# Multi-Channel Noise Reduction for Binaural Hearing Aids by Using Short-Time Spectral Attenuation Combined with Noise Estimators for Non Stationary Noise

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**Abstract**

Multi-channel noise reduction (NR) with binaural output is an ongoing research topic for future hearing aids. Especially the preservation of interaural level and time differences is important in order to keep the spatial impression intact. In this contribution we present an extended noise reduction algorithm based on the two-channel NR scheme from Kim et al [1]. The algorithm has a structure similar to the generalized sidelobe canceler (GSC) proposed by Griffith and Jim [2], however, instead of reducing the noise components by cancellation, short-time spectral amplitude filtering is used. We extended and tested this algorithm with different compensation filters to reduce the estimation error as well as musical noise. As a new approach a compensation filter based on minimum statistics will be presented. This method has the advantage that no voice activity detector (VAD) is necessary. In the evaluation we will compare different filter estimation techniques on a realtime developing platform for hearing aid algorithms. Furthermore, listening tests will confirm, that our algorithm, if well adjusted, can produce noise reduced output signals while preserving the speech signal quality.

## Introduction

Currently, devices for the hearing impaired can be separated into monaural fitting with one aided ear and bilateral fitting with two independent processing hearing aids. Both methods can be improved with additional independent beamformer algorithms on each ear. These algorithms use the spatial distribution of sound sources to improve the Signal-to-Noise-Ratio (SNR) between a frontal source and spatially separated noise sources. Common beamformers in hearing aids have multichannel inputs on one side of the head to calculate a single, monaural output. One standard solution is the delay and subtract beamformer and its adaptive counterpart the adaptive differential microphone beamformer (ADM) [3]. Another solution would be the superdirective beamformer [4].

By the miniaturization of receivers and transmitters the binaural link of hearing aids will be common in hearing aids of the next generation. The benefit of the binaural link could result in more information about the spatial distribution of the noise-field by the larger distance of binaural positioned microphones compared to a monaural setup. Additionally, the communication of the hearing aids will allow a better feedback control [5].

For noise reduction algorithm one requirement of binaural algorithms is that two independent output signals are provided which preserve the interaural level and time differences in order to keep the spatial impression intact. However, in order to limit the needed data transfer rate (and therefore battery power) only a limited number of channels should be transmitted.

May [6] shows that the simple linking of two monaural algorithms can result in a significant higher noise reduction. However, the unfavorable position of beam nulls results in a narrow beam in the desired look direction which can cause signal cancellation. Alternative solutions of the aspired algorithms are the published binaural input output beamformer (BIOB) by Lotter and Vary [7] as well as binaural spectral subtraction beamformer (BSSB) proposed in this contribution. In the following we will present the overall structure of the algorithm and the different versions for the needed compensation filter. In section three the evaluation method will be explained followed by the results. Finally, we will draw some conclusions of our work.

## Binaural spectral subtraction beamformer algorithm

This extended noise reduction algorithm with binaural input and binaural output signals use the spectral subtraction based on minimum statistics by Martin [8] to estimate a compensation filter. The block diagram in figure 1 shows the structure of the introduced algorithm.

The BSSB weights the binaural input channels $S_\mathrm{l}$ and $S_\mathrm{r}$ with the compensation filter $|H_\mathrm{spec}|$ to achieve the headside dependent output signals $Y_\mathrm{l}$ and $Y_\mathrm{r}$ resulting in preserved interaural level and time differences

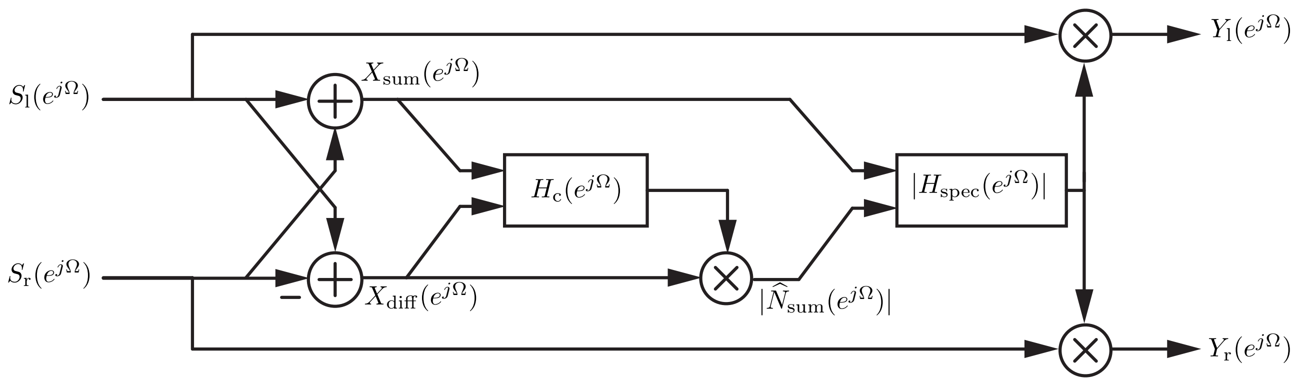


**Figure 1:** Block diagram of the binaural spectral subtraction beamformer algorithm. The binaural microphone signals are filtered with an compensation filter by using the spectral subtraction. Interaural level and time differences will be preserved.

$$Y_l(e^{j\Omega}) = S_l(e^{j\Omega}) \cdot |H_{\text{spec}}(e^{j\Omega})|, \quad (1)$$
$$Y_r(e^{j\Omega}) = S_r(e^{j\Omega}) \cdot |H_{\text{spec}}(e^{j\Omega})|. \quad (2)$$

To compute the compensation filter a frontal signal source in a noisefield is constrained. The subtraction of the binaural input channels leads to the signal $X_{\text{diff}}$ which contains noise only

$$X_{\text{diff}}(e^{j\Omega}) = S_l(e^{j\Omega}) - S_r(e^{j\Omega}). \quad (3)$$

By adding the input channels the signal $X_{\text{sum}}$ contains noise as well as the source signal

$$X_{\text{sum}}(e^{j\Omega}) = S_l(e^{j\Omega}) + S_r(e^{j\Omega}). \quad (4)$$

A compensation of the differences between these signals caused by the spatial position of the microphones is necessary and can be realized by a correction filter $\widehat{H}_c$. In frequency domain $\widehat{H}_c$ can be estimated by the following Wiener-Lee-equations

$$|\widehat{H}_{c,\text{WL1}}(e^{j\Omega})|^2 = \frac{\Phi_{N_{\text{sum}}N_{\text{sum}}}}{\Phi_{N_{\text{diff}}N_{\text{diff}}}}, \quad (5)$$
$$\widehat{H}_{c,\text{WL2}}(e^{j\Omega}) = \frac{\Phi_{N_{\text{sum}}N_{\text{diff}}}}{\Phi_{N_{\text{diff}}N_{\text{diff}}}}, \quad (6)$$

with the auto power densities $\Phi_{N_{\text{sum}}N_{\text{sum}}}$, $\Phi_{N_{\text{diff}}N_{\text{diff}}}$ and the cross power density $\Phi_{N_{\text{sum}}N_{\text{diff}}}$. In this contribution we use a multichannel extension of the minimum statistics algorithm [9] to estimate the needed power densities. This method results in a good estimation in non stationary noise fields and has the advantage that no voice activity detector is necessary.

Now the noise signal $\widehat{N}_{\text{sum}}$ can be estimated by weighting the difference signal $X_{\text{diff}}$ with the correction filter $\widehat{H}_c$ and results in three different estimations of $\widehat{N}_{\text{sum}}$

$$|\widehat{N}_{\text{sum1}}(e^{j\Omega})| = \sqrt{|X_{\text{diff}}(e^{j\Omega})|^2 \cdot |\widehat{H}_{c,\text{WL1}}(e^{j\Omega})|^2} \quad (7)$$
$$|\widehat{N}_{\text{sum2}}(e^{j\Omega})| = \left|X_{\text{diff}}(e^{j\Omega})\right| \cdot \left|\widehat{H}_{c,\text{WL2}}(e^{j\Omega})\right| \quad (8)$$
$$|\widehat{N}_{\text{sum3}}(e^{j\Omega})| = \left|X_{\text{diff}}(e^{j\Omega}) \cdot \widehat{H}_{c,\text{WL2}}(e^{j\Omega})\right|. \quad (9)$$

However, by using the introduced correction filter $\widehat{H}_c$ the algorithm now is independent from non stationary noise but is constrained to a stationary spatial placement.

Finally, the compensation filter $|H_{\text{spec}}|$ can be computed by using the well-known spectral subtraction rule

$$|H(e^{j\Omega})| = \max\left(\frac{|X_{\text{sum}}(e^{j\Omega})| - \alpha|\widehat{N}_{\text{sum}}(e^{j\Omega})|}{|X_{\text{sum}}(e^{j\Omega})|}, \beta\right)^{\gamma} \quad (10)$$

with the overestimation factor $\alpha$ and the noise-floor parameter $\beta$ to reduce the estimation error as well as musical noise [10] and the exponent $\gamma$ to blend between wiener filtering and spectral subtraction.

## Evaluation

To evaluate the presented algorithm in terms of sound quality a paired comparison test was conducted with 16-18 normal hearing subjects between 18 and 30 years. In a natural environment (office room) multichannel test signals were recorded by the microphones of two hearing aid dummies (figure 2) fixed bilateral on an artificial head. The used signals were male voice from a frontal speaker and an additional noise of a circularly moving hairdryer as the disturbing signal. Afterwards these signals were processed by the BIOB and BSSB algorithm on a realtime developing platform for hearing aid algorithms to receive binaural output signals.

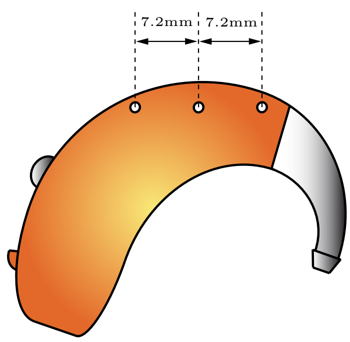


**Figure 2:** for the evaluation used hearing aid dummy, containing a receiver and three microphones in a distance of 7.2mm

The noise-floor parameter for both algorithms was set to

| | parameter | value |
|---|---|---|
| BSSB | $\alpha$ (noisefloor) | -20 dB / -6 dB |
| | $\beta$ (overestimation factor) | 1 |
| | $\gamma$ | 1 |

**Table 1:** Selected parameters for the evaluated BSSB algorithm (see equation 10).

-20 dB. Furthermore, in an additional run a noise floor of -6 dB was used to get information about the quality during a realistic fitting. Additionally, Table 1 shows the used parameters of the BSSB algorithm.

In order to identify the greatest benefit of the noise estimation $\widehat{N}_{\text{sum}}$ for the BSSB the test signals were processed by each introduced filter estimation technique. So, each run of the comparison test contained five test signal including an unprocessed stereo reference signal.

The test signals were presented to the subjects via headphones. Furthermore the speech quality and the noise quality were evaluated during independent runs. To interpret the received answers the rank scale and perception distance in combination with the consistency and concordance were computed by using the Bradley-Terry-Luce-modell [11][12].

# Results

Table 2 shows the results of the comparison test for a noise-floor parameter of -20 dB. Assuming a nearly unchanged signal from the desired look direction a speech quality similar to the quality of the stereo signal was expected. Almost all algorithms confirm this expectation. The speech quality is similar or even better than the unprocessed stereo signal. However, the quality of the noise signal is significant worse compared to the unprocessed stereo signal. This may be due to a high amount of musical noise caused by estimation errors during the spectral filtering.

The additional results for a used noise-floor parameter of -6 dB is shown in Table 3. The quality of the speech signal has a non significant concordance and shows low distances for all algorithms in relation to the unprocessed stereo signal. Therefore, only minor differences in speech quality can be assumed what also confirm our expectation. In contrast to the lower noise-floor the noise quality of the introduced BSSB is similar or better evaluated than the unprocessed stereo signal depending on the used noise estimator. Furthermore, the noise quality of the used BIOB algorithm is worse compared to the BSSB algorithm.

# Conclusions

In this contribution some new and encouraging ideas for binaural linking concerning noise estimation were proposed. We introduced the BSSB algorithm based on spectral subtraction in combination with different noise estimators. The advantage of this algorithms is that the noise source can be stationary as long as the spatial location, and therefore the spatial stationarity, is constant. Furthermore, we avoid the using of a voice activity detector by using the minimum statistics algorithm for the noise estimation.

The evaluation shows good results in terms of sound quality while the noise reduction algorithm uses a moderate noise-floor of -6 dB. However, a reduced noise-floor (below -12dB) is not recommendable because of a high amount of musical noise in the output signal. But an effective noise reduction of -6 dB in combination with a good sound quality is a great benefit for a hearing aid user.

However, further investigation based on the introduced binaural spectral subtraction beamformer algorithm is necessary as well as the measuring of speech intelligibility which is not tested yet.

| | rank | algorithm | distance |
|---|---|---|---|
| speech quality | 1 | BSSB WL2b | 0.00 |
| | 2 | BIOB (Lotter and Vary) | 0.05 |
| | 3 | unprocessed (stereo) | 5.52 |
| | 4 | BSSB WL2a | 6.72 |
| | 5 | BSSB WL1 | 22.76 |
| | consistency | 0.78 | |
| | concordance | significant at 0.99 | |
| | subjects | 18 (consistent results: 13) | |
| noise quality | 1 | unprocessed (stereo) | 0.00 |
| | 2 | BSSB WL2b | 7.81 |
| | 3 | BSSB WL2a | 7.89 |
| | 4 | BIOB (Lotter and Vary) | 10.07 |
| | 5 | BSSB WL1 | 17.39 |
| | consistency | 0.87 | |
| | concordance | significant at 0.99 | |
| | subjects | 17 (consistent results: 14) | |

**Table 2:** Results of a paired comparison test to evaluate the presented algorithms in terms of sound quality. The noise reduction of the used algorithms were limited to a noisefloor of -20 dB.

| | rank | algorithm | distance |
|---|---|---|---|
| speech quality | 1 | BSSB WL1 | 0.00 |
| | 2 | BSSB WL2b | 0.01 |
| | 3 | BSSB WL2a | 0.31 |
| | 4 | unprocessed (stereo) | 0.45 |
| | 5 | BIOB (Lotter and Vary) | 0.76 |
| | consistency | 0.67 | |
| | concordance | not significant | |
| | subjects | 16 (consistent results: 7) | |
| noise quality | 1 | BSSB WL1 | 0.00 |
| | 2 | BSSB WL2a | 0.73 |
| | 3 | unprocessed (stereo) | 6.24 |
| | 4 | BSSB WL2b | 7.06 |
| | 5 | BIOB (Lotter and Vary) | 22.70 |
| | consistency | 0.69 | |
| | concordance | significant at 0.99 | |
| | subjects | 16 (consistent results: 8) | |

**Table 3:** Results of a paired comparison test to evaluate the presented algorithms in terms of sound quality. The noise reduction of the used algorithms were limited to a noisefloor of -6 dB.

## Acknowledgement

This research was (partly) funded by grant 01EZ0741 of the German Federal Ministry of Education and Research (BMBF). The views and conclusions contained in this document, however, are those of the authors.

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