Open Community Platform for Hearing Aid Algorithm Research

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Abstract

The project "Open community platform for hearing aid algorithm research" funded by the National Institutes of Health (NIH Grant 1R01DC015429-01) aims at sustainable, focused research towards improvement and new types of assistive hearing systems. To this end an open-source software platform for real-time audio signal processing will be developed and made available to the research community including a standard set of reference algorithms. Furthermore, novel algorithms for dynamic and frequency compression, auditory-scene-analysis based noise suppression and speech enhancement and feedback management will be investigated. For a realistic assessment of the benefits of hearing aid algorithms and combinations thereof, instrumental measures of performance in virtual acoustic environments of varying complexity will be included in the algorithm design and optimization. With such a quasi-standard set of benchmarks and the means to develop and integrate own signal-processing methods and measures, the platform enables comparative studies and collaborative research efforts. In addition to an implementation for PC hardware, the system will also be made usable for ARM-processor based hardware to allow pre-development of wearable audio devices – so-called "hearables". This contribution will present underlying previous work and the goals and plans of the project that has started midyear 2016.

Introduction

In the development of signal processing methods for hearing aids a major requirement is a reliable real-time capability of the system. While this constraint is not the most prevalent one when starting the development of new algorithms, it becomes more relevant when focus shifts to evaluation in laboratory subject listening experiments or field test, and is most essential in real-world hearing aid application. Reproducibility of the processing results plays an important role – it needs to remain identical in all cases ranging from offline processing on desktop PCs to low-latency real-time processing on particular mobile hardware platforms. The ultimate integration of hearing aid algorithms is widely conducted by hearing aid manufacturers on proprietary systems. These are not accessible to the research community and underlie commercial constraints. Thus, these proprietary systems are a limiting factor for the translation of results and developments from research to real-world application.

A major goal of the project "Open community platform for hearing aid algorithm research" is to provide an open tool to the hearing aid research community that lowers such boundaries and facilitates translation of advances from the research community into widespread use with hearing assistive systems. The open Master Hearing Aid (openMHA) is a software platform for the development and evaluation of hearing aid algorithms. It builds on the HörTech Master Hearing Aid (MHA, [1, 2]) that already has been used successfully by the hearing aid industry and in academic research. It offers a complete chain of hearing aid signal processing methods that can be used as a reference and in combination with newly developed algorithms.

The openMHA enables researchers to perform offline-processing and real-time signal processing with the required low delay between acoustic input and output of less than 10 milliseconds. It comes with a library for common signal processing tasks and commonly needed services in hearing aid signal processing, like support for acoustic calibration and filterbanks.
In February 2017, a pre-release of the openMHA has been published on GitHub under an open-source license (AGPL3).[3] This pre-release, described in more detail in this contribution, features an initial set of reference algorithms for hearing aid processing, which will be expanded in subsequent releases. Thereby, openMHA provides a growing benchmark for the development and investigation of novel algorithms on this platform in the future.

Project Overview

The project "Open community platform for hearing aid algorithm research" started in July 2016. Yearly releases of the openMHA are milestones (M) in the project schedule that extend the software and its compatibility with different hardware and operating systems as well as the set of algorithms included as plugins.

M1 First official software release (June 2017)
Fully functional version that allows algorithm development under Linux on PC platforms, provides Linux realtime runtime support for PC and Beaglebone black ARM platforms, and includes a set of real-time algorithms that is based on the latest set of Master Hearing Aid algorithms [4, 5, 6, 7]

M2 Extended support
Additional Windows operating system support for algorithm development, extended support for multichannel (6/4 channel) AD/DAconverters on the Beaglebone black and runtime support for the other ARM based platforms

M3 Extended algorithms
Version including an extended set of algorithms for extensive subjective evaluations by the community

M4 Updates
Development kit updates based on the feedback of the community as well as updated versions of the algorithms and new experimental algorithms

M5 Updates
Development kit updates based on the feedback of the community as well as updated versions of the algorithms and new experimental algorithms

Structure of the openMHA

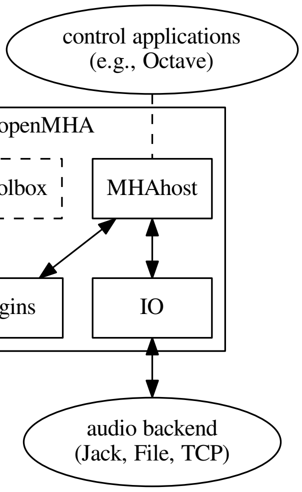

Figure 1: Structure of the openMHA. The openMHA contains a toolbox library "libMHAToolbox", a command line host application, which acts as an openMHA plugin host and provides the configuration interface, and openMHA plugins.

The structure of the openMHA is depicted in Figure 1. It consists of the command line application (MHAhost) which is used to load plugins for signal processing and audio in- and output (IO). Furthermore it can be accessed via external control applications and TCP/IP-based connections and used as runtime configuration interface. A more detailed description of the openMHA software structure and concept can be found in [8].

Pre-release content as of February 2017

As denoted in the project overview above yearly releases of the openMHA are scheduled to fulfill the project milestones. In preparation of the first release (*M1*) a pre-release was made available on GitHub in February 2017 under an open-source license (AGPL3). It provides the full functionality of a binaural, multi-channel hearing aid. Apart from the basic software and signal processing requirements – the openMHA command line application, the toolbox library with filterbanks and other fundamental methods commonly needed for signal processing (not only) in hearing aids and sound IO libraries – a set of plugins that realize basic hearing aid algorithms is included. Furthermore, configuration files that realize the full processing chain depicted in Figure 2 as well as single stages

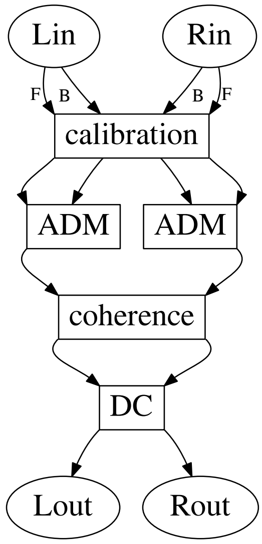

Figure 2: Binaural hearing aid processing chain, realized with the pre-release version of the openMHA.

thereof are provided together with detailed documentation. Given a four-channel binaural hearing aid setup (2 microphones left and right each), a device-dependent sound level calibration is applied in the first step. This is followed by a noise reduction stage, bilateral adaptive differential microphones (ADM), and binaural coherence filtering, to reduce acoustic feedback in the hearing aid devices. At the output of the system, hearing loss and recruitment is compensated, using a multi-band dynamic range compressor (DC). In the following, the latter three plugins are described in more detail.

Adaptive differential microphone plugin

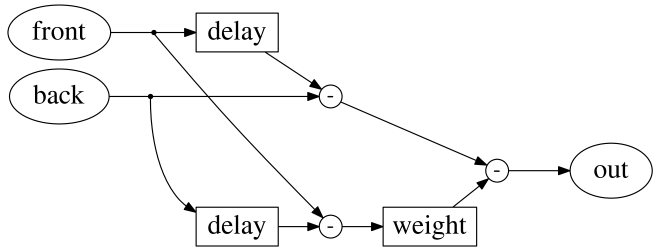

Figure 3: Adaptive differential microphone signal flowchart. The input of the *front* and *back* microphone is combined to a single-channel output after applying a delay and a weighting.

In order to increase the intelligibility of speech in noisy environments adaptive differential microphones (ADM, [9]) aim at the preservation of a target signal while suppressing background noise. In this context, two general assumptions are made: The target speech signal is assumed to be present in the frontal hemisphere of a listener, while noise occurs in the rear hemisphere. ADMs work for pairs of omnidirectional microphones separated by a small distance, and combine a two-channel input to a single-channel output signal by adding up delayed and weighted versions of the input as shown in Figure 3. Here, the input signals from the front and the back microphone of each hearing aid device are used, such that two independent, bilateral ADMs are realized in the given binaural setting. By this means a two-channel, binaural signal is obtained and passed to the following stage.

Binaural coherence filter plugin

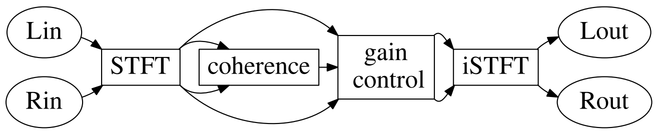

Figure 4: Coherence filter signal flowchart. Binaural coherence-based gain control is applied to the left (Lin) and the right (Rin) input channel in different frequency bands in the STFT domain, transformed back (iSTFT) to a binaural output (Lout, Rout) in the time domain.

At high output levels a sound loop between the hearing aid receivers (outputs) and the closely located microphones (inputs) can emerge, resulting in annoying, self-sustaining beep tones. As feedback develops independently at both sides of a binaural setup, incoherent signal portions in isolated frequency bands are an indicator of developing feedback artifacts. Binaural coherence filtering, i.e., coherence-based gain control is applied to reduce this effect and enable higher gain levels of the hearing device [10]. As shown in Figure 4 the binaural coherence is measured between the left and the right input signals to the hearing aids. From this measure, frequency-dependent gains are derived and applied to both input signals. Coherence filtering also contributes to noise and reverberation reduction, as diffuse, incoherent background sounds are also suppressed. A combination the binaural coherence filtering with preceding bilateral ADMs was shown to be beneficial, i.e., increased SNR [5] and speech intelligibility in subjects [6] with a binaural hearing aid setup.

Dynamic range compression plugin

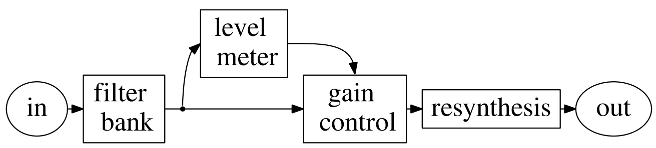

Figure 5: Dynamic compression signal flowchart. The input is split into frequency bands (filterbank). Before a resynthesis to the output signal an input-level dependent gain rule is applied.

At the output of the hearing aid processing chain multiband dynamic range compression [11] is applied. This operation serves two important aspects in a hearing aid: The hearing loss is compensated by defining gain rules between input and output level. Specific gain rules are also used to compensate recruitment effects that often accompany hearing loss, i.e., a decreased range between the percept of a soft sound and the loudest sound with a still comfortable level. To compensate for this effect, soft input sounds are usually amplified with higher gains than loud sounds. Figure 5 shows the signal flow for dynamic compression available in the openMHA: After splitting the input signal in different frequency bands the absolute sound level in each band is measured. Dependent on the level and frequency band gain control is applied according to a gain rule that can be set by the user. It is also possible to configure binaural and inter-frequency interactions of gain derivation. Finally, the time signal is resynthesized.

Discussion

A major goal of the project "Open community platform for hearing aid algorithm research" is the accessibility and high usability of the openMHA as a tool for hearing aid algorithm development and evaluation for the whole community. Hence, a wide range of potential users has to be addressed ranging from programmers acting as plugin developers to audiologists using the openMHA as a pure measurement and evaluation tool. For the latter, additional tools like graphical user interfaces and high-level control is necessary. Plugin developers need well documented concepts to follow when they want to integrate own work into the framework. Application engineers are located somewhat in-between the two user groups decribed above: They can use the openMHA to realize own algortihms with available plugins, manipulate parameters and also compare their work to the reference set of algorithms that is included.

Independent of the level on which work with the openMHA is conducted, the underlying system remains the same on all types of hardware – stationary in the lab or mobile for field testing. Thereby, openMHA accomplishes high reproducibility and supports sustainable research. As the system is open to everyone in the community, a smooth translation of developments across different application layers and groups is possible, and exchange beyond boundaries of different research facilities is enabled.

Up-to-date information, download and installation manuals can be found on `http://www.openmha.org`.

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