Artificial Intelligence for Spatial Sound Processing Evaluation in Hearing Aids

Context:

Hearing loss affects over 1.5 billion people worldwide, leading to psychosocial, physical, and cognitive repercussions. 434 million hearing-impaired individuals use hearing aids. In France, the "100% Santé" reform has significantly improved access to hearing aids, but speech intelligibility in noisy environments remains a major challenge. Manufacturers are developing advanced algorithms integrating adaptive microphone directivity and noise reduction to enhance intelligibility. These algorithms detect and localize sound sources, dynamically adjusting microphone directivity and optimizing noise reduction. However, estimating their effectiveness remains difficult due to limitations in current evaluation methods.

The proposed study aims to evaluate the performance of speech enhancement algorithms in noise, focusing on adaptive directivity combined with noise reduction.

Existing evaluation approaches include subjective methods (e.g. speech audiometry in noise), which are time-consuming, and objective methods (analysis of hearing aid output signals). Psychoacoustic indicators such as HASPI 2.0 and MBSTOI fail to present fully the spatial filtering process performed by the hearing aid in spatially environments. Moreover, MBSTOI does not account for hearing loss, and HASPI does not consider the binaural hearing. Studies by Wu Y-H et al. [1] and Aubreville et al. [2] have proposed adaptive directivity evaluation methods, but they face limitations, particularly regarding the variable position of noise sources.

Objective:

This PhD project aims to develop a new method for obtaining polar sound capture diagrams that represent the spatial filtering performed by modern hearing aids. Inspired by the work of Hagerman and Olofsson [3], this method separates sound energy from different directions in a 360° environment without relying on the interfering signal's direction. Initial results from this method are promising, demonstrating its effectiveness for various directional microphone systems and modern hearing aids using binaural connectivity [4].

Method:

The first phase of the project involves extending the approach of Wu Y-H et al. to complex acoustic scenarios with multiple sources or speakers. Controlled laboratory environments will be used to isolate each sound source and apply the Hagerman and Olofsson method. However, this approach has limited applicability in real-world conditions, where acoustic environments vary unpredictably.

The second step of the project will be dedicated to leveraging AI method to overcome these limitations and propose models that could assess hearing aids directivity performance in real-world scenarios. We propose to rely on existing spatial filtering approaches [5, 6] and simulated acoustics environments [7] to identify canonical acoustic scenarios to be explored during recorded sessions. To train the model, the data recorded with hearing aids will be used together with simulated data. Finally, in order to improve robustness to real-world scenarios, we will explore the possibility to exploit data recorded in ecological conditions using either source separation pre-processing [5, 6] or self-supervised learning [8].

From the obtained polar sound capture diagrams, a correlation with existing psychoacoustic indicators will be established. Furthermore, binaural indices derived from models and measurements will be compared with those from normal-hearing listeners HRTF. Finally, we will eventually validate the approach by confronting it to listening test results.

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Key words:

- Hearing aids
- Adaptive directivity
- Complex sound environment
- Deep learning
- Self-supervised learning
- Spatial sound processing

Expected candidate skills:
Audio signal processing, acoustics, and programming. Skills in deep learning and self-supervised learning in AI would be an asset.

Thesis location:
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