

Technical evaluation of hearing-aid fitting parameters for different auditory profiles

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Summary

Hearing-aid users have reported an increased satisfaction since digital technology and advanced signal processing became available in hearing aids. However, many users still experience difficulties in noisy environments and in complex listening scenarios. Although numerous parameters can be adjusted to provide an individualized hearing solution, hearing-aid fitting currently consists of: 1) the gain prescription and adjustment based on the pure-tone audiogram, 2) the activation of advanced features on-demand, such as beamforming and noise reduction. In a previous study [1], a novel approach for auditory profiling was suggested, where the hearing deficits were characterized according to two types of distortion. This allowed the classification of listeners into four auditory profiles according to a high/low degree of hearing distortions along the two dimensions. The aim of the present study was to evaluate different hearing-aid compensation strategies that may fit the needs of different auditory profiles via technical measures. A hearing-aid simulator, consisting of beamforming, noise reduction, and dynamic range compression, was used to test which parameter spaces and outcome measures may be of interest for a “profile-based hearing-aid fitting”. The simulator consists of two dummy behind-the-ear hearing aids and off-line sound processing performed on a personal computer. Technical measures, such as signal-to-noise ratio (SNR) improvement, envelope degradation, and a metric of spectral distortions, were used to evaluate the effects of different signal processing strategies on the signal at the output of the simulator. Several parameter settings were evaluated using speech in the presence of various interferers at different SNRs. Here, the results of this technical evaluation are presented and discussed, with a view towards identifying the effective compensation strategies for different auditory profiles.

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1. Introduction

Satisfaction reported by hearing-aid users has increased significantly since digital technology became available [2]. This can be attributed to the ability of modern hearing aids (HAs) to deliver non-linear amplification as well as advanced signal processing features, such as beamforming and noise reduction. However, many HA users still experience difficulties in understanding speech in noisy environments and other complex listening scenarios.

While numerous parameters can be adjusted to provide an individualized hearing solution, current hearing-aid fitting procedures are relatively simple. Usually, frequency and level dependent gain is first determined based on the listener's pure-tone sensitivity, i.e., the audiogram. Subsequently, advanced features, including algorithms like beamforming and noise reduction, may be activated depending on personal preferences. Importantly, the fitting procedure does not take supra-threshold performance, e.g., measures of the listener's performance at moderate sound levels and in complex environments, into account. Therefore, listeners with similar audiograms receive similar fitting solutions. The individual listener's needs are addressed during fine tuning, which depends solely on the audiologist's skills and experience. Given the nonlinear nature of many hearing-aid algorithms and their interactions, the design of individualized compensation strategies can be a complex task. This complexity is further increased by a broad range of sound scenarios encountered by individual HA users as well as inherent variability in a given individual's responses.

Evaluating a listener's supra-threshold performance requires tools beyond the pure-tone audiogram. The listener's performance may be estimated using a test battery and individual data can then be used to quantify the degree of perceptual distortions perceived by each listener. Recently, a data-driven approach to characterize individual listeners' hearing along two dimensions has been proposed [1], where each dimension represented an independent type of supra-threshold distortions. Each listener was assigned one of four possible auditory profiles defined by their degree of perceptual distortions in the two dimensions. It is reasonable to assume that the most efficient

compensation of a given hearing loss depends on the type of auditory distortions present, such as the ability to perceive the temporal and spectral features of sounds. Hence, a "profile-based" HA fitting would ideally activate algorithms that can compensate for the specific types of distortions present in each listener. In order to approach this ideal scenario, a technical characterization of how modern HA features can affect specific distortions in the physical signal should be obtained first. Such a characterization was the aim of the present study to help define feature combinations that are adapted to different auditory profiles.

A profile-based HA parameter space may require different directionality, noise reduction, and compression settings. Although the two first types of strategies aim for signal-to-noise ratio (SNR) improvement, directionality applies a spatial filtering that keeps the signal in front unaltered while noise reduction applies spectral filtering on the noisy mixture. The effects of noise reduction and directionality on speech-in-noise perception have been a topic of interest in previous studies [3, 4]. Furthermore, the influence of the parameters used in dynamic range compression [5, 6] has been broadly studied. The characteristics of these processing algorithms in isolation have also been assessed by means of technical measures, such as speech intelligibility prediction or physical measures of the acoustic signal [7, 8], which do not require the participation of a listener. The present study is inspired by the approaches used in these previous studies and focuses on characterizing the effects of the HA algorithms on established metrics reflecting distortions in the physical signal.

In the literature, the SNR improvement and other physical measures at the output of real hearing aids have been explored in connection to speech-in-noise perception [9] as well as perceived quality measurements [10]. In this context, speech intelligibility prediction models and speech quality models are commonly used to quantify the expected performance of specific algorithms [8, 11]. While these objective measures may correlate with the observed perceptual performance of normal-hearing listeners, there is no guarantee that hearing-impaired listeners would exhibit the same behavior. Therefore, in the present study, such model-based objective performance measures were complemented with technical metrics related to

SNR, spectral, and temporal signal distortions. The idea was to characterize how the combination of parameters in HA algorithms affects such metrics rather than predicting HA user performance.

For this purpose, a hearing-aid simulator (HASIM) was designed and evaluated with a set of five objective metrics. The chosen physical measures were the segmental SNR and objective measures of temporal-envelope and spectral distortions. The objective speech intelligibility and quality measures used here were the short-time objective intelligibility (STOI) and the perceptual evaluation of speech quality (PESQ) [12, 13]. The main goal was to characterize the performance of each algorithm in isolation as well as their interaction in several sound scenarios. Additionally, it was of interest to identify the combinations of parameters that lead to the best/worst performance in terms of the five chosen metrics.

2. Hearing-aid simulator (HASIM)

The HASIM was implemented in MATLAB via the combination of three processing algorithms. As shown in Figure 1, the signal recorded from the frontal and rear microphones of a hearing aid was processed by a beamformer, a noise reduction algorithm and a wide-dynamic range compressor.

2.1. Beamformer (BF)

The BF provides an omnidirectional sum of both microphones and two polar patterns, a fixed unilateral BF and a binaural BF. To obtain the optimized beam-patterns for the two BF types, a head and torso simulator (HATS) was placed in the center of an anechoic room facing a speaker at 0 degrees (distance 1.5 m). The impulse responses from the speaker to each of the four microphones were measured with a 5-s maximum length sequence (MLS) with a code length of 11 bit at a sound pressure level (SPL) of about 65 dB. This was repeated for loudspeakers situated in the horizontal plane for angles from 0 to 360 degrees with a resolution of 5 degrees. After the impulse responses were obtained, a linear filter was built for each microphone (front, rear) and optimized in a least-square sense to a predefined beampattern [14]. Optimization was performed only in the frequency region between 1 and 5 kHz. Below 1 kHz, the front microphone signal alone was used as the output, and above 5 kHz unilateral

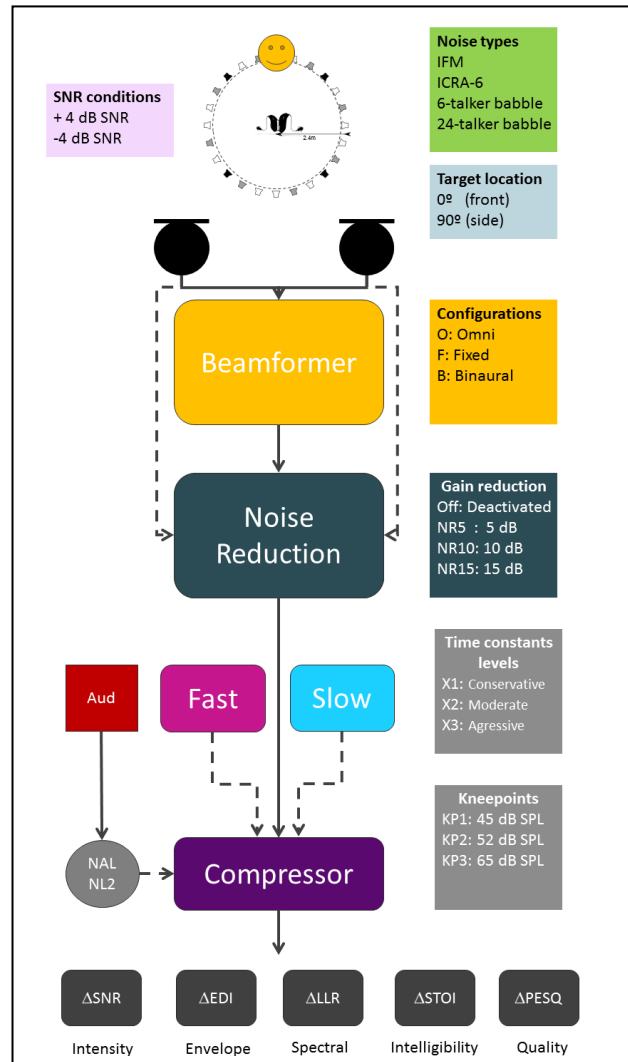


Figure 1: Diagram of the hearing-aid simulator (HASIM) including the sound scenarios tested and the objective measures considered in the study. Panels placed on the right briefly explain the different levels used for each stage.

beamforming was applied. For the binaural BF, the four outputs of the left (L) and right (R) ear devices were processed in a similar fashion. This resulted in a diotic signal. However, the use of a diotic signal removes spatial cues that are important for localization and spatial separation in real environments. Therefore, to improve the acceptance of the binaural beamformer, a portion of the signal from the front microphone was added to each device. In this case, 85% of the processed signal and 15% of the front microphone was considered in the simulations.

2.2. Noise reduction (NR)

The noise reduction system was based on the spatial properties of two closely-spaced microphones and the assumption that the sounds of interest would be primarily located in front of the listener. From the

two microphone signals, two first-order differential arrays (cardioids), pointing in opposite directions (towards the front and back), were created as described in [15]. Hereby, the front-facing cardioid primarily captured sounds in front of the listener (sound of interest) while the rear-facing cardioid primarily captured sounds behind the listener (noise). By comparing the power spectral density estimates of the two cardioids in each time-frequency frame, a binary mask was created which determines if a given time-frequency tile mainly contains energy from the front or the back of the listener. The time-frequency mask was converted into a binary gain, which attenuates time-frequency tiles with more energy in the rear-facing cardioid compared to the front facing cardioid with a fixed amount of attenuation [15].

2.3. Dynamic range compressor (WDRC)

The compressor consisted of a 15-band filterbank (0.1-10 kHz), a percentile estimator, and an amplifier with non-linear gain. The bandwidth of the filterbank was approximately one-third octaves for the eleven mid-frequency bands and half an octave for the four remaining upper and lower bands. The envelope of the individual bands was estimated based on the low-pass filtered squared signal. The envelope was then transformed to the logarithmic domain and passed through the percentile estimator that effectively controlled the time constants of the compression system. The output of the percentile was increased with a fixed-rate attack time if the envelope was greater than the output. Similarly, the output of the percentile was decreased with a fixed-rate release time if the envelope was smaller than the output. The percentile estimator calculated the desired gain in the compressor input-gain function and was set for each of the compressor conditions. The amplifier's gain function was a broken-stick nonlinearity with a single kneepoint used to set the insertion gain for conversational speech level (65 dB SPL). The upper and lower slopes of the function were calculated to match the target gains for soft (50 dB SPL) and loud (80 dB SPL) speech targets. The calculated gain was applied to the individual frequency bands based on the prescription rule corresponding to the individual pure-tone audiometric thresholds. The compressed output signal was formed by the sum of all bands [16].

3. Method

3.1. Sound scenarios

The sound scenarios used in the technical evaluation were recorded in an anechoic chamber with 24 loudspeakers placed in the horizontal plane, in steps of 15°, around a chair located in the middle of the chamber. A HATS was placed on the chair while wearing HA satellites consisting of a HA housing with a front and a rear microphone. The international speech test signal (ISTS) [17] was used as the target signal, which was recorded when played from the loudspeakers located at 0° and 90° degrees at 65 dB SPL. Two noises were used; the international female noise (IFN), a stationary noise with the same long-term average spectra (LTAS) as the ISTS [17], and ICRA-6 [18], a fluctuating noise composed of the envelope of six talkers and the fine structure of a random noise. The two noise maskers were recorded from the two loudspeakers located at ±45°. Additionally, two multi-talker noise environments were constructed using recordings of real conversations[19]. A 6-talker babble was recorded from loudspeakers located at ±15°, ±30°, and ±45°. A 24-talker babble was recorded by playing the speech of one independent talker from each of the 24 loudspeakers.

The sound scenes were prepared by combining the signal from each of the microphones of the target signal and each of the sound environments. The conditions considered for each of the noise environments were:

1. Target at 0° and +4 dB SNR.
2. Target at 90° and +4 dB SNR².
3. Target at 0° and - 4 dB SNR.
4. Target at 90° and - 4 dB SNR².

In addition, each of the sound scenes was constructed either with the target in phase (S0N0) or in antiphase (S π N0). This was done to enable the extraction of the target and the noise signals in each stage of the HASIM using the Hagerman-Olofsson separation technique [20].

3.2. Hearing-aid parameter spaces

Each of the three HASIM stages was tested in a number of conditions. The BF was tested in three modes: Omni (O), Fixed (F), and Binaural (B). The NR algorithm was tested with attenuations of 5

² The SNR is referred to the tested device (left) only.

(NR5), 10 (NR10), and 15 dB (NR15), as well as when the algorithm was deactivated (Off). The parameters of the WDRC adjusted in the simulations were the kneepoint (KP) and the time constants (TC). The KP was set at either 45, 52, or 65 dB SPL. The TC were divided into ‘fast’ and ‘slow’ options and tested with three levels in each case:

1. Fast1: Attack = 15 ms; Release = 50 ms.
2. Fast2: Attack = 10 ms; Release = 10 ms.
3. Fast3: Attack = 5 ms; Release = 10 ms.
4. Slow1: Attack = 40 ms; Release = 400ms.
5. Slow2: Attack = 100 ms; Release = 800ms.
6. Slow3: Attack = 250 ms; Release = 1250ms.

The compression ratio was determined by applying the NAL-NL2 [21] prescription rule to different audiometric profiles based on the proposed standard audiograms [22]. The audiometric thresholds of the audiograms N1, N2, N3, N4, S1, S2, and S3 were entered into the NAL-NL2 software and the target gains at 50, 65, and 80 dB SPL were transferred to the compressor algorithm. Additionally, the 0-dB linear-gain condition was tested in order to explore the processing algorithms (BF and NR) in isolation. In total, 216 different parameter combinations (3 BF x 4 NR x 3 KP x 6 TC) were tested per audiometric profile.

3.3. Procedure

The simulations were carried out in the same way for each of the sound scenarios and set of HA parameters. Once the sound scenario at the input of the frontal and rear microphones was constructed, the resulting signals were used as the input to the BF. As mentioned above, this was done for both the SON0 and $S\pi N0$ versions of each sound scenario. After the BF stage, the resulting signal as well as the original signals from the frontal and rear microphones were input to the NR algorithm. The last stage was the WDRC which was fed with the signal obtained at the output of the NR. Once this was done, the reference signal for the evaluation, corresponding to the omni-directional and linear condition (OmLin), was obtained by performing a simulation in which the prescribed gain per frequency band corresponded to the long-term spectrum of the output signal. This was done in order to 1) minimize the effect of the spectral shape of the output signal for each audiogram and more clearly observe the effects of the WDRC parameters, and 2) reduce the effect of the input SNR. Moreover, using OmLin as a reference

yielded a reference output signal that had been processed by the whole HASIM but was not influenced by the distortions and enhancements created by each algorithm.

3.4. Objective measures

The technical evaluation involved three physical measures of the acoustic signal. These were the segmental signal-to-noise ratio (segSNR) at the output of the HASIM, the log-likelihood ratio (LLR) between the unprocessed and processed signals, and the envelope distortion index (EDI) between the unprocessed target and the isolated target at the output of HASIM as defined in [6], [7]. Additionally, two performance measures were considered: STOI and PESQ. In both performance measures, the reference signal was the clean target from the OmLin condition and the test signal was the noisy speech at the output of HASIM.

4. Results and Discussion

The simulations were first carried out for the processing algorithms (BF & NR) and the fitting algorithm (WDRC) in isolation. A multi-way ANOVA for all the sound scenarios showed a significant effect of NR [$F(3,191)=3.23$, $p=0.02$] and BF [$F(2,191)=9.73$, $p<0.01$] on the segSNR but not their interaction, which was only significant on the LLR [$F(6,191)=3.29$, $p<0.01$]. In contrast, only BF had a significant influence on EDI [$F(2,191)=171.8$, $p<0.001$]. When comparing the different sound scenarios, NR had no effect when the noise was located in front, due to the inefficiency of the SNR estimation algorithm in such a setting. In contrast, for the 24-talker babble, BF had a significant effect on the three physical measures and NR affected segSNR and LLR significantly. In the following, only the results for the 24-talker babble scenario in its four conditions are reported and discussed.

Figure 2 shows the changes in segSNR, EDI, and LLR scores, relative to the OmLin condition. The left panel shows the performance of BF and NR for different SNR and target location conditions. While the segSNR scores increased when BF and NR were activated and the target was located at 0° , the scores of the binaural BF (B) were 2.5 dB lower when the target was located at 90° . This was also observed in the EDI scores, which increased dramatically when the binaural BF was activated and the target was located at 90° . Furthermore, the LLR increased

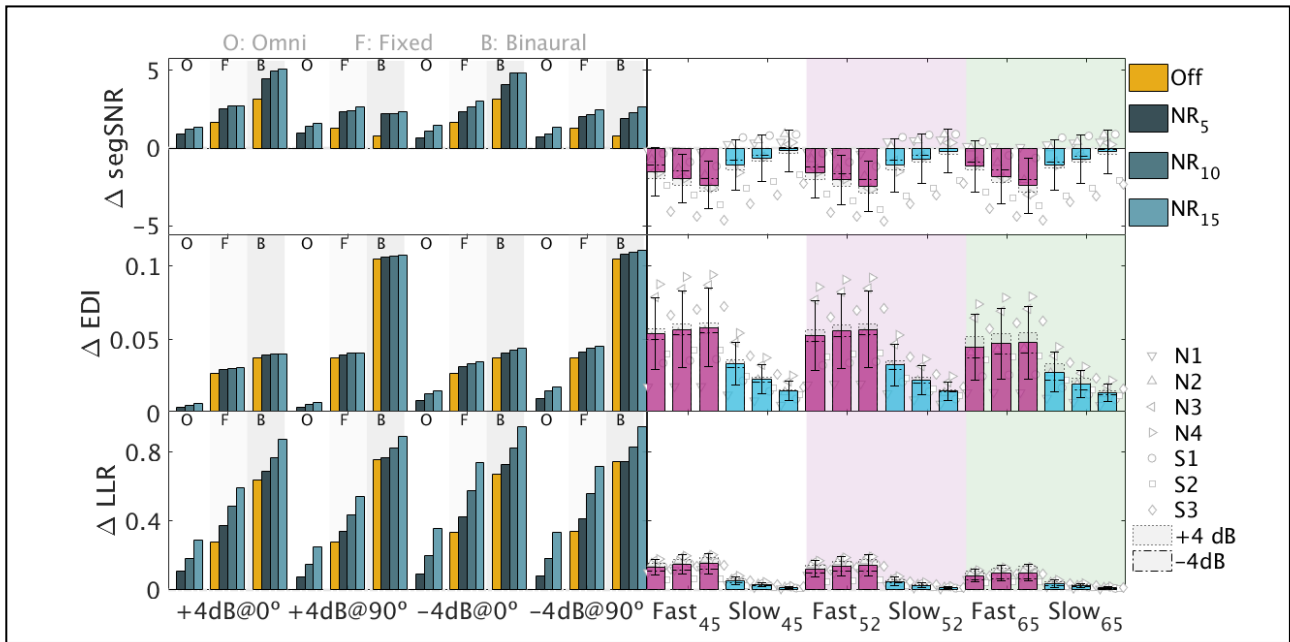


Figure 2: SNR improvement (ΔsegSNR), envelope (ΔEDI), and spectral distortions (ΔLLR), of the the HASIM algorithms in isolation. **Left panel** shows the beamformer (BF) and noise reduction (NR) for the different sound scenarios and **Right panel** the fitting algorithm (WDRC) for the different combinations of parameter kneepoint (KP=45,52 or 65 dB SPL) and time constants (TC=Fast or Slow in their three levels) for positive and negative SNRs as well as the mean of each audiometry.

when the algorithms were more aggressive, regardless of the target location. Moreover, the condition B and NR15 yielded the largest change in segSNR (5 dB) but also the highest amount of spectral distortion (>0.8).

The right panel of Figure 2 shows the mean results for the different WDRC conditions (colored bars), the average results for positive and negative SNRs (shaded bars), as well as the mean of each audiometric configuration (markers). In contrast to the results shown in the left panel, WDRC reduced the segSNR, particularly for the fast-acting compression settings. While the influence of KP on the EDI scores was small but significant [$F(2,495)=18.9$ $p<0.001$], there were large differences between the slow and fast-acting configurations [$F(1,495)=1033.8$, $p<0.001$]. The effective compression became more linear with increasing time constants (slow-acting) showing a reduced amount of distortions and a smaller SNR reduction. Therefore, the selection of fast-acting compression may counteract the SNR enhancement provided by the processing stages and can introduce additional distortions in the temporal envelope (i.e. higher EDI scores). When comparing the results for the individual audiometric configurations, the audiograms with a higher degrees of hearing loss, particularly at low frequencies (N3 and N4), led to even larger envelope distortions. On the other hand,

the spectral distortions (LLR) introduced by the WDRC were much lower than the ones introduced by the processing algorithms (BF & NR).

Figure 3 illustrates the differences in performance due to the combined effects of the BF, NR, and WDRC. Only the condition with the target in front

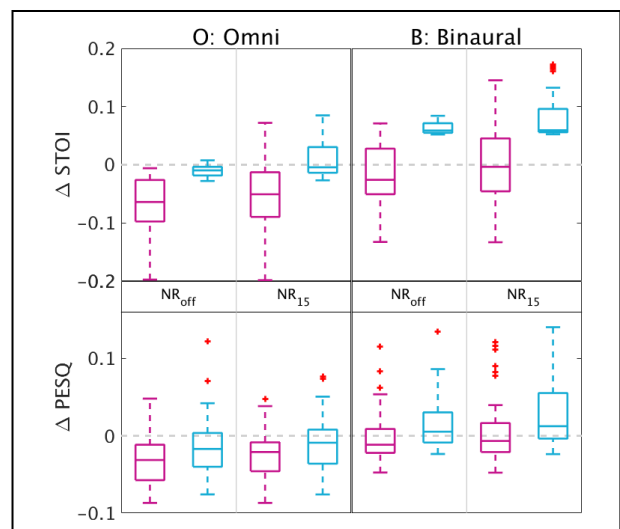


Figure 3: Objective performance measures (STOI and PESQ) of the HASIM with the target direction at 0° . Each result corresponds to a combination of the three algorithms. The magenta boxplots show results of the fast-acting compressor and cyan the slow-acting compressor.

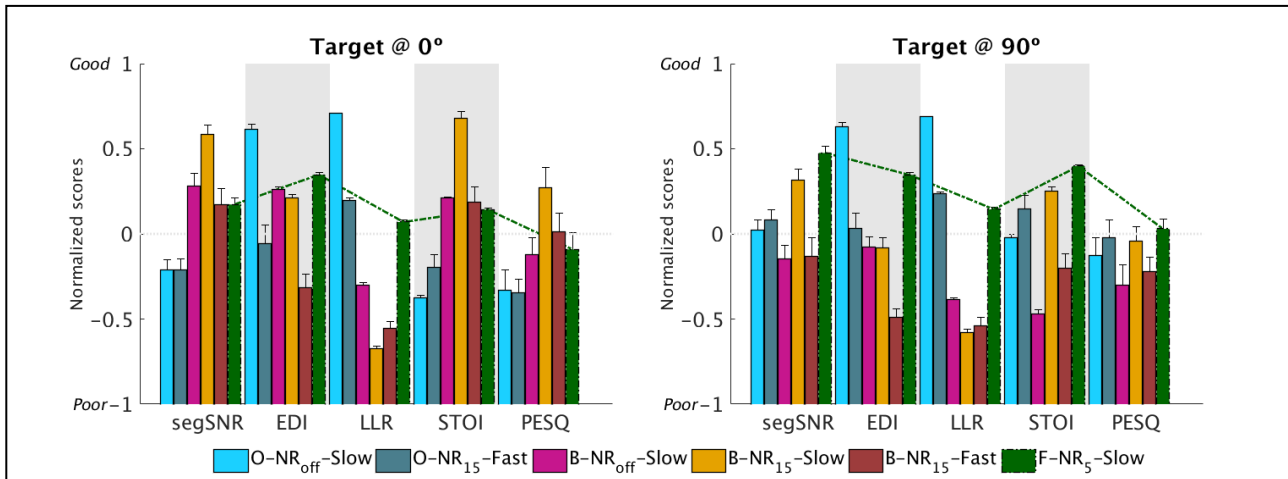


Figure 4: Normalized scores in the five chosen objective measures for six HA parameter settings. Results are divided in low (mild) and high (sev) degree of hearing loss and by target direction. The results were normalized between the 10th and 90th percentiles. The normalized scores of EDI and LLE were multiplied by a factor (-1) so -1 always corresponds to a poor performance and 1 to a good performance.

was considered here, and the boxplots show the scores relative to the OmLin condition for all noise types and audiometric configurations. The results suggested a clear improvement of the STOI scores when BF was binaural and an additional improvement when NR was activated with the highest attenuation (NR15). One should note that the variance of the results of the fast-acting compression was higher than for the slow-acting HA configuration. This is mainly due to the slow-acting compression linearizing the long-term response and acting as a gain reduction that does not affect the spectro-temporal features of the signal. However, fast-acting compression has different effects depending on the compression ratio applied, which depends on the audiometric thresholds. In contrast, the results for the PESQ metric did not show significant differences neither in terms of the mean values nor the variance.

To test different profile-based compensation strategies, it is of interest to explore HA parameter spaces that differ widely from one another, not only in terms of performance, but also in terms of spectral and temporal distortions. Therefore, six HA parameter settings were chosen for that purpose. Figure 4 shows the average results across noise types, SNR conditions, and hearing profiles for these six settings. The normalized results of the five objective measures are shown for the 0° (left panel) and 90° (right panel) target condition. As expected, the HA setting with no processing activated and slow-acting compression (O-NR_{off}-Slow) provided good scores for the distortion measures (i.e., EDI, and LLR), but slightly negative

scores for the segSNR, STOI, and PESQ metrics. In contrast to the unprocessed HA setting, a HA setting with all the algorithms activated at their most aggressive level (B-NR₁₅-fast) showed clear spectral and temporal distortions. In addition, B-NR₁₅-fast showed an improvement in SNR and STOI when the target was located in front but poorer scores when it was located at 90°. For the fourth HA setting (B-NR₁₅-slow), this improvement was even higher and exceeded the unprocessed HA setting in all cases. The HA setting with moderate processing parameters (F-NR₅) and slow-acting compression showed positive scores for both target directions, suggesting an improvement in speech intelligibility compared to most of the other HA settings considered here.

5. Conclusions

Several HA parameter spaces were characterized by using objective physical measures at the output of a HA simulator. While the processing algorithms (BF and NR) tended to enhance the SNR and introduce spectral distortions, fast-acting compression had a detrimental effect on SNR improvement and temporal distortion. Parameter spaces towards a profile-based HA fitting were proposed by choosing combinations of parameters that provided different results in terms of SNR benefit, physical distortions and performance predictors. Overall, a perceptual evaluation using these identified parameters spaces should provide meaningful differences among the different HA settings and may help in the implementation of a profile-based compensation of the hearing deficits.

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