Smart Microphone Array

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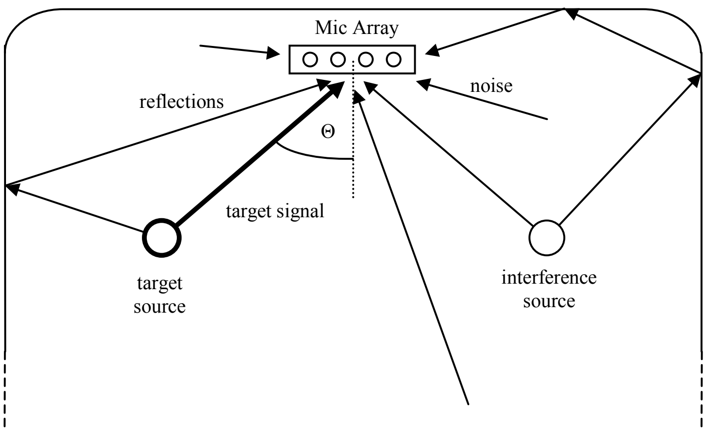
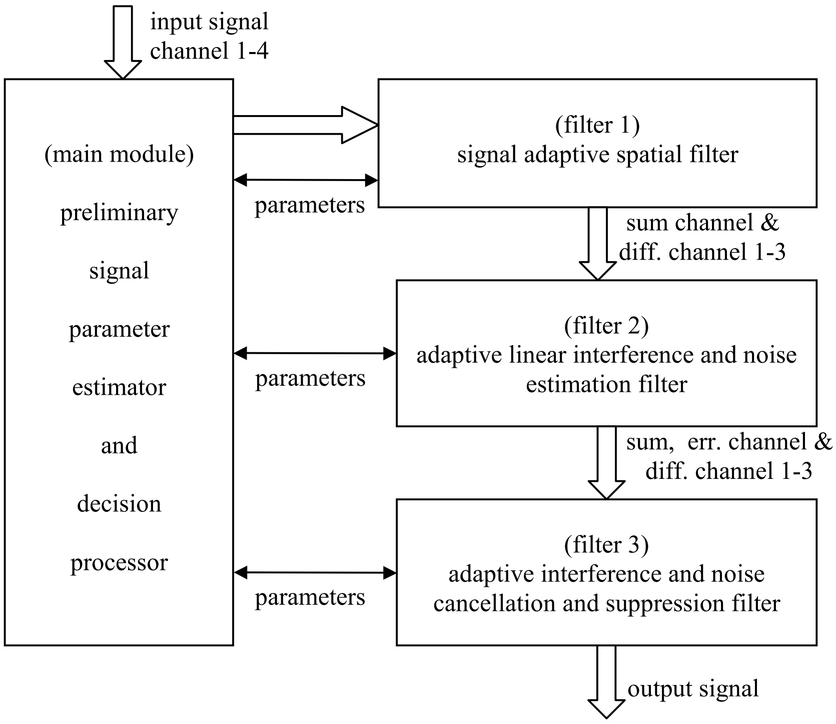
Subject of the project is the development of a method to process multiple audio signals being recorded using a microphone array. The signals, comprising a target superimposed by various interferences, are processed by a multi stage, adaptive filtering method to extract all undesired signal parts and subtract them from the entire spectrum. As a result, a qualitatively enhanced target signal can be achieved. The implementation of microphone arrays enables the application of simple omni-directional instead of directive microphones. Due to adaptive beamforming the overall system is made sensitive to a pre-determined steering direction, though the position of the array does not affect the quality of the output. Thus, it can also be placed in the far field.
An actual application example denotes the enhancement of speech intelligibility for in car communication. Here, the interfering signal comprises background noise (e.g. engine noise, tyre/road interaction noise, etc.) and additional speakers (other passengers). The situation is depicted below:



The processing unit consists of a main module, providing parameter definitions and decisions regarding the conditions of the subsequent program parts, and three cascaded adaptive filter stages that work independent of one another. The first stage performs signal separation yielding a "sum signal", containing target and interference with an improved signal-to-noise ratio (SNR), and three "difference signals", mainly containing interference. These signals comprise the input to the next filter the aim of which is to further enhance the estimation of the interference signal. The third filter implies various computation to obtain an improved SNR and a SNR-dependent, non-linear gain factor that, multiplied with the previously estimated sum signal, results in an interference cancelled and suppressed target signal.



The developing phase includes the generation of a MATLAB program, which is tested with artificial and real audio signals to obtain an optimal adjustment of the parameters. Then the algorithm is implemented into the real time computer music software "pure data" (pd) to ease the demonstration of the overall system's performance and to serve as preparation for a possible DSP implementation.