PERCEPTUALLY MOTIVATED COHERENCE PRESERVATION IN MULTI-CHANNEL WIENER FILTERING BASED NOISE REDUCTION FOR BINAURAL HEARING AIDS

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ABSTRACT

Besides noise reduction an important objective of binaural speech enhancement algorithms is the preservation of the binaural cues of both desired and undesired sound sources. Recently, an extension of the binaural Multi-channel Wiener filter (MWF), namely the MWF-IC, has been presented, which aims to preserve the Interaural Coherence (IC) of the noise component. Since for the MWF-IC a substantial trade-off between noise reduction and IC preservation exists, in this paper we propose a perceptually constrained version of the MWF-IC, where the amount of IC preservation is controlled based on psychoacoustic criterias of the IC discrimination ability of the human auditory system. In addition, we present a simplified version of the MWF-IC, resulting in a decrease of computational complexity. Experimental results show that the perceptually motivated MWF-IC and its simplified version yield a very similar performance and the loss in intelligibility weighted output SNR compared to the binaural MWF can be limited to 0.5 dB, whereas the spatial separation between the output speech and noise component is increased leading to better perceptual results.

Index Terms— Hearing aids, binaural cues, noise reduction

1. INTRODUCTION

Noise reduction algorithms in hearing aids are crucial to improve speech understanding in background noise for hearing impaired persons. For binaural hearing aids, algorithms that exploit the microphone signals from both the left and the right hearing aid are considered to be promising techniques for noise reduction, because in addition to spectral information spatial information can be exploited [1]. In addition to reducing noise and limiting speech distortion, another important objective of binaural noise reduction algorithms is the preservation of the listener's impression of the acoustical scene, in order to exploit the binaural hearing advantage and to avoid confusions due to a mismatch between the acoustical and the visual information. This can be achieved by preserving the binaural cues of all sound sources in the acoustical scene.

To achieve binaural cue preservation, two main concepts for binaural noise reduction have been developed. In the first concept, the multi-channel signals are used to calculate a real-valued gain, where the same gain is applied to the reference microphone in the left, respectively right hearing aid [2, 3, 4]. This processing strategy allows perfect preservation of the binaural cues of both the speech and the noise component, but typically suffers from limited noise reduction performance and possible single-channel noise reduction artifacts. The second concept is to apply a complex-valued filter to all available microphone signals on the left and the right hearing aid, combining spatial and spectral filtering. Using this processing strategy, generally a large noise reduction performance can be achieved, but the binaural cues of the residual noise component are not guaranteed to be preserved.

In [1] the binaural Speech Distortion Weighted Multi-channel Wiener Filter (MWF) has been presented. It has been theoretically proven in [5] that in case of a single speech source the binaural MWF preserves the binaural cues of the speech component but distorts the binaural cues of the noise component such that both speech and noise components comprise the same binaural cues and hence are perceived as coming from the speech direction. Due to this perceptual disadvantage, several extensions of the binaural MWF in [5, 6] and the binaural TF-LCMV [7] have been presented in order to also preserve the so-called Interaural Transfer Function (ITF) of directional interferences. However, for diffuse noise whose characteristics can not be properly described by the ITF but rather by the Interaural Coherence (IC), these extensions are not able to preserve the spatial characteristics. Hence in [8] a MWF-based IC preservation filter, namely the MWF-IC, has been presented allowing to preserve the IC of the residual noise in diffuse noise fields. Since for the MWF-IC a trade-off between IC preservation and output SNR exists in this paper we propose to control the amount of IC preservation based on the IC discrimination abilities of the human auditory system. Furthermore we define a simplified version of the MWF-IC cost function aiming to minimize the interaural correlation between the output noise component of the left and the right hearing aid. In addition, the impact of a rank-1 approximation of the speech correlation matrix on the performance of the MWF, MWF-IC and its simplified version has been investigated.

Experimental results in a cafeteria scenario show that the MWF-IC and its simplified version show a very similar performance in preserving the IC and speech intelligibility weighted output SNR. The degradation of the output SNR compared to the binaural MWF can be limited to 0.5 dB due to the perceptually motivated IC preservation boundaries.

2. CONFIGURATION AND NOTATION

Consider the binaural hearing aid configuration in Figure 1, consisting of a microphone array with M microphones on the left and the right hearing aid. The m-th microphone signal in the left hearing aid $Y_{0,m}(k,l)$ can be written in the frequency-domain as

$$Y_{0,m}(k,l) = X_{0,m}(k,l) + V_{0,m}(k,l)\,, \quad m = 1 \dots M,$$

with $X_{0,m}(k,l)$ and $V_{0,m}(k,l)$ representing the speech and the noise component, k denoting the frequency index and l the block in-

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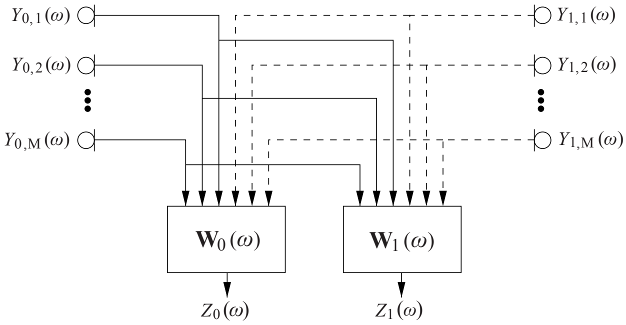

Fig. 1. Binaural hearing aid configuration

dex. The m-th microphone signal in the right hearing aid $Y_{1,m}(k,l)$ is defined similarly. For conciseness we will omit the frequency variable k and the block index l in the remainder of the paper, except where explicitly required. We define the $2M$-dimensional signal vector $\mathbf{Y}$ as

$$\mathbf{Y} = [Y_{0,1} \dots Y_{0,M}\, Y_{1,1} \dots Y_{1,M}]^T. \tag{1}$$

which be written as $\mathbf{Y} = \mathbf{X} + \mathbf{V}$, where $\mathbf{X}$ and $\mathbf{V}$ are defined similarly as $\mathbf{Y}$. Furthermore, we define the $4M$-dimensional stacked weight vector $\mathbf{W}$ as

$$\mathbf{W} = \begin{bmatrix} \mathbf{W}_0 \\ \mathbf{W}_1 \end{bmatrix}. \tag{2}$$

The output signal at the left hearing aid Z_0 is equal to

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

where $Z_{x,0}$ represents the speech component and $Z_{v,0}$ represents the noise component. The output signal at the right hearing aid Z_1 can be defined similarly. The correlation matrices of the signal components are defined as

$$\mathbf{R}_y = \mathcal{E}\left\{\mathbf{Y}\mathbf{Y}^H\right\}, \mathbf{R}_v = \mathcal{E}\left\{\mathbf{V}\mathbf{V}^H\right\}, \mathbf{R}_x = \mathcal{E}\left\{\mathbf{X}\mathbf{X}^H\right\}. \tag{4}$$

which in the remainder of the paper are estimated as

$$\mathbf{R}_y(k) = \frac{1}{L_y}\sum_{i=0}^{L_y-1} \mathbf{Y}(k,i)\mathbf{Y}^{\mathrm{H}}(k,i) \quad \text{speech present}, \tag{5}$$

$$\mathbf{R}_v(k) = \frac{1}{L_v}\sum_{i=0}^{L_v-1} \mathbf{V}(k,i)\mathbf{V}^{\mathrm{H}}(k,i) \quad \text{speech absent}, \tag{6}$$

i.e. the mean value of the L_y available signal vectors when speech and noise is present, respectively the L_v available signal vectors when speech is absent. Assuming the speech and noise component to be uncorrelated the speech correlation matrix $\mathbf{R}_x$ can then be estimated as

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

Due to estimation errors, the speech correlation matrix for a single speech source is not guaranteed to be positive definite and rank-1. We compute a rank-1 representation of the speech correlation matrix based on the eigenvalue decomposition (EVD) of $\mathbf{R}_x$ which has been shown to improve the output SNR [9]. The speech correlation matrix can be decomposed as

$$\mathbf{R}_x = \underbrace{\sigma_1 \mathbf{q}_1 \mathbf{q}_1^H}_{\mathbf{R}_x^1} + \sum_{i=2}^{M} \sigma_i \mathbf{q}_i \mathbf{q}_i^H \tag{8}$$

with σ_i denoting the sorted eigenvalues of $\mathbf{R}_x$ and $\mathbf{q}_i$ denoting the corresponding eigenvectors. σ_1 denotes the largest eigenvalue of $\mathbf{R}_x$. To assure that $\mathbf{R}_x$ is positive semi-definite we set all negative eigenvalues in (8) to 0. The rank-1 matrix $\mathbf{R}_x^1$ can then be used as an approximation of the speech correlation matrix. The input Interaural Coherence (IC) of the noise component is defined as

$$IC_v^{in} = \frac{\mathcal{E}\{V_0 V_1^*\}}{\sqrt{\mathcal{E}\{V_0 V_0^*\}\,\mathcal{E}\{V_1 V_1^*\}}} = \frac{\mathbf{e}_0^T \mathbf{R}_v \mathbf{e}_1}{\sqrt{\mathbf{e}_0^T \mathbf{R}_v \mathbf{e}_0\, \mathbf{e}_1^T \mathbf{R}_v \mathbf{e}_1}}. \tag{9}$$

The vectors $\mathbf{e}_0$ and $\mathbf{e}_1$ are zero column vectors with $\mathbf{e}_0(1) = 1$ and $\mathbf{e}_1(M+1) = 1$ such that $V_0 = \mathbf{e}_0^T \mathbf{V}$ and $V_1 = \mathbf{e}_1^T \mathbf{V}$ are the noise components in the reference microphones.
The interaural correlation between the output noise component of the left and right hearing aid is defined as

$$COR_v^{out} = \mathcal{E}\left\{Z_{v,0} Z_{v,1}^*\right\} = \mathbf{W}_0^H \mathbf{R}_v \mathbf{W}_1. \tag{10}$$

The output IC of the noise component is equal to the normalized interaural correlation in (10) and can be written as

$$IC_v^{out} = \frac{\mathbf{W}_0^H \mathbf{R}_v \mathbf{W}_1}{\sqrt{\mathbf{W}_0^H \mathbf{R}_v \mathbf{W}_0 \mathbf{W}_1^H \mathbf{R}_v \mathbf{W}_1}}. \tag{11}$$

The (real-valued) Magnitude Squared Coherence (MSC) is defined as $MSC = |IC|^2$. The IC of the input and output speech component can be defined similarly as for the noise component.

3. BINAURAL NOISE REDUCTION ALGORITHMS

In this section we briefly review the cost functions for the binaural MWF [1] and the MWF-IC [8]. In addition, we propose a simplified version of the MWF-IC, named MWF-COR, minimizing the interaural correlation of the noise component.

3.1. Binaural multi-channel Wiener filter (MWF)

The binaural MWF produces a minimum mean-square error (MMSE) estimate of the speech component in the reference microphone signal for both hearing aids. The binaural MWF cost function estimating the speech components X_0 and X_1 in the left and the right hearing aid is equal to

$$J_{MWF}(\mathbf{W}) = \mathcal{E}\left\{\left\| \begin{bmatrix} X_0 - \mathbf{W}_0^H \mathbf{X} \\ X_1 - \mathbf{W}_1^H \mathbf{X} \end{bmatrix} \right\|^2 + \mu \left\| \begin{bmatrix} \mathbf{W}_0^H \mathbf{V} \\ \mathbf{W}_1^H \mathbf{V} \end{bmatrix} \right\|^2 \right\}, \tag{12}$$

where the parameter μ enables to provide a trade-off between noise reduction and speech distortion and without loss of generality the first microphone has been used as reference microphone. The filter minimizing $J_{MWF}(\mathbf{W})$ is equal to

$$\mathbf{W}_{MWF} = \mathbf{R}^{-1} \mathbf{r}_x, \tag{13}$$

with

$$\mathbf{R} = \begin{bmatrix} \mathbf{R}_x + \mu \mathbf{R}_v & \mathbf{0}_{2M} \\ \mathbf{0}_{2M} & \mathbf{R}_x + \mu \mathbf{R}_v \end{bmatrix}, \quad \mathbf{r}_x = \begin{bmatrix} \mathbf{R}_x \mathbf{e}_0 \\ \mathbf{R}_x \mathbf{e}_1 \end{bmatrix}. \tag{14}$$

Based on the theoretical analysis in [5], it has been shown in [8] that in case of a single speech source the output IC of the speech and the noise component are the same and equal to the IC of the speech source, i.e.

$$IC_x^{out} = IC_v^{out} = IC_x^{in} = e^{j\angle \frac{A_0}{A_1}}. \tag{15}$$

with A_0 and A_1, the Acoustic Transfer Functions from the speech source to the left and right reference microphone. Equation (15) also implies that $MSC_v^{out} = MSC_x^{out} = 1$, such that in the case of a diffuse noise field the residual noise component would be perceived as a point source coming from the speech direction, which is obviously undesired.

3.2. MWF with Interaural Coherence preservation (MWF-IC)

Aiming at preserving the Interaural Coherence of diffuse noise fields, a coherence preservation term has been defined in [8] as

$$J_{IC}(\mathbf{W}) = \left| \frac{\mathbf{W}_0^H \mathbf{R}_v \mathbf{W}_1}{\sqrt{\mathbf{W}_0^H \mathbf{R}_v \mathbf{W}_0 \mathbf{W}_1^H \mathbf{R}_v \mathbf{W}_1}} - IC_v^{des} \right|^2, \quad (16)$$

where IC_v^{des} represents the desired output IC. When adding this term to the MWF cost function, i.e.

$$J_{MWF-IC}(\mathbf{W}) = J_{MWF}(\mathbf{W}) + \lambda\, J_{IC}(\mathbf{W}), \quad (17)$$

a trade-off between noise reduction and IC preservation arises which can be controlled by the trade-off parameter λ. Since no closed-form expression is available for the filter $\mathbf{W}(\lambda)$ minimizing the cost function J_{MWF-IC}, an iterative numerical optimization method has been used. In order to improve the numerical robustness and the convergence speed, analytical expressions for the gradient and the Hessian of the cost function $J_{MWF-IC}(\mathbf{W})$ have been provided. To avoid the rather computationally complex minimization of (17), in the next section we define a simplified cost function also aiming to preserve the IC of the residual noise component.

3.3. MWF with Correlation Minimization (MWF-COR)

Since the perceived width of a diffuse sound field is mainly determined by the absolute value of the IC [10], we propose a simplified cost function that instead of exactly controlling the complex-valued output IC as in (16) allows to control the output MSC of the noise component by minimizing the squared absolute value of the interaural correlation of the output noise component, i.e.

$$J_{MWF-COR}(\mathbf{W}) = J_{MWF}(\mathbf{W}) + \delta \left| \mathbf{W}_0^H \mathbf{R}_v \mathbf{W}_1 \right|^2. \quad (18)$$

Since the MSC of the output noise component of the MWF is equal to 1 (cf. section 3.1), the output MSC of the MWF-COR can be adjusted to lie between 0 and 1 by carefully controlling the trade-off parameter δ. Similarly as for the MWF-IC, we still need to resort to iterative optimization techniques to find the filter $\mathbf{W}(\delta)$ that minimizes the cost function $J_{MWF-COR}$. Again, the gradient and the Hessian of $J_{MWF-COR}$ have been provided which are computationally less complex compared to the gradient and the Hessian of the MWF-IC cost function in (17).

4. PERCEPTUAL OPTIMIZATION OF THE TRADE-OFF PARAMETERS λ AND δ.

It has been shown in [8] that for the MWF-IC a trade-off between noise reduction and MSC preservation exists. Hence it is crucial to find a suitable trade-off parameter λ in (17) which provides a reasonable trade-off between noise reduction and MSC preservation. Without loss of generality we will only consider the cost function J_{MWF-IC}, but the same procedure can be used for the cost function $J_{MWF-COR}$ in (18) where the trade-off between noise reduction and MSC preservation is determined by the parameter δ.

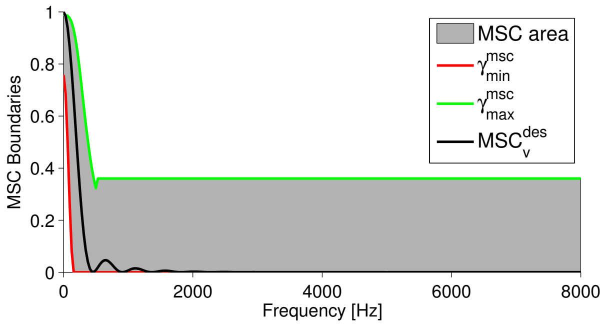

Fig. 2. MSC constraint boundaries for the MWF-IC and MWF-COR

To control the amount of MSC preservation we limit the possible solutions of the optimization problem in (17) by imposing a constraint on the MSC of the output noise component, i.e.

$$\gamma_{min}^{msc} \leq MSC_v^{out}(\mathbf{W}(\lambda)) \leq \gamma_{max}^{msc} \quad (19)$$

where γ_{min}^{msc} and γ_{max}^{msc} are lower and upper bounds for the MSC of the output noise component and $MSC_v^{out} = \left| IC_v^{out} \right|^2$. This inequality constraint limits the range of the trade-off parameter λ such that the output MSC lies within the boundaries γ_{min}^{msc} and γ_{max}^{msc}. The constraint boundaries γ_{min}^{msc} and γ_{max}^{msc} can be defined based on subjective listening experiments evaluating the IC discrimination abilities of the human auditory system in a diffuse noise field. In [11] frequency-dependent IC discrimination thresholds in a diffuse noise field have been measured. It has been shown that the sensitivity to changes in IC from a reference is strongly dependent on the reference IC. For a reference IC close to 1 small changes can be perceived, whereas for a reference IC close to 0 the human auditory system is less sensitive to changes in the IC, which is consistent with the subjective results in other IC discrimination studies [12, 13]. In [14] the IC discrimination sensitivity in a diffuse noise field was examined where the IC below 500 Hz was set to 1 and the IC above 500 Hz was set to 0 approximating a diffuse noise field. The IC above 500 Hz was changed between -1 and 1 with a step size of 0.2 and the results indicate that for frequencies above 500 Hz a deviation of the IC of ± 0.6 is not discriminable from the reference IC of 0. Based on the subjective results from [11] and [14] we define the constraint boundaries γ_{min}^{msc} and γ_{max}^{msc}, which are depicted in Figure 2. Based on the subjective listening tests in [11] and [14] it is assumed that if the output MSC lies within the gray area in Figure 2 the spatial impression of the output noise component is perceptually not discriminable from the spatial impression of a diffuse noise field. The trade-off parameter λ is then determined in an exhaustive search, such that the inequality constraint in (19) is satisfied.

5. EXPERIMENTAL RESULTS

In this section we present simulation results for a cafeteria scenario to compare the performance of the MWF, MWF-IC and the MWF-COR with respect to the intelligibility weighted output SNR and the broadband MSC error of the noise component.

5.1. Setup

Binaural Behind-The-Ear Impulse Responses (BTE-IR) measured in a cafeteria from [15] have been used to generate the speech component in the signals. Each hearing aid was equipped with 2 microphones, therefore in total 4 microphone signals are available. The speaker was located in front of the listener at a distance of 1m. Recorded ambient noise from the same environment was added to the speech component at an intelligibility weighted input SNR of 0 dB. The signals were processed at $f_s = 16$ kHz using an weighted overlap-add framework with a block size of $N = 512$ samples and an overlap of 75% between successive blocks. The noisy signal had a length of 10 s and was preceded by a noise-only signal of 3 s length. The noise-only part was not taken into account during evaluation. The desired IC in the MWF-IC was calculated as a modified sinc-function according to [16]. The parameter μ in all algorithms was set to 1 and the trade-off parameters λ and δ were determined as described in section 4.

5.2. Performance measures

The output *intelligibility weighted SNR* [17] is defined as

$$iSNR = \sum_k I(k) 10 \log_{10} \left(\frac{P_x(k)}{P_v(k)} \right), \quad (20)$$

where $P_x(k)$ and $P_v(k)$ are the PSDs of the speech component, respectively noise component of the output signal. $I(k)$ is a weighting function that takes the importance of different frequency bands for the speech intelligibility into account. The output MSC of the noise component is calculated from IC_v^{out} using (11) where the noise correlation matrix was calculated from the noise component during the 10 s speech + noise period. The broadband MSC error is calculated by averaging the frequency-dependent MSC errors, i.e.

$$MSC_v^{err} = \frac{1}{N-1} \sum_{k=1}^{N-1} \left| MSC_v^{des}(k) - MSC_v^{out}(k) \right|. \quad (21)$$

5.3. Performance Results

The MSC error of the noise component for the different estimates of the speech correlation matrix $\mathbf{R}_x$ and $\mathbf{R}_x^1$ is depicted in Fig. 3. For the MWF the MSC error is noticeable increased if the rank-1 approximation of the speech correlation matrix $\mathbf{R}_x^1$ is used compared to the (possible) full-rank speech correlation matrix $\mathbf{R}_x$. It has been shown in [8] that in case of a rank-1 speech correlation matrix the output MSC is always 1 resulting in a high broadband MSC error as depicted in Fig. 3. If the full-rank speech correlation matrix $\mathbf{R}_x$ is used the output MSC may differ from the expected value of 1 since the rank-1 assumption is violated as already shown in [8]. Nevertheless, for both estimates of the speech correlation matrix the MSC error can be significantly reduced using the MWF-IC and the MWF-COR where the MWF-COR shows a slightly larger decrease compared to the MWF-IC.

The output iSNR in the left and the right HA is depicted in Fig. 4. Using the rank-1 approximation of the speech correlation matrix $\mathbf{R}_x^1$ the output iSNR for the MWF is increased by 1.6 dB in the left HA and 1.2 dB in the right HA compared to the output iSNR using the full-rank speech correlation matrix $\mathbf{R}_x$. Using the full-rank speech correlation matrix $\mathbf{R}_x$, the decrease in output iSNR in the left HA compared to the MWF is 0.2 dB for the MWF-IC and 0.1 dB for the MWF-COR. In the right HA the decrease in output iSNR is 0.3 dB for the MWF-IC and 0.1 dB for the MWF-IC.

If the rank-1 approximation of the speech correlation matrix $\mathbf{R}_x^1$ is used the decrease in output iSNR for the MWF-IC and the MWF-COR compared to the output iSNR of the MWF is up to 0.5 dB since more weight must be put on the IC preservation term in (17),

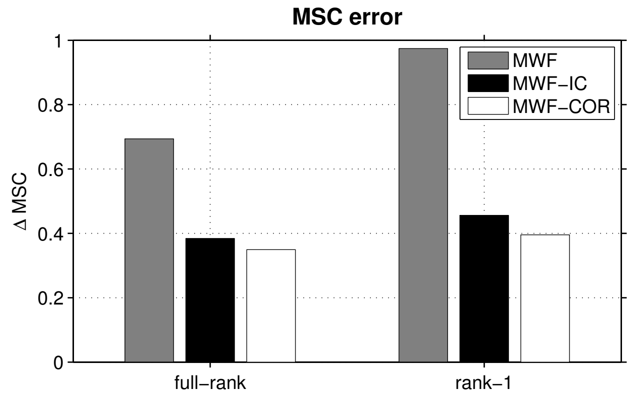

Fig. 3. MSC error of the output noise component for the MWF, MWF-IC and MWF-COR algorithms. The left column shows the result for the full-rank speech correlation matrix $\mathbf{R}_x$ and the right column shows the results for the rank-1 speech correlation matrix $\mathbf{R}_x^1$.

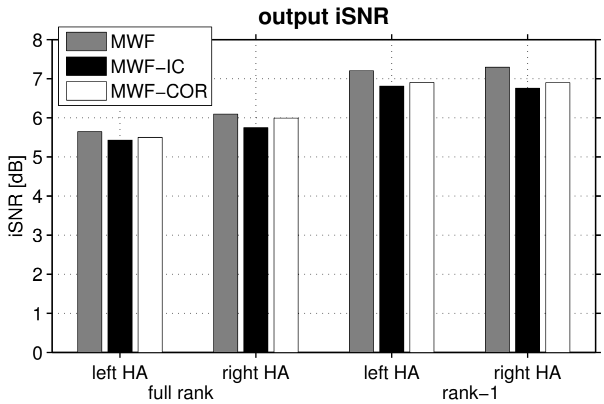

Fig. 4. Output iSNR in the left and right HA for the MWF, MWF-IC and MWF-COR algorithms. The left column shows the result for the full-rank speech correlation matrix $\mathbf{R}_x$ and the right column shows the result for the rank-1 speech correlation matrix $\mathbf{R}_x^1$.

respectively the correlation minimization term in (18) to achieve the same performance in MSC preservation as for the full-rank speech correlation matrix. Nevertheless, the output iSNR for the MWF-IC and the MWF-COR is still larger if the rank-1 speech correlation matrix is used compared to the output iSNR if the full-rank speech correlation matrix is used as depicted in Fig. 4.

6. CONCLUSION

In this paper we have shown that for an realistic noise field in a cafeteria environment the MWF-IC and the MWF-COR yield similar results whereas a slightly better noise reduction and cue preservation performance can be achieved using the MWF-COR. The output iSNR for all algorithms could be significantly increased by using a rank-1 approximation of the speech correlation matrix. The impact of the perceptually constrained MSC boundaries on spatial awareness and speech intelligibility needs to be further investigated in subjective listening experiments.

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