The Institute for Data Processing at the Technical University of Munich develops an online teleconference system. At the moment the system consists of separate devices and algorithms, which were implemented in previous works. This includes the recording hardware such as the microphone array and the software for the speaker localisation, separation and recognition. In this thesis I connected the algorithms at the recording side to make channel assignment possible. Channel assignment means to put every speaker at an own audio channel to support the human hearing system at the playback side via 3D-sound. The other task of this thesis is to evaluate the teleconference system, in special the speaker identification approach. This is done in two ways. The first compares the developed recognition system to an offline speaker diarization algorithm from the ICSI, to have a minimal reachable threshold for the Diarization Error Rate. This is also used to find the best parameters for the recognition task. In the second evaluation simulated and real conferences are tested to achieve a quality statement about the whole teleconference system in view of the channel assignment.
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