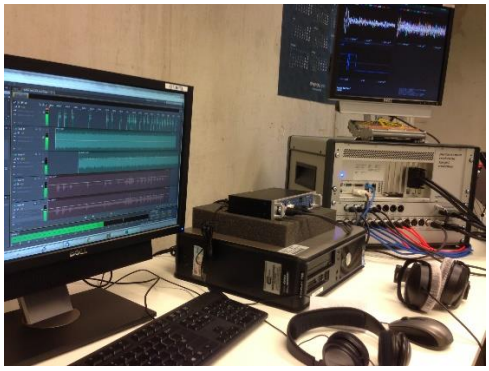
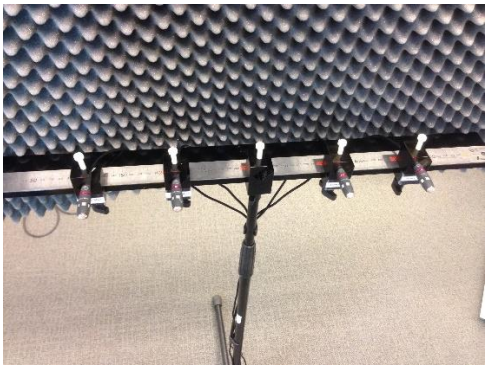
Real-Time Implementation and Evaluation of Distributed Multi-Microphone Beamforming Techniques







Multi-microphone speech enhancement techniques have attracted the attention of many researchers and developers for the last decades, for instance in the context of hearing aids and hands-free communication tasks. Such techniques in general allow exploiting the spatial characteristics of the acoustic scenario, potentially leading to a substantial performance improvement.

The most well-known multi-microphone techniques are so-called *beamforming*. Beamforming refers to any technique that *algorithmically*, rather than *physically*, steers a *beam* towards the location of target speech source(s). To efficiently enhance a recorded speech signal, however, such beamforming might not be sufficient, e.g., in acoustic scenarios where the received microphone signals are also polluted by other interfering speech signals and noise components. In such cases, besides steering beams to enhance the target speech source(s), steering *nulls* should be considered to cancel out the interfering speech signals and noise components. To achieve this goal, an adaptive Linearly Constrained Minimum Variance (LCMV) beamforming can be utilized. A LCMV beamformer essentially aims at minimizing the total power of the beamformer output, under a set of linear constraints that steer the beams and the nulls such that the signals coming from the target directions remain undistorted while signals coming from the interfering directions are rejected.

We can further increase the performance of the LCMV beamformer by utilizing multiple *distributed* microphone arrays. In such cases often these devices exchange pre-processed (compressed) signals with each other instead of exchanging all raw microphone signals. The former is more preferable in many practical scenarios, e.g., when data have to be transmitted via wireless links with a limited bandwidth (context of Wireless Acoustic Sensor Networks).

Your work:

Possible techniques for implementing a distributed adaptive LCMV beamforming in real-time will be first studied. Note that this phase also requires a study of robust Voice Activity Detection (VAD) techniques that are able to work in conjunction with the subspace estimation procedure. The literature study will be followed by a Matlab-and-Simulink-based real-time hardware implementation of the developed distributed techniques using two or three distributed microphone arrays (connected via cable). The real-time setup will be performed and evaluated in the real-time audio lab using a high-performance audio Xpc target machine.

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Number of students: 1 or 2**, Work load:** literature: 20%, implementation: 60%, validation: 20%

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