Evaluation of noise reduction in digital hearing aids in situations with multiple signal sources

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Features of modern hearing aids such as the digital noise reduction and directional microphones can enhance speech. One way to evaluate the effect of these features is to measure the increase of the signal-to-noise ratio (SNR). To this end, the method of Hagerman and Olofsson is often used. However, only two signals can be distinguished with this method, e.g., speech and noise. Since many realistic situations include more than two signals, an extension of the method of Hagerman and Olofsson for an arbitrary number of signals is introduced. To proof the concept, this extended method is applied to a setup with 9 different signals presented by 8 speakers. This study considers a separation of speech and noise for 8 signal sources. All speakers are positioned around a hearing aid on a circle with a radial distance of $r = 1\,\mathrm{m}$ and an angular distance of 45° between 0° and 360°. Speech is presented from 0° and noise from all 8 directions $(0°, 45°, ..., 360°)$. With this setup, a state-of-the-art hearing aid is analysed for different settings where the digital noise reduction and/or the directional microphones are turned on or off. As a result, the SNR for all directions can be investigated individually. This demonstrates the practicability of the extended method.

INTRODUCTION

Speech intelligibility in noisy situations decreases depending on the characteristics of speech and noise such as the frequency spectrum and the signal-to-noise-ratio (SNR). As a result, the listening effort for hearing impaired people increases, and leads to a faster exhaustion than for normal hearing people (Holube *et al.*, 2005).

To increase the SNR, the distance between speaker and listener can be reduced or the listening environment can be changed by the listener. Also using hearing aids can lead to a higher speech intelligibility within the same environment. Hearing aids with adaptive features such as digital noise reduction and microphone directionality enhance the SNR. Microphone directionality enhances the spatial SNR and digital noise reduction analyses the signal temporally or spectrally (Chung, 2004). This leads to an increase of speech intelligibility (Bentler, 2005; Brons *et al.*, 2014).

To objectively evaluate speech enhancement in hearing aids, there exists several approaches such as computing the modulation transfer function (Holube *et al.*, 2005),

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performing a percentile analysis (Harries, 2010) or using the method of Hagerman & Olofsson (Hagerman and Olofsson, 2004).

Among these three examples, the method of Hagerman & Olofsson can be used to separate the speech and the noise signal to directly calculate the SNR. However, two signals can be separated only.

To investigate the speech enhancement of hearing aids in a more realistic listening environment, more than two signals from various directions should be considered. This study introduces an extended version of the method of Hagerman and Olofsson. With this version, an arbitrary number of N signals can be considered, Therefore, a maximum number of N directions can be evaluated.

To proof the concept, this extended method is applied to a setup with 9 different signals presented by 8 speakers. Speech is presented from 0°, and noise from all 8 directions $(0°, 45°, ..., 315°)$. With this setup, a state-of-the-art hearing aid is analysed for different settings where the digital noise reduction and/or the directional microphones are turned on or off.

This paper is organised as follows: First, the method of Hagerman & Olofsson is shortly introduced, and then the extension is presented. Next, the measurement setup is explained, and the results are discussed. Finally, the results are summarized and a conclusion is given.

HAGERMAN & OLOFSSON METHOD

Two superpositions $a_{\mathrm{in},1}(t), a_{\mathrm{in},2}(t)$ of two signals $v_1(t), v_2(t)$ are used, in which one superposition takes a phase delay of 180° for $v_2(t)$ into account, where

$$a_{\mathrm{in},1}(t) = v_1(t) + v_2(t), \quad \text{(Eq. 1)}$$
$$a_{\mathrm{in},2}(t) = v_1(t) - v_2(t). \quad \text{(Eq. 2)}$$

Both superpositions are presented to a system, e.g., a hearing aid, successively. $a_{\mathrm{in},1}(t)$ and $a_{\mathrm{in},2}(t)$ are modified by the system, Therefore, two output signals $a_{\mathrm{out},1}(t), a_{\mathrm{out},2}(t)$ are produced.

$v'_1(t)$ and $v'_2(t)$ are calculated out of the output signals with

$$v'_1(t) = \frac{1}{2}(a_{\mathrm{out},1}(t) + a_{\mathrm{out},2}(t)), \quad \text{(Eq. 3)}$$
$$v'_2(t) = \frac{1}{2}(a_{\mathrm{out},1}(t) - a_{\mathrm{out},2}(t)). \quad \text{(Eq. 4)}$$

$v'_1(t), v'_2(t)$ are the input signals processed by the hearing aid. Therefore, they are similar to $v_1(t), v_2(t)$.

Ricketts (2000) showed that traditional test environments with a single noise source located at 180° azimuth cannot be used to accurately predict directional benefit in comparison to other tests with more than one noise sources or real-world environments. Therefore, it is necessary to include more than two competing sound sources, e.g., one speech source and more than one noise sources.

EXTENDED METHOD

An arbitrary number of N signals is given, e.g., $v_1(t), v_2(t), ..., v_N(t)$. The number of input signals equals the number of measurement rounds. By that, N times a superposition of all signals is built. The phase of one signal is inverted within one input signal. Therefore, N input signals are built with

$$\mathbf{a}_{\text{in}}(t) = \begin{pmatrix} a_{\text{in},1}(t) \\ a_{\text{in},2}(t) \\ \vdots \\ a_{\text{in},N}(t) \end{pmatrix} = \begin{bmatrix} -1 & 1 & \dots & 1 \\ 1 & -1 & \dots & 1 \\ \vdots & 1 & \ddots & \vdots \\ 1 & \dots & 1 & -1 \end{bmatrix} \begin{pmatrix} v_1(t) \\ v_2(t) \\ \vdots \\ v_N(t) \end{pmatrix} = \mathbf{Av}(t). \qquad \text{(Eq. 5)}$$

A is the system matrix and of type NxN. Its i-th column changes the phase of signal $v_i(t)$ in one input signal and its j-th row defines the input signal $a_{\text{in},j}(t)$ at the hearing aid input. The system Matrix shall only change the phase and not weight signals. Therefore, its values are only -1 or 1 respectively.

The rank of **A** is N for all $N > 2$, Therefore, the inverse of **A** exists with

$$\mathbf{A}^{-1} = \begin{bmatrix} -\frac{N-3}{2(N-2)} & \frac{1}{2(N-2)} & \cdots & \frac{1}{2(N-2)} \\ \frac{1}{2(N-2)} & -\frac{N-3}{2(N-2)} & \cdots & \frac{1}{2(N-2)} \\ \vdots & \frac{1}{2(N-2)} & \ddots & \vdots \\ \frac{1}{2(N-2)} & \cdots & \frac{1}{2(N-2)} & -\frac{N-3}{2(N-2)} \end{bmatrix} \qquad \text{(Eq. 6)}$$

The condition of the system matrix **A** for all $N > 3$ is $\varphi(\mathbf{A}) = \frac{N-2}{2}$. Therefore, as the impact of measurement tolerances increases, the more signals are used for a setup.

At the output of the hearing aid, all signals can be reconstructed with

$$\begin{pmatrix} v'_1(t) \\ v'_2(t) \\ \vdots \\ v'_N(t) \end{pmatrix} = A^{-1} \begin{pmatrix} a_{\text{out},1}(t) \\ a_{\text{out},2}(t) \\ \vdots \\ a_{\text{out},N}(t) \end{pmatrix}. \qquad \text{(Eq. 7)}$$

With this extension, various complex listening situations with multiple noise and/or speech signals from multiple directions can be analysed. After the separation of all signals $(v_1(t), ..., v_N(t))$, it is possible to compute the absolute SNR between two or multiple signals. Moreover, the SNR at the output can be compared with the SNR at the input. A positive value indicates an enhancement of the desired signal, e.g., speech. In this way the benefit of hearing aid features can be analysed in complex listening situations.

Fig. 1: The measurement setup consists of 8 loudspeakers, one reference microphone and one state-of-the-art hearing aid (BTE) connected to an ear simulator. The BTE is facing the loudspeaker in 0°. Not shown is the RME fireface 800 soundcard and the PC.

MEASUREMENT SETUP

The measurement setup consists of 8 speakers, which are equally distributed on a circle around the hearing aid. The radius of this circle is 1 m and the angular distance between the speakers is 45° (see Fig. 1). In this study, a speech signal, such as the International Speech Test Signal (ISTS), is presented from an angle of 0° and 8 different noise signals are presented from all 8 directions so that N equals 9. The noise signals are incoherent and built out of the ISTS so that the long term average spectrum is equal to the ISTS. The ISTS is presented with 65 dB SPL and an overall SNR of +5 dB is chosen. Thus, the individual level for each noise signal is 51 dB. The hardware of the setup consists of 8 GENELEC speakers of type 8020, a RME Fireface 800 soundcard, a Bruel & Kjaer (B&K) ear simulator according to IEC 60318-4, a reference microphone from B&K of type 4190, and a PC. All measurements as well as the signal analysis are performed with Matlab version 2017a.

The measurement signals $a_{\text{in},1}(t),\ldots,a_{\text{in},9}(t)$ for each loudspeaker are set in a vector with $x(t) = (a_{\text{in},1}(t), 0_{5\,\text{s}}(t), a_{\text{in},2}(t),,\ldots,0_{5\,\text{s}}(t), a_{\text{in},9}(t))^T$, Therefore, an output vector $y(t)$ is recorded. With the superposition of each input signal, a separation of the output signals is possible. With the inverse of the system matrix the signals $v'_1(t),\ldots,v'_9(t)$ are reconstructed.

As transient effects take place within the first 15 seconds, the mean power is calculated in a time window from 15 s to 60 s. The SNR between the ISTS and each of the 8 noise signals is computed separately. Therefore, a spatial analysis of the SNR is possible.

To check the measurement setup, the extended method is applied to the signals of the reference microphone. The polarplot in Fig. 2 shows that the desired SNR of $+5\,\text{dB}$ is measured with an accuracy from $-0.4\,\text{dB}$ to $+1.3\,\text{dB}$. The SNR indicates random behavior independent to a hearing aid setting or the direction. Thus, the results indicate negligible modification of the given SNR for each measurement round.

For the measurements, a state-of-the-art behind the ear (BTE) hearing aid is used. The gain of the device is adjusted by simulating an auditory threshold of type N3 as defined in **?**. For this setup the maximum pressure output, the compression ratio as well as adaptive features such as feedback reduction, wind control are deactivated. As parameters, noise reduction and a fixed microphone directionality are investigated. Therefore, 4 test settings are evaluated:

1. noise reduction off & omnidirectional microphone settings,
2. noise reduction on & omnidirectional microphone settings,
3. noise reduction off & directional microphone settings, and
4. noise reduction on & directional microphone settings.

RESULTS AND DISCUSSION

The output SNR in relation to the input SNR is calculated for each sound source, so that the 4 hearing aid settings are evaluated independently. Figure 3 shows the results for all 4 hearing aid settings as a polar plot. The curves are linearly interpolated between the measurement points. A negative SNR shown in the polar plot indicates an improvement of speech intelligibility. The test setting with a deactivated noise reduction and an omnidirectional microphone setting indicates no modification of the SNR for all directions. This result indicates a good reproduction of the defined SNR for the proposed extended method. The result for an activated noise reduction and an omnidirectional microphone setting shows a negative SNR independent to the direction (see dashed and dotted line in Fig. 3). This is expected due to a working noise reduction in the temporal or frequency domain (Chung, 2004).

Furthermore, the test setting with a deactivated noise reduction and a directional microphone setting indicates a negative SNR dependent to the direction (see continuous

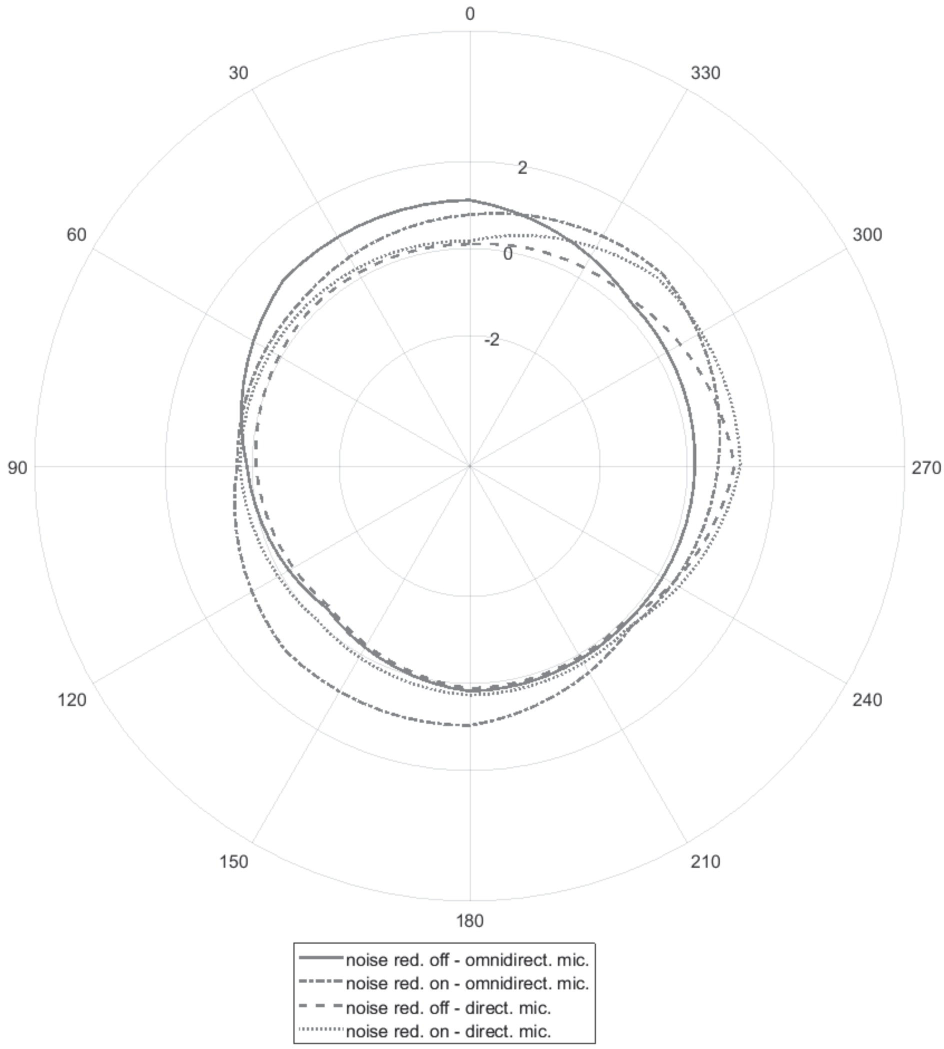

Fig. 2: Output SNR of the reference microphone in relation to the input SNR of the given signal. Each line represents the result of one of the four hearing aid settings. Its allocations are shown in the legend.

line). As the SNR is hardly modificated in 0° azimuth, the lowest SNR can be found in 180° azimuth. This indicates a subcardioid polar pattern, which can typically be found in hearing aids (Bentler, 2005; Chung, 2004).

In addition, an activated noise reduction as well as a directional microphone characteristic show a maximum amount of SNR reduction in 180°. These findings also support Ricketts, who stated in 2000, that a spatial analysis of features should be investigated with more than two sound sources (Ricketts, 2000).

The results in Fig. 3 indicate the proof of the concept of the extended method.

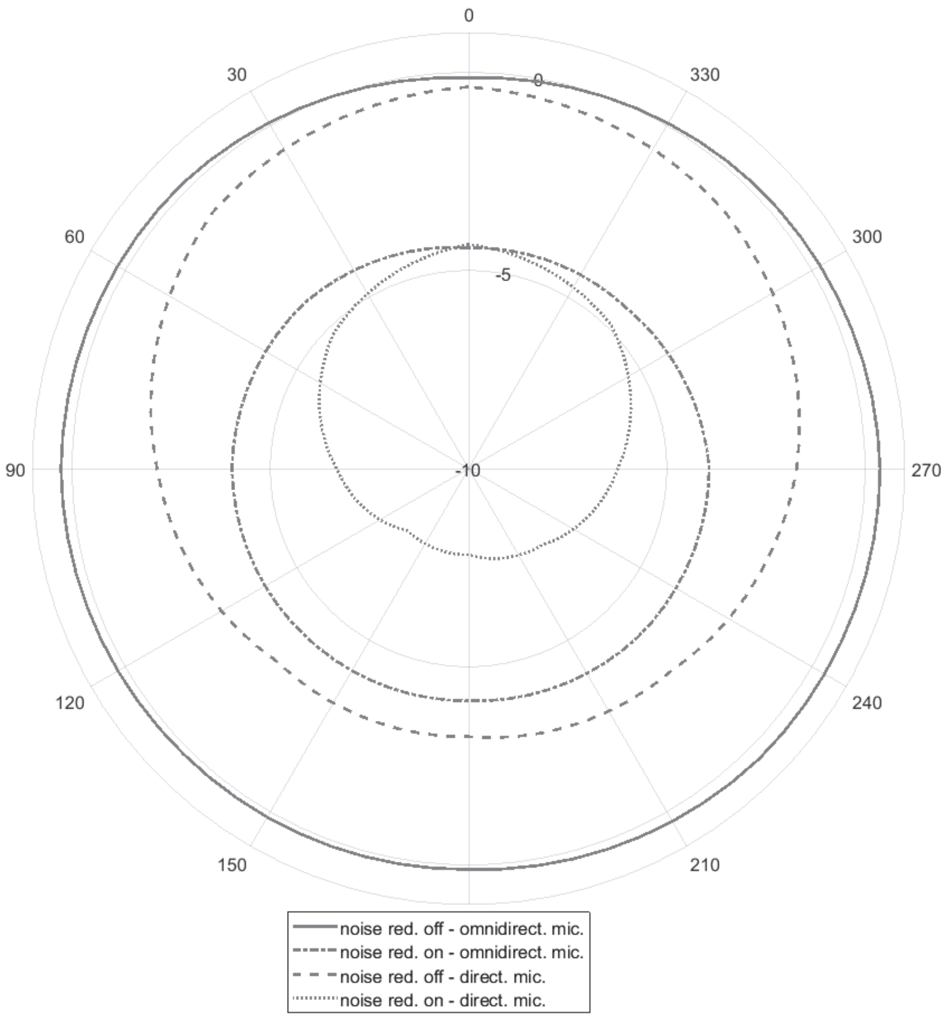

Fig. 3: Output SNR of the hearing aid in relation to the input SNR measured at the reference microphone. The directional microphones of the hearing aid are orientated in 0° direction. Each line represents the result of one of the four hearing aid settings. Its allocations are shown in the legend.

CONCLUSIONS

This paper presents an extended method of the concept presented by Hagerman & Olofsson in 2004. Hagerman and Olofsson introduced a method in which two signals are superpositioned at the input of system and can be reconstructed at the output of a system. As only two signals can be distinguished in this concept, a maximum number of two sound sources can be evaluated. This paper presents an extended method, in which an arbitrary number of N signals can be distinguished. Within the concept, a system matrix is introduced, which describes the phase of each signal for every superpositioned input signal. An inverse of the system matrix can be calculated for more than two signals. Therefore, the system matrix is used to reconstruct the signals at the output of the system. A measurement setup was designed to proof the concept. The results show an enhancement of the SNR independent to the direction for an activated noise reduction and an omnidirectional microphone setting. Also an enhancement of the SNR for a fixed microphone directionality dependent on the direction is measured. A maximum amount of SNR enhancement can be found in 180° for a test setting with an activated noise reduction and a fixed microphone directionality. These test results demonstrate the practicability of the extension of the method by Hagerman and Olofsson.

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