Multi-channel algorithms for wind noise reduction and signal compensation in binaural hearing aids

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Abstract—This paper contains a new approach for wind noise reduction in binaural hearing aids. This approach combines the "single-channel low frequency reduction" algorithm [1] and the "two-channel correlation detector" [2] with a multi-channel binaural hearing aid to compensate the desired signal. The algorithm benefits from the fact that if wind comes from one side, the wind noise mostly occurs in the ear facing away from the wind. As a result of the binaural coupling the signal quality can be improved by the noise reduction and speech compensation.

The evaluation compares the introduced algorithms in terms of signal quality. Sentences from the Oldenburger Sentence Test were recorded in an acoustically optimized wind tunnel at different wind speeds and wind directions. The noise quality of the processed audio-files as well as the ease of listening for the speech was tested with a modified MUSHRA test. Furthermore, the overall quality was tested with a two-alternative forced choice (2AFC) paired comparison test. For all test conditions, the new algorithm performed better than previous approaches.

I. INTRODUCTION

When air flows around obstacles such as head and ear, this air flow gets turbulent and causes wind noise in hearing aids. The spectrum of this signal is dominant at low frequencies and has a noise-like character. The strongest air turbulence occurs above the concha at the microphones of behind-the-ear hearing aids. Levels up to 110dB SPL at a wind speed of 5 m/s can be reached here [3]. Hearing aid users describe the wind noises as irritating. Furthermore, the noise leads to masking of the speech signal.

Common wind noise reduction algorithms use the following properties in order to detect and reduce wind noise

• spectrum is dominant at low-frequencies,
• noise-like character and
• spatial correlation decreases rapidly with the distance of the microphones [4] [5] [6]

Future hearing aids will be linked binaurally. One application for the binaural link is the reduction of wind noise, which can be achieved by making use of the fact that if wind comes from one side, the wind noise mostly occurs in the ear facing away from the wind. This quality is used by the binaural algorithm to reduce wind noise and restore distorted speech.

II. ALGORITHMS

The equations of the following algorithms are described in the short-time discrete fourier transformation domain, where $k$ represents the frequency bin index and $\ell$ the block index.

A. Single-channel low frequency reduction

In single-channel hearing aids no information about the spatial correlation is available. Instead, the spectral shape or the low frequency sound intensity have to be used as indicators for such a system. Therefore, the single-channel low frequency reduction algorithm (1CLR) [1] (figure 1) monitors the shape of the long-term spectrum and calculates the fraction of the total signal power at low frequencies. Wind noise is detected when this fraction exceeds a certain threshold and as a consequence the gain at low frequencies is reduced. [6]

B. Two-channel correlation detector

In monaural hearing aids with two or more microphones the spatial correlation of the signals can be used as an indicator for wind noise. The two-channel correlation detector algorithm (2CCD) [2] (see figure 2) performs in the frequency domain by calculating the Magnitude Squared Coherence (MSC)

$$
MSC_{nm}(k, \ell) = \left| \frac{\phi_{nm}(k, \ell)}{\phi_{nn}(k, \ell)\phi_{mm}(k, \ell)} \right|^2 \in [1, 2], (1)
$$

where $\phi_{nm}(k, \ell)$ indicates the cross power spectral density (CPSD) between the two microphones $n$ and $m$ and $\phi_{nn}(k, \ell)$ and $\phi_{mm}(k, \ell)$ are the auto power spectral densities (APSD) at microphones $n, m$. The PSDs are estimated by the recursive Welch periodogram

$$
\phi_{nm}(k, \ell) = \alpha\phi_{nm}(k, \ell - 1) + (1 - \alpha)X_n(k, \ell)X_m^*(k, \ell). \quad (2)
$$

where $X$ represents the short time spectrum and $\alpha$ is a smoothing time constant. MSC close to one indicates an undisturbed signal and the gain of the current frequency bin is set close to one. On the other hand, MSC decreases with a higher amount of wind noise and the gain is set close to zero. Subtracting the two input signals causes the desired directional behavior of the microphones. Finally, the gains are applied to the spectrum of this directional microphone output. [6]
C. Binaural wind noise compensation

The binaural wind noise compensation algorithm (BWNC) combines the presented algorithms and additionally uses noise-level differences for a binaural compensation of the desired signal. Figure 4 shows the block diagram of our new algorithm.

For the detection of disturbed frequency bins we use the MSC as a basic tool which is close to one in the noise-free case. However, the MSC decreases with a higher amount of wind noise.

Based on a monaural hearing aid with three microphones a wind noise indicator WNI for the channel \( n \) is calculated by determining the maximum of the MSCs between channel \( n \) and all other channels \( m \)

\[
WNI_n(k, \ell) = \max(\text{MSC}_{nm}(k, \ell)) \quad \forall n \neq m \in [1, 2, \ldots, N]. \tag{3}
\]

Under real-world conditions sometimes only one channel is disturbed depending on the wind direction. Therefore, the maximum operator of the MSCs avoids the reduction of undisturbed channels in such a situation when more than two microphones are in use.

The gain \( G \) for the input channels is a function of the WNI

\[
G_n(k, \ell) = \max(\min(\frac{WNI_n(k, \ell) - TH_{\min}}{TH_{\max} - TH_{\min} + G_{\min}}, 1), G_{\min}) \tag{4}
\]

with the thresholds \( TH_{\max} \), \( TH_{\min} \) and the maximal attenuation \( G_{\min} \). Frequency bins with a WNI above \( TH_{\max} \) will pass unmodified, and values lower than \( TH_{\min} \) will be attenuated by \( G_{\min} \). Finally, the output spectrum is computed by

\[
Y_n(k, \ell) = X_n(k, \ell) \cdot G_n(k, \ell). \tag{5}
\]

Spectral smoothing [7] of WNI or \( G \) is recommended to reduce musical noise and allows a shorter averaging time constant \( \alpha \) for computation of the required PSDs \( \phi(k, \ell) \).

Additionally, WNI is the indicator for the binaural compensation which is performed as a cross-copy of frequency bins with a high WNI at one head-side to the bins with a low WNI at the opposite head-side when the difference of the WNI exceeds a given threshold \( d \)

\[
Y_{rn}(k, \ell) = Y_{rn}(k, \ell)|_{WNI_n(k, \ell) < TH_{\min}} \quad \text{and} \tag{6}
\]

\[
Y_{rn}(k, \ell) = Y_{rn}(k, \ell)|_{WNI_n(k, \ell) < TH_{\min}} \quad \text{with the microphones } n \text{ of the hearing aid } r \text{ and } l.
\]
The dominance of the low frequencies of the wind noise spectra is obvious for all directions. The sound pressure level differences between the head-sides are more than 20 dB SPL for $\phi = 90^\circ$ and decrease with a decreasing angle. Hence, our assumption is confirmed.

Figure 6(a) shows channel-wise the spectra of unprocessed noisy speech at a wind speed of 9.1 m/s. In both channels the speech signal is visible. Additionally, the left channel shows distortions at low frequencies caused by the wind. In figure 6(b) the output spectrum of 2CCD is displayed. The wind detection recognizes the distortions at the left channel and attenuates affected frequency bins. Therefore, the left channel is attenuated almost completely. In contrast, the BWNC performs the cross copy of distorted bins and restores the speech signal on the left channel. The output spectrum is depicted in figure 6(c).

**D. Evaluation procedure**

For evaluation purposes we selected the following recordings for independent trials of the modified MUSHRA

- wind speed of 2.8 m/s at $\phi = 0^\circ$,
- wind speed of 5.9 m/s at $\phi = 60^\circ$,
- wind speed of 5.9 m/s at $\phi = 90^\circ$ and $\phi = 90^\circ$,
- wind speed of 9.1 m/s at $\phi = 90^\circ$.

In order to produce reference signals, the selected recordings were processed by the DSB without any wind reduction algorithm. As hidden anchors a 3.5 kHz and a 1.5 kHz low-pass filtered version of those reference signals were used.

The subjects had to score the quality of the remaining noise signal in relation to the reference on the scale -100 (clearly poorer), -50 (poorer), 0 (equal), 50 (better) to 100 (clearly better) with a step-size of one for all trials. One rating of 0 was obligatory. When a subject was not able to recognize the hidden reference, those rankings were excluded to improve the consistency of the results. The same procedure was performed to evaluate the ease of listening for the speech signal, where the subjects were asked to concentrate only on the speech in the noisy signals. Finally, a 2AFC pair comparison test was performed to determine the overall quality of the processed audio signals.

**E. Results**

The results of the noise quality rating are displayed in figure 7(a). For all algorithms and all tested conditions an improvement of the noise quality can be seen. Furthermore, 2CCD and BWND show a similar result for $\phi = 0^\circ$ and $v_w = 2.6m/s$. A similar outcome is obvious for the results of...
the ease of listening test (see figure 7(b)).

The 2AFC test for the evaluation of the overall quality results in the following ranking

- Rank 1: BWNC
- Rank 2: 2CCD
- Rank 3: 1CLR
- Rank 4: unprocessed

with an accordance at a significance level of 0.99 and a consistency of 0.80.

An improvement for all tested conditions can be seen. However, the MUSHRA results show a large interquartile distance of the ratings especially for higher wind speeds for both tests. The cause of this high deviation could be the different rating standards of the subjects during the test. This assumption is supported by the fact that even for the anchor signals a high variation can be seen.

IV. CONCLUSION

In this contribution we introduced a binaural wind noise reduction and signal compensation extension for hearing aids. This BWNC algorithm was compared with well-known algorithms. The results show an improvement through the BWNC at laterally incoming wind. Furthermore, an acceptable quality of the remaining noise can be achieved. For frontal wind the advantages of the BWNC decreases compared to the 2CCD.

At higher wind speeds the microphone membranes can hit the body or the pre-amplifier overload. Both can cause non-harmonic distortions. Fortunately, the BWNC will compensate these distortions if they occur at a single microphone.

Unfortunately, the speech compensation can disturb the stereo image at low frequencies. However, since undisturbed and high frequencies will remain unchanged the binaural cues for localization may be preserved. On the other hand, the disturbance of binaural cues is acceptable in windy situations. The effect on the binaural cues will be a topic in further investigations.

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Processed audio samples can be found under http://projekte.fh-oow.de/hal/windnoisereduction.

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