Hearing aid processing of loud speech and noise signals: Consequences for loudness perception and listening comfort. - DTU Orbit (13/04/2019)

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Sound processing in hearing aids is determined by the fitting rule. The fitting rule describes how the hearing aid should amplify speech and sounds in the surroundings, such that they become audible again for the hearing impaired person. The general goal is to place all sounds within the hearing aid users' audible range, such that speech intelligibility and listening comfort become as good as possible. Amplification strategies in hearing aids are in many cases based on empirical research - for example investigations of loudness perception in hearing impaired listeners. Most research has been focused on speech and sounds at medium input-levels (e.g., 60-65 dB SPL). It is well documented that for speech at conversational levels, hearing aid-users prefer the signal to be amplified by approximately half the amount of the hearing loss (in dB). This places the amplified speech signal approximately in the middle of the users' audible range, at a comfortable listening level. However, there has been little research on the optimal gain-prescription for soft and loud sounds. At present, such prescriptions are based mainly on logic, as there is limited evidence on what type of amplification is best for these input-levels. The focus of the PhD-project has been on hearing aid processing of loud speech and noise signals. Previous research, investigating the preferred listening levels for soft and loud sounds, has found that both normal-hearing and hearing-impaired listeners prefer loud sounds to be closer to the most comfortable loudness-level, than suggested by common non-linear fitting rules. During this project, two listening experiments were carried out. In the first experiment, hearing aid users listened to loud speech and noise signals with built-in level-variation (62 – 82 dB SPL). The signals had been compressed with seven different compression ratios, in the range from 1:1 to 10:1, yielding different degree of overall level-variation in the processed signals. Subjects rated the signals in regard to perceived level variation, loudness and overall acceptance. In the second experiment, two signals containing speech and noise at 75 dB SPL RMS-level, were compressed with six compression ratios from 1:1 to 10:1 and three release times from 40 ms to 4000 ms. In this experiment, subjects rated the signals in regard to loudness, speech clarity, noisiness and overall acceptance. Based on the results, a criterion for selecting compression parameters that yield some level-variation in the output signal, while still keeping the overall user-acceptance at a tolerable level, is suggested. It is also discussed how differences in speech and noise components seem to influence listeners ratings of the test signals. General recommendations for a fitting rule, that takes into account the spectral and temporal characteristics of the input signal, is given together with suggestions for further studies. Finally, the experimental methods used for the listening tests in this project are discussed.« « «

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