Exam project proposal: "Optimization of an adaptive algorithm in a complex system".

A hearing aid is a complex system, which employs various techniques, to amplify the target sound and to improve the life quality of hearing impaired people. In the recent years, a large part of the techniques used in hearing aids lies in the digital domain, where various digital signal processing algorithms enhances the sound experience for the hearing impaired.

The signal processing algorithms in a hearing aid system can generally be divided into the following categories

1. Hearing loss compensation system, which calculates gain and amplifies the input signal.
2. Noise reduction and directionality system, which aims to remove the noise from the target signal.
3. Feedback management system, which is the focus of this project.

In a hearing aid, because the receiver (loudspeaker) and the microphones are located close to each other, the output signal of the hearing aid is partly returned to the microphone via an acoustic coupling through the air. This leads to the acoustic feedback problem, which often causes significant sound quality degradations, and in the worst case, it makes the hearing aid system unstable and howling occurs. As hearing aids amplify the input microphone signals, the likelihood of feedback artifacts increases. Therefore, the feedback management systems in hearing aids aims to reduce the acoustic feedback problems.

Figure 1 shows a simplified block diagram of a hearing aid in the frequency domain.

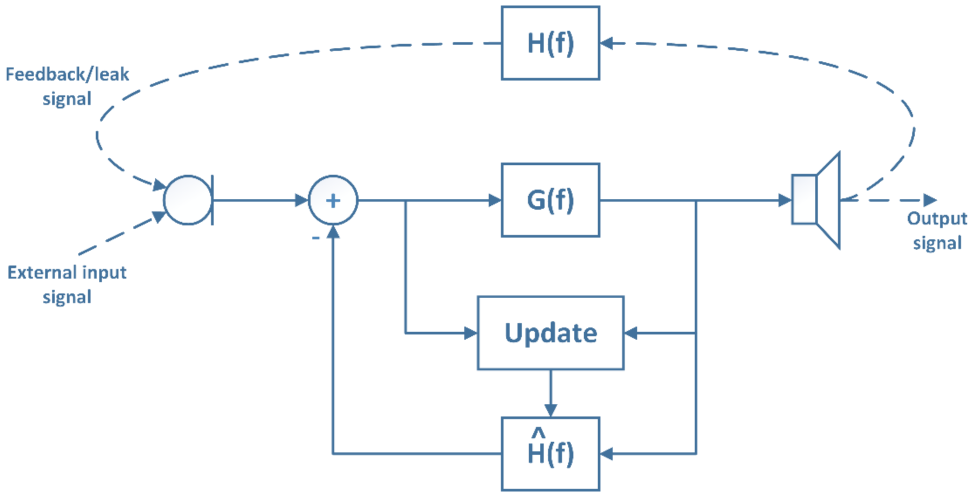

Figure 1 Simplified block diagram of a hearing instrument.

Here $H(f)$ represents a model of the acoustical feedback (sound from receiver to microphone). $G(f)$ is the gain, which is calculated by different components in the hearing aid and is applied on the input signal, and $\widehat{H}\ (f)$ is an estimation of $H(f)$ and is estimated by "*Update*" block in the feedback management system and will be used to remove the feedback signals from the input signals.

As both the feedback model and the gain are changing over time and frequency, a modern hearing aid typically implements an adaptive feedback cancellation system that dynamically estimates the acoustical feedback $H(f)$.

An adaptive system typically consists of many free parameters that controls the overall system performance and adjusting these parameters can lead to optimal performance in some situations and sub-optimal performance in other situations. As with many other real world applications, tradeoffs are needed.

Manually optimizing an adaptive system for optimal performance in different situations can be a long and error-prone process. The manually optimization has the advantage that the tweaking process is "intelligent" controlled by an experienced developer that often generalize for several situations at the same time, however, these processes are still error-prone and might not provide optimal solutions. Also, it can be hard to judge if some parts of the model are needed. If these parts are not needed, the model can be made simpler without sacrificing performance.

The overall process of optimizing an adaptive system one has to balance the following

1. Good sound quality.
2. Good speech understanding
3. Robustness to different acoustical conditions
4. Whistling free (open loop gain < 0)

The system must handle different acoustical situations well. As noted in the introduction several algorithms are active and can affect the overall gain in the hearing instrument, which can lead to different behavior of the system. To capture the interaction with other algorithms in the system, it is important that use cases are defined which causes these algorithms to be "active" to ensure generalization is good.

The main purpose of this master project is to investigate methods for optimizing an adaptive feedback canceller and assess the quality of the solution.
The optimization process can be seen as a non-linear optimization problem and the cost function could be a weighted combination of the following use cases

1. Acoustical feedback changes in situations with different challenging musical pieces
2. Acoustical feedback changes in situations with reverberant rooms and speech
3. Acoustical feedback changes in situations with front speaker and background noise sources
4. Static feedback conditions in situations with different types of speakers

Ideally, the candidate should have a background in signal processing and statistical analysis.