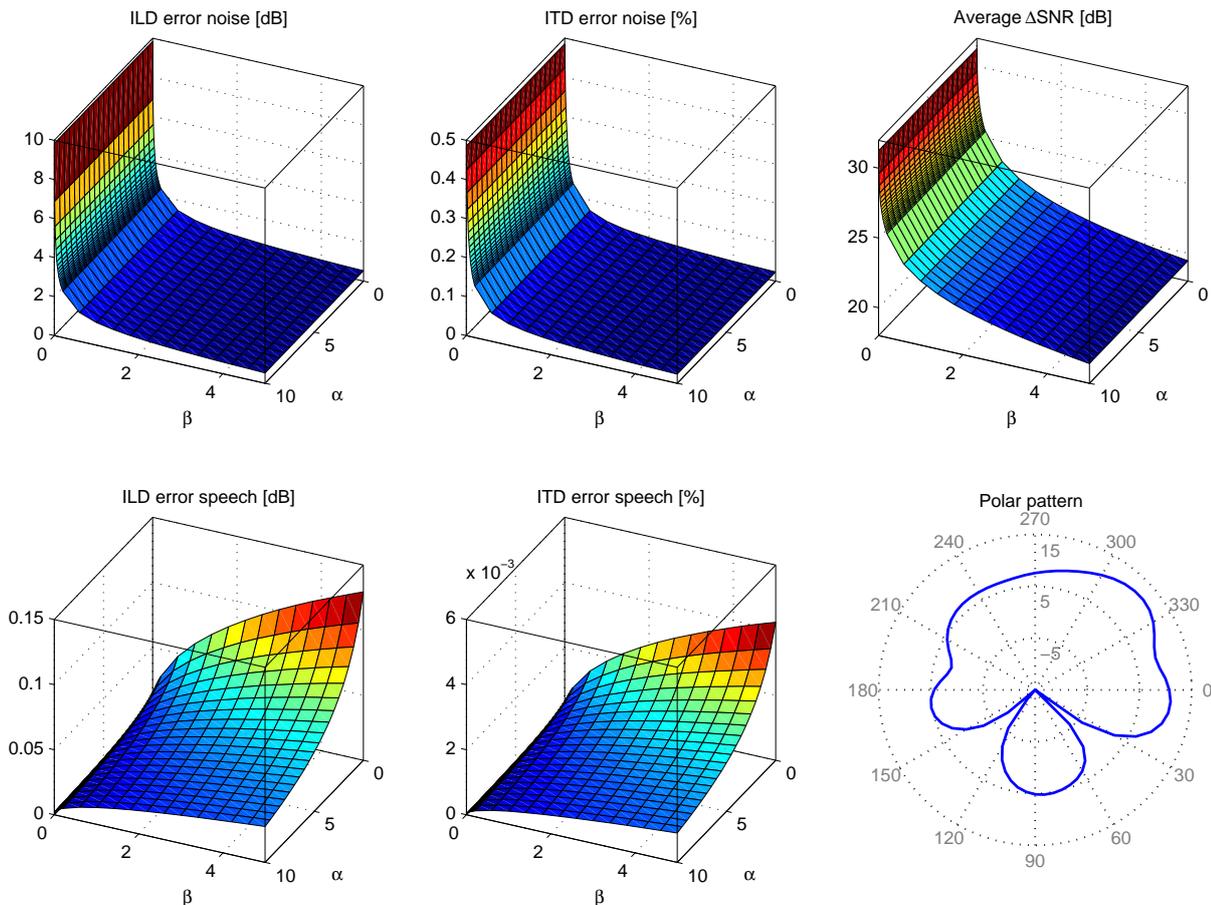


Figure 2: ILD and ITD error for the speech and the noise component and SNR improvement for different values of α and β ; polar pattern for \mathbf{W}_0 ($f = 2000$ Hz, $\alpha = 10$, $\beta = 1$)

the ITD/ILD error for the speech component is equal to zero, but the ITD/ILD error for the noise component is quite large, since the binaural noise cues are equal to the binaural speech cues. By increasing β , the ITD/ILD error for the noise component decreases substantially, whereas the SNR improvement decreases and the ITD/ILD error for the speech component marginally increases. The parameter α can be used for reducing the ITD/ILD error for the speech component caused by increasing β (although this does not appear to be necessary in this scenario). Figure 2 also depicts the polar pattern for the filter \mathbf{W}_0 ($\alpha = 10$, $\beta = 1$). Obviously, a sharp null is present in the direction θ_v .

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