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Multi-microphone noise reduction techniques for hearing aids [1]

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Multi-microphone noise reduction techniques for hearing aids

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Summary: Noise reduction algorithms are crucial to improve the speech intelligibility for hearing impaired people in background noise. Multi-microphone systems exploit spatial in addition to spectral information of the desired and noise signal, and are thus preferred to single microphone procedures.
This study evaluates the performance achievable with a small-sized endfire array. Two different adaptive multi-microphone algorithms are studied: a classical beamforming algorithm, the Generalized Sidelobe Canceller (GSC), and a new, more robust subband algorithm based on a Generalized Singular Value Decomposition (GSVD). In contrast with the classical technique, the GSVD-based algorithm does not make any a priori assumptions about speaker location, microphone characteristics and positions. Deviations from these assumptions are harmful to the performance of the classical approach.
The influence of the number of microphones and their interdistance is investigated. In realistic scenarios, the use of an additional third microphone in the new algorithm results in an additional improvement in intelligibility weighted SNR of 2 to 4 dB compared to 2-microphone systems with the same dimensions.

1 Introduction

Hearing impaired people experience difficulties to understand speech when background noise is present. Compared to normal hearing people, they may need a Signal-to-Noise Ratio (SNR) which is up to 15 dB (on the average 4.5 dB) higher. An increase in SNR of 1 dB may correspond to 15% improvement in intelligibility. Noise reduction algorithms are as a consequence crucial to improve the intelligibility.
In the past, hearing aids were only equipped with one single microphone. However, single microphone noise reduction techniques can only differentiate between signals that have a different spectrum. Quite often the disturbing signals are however also speechlike (e.g. cocktail party situation), such that these techniques will fail.
Multi-microphone techniques offer the advantage of differentiation in space: a beam can be steered towards a desired source and/or zeroes can be steered to undesired directions. These noise reduction techniques are often referred to as *beamforming techniques*. Two classes of *beamforming* techniques can be distinguished: fixed beamforming and adaptive beamforming. In the former, the filter weights in the signal processing algorithm are kept fixed. They are optimized such that a focused beam is steered to the target direction (i.e. the forward-look direction) while reducing background noise coming from other directions as much as possible. In adaptive beamformers, the filter weights are adapted/changed to respond to varying environments. A limited number of interferers with changing characteristics can efficiently be attenuated by steering zeroes towards their directions.
Thanks to the ongoing miniaturization in the electronics, more microphones and more processing power can be built-in in future Behind-The-Ear (BTE) hearing aids. In this paper, we assess the practical improvement in the performance of some adaptive algorithms thanks to additional microphones for a small-sized endfire array having the dimensions of a BTE hearing aid. The

effect of the microphone interdistance is also investigated. The performance assessment is based on physical measures. In a later study, perceptual tests with normal hearing and hearing impaired persons will be done. Section 2 describes the physical performance measures that are used. Section 3 explains the *2* different adaptive noise reduction algorithms that will be evaluated: a classical adaptive beamforming algorithm, called Generalized Sidelobe Canceller (GSC) [1],[2], will be compared with a new, more robust adaptive noise reduction algorithm based on a Generalized Singular Value Decomposition (GSVD) [3]. Section 4 investigates the influence of the number of microphones and their interspacing on the performance.

2 Performance measures

2.1 Improvement in intelligibility

Since the major goal of a noise reduction algorithm is to improve the speech intelligibility, performance measures incorporating speech intelligibility such as the intelligibility weighted SNR, $\text{SNR}_{\text{intellig}}$, have been proposed [4]

$$\text{SNR}_{\text{intellig}} = \sum_i I_i \rho_i, \tag{1}$$

with

- I_i = the band importance of the i-th one-third octave band [5],
- ρ_i = the signal-to-noise ratio SNR_i in the i-th one-third octave band.

2.2 Noise sensitivity and robustness

In addition to the improvement in intelligibility, the sensitivity of a noise reduction algorithm to assumptions and internal noise -also called the noise sensitivity- is an important issue. When the interspacing between adjacent microphones gets smaller than half the acoustical wavelength, large filter weights are needed to reduce environmental noise. Such small arrays are also called *superdirective arrays* [6],[7]. A drawback of these large weights however, is an amplification of uncorrelated noise such as internal noise of amplifiers and microphones, internal and quantization noise of the Analog-to-Digital Convertors (ADC) and wind turbulence. A measure used to quantify the amplification of uncorrelated noise by the array is the noise sensitivity, often defined as the ratio of the array response to mutually uncorrelated white noise sources and the array response to desired signals [8],[9]. In the sequel, we propose the following measure to calculate the noise sensitivity

$$\Psi_{\text{intellig}} = \sum_i I_i \Psi_i, \quad \Psi_i = G_{\text{white},i} - G_{\text{desired},i} \tag{2}$$

with

- $G_{\text{desired},i}$ the gain (in dB) of the desired signal in the i-th one-third octave band i.e. $E_{\text{out},i}^{\text{desired}} - E_{\text{in},i}^{\text{desired}}$, where $E_{\text{out},i}^{\text{desired}}$ and $E_{\text{in},i}^{\text{desired}}$ equal the energy (in dB) of the desired signal at the output of the algorithm and the front microphone signal respectively,
- $G_{\text{white},i}$ the gain (in dB) of uncorrelated, white noise in the i-th one-third octave band.