A Study on Microphone Array Position Calibration for Hearing Aids

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Abstract

Microphone arrays allow for tasks such as beamforming or direction of arrival (DoA) estimation. In realistic hearing aid scenarios, however, the bilateral array is not fixed since the sub-array on each side of the human head can be moved by the user, possibly leading to a decrease in algorithm performance. These displacements can be accounted for by robust methods taking into account location uncertainties or perturbations. Calibration of the array might be necessary before deployment or on a regular basis due to factors such as hardware aging and environmental effects.

Hence, array calibration for hearing aids is a relevant topic with several constraints to consider. Firstly, hearing aids are usually customisable in order to fit users' heads, which results in altered microphone position setups. Secondly, the use cases of the hearing aids are often evaluated in shoebox-room settings, where background noise and sound reflections can be interferences of the sound sources degrading calibration. This experiment discusses microphone array calibration for hearing aids. The focus of the experiment is on the application of calibration techniques that use a microphone array setup of hearing aids and a set of calibration sources in a home environment, to estimate the actual position of each microphone in the array.

Pyroomacoustics is used to simulate hearing aids in a shoebox room setting with background noise and sound reflections from the floor, walls, and ceiling [1]. Blue dots in Figure 1 indicate the microphone array setup (denoted as *original*) on a B&K HATS [2]. The experiment was repeated 10 times, each time the microphone positions are randomly perturbed with a 40 mm displacement from origins (cf. orange dots in Figure 1). The experiment adopts the Pilot Calibration approach [3], which requires the use of at least 3 sources operating one at a time and from at least 2 different azimuth angles. In this experiment, 4 sources (with a signal length of $10,000$ samples at a sampling rate of 16 kHz and an SNR of 40 dB) are generated for calibration; the sources are located at DoA angles of $(-70°, 20°)$, $(-20°, -40°)$, $(50°, 60°)$ and $(°80, 0°)$ azimuth and elevation.

After 10 calibrations, the average position error was estimated. It could be found that the calibration has reduced the average position error to 0.01318 mm from the perturbed position error of 40 mm. Experimental results demonstrate that the calibration technique can significantly reduce the position error of the microphone array and thus improve the accuracy and consistency of sound amplification of hearing aids.

Further experiments were carried out to examine the perturbation tolerance of the calibration method and the impact of signal length on the calibration accuracy. Figure 2 shows that when the microphone array is perturbed with a displacement of over 320 mm, the calibration error increases significantly such that the calibration method is no longer effective. Figure 3 shows that the calibration error decreases as the signal length (number of samples per signal) increases; it also shows that a signal with $1,000$ samples (62.5 ms at 16 kHz) can still result in a reasonable calibration with an average position error of 0.1364 mm, which illustrates that the calibration method can be implemented at low latency.

The experiment concludes with a discussion of the potential applications of the proposed technique in the field of hearing aids and related technologies.

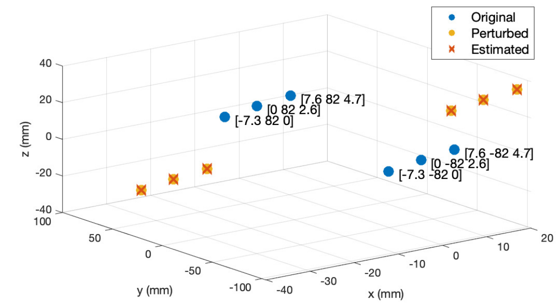

Figure 1: *Microphone Array Calibration*

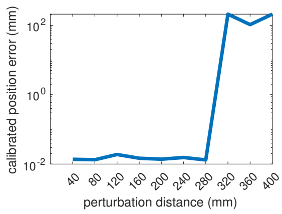

Fig. 2: *Perturbation distance vs. Calibrated Position Error*

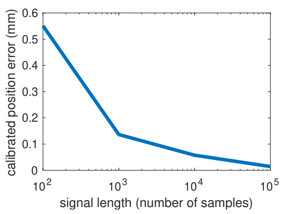

Fig. 3: *Signal Length vs. Calibrated Position Error*

Index Terms: direction finding, array calibration, hearing aids, speech enhancement

1. References

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