Abstract

In some applications might be important to record the sound of an acoustic source as clear as possible. These applications can be found in cars, smart houses and other electronic devices which can be actuated by the sound of a person. Another possible case is when we need to record a conference or a presentation and the quality of the sound is crucial.

In my thesis I will present an application, which samples and reconstructs the sound of a given source, mainly speech, in a room. Because of the attributes of a room, the signal-to-noise and the signal-to-interference ratios are low, so we have to consider some kind of algorithm to make it better, to make the speech of a person more understandable by suppressing the noise. In my thesis I dealt with two different types of these algorithms, these were the narrowband and wideband filters, which I will present here. Generally, these programs work in time domain and can be applied as spatial and/or frequency filters.

In my thesis I will show how these algorithms work, and how they can be implemented, and used in my model and in real environments.