Mapping aided speech recognition thresholds for model-based hearing aid fitting

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Introduction

In common practice, the procedure of fitting a hearing aid to an individual user starts with applying a standard prescription rule, i.e. setting various signal processing parameters especially frequency-dependent gain and compression. The prescription rules mainly use the audiogram to derive settings which focus on optimized speech intelligibility while keeping loudness in reasonable limits, e.g. [1] [2]. This first step in the fitting procedure ("first fit") is usually followed by a phase of an individual fine-tuning based on the feedback of the user. However, this fine-tuning process is very time-consuming and often not leading to a fit that is optimal for the user with regards to e.g. speech intelligibility, loudness, and the preferences of the user.

In order to improve the fitting process by stronger involving the user, several approaches have been made to enable the user to self-adjust the settings of the hearing aid without extensive expert-knowledge on the different parameters, e.g. [3] [4] [5]. These procedures are mainly used for fine-tuning the amplification scheme, i.e. adjusting the frequency-dependent gain and the overall level. However, the parameter space of fitting parameters to be employed, e.g. for complex conditions offered to a specific listener for fine-tuning is not systematically explored and optimized. Usually, the current approaches allow for adjusting parameters in a certain range around the settings of the first-fit. The suitability of the self-adjusted conditions in terms of, e.g., speech intelligibility is usually only evaluated in the aftermath. On the other hand, model approaches simulating the results of listening tests are well suited to simulate and compare a large number of hearing aid settings, and thus selecting appropriate conditions, if they include hearing impairment and aided hearing, i.e. hearing with a hearing aid.

The aim of this study therefore is to develop a general parameter space for self-adjusting a hearing aid using two basically orthogonal metaparameters (i.e., aggregated parameters simultaneously altering several parameters in a prescribed way) and individualize the corresponding parameter space for a specific listener characterized by an audiogram. The individualized parameter space is designed to ensure a certain speech intelligibility on the one hand and limit the loudness to acceptable level on the other hand using validated models: The speech intelligibility is modeled using the simulation framework for auditory discrimination experiments (FADE) [6], while the dynamic loudness model (DLM) [7] is applied for modeling the loudness.

Methods

Defining the general parameter space

The first step is the definition of a general parameter space that includes a variety of different hearing aid settings and is represented by two independent metaparameters. These were defined to systematically vary the frequency-specific and non-linear amplifications schemes of hearing aids:

The first metaparameter controls the gain and compression settings (x-axis). The discrete settings were derived compensating different degrees of hearing impairments with a standard prescription rule. Specifically, we used standard audiograms as defined in [8] and applied NAL-NL2 as prescription rule [1].

The second metaparameter is a sound balance varying the amount of high- and low-frequency gain and refers to individual sound preferences of the user (y-axis). The sound balance is applied in addition to the general compensation of the hearing loss, i.e., the reference condition. This second dimension only changes the frequency-dependent gain function by increasing high frequency gain and simultaneously decreasing low frequency gain or vice versa. The compression, i.e. level dependent gain, is not altered but preserved as derived with the prescription rule.

For the additional gain of the sound balance, normalized gain functions are defined by setting normalized gain factors for three frequency bins. The normalized gain factor for the pivot point of the sound balance, i.e. the mid frequency which divides high and low frequencies, is set to zero, i.e. no additional gain. Additional values are set for two and four frequency bins away from the pivot point for eight different normalized gain curves with increased high and decreased low frequency gain. These eight curves are used with inverted normalized gains for increased low and decreased high frequency gain. This led to 16 normalized gain functions in total as shown in Figure 1. These normalized gain functions are adjusted for each standard audiogram by shifting the pivot point to the middle of the usable frequency range. This ensures, that the effect of the sound balance is perceivable with regards to the respective audiogram. Absolute gains that could be applied with a hearing aid have to be derived from the normalized gain functions by multiplying the normalized gain functions that ranges between -1 and 1 with the value of the maximum added gain (MAG). To limit the loudness of the processed signal, this is also done separately for each standard audiogram using the DLM [7]. The MAG for each standard audiogram is set by iteratively maximizing the value for the MAG that results in a loudness smaller than 45 categorical units (CU) for the normalized gain function that adds most high frequency gain.

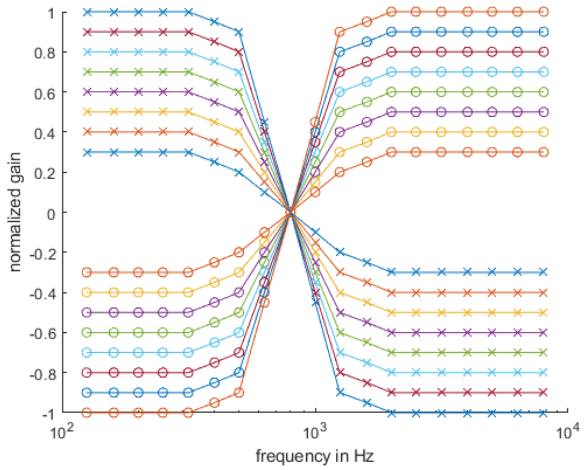

Figure 1: Exemplary normalized gain functions that represent the variations along the y-axis of the parameter space for the compensated standard audiogram N5 according to [8]. Curves marked with a circle add high frequency gain and reduce low frequency gain, curves marked with an x add low frequency gain and reduce high frequency gain of the prescribed reference gain function.

Estimating the individual parameter space

The individualization of the parameter space aims at offering only conditions, i.e. hearing aid settings, to a specific listener, that ensure a certain speech intelligibility and the loudness to be not too loud. For the estimation of speech intelligibility and loudness in the different conditions, models are applied. The first step towards an individualized parameter space is to characterize the specific listener by an audiogram. For this proof-of-concept study we chose the standard audiogram N5 [8], a moderate mildly sloping hearing loss, to represent the listener.

For all conditions, i.e. tested hearing aid settings, the speech recognition threshold (SRT) was simulated with the Oldenburg matrix sentence test (OLSA) [9] using FADE [6]. The background noise was the ICRA noise [10] with a level of 65 dB SPL, which is a noise with speech-like spectral and temporal characteristics. FADE employs an automatic speech recognizer (ASR) which is trained with signals that consist of speech and noise at different signal-to-noise-ratios, i.e. noise at 65 dB SPL and varying levels of the speech signal. The hearing aid processing was implemented with the open Master Hearing Aid [11]. The ASR uses features from a separable gabor filterbank as described in [6]. The loudness was modeled using the DLM [7] which is capable of considering hearing loss by means of individual frequency dependent thresholds. The mean loudness in sone modeled with the DLM was transformed into categorical units (CU) according to [12].

All conditions with an SRT larger than -7 db SNR were discarded as well as all conditions with a loudness larger than 45 CU, which corresponds to the label “very loud” [13]. The remaining conditions constitute the individual parameter space where all conditions can be assumed to result in a satisfactory speech intelligibility and loudness. However, due to the variations in the level- and frequency-dependent gain functions they sound differently and users could choose one setting based on their sound preference.

Results

SRT and loudness values were obtained for a virtual listener with a hearing loss as defined in the standard audiogram N5 as a proof-of-concept. The individualized parameter space for this listener is shown in Figure 2. It includes all tested conditions that have a modeled SRT of -7 dB SNR or lower and a maximum loudness of 45 CU.

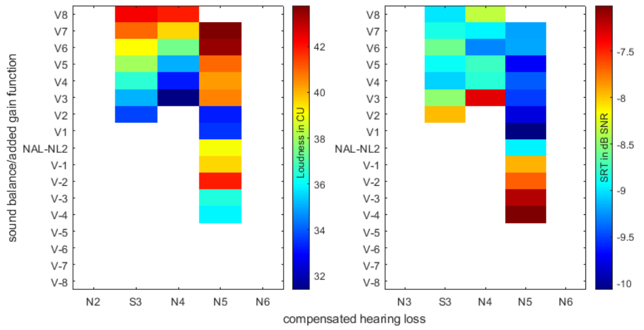

Figure 2: Simulated loudness (left panel) and speech recognition thresholds (SRTs, right panel) for conditions, i.e. hearing aid settings, constituting the individualized parameter space for a listener with a hearing loss as defined in the standard audiogram N5 (mildly sloping moderate hearing loss). The conditions are defined by two metaparameters, one for combined variation of gain and compression (x-axis) and one implementing a sound balance (y-axis), which have a simulated SRT of -7 dB SNR or better and a maximum modeled loudness of 45 CU.

The individualized parameter space includes 25 conditions out of which 13 are variations of the prescribed gain for other audiograms with decreased hearing loss compared to our virtual listener (S3 and N4). In all of these conditions, additional gain is applied for high frequencies, while the gain for low frequencies is reduced (V2/V3 to V8). On trend, the increased high frequency gain leads to slightly increased speech intelligibility of about 1 to 1.5 dB SNR and increased loudness of about 8 to 10 CU. All conditions derived from prescription for audiograms with an increased hearing loss compared to our virtual listener (N6 and N7) lead to a modeled loudness of more than 45 CU and are not included in the individualized parameter space. Most conditions in the individualized parameter space are derived from the prescribed gain for the audiogram of our virtual listener: Seven conditions were variations with more high-frequency-gain (V1 to V7), the prescription for the given audiogram and four variations with increased low-frequency-gain (V-1 to V-4). For the increased high frequency gain, loudness is increased by 10 to 12 CU and condition V8 exceeds the limit, i.e. the additional gain function with the most increased high frequency gain. For these conditions the speech intelligibility is rather stable. For the conditions with increased low frequency gain, loudness also increases but only about 6-8 CU, while speech intelligibility is decreased by 1.5 to 2 dB SNR.

Discussion

The concept and gain functions presented in the method section and the results demonstrate that the individualized parameter space is on the one end limited by high loudness, when increasing the gain especially for high frequencies and on the other end by low speech intelligibility, when the sound balance adds more gain for low frequencies and reduces the gain for high frequencies. This is in line with expectations as medium and high frequencies add spectral information that is particularly relevant for speech intelligibility. The estimated individual parameter space illustrates, that a variety of different hearing aid settings can potentially be the optimal fit for a specific listener. It can be applied for self-adjustment procedures in the context of hearing aid fitting and puts the user in control while limiting the complexity of the process.

The individualized parameter space can also be relevant in the context of mobile health research, where applications are being developed and also already available offering remote hearing diagnostics [14]. Due to, e.g. uncalibrated equipment, background noise or distractions during the measurements, results from such diagnostical procedures might not be as accurate as in clinical procedures. In these cases, individual parameter spaces based on standard audiograms can be used to offer a variety of settings without relying on an exact audiogram but rather referring to an estimate.

However, to make these different options available to the user requires simplifying the representation of the different parameters and enabling the user to choose between settings in a reasonable range of speech intelligibility and loudness. These ranges are defined based on models and even if the applied models are capable of considering individual hearing thresholds, suprathreshold parameters describing the individual hearing abilities of the user are not considered, yet. The main reason for this gap is the lack of validated procedures to obtain values for descriptive parameters of suprathreshold hearing and their integration into models for speech intelligibility and loudness. Consequently, the limits of and the variations within the individualized parameter space have to be authenticated with experimental data.

It also is of interest to shed light on the criteria underlying the choice of hearing aid settings by the user beyond speech intelligibility and loudness and this individualized parameter space offers a good base for corresponding research. Moreover, as frequency- and level dependent gain don't represent the full range of adjustable parameters in hearing aids, it might also be appropriate to investigate other (meta)-parameters, e.g., for setting other parameters such as noise reduction schemes or using another prescription rule to derive the individual reference condition.

Conclusions

This proof-of-concept study has shown, that a parameter space using two metaparameters including a variation of complex hearing aid settings can be individualized for a listener with a specific audiogram by employing models to limit admissible ranges for loudness and speech recognition thresholds. This reduces the conditions and thus the complexity of the self-adjustment procedure. The results confirm, that the process of optimizing hearing aid settings is not determined, but can lead to different results based on the preferences of the specific listener. The process of defining an individualized parameter space developed in this study can be used to confine the options offered to the listener for self-adjusting hearing aid settings, but further research is necessary to validate the results with experimental data.

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