PSYCHOACOUSTICS AND HEARING AIDS

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ABSTRACT

The past years have shown a dramatic development in hearing instruments technology. The introduction of digitally programmable and now fully digital hearing instruments has lead to an increasing potential to improve the listening performance of hearing impaired persons.
Psychoacoustics play an important role in this development. New audiological measurement methods reveal more information about the individual hearing deficits. The development of new signal processing strategies is guided by the increasing knowledge about the hearing system.
This paper will show how psychoacoustics can contribute to the assessment of hearing deficiencies and to the signal processing used in hearing instruments.

INTRODUCTION

Over the last decade, hearing intruments technology has shown a quick and dramatic development. First, digitally programmable hearing instruments brought advanced signal processing schemes and much greater flexibility for adjustment and fitting. Programmability enabled multi-memory instruments that allowed for optimization of the parameter settings for multiple listening situations. The introduction of hearing instruments based on digital signal processing led to further improvements in processing capabilities.
This development provides great technological potential that will only be worth the effort if it is used to create the best possible benefit for any individual hearing impaired subject. More complex processing schemes need more complex knowledge in designing and programming these schemes as well as more complex approaches for the fitting process. This paper shows some areas in which psychoacoustics can help in that process.

PSYCHOACOUSTICS AND HEARING IMPAIRMENT

The most common type of hearing loss can be characterized as sensorineural hearing loss. In a sensorineural hearing loss, the conversion of the mechanical vibration of the inner ear liquid in neural pulses and the transmission is more or less deteriorated. The functionality of both inner and outer hair cells is worse than normal. The active processes that lay ground for the fascinating performance of the normal ear in regard to sensitivity as well as frequency and time resolution are reduced. The effect of this type of hearing loss is not only that hearing impaired subjects hear sounds softer, but the sounds are distorted on the neural level, causing a loss of speech intelligibility (Plomp 1986). Another characteristic consequence of sensorineural hearing loss is a reduced dynamic range between the elevated threshold and the loudness discomfort level (LDL), which tends to be rather normal. Soft sounds are

not heard due to the threshold shift, but loud sounds are heard more or less normal. This can be interpreted as an acoustical contrast enhancement enlarging the difference between the perceived loudness of soft and loud sounds. In combination with masking this makes it difficult to hear soft speech components (such as some of the consonants) close to relatively loud components (such as vowels). Once some speech components remain unheard, they cannot be used for speech intelligibility. As a consequence, speech intelligibility often is rather low even when the average speech level is in a comfortable range.
The individual residual dynamic range can not accurately be predicted only from puretone threshold data (Kinkel and Moser 1998). As an alternative, the individual loudness perception can be measured with loudness scaling procedures (Hellbrück and Moser 1985; Allen, Hall et al. 1990; Kollmeier 1997). Typically, listeners are asked to rate the loudness of narrow-band noise signals on a categorical scale („not heard – very soft – soft – medium – loud – very loud – extremely loud"). A loudness growth function can be derived fitting a linear function to the loudness ratings for different input levels. This function can be described by its slope and by the level that corresponds to the loudness 0 („not heard"). This level is a good estimate of the pure tone threshold for mild to severe hearing losses (Kinkel and Moser 1998). The difference between the individual loudness function and the normal loudness function at different levels can be used to calculate target gain for amplification as a function of frequency and input level. For compensation of a reduced dynamic range, the amplification at low levels must be higher than at high levels, in some cases at very high levels no amplification at all might be necessary.
The loss of frequency resolution and the loss of temporal resolution is also frequently associated with sensorineural losses. Although it is known that both factors have an adverse effect on speech intelligibility, they are not assessed in routine audiometric measurements today.

PSYCHOACOUSTICS IN HEARING AID TECHNOLOGY

Many of the signal processing schemes used in contemporary hearing instruments are based on psychoacoustic ideas and models.This chapter describes the most common approaches to restore normal sound perception.

Frequency processing

To compensate for frequency dependent hearing losses, most hearing instruments separate incoming signals into different frequency bands allowing separate gain adjustment or even more complex processing schemes. In commercially available hearing instruments, the number of bands can vary between only a few and about 20, following the concept of the Critical Bands (Zwicker and Fastl 1990). Gain adjustment in more frequency bands allows for a finer adjustment to the frequency characteristics of any particular hearing loss. Another positive side-effect of processing sounds in many narrow bands is that they allow for a finer feedback management reducing the gain in a single band in which feedback occurs.

Restoring loudness perception

The first important step is to transform the input signal in a way that all speech components become audible again and that no sounds exceed the LDL. For all subjects suffering from reduced dynamic range this means that linear amplification cannot deliver the suitable amount of gain for all signals, since soft signals require more gain than loud sounds. Therefore, since many years, hearing aids have involved automatic gain control circuits (AGC). In such circuits, amplification is linear up to a certain level („kneepoint") and compressed or even limited by a certain factor above this level. The kneepoint often was at 65 dB, which is right in the middle of the normal speech level range. So in a normal conversation, the AGC was switched on and off permanently, causing audible distortion. As a solution, the kneepoint was lowered below the normal speech level, e.g., to 45 dB. With such a low kneepoint, much less compression was necessary without providing too much amplification for loud sounds. Since amplification is controlled over a great level range, this type of compression often is referred to as „WDRC" (wide dynamic range compression). The fact, that the compression circuit normally is settled and that rather low compression ratios can be used, helps to avoid distortion. Most subjects rate the sound of WDRC hearing instruments much clearer and better as hearing instruments involving „classic" AGC circuits. WDRC hearing aids are rated better than linear hearing aids both in speech perception and in loudness growth measures. WDRC instruments resulted in consistent speech perception scores in different listening situations, whereas linear aids gave good scores only in average

speech situations. Scores degraded both for soft and loud situations. WDRC provided a significant larger and more normalized input dynamic range than linear amplification without volume control adjustments (Jenstad, Seewald et al. 1999; Jenstad, Pumford et al. 2000).
Considering loudness in more detail reveals that loudness is not only a function of level, but also of bandwidth and frequency composition of the signal. With many channels acting independently, some soft signal components that would be masked in normal hearing, become audible, because a low level is detected in the respective frequency band causing a relatively high gain. One way to avoid these effects is the introduction of channel interaction: the gain in each channel is influenced by the gain in adjacent channels. Totally independent processing could also work against speech intelligibility in certain acoustical situations. When a loud high-frequency signal is present, the compression in that channel is activated and the gain is reduced. If the gain in lower frequency bands is not affected, low-frequency signals can potentially mask the speech components in the high-frequency bands. Channel interaction can help to reduce this potential detrimental effect of upward spread of masking. If the gain of some channels is reduced, e.g., to avoid feedback, the overall gain should be readjusted to avoid overall level fluctuations.
Some of the latest hearing instruments include loudness models considering loudness summation and masking effects for the calculation of channel gains and the overall amplification.

Temporal characteristics

Regarding the temporal behavior of compression systems, three main approaches can be found in hearing instruments: compression limiting, automatic volume control, and syllabic compression.
Compression limiting is implemented with high compression thresholds, high compression ratios and short time constants. Only the amplification at high levels is restricted in order to avoid the output level to exeed the LDL. Peak clipping can be interpreted as compression limiting with an compression ratio ∞:1 above threshold.
To control the overall loudness, automatic volume control (AVC) uses low thresholds, low-to-medium compression ratios and long time constants. AVC can replace the manual volume control. The advantage of AVC is that it preserves fast changes in volume like the modulation within speech. A disadvantage is that AVC may act too slowly when sudden loud sounds (such as a slamming door) occur.
Syllabic compression systems realize a low kneepoint, low-to-medium compression ratios and short time constants to adjust the gain for different speech syllables. The main goal of syllabic compression is to restore the loudness relation between different speech elements to reduce the acoustical contrast enhancement caused by recruitment. Another advantage of syllabic compression is that it can quickly reduce gain when sudden loud sounds occur and can quickly recover from gain reduction after the loud sound has stopped. A disadvantage can be, that the natural level variations within speech are reduced, which might result in poorer speech intelligibility.
A compromise between the latter two principles can be to combine long and short time constants depending on the time course of the signal level. Short time constants are activated only when sudden loud sounds occur, otherwise the slow time constants are active. This strategy acts like an AVC circuit, but prevents loud sounds from being amplified too much.

Speech processing

Some hearing instruments involve systems for speech enhancement. These systems try to increase the relative intensity of some segments of speech. Current processing strategies identify and enhance speech based either on temporal, or more recently, spectral content. Speech enhancement in hearing aids is still relatively new, and its effectiveness is largely unknown. Most studies show only a marginal average effect, but there is some evidence that speech enhancement has significant positive effect only for some kinds of hearing impairment.

Automatic program selection

A few years ago, one main tendency was the development of so called „automatic“ hearing instruments. The basic idea is rather simple: since the ear itself does not have any trimmers or switches, a hearing instrument should not need them either. Regarding the loudness processing, technologies like WDRC or the K-Amp™ (Killion 1993) reached this goal quite well. However, a closer look reveals that even in normal hearing different strategies are used in different listening situations and acoustical envi-

ronments. In hearing instruments this strategy was realized in programmable hearing aids having multiple memories. This allowed for an optimized fitting for different situations. In the early days of such hearing instruments, the handling was a major problem due to tiny switches and lacking feedback for „successful“ program changes. A remote control made the handling much easier, but caused higher costs.
An obvious idea is a mechanism that allows automatic program changes. This requires an analysis of the acoustical situation by the hearing instrument and decisions about the type of situation and the selection of an appropriate program or the activation of the appropriate signal processing elements (e.g., noise suppression or directional microphones in noisy situations).
An first attempt to distinguish speech from non-speech signals used the characteristic modulation spectrum of speech: speech signals show a characteristic temporal behavior independent of the speaker or the spoken language: the fluctuations (modulation spectrum) of the speech envelope have a chararacteristic maximum at about 4 Hz. Hearing instruments can analyze the envelope of the signal in each band, if modulation frequencies typical for speech are found, the signal is classified as speech-like and the gain is kept constant. Otherwise the signal is classified as noise-like and the gain in that channel is reduced. More complex hearing instruments analyze up to four different parameters, which allows a more complex classification.

Other processing approaches

Some amount of the acoustical energy delivered to the ear canal escapes through openings in the earmold or ITE shell and enters the microphone of the hearing aid again. Especially with high power hearing aids and with open earmolds, amplification can lead to feedback. Several strategies for feedback reduction are used in hearing instruments. Some instruments include notch filters that can be adjusted to the feedback frequency. In multiband instruments, the gain of the channel in which feedback occurs, can be reduced. Feedback reduction is also a side effect of the speech processing mentioned above: if feedback occurs, the feedback tone has a high level, but no modulation. Therefore it is classified as „non-speech“ signal, the gain in that channel is reduced and the feedback stops.
Most hearing impaired persons suffer from a reduced speech intelligibility in noisy situations. The ability to understand speech in noisy situations depends primarily on the binaural hearing system processing the signals from both ears. This implies that persons with a symmetrical hearing loss should be fitted bilateral with two hearing instruments. Another approach to improve the signal-to-noise ratio is to use directional microphones. Directionality can be realised using two unidirectional microphones, thus allowing to switch between direcional mode for noisy situations and omnidirectional mode for situations, in which the wearer does not only want to hear sounds from the front. Further processing of the signals from two microphones can be used to control the shape of the directional pattern. An expansion circuit can help to reduce the additional circuit noise generated by directional microphones.
It is well known that the capacity of the brain to filter wanted speech sounds from the unwanted background sounds also degrades with age. Therefore it would be beneficial if the hearing aid could distinguish „wanted“ from „unwanted“ sounds. Some attempts have been made to reduce noise electronically. One strategy is to reduce gain, either in the low frequencies or in specific bands, when steady-state signals (noise) are detected. Although research findings supporting the efficacy of digital noise reduction systems are mixed, they do indicate that the noise reduction can work to reduce annoyance and possibly improve speech recognition in the presence of non-fluctuating noise. This type of noise reduction is sometimes advocated as complementary processing to directional microphones. While directional microphones can reduce the levels of background noise regardless of its temporal content, they are limited to reducing noise from behind or to the sides of the user.

SUMMARY

Hearing instruments technology has developed quickly and dramatically over the past few years thus providing dramatic potential improvements for hearing impaired subjects.
This paper describes how psychoacoustics can contribute to the assessment of hearing deficiencies. New measurement methods like loudness scaling and new speech tests reveal more information about hearing defiencies that are valuable for the fitting process. The signal processing schemes used in contemporary hearing instruments use psychoacoustic knowledge to improve listening abilities and speech intelligibility in noisy situations, thus creating best possible benefit for the hearing impaired.

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