Abstract

Localizing the source of the audio sound might be important in some kinds of applications, for example in a house or just in one room. Smart homes and other applications where the position of the acoustic sound is critical are some of them.

In this thesis I will present how the processing and filtering of the sound was designed and implemented. The used acoustic localization algorithms will be introduced, how they work, and the fusion of the methods which helps for a more reliable and robust operation. Algorithms for localization are based on delaying the received sampled signal between each other, and summarize them somehow, so that they can define from which way the sound has been arrived. Detecting the position of the source may be incorrect because of the reflection of the sound or the noisy sampled signal, therefore I examined the properties of the human sound and the chosen methods (correlation, beamforming) so that the application could operate in a more reliable and consistent way in real-time.