Abstract

The comfort in using speech communication equipments can be considerably increased with hands-free devices. In this case however, the problem arises that also background noise is captured by the microphones. By using a microphone array, the spatial sampling of the acoustical waves is possible. The subsequent signal processing (beamforming) can then amplify signals arriving from the desired direction of the speaker in contrast to signals coming from other directions. Thus it is possible to enhance the desired speech signal.

In this thesis algorithms for adaptive beamforming based on the solution of an eigenvalue problem in the frequency domain are developed and evaluated. The eigenvalue problem is a result of the maximization of the signal-to-noise ratio (Max-SNR) at the output of the beamformer. Two alternative beamformer structures are presented: a filter-and-sum beamformer (FSB) and a generalized sidelobe canceller (GSC).

The FSB shows fast adaptation behaviour and is therefore well suitable to follow a moving speaker. The speech distortion coming along with the Max-SNR criterion, applied separately for every frequency bin, is reduced significantly by some novel post filter methods.

Higher noise suppression in comparison to the FSB can be achieved by the GSC, when the speaker position is nearly constant. For the GSC structure novel realizations of the fixed beamformer and the blocking matrix, also based on the solution of the eigenvalue problem in the frequency domain, are proposed.

The beamforming methods presented here are characterized by their blind adaptation properties. This means that no speaker localization is necessary and the geometrical assembly of the microphones need not to be known. Furthermore, adaptation is possible even in presence of strong permanent active stationary noises.