EXTENSION OF THE MULTI-CHANNEL WIENER FILTER WITH ITD CUES FOR NOISE REDUCTION IN BINAURAL HEARING AIDS

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ABSTRACT

This paper presents a novel extension of the multi-channel Wiener filter (MWF) for noise reduction in binaural hearing aids, taking into account binaural localisation cues. By adding a term related to the interaural time difference (ITD) cue of the noise component to the cost function of the MWF, both the ITD cues of the speech and the noise component can be preserved, in addition to significantly improving the signal-to-noise ratio of the microphone signals.

1. INTRODUCTION

Noise reduction algorithms in hearing aids are crucial to improve the speech intelligibility in background noise for hearing impaired persons. Multi-microphone systems are able to exploit spatial in addition to spectral information and are hence preferred to single-microphone systems. Commonly used multi-microphone noise reduction techniques for - monaural and binaural - hearing aids are based on fixed beamforming [1], adaptive beamforming [2, 3, 4, 5], or multi-channel Wiener filtering [6, 7, 8, 9].

In a dual hearing aid system, output signals for both ears are generated, either by operating both hearing aids independently (i.e. a bilateral system) or by sharing information between the hearing aids (i.e. a binaural system). In addition to reducing background noise and limiting speech distortion, another important objective of a binaural algorithm is to preserve the listener's impression of the auditory environment in order to exploit the natural binaural hearing advantage. This can be achieved by preserving the binaural cues, i.e. the interaural time and level difference (ITD, ILD), of the speech and the noise components.

In [1], a fixed beamforming technique has been proposed where the filter weights are optimised in order to maximise the directivity index while restricting the ITD error below some threshold. Binaural adaptive beamforming techniques, based on the Generalised Sidelobe Canceller (GSC), have been proposed in [2, 5]. In [2], the low frequencies of the left and the right signal are passed through unaltered in order to preserve the ITD cues, whereas the high frequencies are adaptively processed using the GSC and then added to the low frequencies. A major drawback of this approach is that not only the speech but also the noise in the low-frequency portion is passed through, significantly comprimising the noise reduction performance. In [5], the preservation of the ITD and the ILD cues is restricted to an angular region around the front, while at other angles the background noise is reduced.

In [8], a binaural multi-channel Wiener filter, providing an enhanced output signal at both ears, has been discussed. In addition to significantly suppressing the background noise, it has been shown experimentally that this algorithm preserves the ITD cues of the speech component. However, the binaural cues of the noise component may be distorted. An extension of the binaural MWF that partially preserves these binaural noise cues has been proposed in [9], but this technique results in a considerable reduction of the noise reduction performance. This paper describes a novel extension of the MWF, adding a term related to the binaural cues of the noise component to the cost function of the MWF. Experimental results show that both the speech and the noise ITD cues can be preserved without compromising the noise reduction performance.

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2. CONFIGURATION AND NOTATION

Consider the binaural hearing aid configuration depicted in Fig. 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 mth microphone signal in the left hearing aid $Y_{0,m}(\omega)$ can be decomposed as

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

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

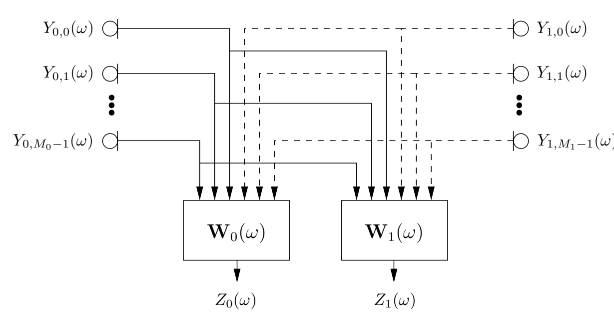

Figure 1: Binaural hearing aid configuration

Assuming that some sort of communication (e.g. wireless link) exists between both hearing aids, we are able to use all microphone inputs from both the left and the right hearing aid 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

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

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 hearing aid $Z_0(\omega)$ and $Z_1(\omega)$ are equal to

$$\begin{aligned} Z_0(\omega) &= \mathbf{W}_0^H(\omega)\mathbf{Y}(\omega) = \mathbf{W}_0^H(\omega)\mathbf{X}(\omega) + \mathbf{W}_0^H(\omega)\mathbf{V}(\omega) , \\ Z_1(\omega) &= \mathbf{W}_1^H(\omega)\mathbf{Y}(\omega) = \mathbf{W}_1^H(\omega)\mathbf{X}(\omega) + \mathbf{W}_1^H(\omega)\mathbf{V}(\omega) , \end{aligned}$$

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

$$\mathbf{W}(\omega) = \begin{bmatrix} \mathbf{W}_0(\omega) \\ \mathbf{W}_1(\omega) \end{bmatrix} , \tag{2}$$

and the $4M$-dimensional real weight vector $\widetilde{\mathbf{W}}(\omega)$ as

$$\widetilde{\mathbf{W}}(\omega) = \begin{bmatrix} \mathbf{W}_R(\omega) \\ \mathbf{W}_I(\omega) \end{bmatrix} , \tag{3}$$

with $\mathbf{W}_R(\omega)$ and $\mathbf{W}_I(\omega)$ the real and the imaginary part of $\mathbf{W}(\omega)$. For conciseness, we will omit the frequency-domain variable ω in the remainder of the paper.

3. BINAURAL MULTI-CHANNEL WIENER FILTERING

The *multi-channel Wiener filter (MWF)* produces a minimum mean-square error (MMSE) estimate of the speech component in one of the microphone signals, hence simultaneously reducing residual noise and limiting speech distortion [6, 7]. Moreover, it has been experimentally shown in [8] that a binaural MWF, producing an estimate of a speech component at both the left and the right hearing aid, preserves the binaural cues of the speech component.

The MSE cost function for the filter $\mathbf{W}_0$ estimating the speech component X_{0,r_0} in the r_0th microphone signal[1] of the left hearing aid is equal to

$$\begin{aligned} J_{MSE,0}(\mathbf{W}_0) &= \mathcal{E}\{|X_{0,r_0} - Z_0|^2\} & (4)\\ &= \mathcal{E}\{|X_{0,r_0} - \mathbf{W}_0^H\mathbf{X}|^2\} + \mathcal{E}\{|\mathbf{W}_0^H\mathbf{V}|^2\}, & (5) \end{aligned}$$

assuming independence between the speech and the noise components. In order to provide a trade-off between speech distortion and noise reduction, the *speech distortion weighted multi-channel Wiener filter (SDW-MWF)* minimises the weighted sum of the residual noise energy and the speech distortion energy [6, 7]. The SDW cost function for the left hearing aid then becomes

$$J_{SDW,0}(\mathbf{W}_0) = \mathcal{E}\{|X_{0,r_0} - \mathbf{W}_0^H\mathbf{X}|^2\} + \mu_0\mathcal{E}\{|\mathbf{W}_0^H\mathbf{V}|^2\} ,$$

where μ_0 provides a trade-off between noise reduction and speech distortion. The SDW cost function $J_{SDW,1}(\mathbf{W}_1)$ for the right hearing aid is defined similarly. The total SDW cost function is

$$\boxed{J_{SDW}(\mathbf{W}) = J_{SDW,0}(\mathbf{W}_0) + J_{SDW,1}(\mathbf{W}_1)} \tag{6}$$

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

where $J_{SDW,0}(\mathbf{W}_0)$ and $J_{SDW,1}(\mathbf{W}_1)$ can be written as

$$\begin{aligned} J_{SDW,0}(\mathbf{W}_0) &= P_0 + \mathbf{W}_0^H(\mathbf{R}_x + \mu_0\mathbf{R}_v)\mathbf{W}_0 - \mathbf{W}_0^H\mathbf{r}_{x0} - \mathbf{r}_{x0}^H\mathbf{W}_0, \\ J_{SDW,1}(\mathbf{W}_1) &= P_1 + \mathbf{W}_1^H(\mathbf{R}_x + \mu_1\mathbf{R}_v)\mathbf{W}_1 - \mathbf{W}_1^H\mathbf{r}_{x1} - \mathbf{r}_{x1}^H\mathbf{W}_1, \end{aligned}$$

with

$$\begin{aligned} \mathbf{R}_x &= \mathcal{E}\{\mathbf{X}\mathbf{X}^H\} \quad \mathbf{r}_{x0} = \mathcal{E}\{\mathbf{X}X_{0,r_0}^*\} \quad P_0 = \mathcal{E}\{|X_{0,r_0}|^2\} \\ \mathbf{R}_v &= \mathcal{E}\{\mathbf{V}\mathbf{V}^H\} \quad \mathbf{r}_{x1} = \mathcal{E}\{\mathbf{X}X_{1,r_1}^*\} \quad P_1 = \mathcal{E}\{|X_{1,r_1}|^2\} . \end{aligned}$$

In practice, we assume that the noise correlation matrix $\mathbf{R}_v$ can be estimated during noise-only periods, and the speech correlation matrix can be computed as

$$\mathbf{R}_x = \mathbf{R}_y - \mathbf{R}_v , \tag{7}$$

where the matrix $\mathbf{R}_y$ is estimated during speech-and-noise periods. Using (2), the total SDW cost function in (6) can be written as

$$\boxed{J_{SDW}(\mathbf{W}) = P + \mathbf{W}^H\mathbf{R}\mathbf{W} - \mathbf{W}^H\mathbf{r} - \mathbf{r}^H\mathbf{W}} \tag{8}$$

with

$$P = P_0 + P_1, \quad \mathbf{r} = \begin{bmatrix} \mathbf{r}_{x0} \\ \mathbf{r}_{x1} \end{bmatrix}, \tag{9}$$

$$\mathbf{R} = \begin{bmatrix} \mathbf{R}_x + \mu_0\mathbf{R}_v & \mathbf{0}_M \\ \mathbf{0}_M & \mathbf{R}_x + \mu_1\mathbf{R}_v \end{bmatrix} . \tag{10}$$

By setting the gradient of $J_{SDW}(\mathbf{W})$ equal to $\mathbf{0}$, the optimal filter minimising $J_{SDW}(\mathbf{W})$ is obtained, i.e.

$$\mathbf{W}_{SDW} = \mathbf{R}^{-1}\mathbf{r} . \tag{11}$$

Using (3), the cost function in (8) can also be written as

$$J_{SDW}(\widetilde{\mathbf{W}}) = P + \widetilde{\mathbf{W}}^T\widetilde{\mathbf{R}}\widetilde{\mathbf{W}} - 2\widetilde{\mathbf{W}}^T\tilde{\mathbf{r}} , \tag{12}$$

with

$$\widetilde{\mathbf{R}} = \begin{bmatrix} \mathbf{R}_R & -\mathbf{R}_I \\ \mathbf{R}_I & \mathbf{R}_R \end{bmatrix}, \quad \tilde{\mathbf{r}} = \begin{bmatrix} \mathbf{r}_R \\ \mathbf{r}_I \end{bmatrix} . \tag{13}$$

4. PRESERVATION OF BINAURAL CUES

Since the SDW-MWF produces an optimal estimate of the speech component in the reference microphone signals at both hearing aids, the binaural cues, i.e. ITD and ILD, of the speech component are generally well preserved [8]. On the contrary, the binaural cues of the noise component may be distorted. In addition to reducing the noise level, it is however also important to (partially) preserve these binaural noise cues in order to exploit the binaural hearing advantage of normal hearing and hearing impaired persons or in order to further process the binaural output signals with a speech enhancement procedure that is based on a difference between speech and noise cues [10, 11].

4.1. Partial estimation of the noise component

An extension of the MWF that partially preserves the binaural noise cues has been proposed in [9]. The objective of the filters is to produce an MMSE estimate of a desired signal that is equal to the sum of the speech component and a scaled version of the noise component in one of the microphone signals, i.e. the cost function for the left hearing aid becomes

$$\bar{J}_{MSE,0}(\mathbf{W}_0) = \mathcal{E}\{|(X_{0,r_0} + \lambda_0V_{0,r_0}) - \mathbf{W}_0^H\mathbf{Y}|^2\} , \tag{14}$$

with $0 \leq \lambda_0 \leq 1$. When $\lambda_0 = 0$, this cost function reduces to $J_{MSE,0}(\mathbf{W}_0)$. When $\lambda_0 = 1$, the optimal filter obviously is equal to a vector consisting of zeros, except for the r_0th element that is equal to 1, resulting in no noise reduction, but complete preservation of the binaural noise cues. It can be easily shown that all expressions derived in Section 3 remain valid when replacing $\mathbf{r}$ in (9) with

$$\mathbf{r} = \begin{bmatrix} \mathbf{r}_{x0} + \mu_0\lambda_0\mathbf{r}_{v0} \\ \mathbf{r}_{x1} + \mu_1\lambda_1\mathbf{r}_{v1} \end{bmatrix} , \tag{15}$$

with $\mathbf{r}_{v0}$ defined similarly as $\mathbf{r}_{x0}$. As will be experimentally shown in the simulations in Section 5, the ITD cue of both the speech and the noise component can be preserved using this technique. However, this can not be achieved without considerably reducing the noise reduction performance.

4.2. Extension of SDW-MWF with noise ITD cue

In this paper we present a different way to preserve the binaural noise cues by adding a term to the SDW cost function that is related to the ITD cue of the noise component. The total cost function can then be expressed as

$$\boxed{J_{tot}(\mathbf{W}) \;=\; J_{SDW}(\mathbf{W}) + \beta\, \underbrace{|ITD_{out}(\mathbf{W}) - ITD_{in}|^2}_{J_{ITD}(\mathbf{W})}} \tag{16}$$

where β is a weight factor[2]. In this paper we will only consider the ITD cue, but it is also possible to add a term related to the ILD cue. The main challenge is to come up with a perceptually relevant mathematical expression for these binaural cues.

We will express the ITD in the frequency-domain using the phase of the cross-correlation between two signals. The input cross-correlation between the noise components in the reference microphone signals is equal to

$$s = \mathcal{E}\{V_{0,r_0}V_{1,r_1}^*\} = \mathbf{R}_v(r_0, r_1) . \tag{17}$$

Similarly, the output cross-correlation between the noise components in the output signals is equal to

$$\mathcal{E}\{Z_{v0}Z_{v1}^*\} = \mathbf{W}_0^H\mathbf{R}_v\mathbf{W}_1 . \tag{18}$$

We now define the cost function $J_{ITD}(\mathbf{W})$ using the cosine of the phase difference $\phi(\mathbf{W})$ between the input and the output noise cross-correlation[3], i.e.

$$\boxed{\begin{aligned} J_{ITD}(\mathbf{W}) &= 1 - \cos\big(\phi(\mathbf{W})\big) \\ &= 1 - \frac{s_R\,(\mathbf{W}_0^H\mathbf{R}_v\mathbf{W}_1)_R + s_I\,(\mathbf{W}_0^H\mathbf{R}_v\mathbf{W}_1)_I}{\sqrt{s_R^2+s_I^2}\sqrt{(\mathbf{W}_0^H\mathbf{R}_v\mathbf{W}_1)_R^2+(\mathbf{W}_0^H\mathbf{R}_v\mathbf{W}_1)_I^2}} \end{aligned}} \tag{20}$$

where $\cdot_R$ and $\cdot_I$ denote the real and the imaginary part.

[2]The weight factor β could be frequency-dependent, since it is well known that e.g for sound localisation the ITD cue is more important at low frequencies than at high frequencies.

[3]Instead of using the input cross-correlation in (17) as the desired output cross-correlation, it is also possible to use other values. If the output noise components should be perceived as coming from the direction θ, the desired output cross-correlation, incorporating the head shadow effect, is

$$s(\omega) = HRTF_0(\omega,\theta)\,HRTF_1^*(\omega,\theta) , \tag{19}$$

where $HRTF_0(\omega,\theta)$ and $HRTF_1(\omega,\theta)$ are the head related transfer functions for the left and the right ear.

Note that this function is scale-independent, i.e. $J_{ITD}(\mathbf{W}_0, \mathbf{W}_1) = J_{ITD}(\rho_0\mathbf{W}_0, \rho_1\mathbf{W}_1)$, $\forall\rho_0, \rho_1$, and that $0 \leq J_{ITD}(\mathbf{W}) \leq 2$. Using (3), this cost function can be written as

$$J_{ITD}(\widetilde{\mathbf{W}}) = 1 - \frac{\widetilde{\mathbf{W}}^T\widetilde{\mathbf{R}}_{vs}\widetilde{\mathbf{W}}}{\sqrt{\big(\widetilde{\mathbf{W}}^T\widetilde{\mathbf{R}}_{v1}\widetilde{\mathbf{W}}\big)^2 + \big(\widetilde{\mathbf{W}}^T\widetilde{\mathbf{R}}_{v2}\widetilde{\mathbf{W}}\big)^2}} , \tag{21}$$

with

$$\widetilde{\mathbf{R}}_{vs} = \frac{s_R\widetilde{\mathbf{R}}_{v1} + s_I\widetilde{\mathbf{R}}_{v2}}{\sqrt{s_R^2 + s_I^2}} . \tag{22}$$

$$\widetilde{\mathbf{R}}_{v1} = \begin{bmatrix} \bar{\mathbf{R}}_{v,R}^{01} & -\bar{\mathbf{R}}_{v,I}^{01} \\ \bar{\mathbf{R}}_{v,I}^{01} & \bar{\mathbf{R}}_{v,R}^{01} \end{bmatrix}, \quad \widetilde{\mathbf{R}}_{v2} = \begin{bmatrix} \bar{\mathbf{R}}_{v,I}^{01} & \bar{\mathbf{R}}_{v,R}^{01} \\ -\bar{\mathbf{R}}_{v,R}^{01} & \bar{\mathbf{R}}_{v,I}^{01} \end{bmatrix}$$

$$\bar{\mathbf{R}}_v^{01} = \begin{bmatrix} \mathbf{0}_M & \mathbf{R}_v \\ \mathbf{0}_M & \mathbf{0}_M \end{bmatrix} . \tag{23}$$

Using (12) and (21), the total cost function is equal to

$$J_{tot}(\widetilde{\mathbf{W}}) = J_{SDW}(\widetilde{\mathbf{W}}) + \beta\, J_{ITD}(\widetilde{\mathbf{W}}) . \tag{24}$$

Since no closed-form expression is available for the filter minimising this cost function, we will resort to iterative optimisation techniques. Many of these techniques (e.g. quasi-Newton method) are able to exploit the analytical expressions for the gradient and the Hessian of $J_{tot}(\widetilde{\mathbf{W}})$, which can be derived using (12) and (21).

As will be experimentally shown in Section 5, both the binaural speech and noise ITD cues can be preserved using this technique without comprimising the noise reduction performance.

5. EXPERIMENTAL RESULTS

5.1. Set-up and performance measures

The recordings used in the simulations were made in a room with dimensions 11'×11'×8'6", having a relatively low reverberation time ($T_{60} \approx 150$ ms). Two Knowles FG microphones were placed horizontally inside both ears of a KEMAR mannequin ($M_0 = M_1 = 2$), with a microphone spacing of 1 cm. The desired speech source is positioned in front of the head (0°) and consists of English sentences. The noise scenario consists of a multi-talker babble source positioned at 45°. All recordings were performed at a sampling frequency of 16 kHz. For evaluation purposes, the speech and the noise signal were recorded separately. The unbiased broadband SNR of the reference microphone signals at the left and the right hearing aid ($r_0 = r_1 = 0$) is 0 dB and −3.2 dB.

The FFT-size used for frequency-domain processing is $N = 256$. As already mentioned in Section 3, the noise correlation matrices $\mathbf{R}_v^n, n = 0 \ldots N-1$, are estimated during noise-only periods, the matrices $\mathbf{R}_y^n$ are estimated during speech-and-noise periods, and the speech correlation matrices are computed as $\mathbf{R}_x^n = \mathbf{R}_y^n - \mathbf{R}_v^n$. For all simulations we used $\mu_0 = \mu_1 = 1$.

As performance measures we use the *SNR improvement* between the input and the output signal at the left and the right hearing aid, and the *ITD cost function* for the noise and the speech component. The SNR improvement for the left hearing aid is defined as the mean of the SNR improvement in dB over all frequencies, i.e.

$$\Delta\text{SNR}_0 = \frac{10}{N}\sum_{n=0}^{N-1} \log_{10}\frac{\mathbf{W}_0^{n,H}\mathbf{R}_x^n\mathbf{W}_0^n}{\mathbf{W}_0^{n,H}\mathbf{R}_v^n\mathbf{W}_0^n} - \log_{10}\frac{\mathbf{R}_x^n(r_0, r_0)}{\mathbf{R}_v^n(r_0, r_0)} .$$

The SNR improvement for the right hearing aid is defined similarly. The ITD cost function for the noise component is defined as the mean of the cost function $J_{ITD}(\mathbf{W}^n)$ in (20) over all frequencies. The ITD cost function for the speech component is defined similarly, by replacing $\mathbf{R}_v$ with $\mathbf{R}_x$ in (17) and (20).

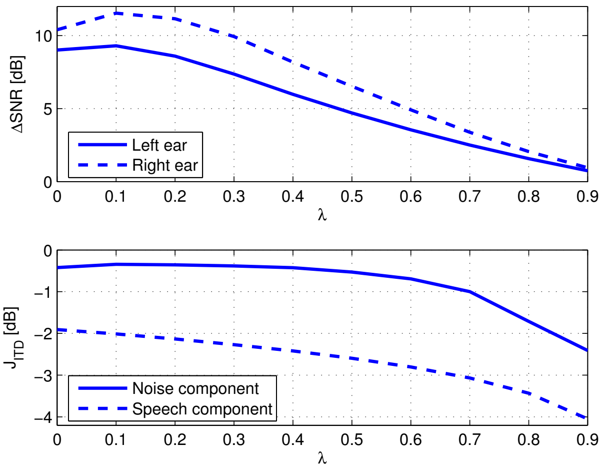

Figure 2: *SNR improvement and ITD cost function using partial estimation of the noise component ($M = 4$, $\mathrm{SNR}_0 = 0\,dB$, $\beta = 0$)*

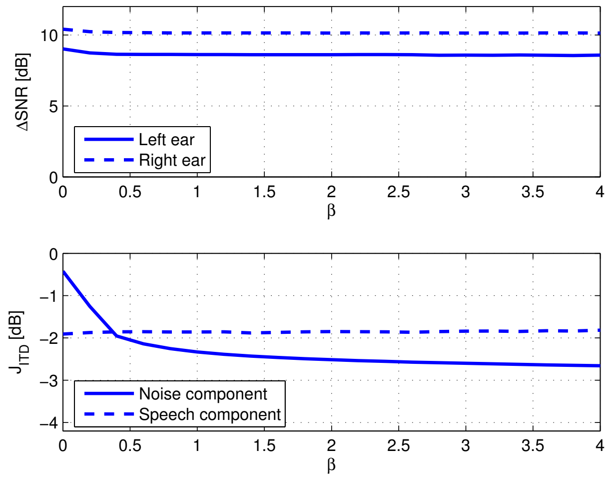

Figure 3: *SNR improvement and ITD cost function using extension of MWF with ITD cues ($M = 4$, $\mathrm{SNR}_0 = 0\,dB$, $\lambda_0 = \lambda_1 = 0$)*

5.2. SNR improvement and preservation of ITD cues

In the *first experiment*, we used the technique described in Section 4.1. Figure 2 shows the SNR improvement and the ITD cost function for different values of the parameter λ ($\lambda_0 = \lambda_1 = \lambda$). For the standard MWF, i.e. $\lambda = 0$, the ITD cost function for the speech component is quite low, but the ITD cost function for the noise component is relatively high, implying that the ITD cue for the speech component is preserved and the ITD cue for the noise component is distorted. As λ increases, the ITD cost function for both the noise and the speech component decreases, but the SNR improvement also significantly degrades (for $\lambda = 1$, $\Delta\mathrm{SNR} = 0$ and $J_{ITD} = -\infty$).

In the *second experiment*, we used the technique described in Section 4.2. Figure 3 shows the SNR improvement and the ITD cost function for different values of the parameter β. As β increases, the ITD cost function for the noise component decreases, the ITD cost function for the speech component slightly increases, and the SNR improvement slightly decreases. Hence, we can conclude that both the speech and the noise ITD cues can be preserved without significantly reducing the noise reduction performance.

6. CONCLUSION

In this paper we have presented an extension of the MWF for binaural hearing aids, which is able to achieve a significant noise reduction while not distorting the ITD cues for both the speech and the noise components. A further extension consists of also adding a term related to the ILD cue to the cost function of the MWF.

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