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Senso: Audiological background[1]

Introduction

The SENSO hearing instrument from Widex represents a technological breakthrough in the hearing aid industry given the utilization of digital signal representation throughout the entire signal path from microphone to output transducer. By developing digital technology to a level applicable for use in hearing aids, most of the limitations previously imposed on the complexity of hearing aid circuitry have been eliminated. This degree of freedom has been utilized to address a number of unsolved problems experienced by hearing aid users.

SENSO utilizes this digital technology in three different areas. First, to increase speech intelligibility in difficult listening environments by incorporating a number of digital signal processing algorithms. Secondly, to alleviate some of the problems traditionally known to bother hearing aid users: acoustic feedback and internal noise from the microphone, and thirdly, to optimize the fitting for the individual patient by providing the audiologist/hearing aid professional with a versatile fitting tool.

This article treats the first of these issues by reviewing the audiological background which constitutes the basis for the SENSO signal processing. The audiological principles take their starting point in the functional differences between the normal and the impaired ear.

Normal Hearing

Normal hearing is based upon the conversion of an incoming sound into a nerve code in the sensory cells of the Organ of Corti in the inner ear. It is well known that this transformation is made by the interaction of the two types of sensory cells: the inner hair cells (IHC) and the outer hair cells (OHC). These hair cells are arranged in four rows along the basilar membrane: one row of inner hair cells and three rows of outer hair cells.

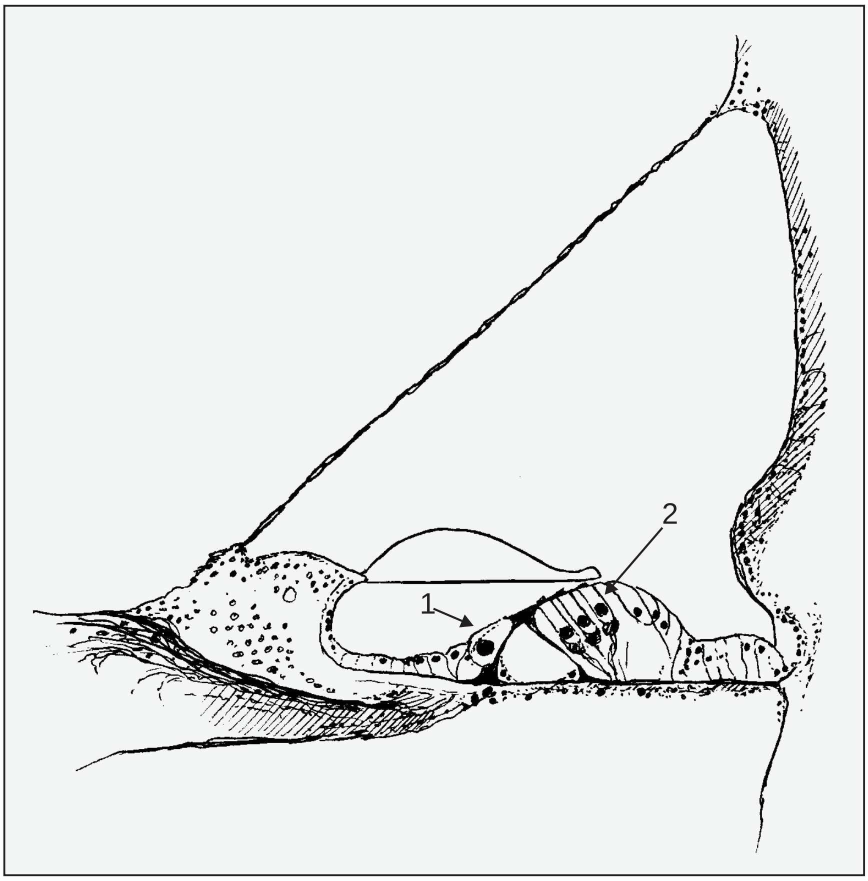

Fig. 1: Cross section of the Organ of Corti in the inner ear. Inner (1) and outer (2) hair cells are indicated with arrows.

1. This article is a slightly expanded version of “Basic Amplification rationale of a DSP-hearing Instrument,” Hearing Review, March 1997.

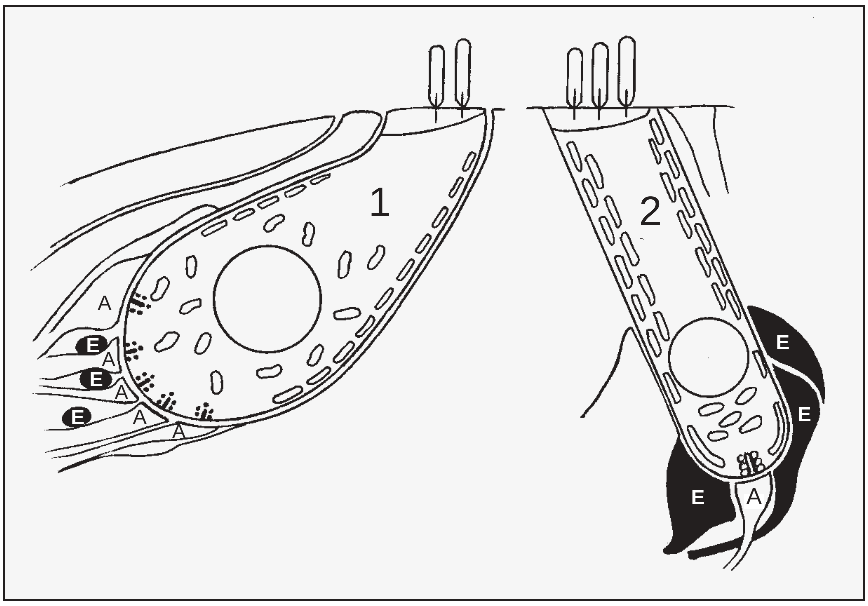

Fig. 2: Inner (1) and outer (2) hair cell with afferent (A) and efferent (E) nerve cells, see also Fig. 1.

During the past decade it has become evident that the inner hair cells are the factual sensory cells. Thus, it is the inner hair cells which, during acousto-mechanical excitation, transform the mechanical movement of the sound stimulus into a complicated pattern of nerve impulses. These nerve impulses are transmitted to the auditory cortex via a structured network of nerve fibers and synapses.

The outer hair cells function, in many aspects, like a servo-control. This servo-control mechanism has the effect that a low level or soft sound is able to evoke sufficiently large vibrations of the basilar membrane, which in turn will excite the inner hair cells. This specialized function of inner and outer hair cells is reflected in several observations. E.g. in a close-up picture of the hair cells (figure2), it is observed that the nerve cells which innervate the inner hair cells are predominantly of the afferent type (marked with A). These afferent fibers transmit impulses from the sensory cell to the brain. The outer hair cells, on the other hand, are innervated by efferent nerve fibers (marked with E). These efferent fibers transmit nerve pulses from the central part of the hearing organ to the sensory cell. Moreover, chemical analyses have shown that the outer hair cells contain bio-chemical compounds like actin, which are similar to those found in muscle tissue. When outer hair cells are stimulated, they have the potential of changing their length and in so doing, they enhance the vibrations of the basilar membrane. The outer hair cells react almost instantaneously to stimulation and they are able to change their shape with a very fast rate, i.e. more than 20.000 times per second.

The function of the outer cells is non-linear in the sense that at very weak sound levels there is a significant effect of outer hair cell function, whereas more intense signals generate almost no effect in function.

Hearing Loss

A sensorineural hearing loss may have a number of causes. Typically, a loss of hair cells is the primary causative factor for a sensorineural hearing impairment.

Some psycho-acoustic consequences of a loss of hair cells

A loss of outer hair cells will create a malfunction of the characteristic servo mechanism, causing weak sounds to be inaudible, while more powerful sounds (which in normal hearing do not involve the contribution of the outer hair cells) will be perceived with normal loudness. This well-known recruitment phenomenon is thus characterized by an increased threshold of hearing, HTL, without a similar shift in the threshold where the signal becomes uncomfortably loud, UCL. In this way, the effect of a loss of outer hair cells can be thought of as a loss of compression.

Inner hair cells are less vulnerable than the outer hair cells. However, when inner hair cells are damaged, one consequence of this is a loss of selectivity at all input levels. Reduced frequency selectivity, and the corresponding spread of masking, reduces the ability to communicate in background noise.

Recruitment
The vast majority of hearing losses include a loss of outer hair cells and, as a consequence, will involve loudness recruitment. (It should be noted that recruitment is normally not associated with retrocochlear lesions nor the presence of a conductive hearing impairment).

Loudness perception by hearing impaired persons
The loudness with which a sound is perceived is stated in a subjectively defined scale: the Sone[2] scale. For sounds well above the hearing threshold, i.e., at medium to high levels, experiments have shown that, in normal hearing, the loudness will double as the level increases with 10 dB (i.e., at high suprathreshold levels, an increase of 10 dB will result in a doubling of the per-

2. The Sone scale is related to the physical dB scale by defining that the loudness 1 Sone corresponds to the average loudness perceived by normal hearing listeners when listening to a 1000 Hz pure tone at a level of 40 dB SPL. The Sone scale is a so-called ratio scale, and n Sones denotes the loudness of a sound which appears n times as loud as a sound with the loudness of 1 Sone.

ceived loudness). In figure 3, recruitment is exemplified by showing the perceived loudness (in Sones) of a 1000 Hz tone for a normal hearing individual versus an individual presenting a 40 dB hearing impairment.

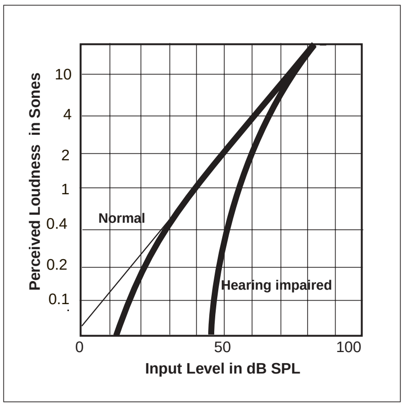

Fig. 3: Loudness versus the level of a 1000 Hz pure tone. Curves are for a normal hearing person and a person with a 40 dB hearing loss.

In figure 3 it can be seen that a 1 kHz tone of 40 dB SPL generates a perceived loudness of 1 Sone for a normal hearing person. However, the tone must be 15 dB more intense, i.e. 55 dB SPL, in order to give the 40 dB hearing impaired person a similar loudness perception. Thus, at this intensity level, a 15 dB insertion gain will normalize the loudness of this tone. At higher input levels, however, the loudness perception tends to normalize for the hearing impaired person when compared to the normal hearing individual. For example, in figure 3, at an input level of 75 dB SPL, both the normal hearing and the hearing impaired obtain a loudness of 10 Sones. Thus, at this level, no additional gain is required at 1000 Hz. Figure 3 may, therefore, be used to calculate the insertion gain as a function of the input sound pressure level which is required to obtain 'loudness normalization' for a 1000 Hz tone.

Recruitment also reduces the range of sound which can be detected by the recruiting ear. The area between the threshold of audibility and the threshold of discomfort defines the dynamic range of hearing. It is well known from the literature that individuals suffering from re-

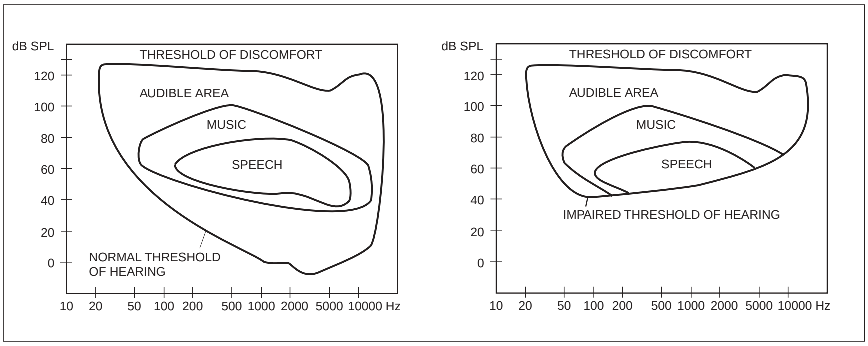

Fig. 4: The auditory range for a normal hearing and a person with a cochlear hearing loss.

cruitment due to a sensorineural hearing impairment generally have a reduced dynamic range as compared to normal hearing individuals.

This is illustrated in figure 4 where the dynamic range for a normal versus a hearing impaired individual is illustrated. It is evident that for the hearing impaired individual, the dynamic range is reduced relative to that of the normal hearing individual.

The differences in loudness perception between the normal versus the hearing impaired individual are used in the development of the algorithms which determine the amplifier characteristics of the SENSO.

Masking

Masking designates the phenomenon whereby the presence of one sound may cause another sound to be inaudible or to reduce its loudness, the latter being known as partial masking. Masking may take place in frequency and in time. Thus, a tone with a certain frequency may mask a tone of another frequency present at the same time, or up until 200 ms after the sound has ended.

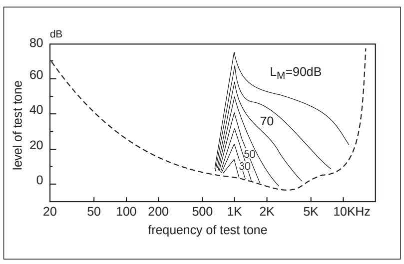

Fig. 5: Masked thresholds with 1000 Hz pure tone maskers at levels ranging from 20 to 90 dB SPL.

Spread of Masking and frequency resolution

Figure 5 shows the result of a masking experiment. In the experiment is determined the presentation level required of tones of different frequencies in order to be heard when a simultaneous 1000 Hz masker tone is present. When the frequency of the tone is close to 1000 Hz, the masking is significant while no masking occurs at remote frequencies. Figure 5 also indicates that the masking curves are asymmetrical and that the masking effects are greatest toward the higher frequencies. This phenomenon is referred to as an upward spread of masking. It is widely acknowledged that many hearing impaired individuals experience excessive upward spread of masking, which has the consequence that intense low frequency sounds such as noise may mask weak speech components to a higher degree than in normal hearing.

The excessive spread of masking is closely related to other functional deficits associated with hair cell loss. These include impaired frequency resolution which denotes that the ability to separate frequency components which appear at the same time is reduced. These symptoms are likely to involve listening problems for hearing impaired individuals when listening to speech in background noise.

Speech Understanding

Speech in quiet

Speech understanding in quiet seldom causes any significant problems for the hearing impaired, sometimes even in the absence of a hearing instrument. This is primarily due to the presence of superfluous information or cues i.e., visual, contextual, structural, situational, phonemic etc., denoted as redundancy. In a quiet environment, the redundancy may compensate adequately for the loss of acoustic information caused by the hearing impairment. Severe problems arise when additional acoustic information becomes unavailable, e.g. due to masking by background noise.

Speech in noise
It is well known that most consonant sounds are much weaker in acoustic energy than vowel sounds. Normal speech intelligibility requires the perception of weaker, high frequency consonants as well as more powerful low frequency speech components which are higher in acoustic energy. In a situation where low frequency background noise is present, the ability to perceive these high frequency components will be disproportionately more difficult for the hearing impaired individual with impaired frequency resolution than for the normal hearing individual. The excessive masking, experienced by the hearing impaired individual, will add to the elevated threshold and result in a loss of "speech understanding" or "distinctness" of speech due to the loss of consonantal speech segments.

Overamplification
An increase in the speech level does not always lead to improved speech understanding. When the speech level is raised above a certain level, a more or less pronounced drop in intelligibility may be the consequence. This effect is more pronounced for hearing impaired listeners than for listeners with normal hearing. Figure 6 illustrates the negative effect on speech intelligibility for hearing impaired *and* normal hearing individuals when the overall level of speech and noise is increased beyond a certain limit, and that the negative effect is more pronounced for hearing impaired listeners than for normal hearing individuals.

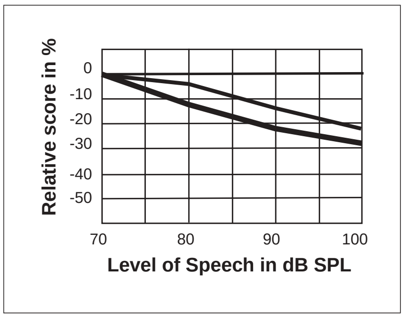

Fig. 6: Intelligibility of speech in noise as a function of overall level. Lower bold curve is for hearing impaired listeners, upper thin curve is for normal hearing listeners (adapted from Studebaker, Sherbecoe & McDaniel 1995).

Thus, it can be seen that the hearing aid user experiences benefit from amplification only to a certain limit, and that overamplification results in negative effects.

Influence of noise type
It is a well known experience from studies of multiple program hearing aids that hearing impaired listeners prefer a different frequency response in different noise environments. Keidser (1995) has studied this phenomenon by comparing the preference among three frequency characteristics and found (see figure 7) that the frequency response defined by the NAL fitting rule was preferred when listening to speech in a background of babble noise while a steeper slope with less bass and more treble than NAL was preferred in a background of traffic noise and a shallower response with more bass and less treble was preferred in a quiet background. In our own research, we have observed similar effects provided the reduction of gain did not affect the audibility of the speech signal. These observations have been utilized in the formulation of the SENSO signal processing.

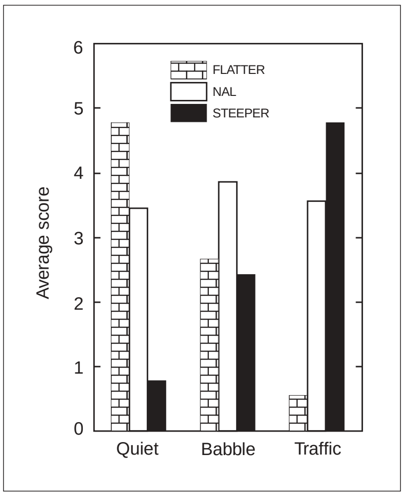

Fig. 7: Preference of frequency response in different listening environments (adapted from Keidser, 1995).

The SENSO Hearing Instrument

Basic Amplification Template of SENSO

The signal processing of SENSO aims to compensate for the negative effects which result from a sensorineural hearing impairment. More specifically, SENSO's signal processing algorithms, which determine its amplification as a function of time and frequency, aim to compensate for loudness recruitment and excessive masking. This is accomplished by developing digital signal processing algorithms which operate according to two underlying, continuously operating principles, *Loudness Mapping* and *Noise Reduction.* According to the first principle, the presentation level and frequency balance will be continuously adjusted to compensate for the long term effects of recruitment. According to the second principle, noise reduction, the sound image will be further shaped in order to minimize the effect of a detrimental listening situation e.g. with heavy background noise. In different noise backgrounds, this latter compensation gives a change in the frequency response which corresponds to the change in preferred frequency response in different listening environments.

The Loudness Mapping principle

The goal of Loudness Mapping is to present complex signals like speech to the hearing aid user in such a way that their different components are perceived with the same loudness as a normal hearing person perceives them. Thus, it is necessary over the entire frequency range, to transform the wide dynamic range of speech (in different environments) to fit within the narrow dynamic range of the impaired listener (i.e., restoring a sense of “natural loudness”). Loudness of a complex sound depends, in a complicated way, on its spectral content and its temporal course, and most attempts to restore natural loudness have disregarded this summation of loudness over frequency and time.

The first attempts to compensate for loudness recruitment in the early 70es, used compression with very short time constants (Villchur 1973). This fast regulation may seem the logical choice when compensating the lost function of the fast acting outer hair cells. However, experiments have shown that a fast acting compression has a tendency to create distortion and to enhance noise during speech pauses. This so-called pumping effect was perceived as annoying by most users. With regard to speech intelligibility, the fast acting compression did not provide quite as good speech intelligibility in most situations as did a slow acting compression and our own research showed that a slow regulation was preferred by the majority of hearing aid users.

Thus we concluded that the loudness mapping should predominantly include an adjustment of level in each frequency band according to the long term level of the speech signal. This slow adjustment may be implemented in a way which provides audibility of the speech signal without blurring its temporal structure and without creating severe pro- cessing artifacts like pumping or distortion.

Consequently, the loudness mapping strategy implemented in SENSO is primarily based on the long term properties of the incoming signal. Since it is known that in fast varying acoustic environments a slow regulation may have some drawbacks, separate algorithms have been developed to minimize these side effects.

Practical implementation

The starting point for SENSO’s loudness mapping is an in-situ determination of the hearing threshold in each of SENSO’s three frequency bands. This is obtained by letting the hearing aid generate and present complex tone signals to the user. The advantage of this method is that the individual acoustic properties of the user’s ear and earmould are included in the threshold values and these are more suitable for hearing aid fitting than normal audiogram values obtained with a transducer different from the one used in the hearing aid. The three thresholds obtained in this way are used to determine the target input/output (I/O) curves for each of the three frequency bands. This is accomplished according to a Loudness Mapping scheme, which is based on the measured differences in loudness perception for normal hearing versus hearing impaired listeners reported by Pascoe (1988).

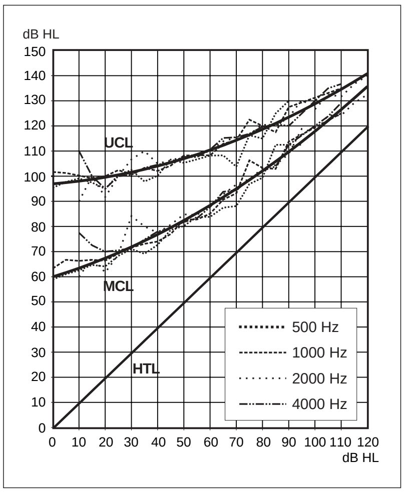

Fig. 8: The uncomfortable level, UCL, and the most comfortable level, MCL, for several frequencies recorded as a function of HTL, from Pascoe (1988).

Pascoe's (1988) data explored the auditory dynamic range and recorded MCL and UCL judgments in reference to hearing threshold levels for a large group of hearing impaired individuals. The data revealed that the relations between MCL or UCL and the hearing loss, were independent of frequency (when measuring the input level in dB HL, see figure 8) i.e. the mean MCL and UCL judgments for each hearing threshold level did not show a significant effect of frequency. We found that Pascoe’s relation between average HTL and UCL was well approximated by a second order polynomial and we utilize this relation in the SENSO fitting algorithm for estimating UCL thresholds from the individual HTL values. Pascoe's mean UCL values are slightly lower (approximately 10 dB) than those found in most studies (for a recent discussion see Elberling and Nielsen, 1993), but, by utilizing Pascoe's data, we are choosing more conservative values which provide a margin to fine tune for individual deviations from the mean UCL. This possibility to fine tune calculated UCL values is of particular significance, clinically, given the range or variation of individual UCL thresholds observed for a given audiological configuration.

Studies of loudness perception of hearing impaired listeners have shown that recruitment takes place, in particular, at levels just above the hearing threshold (e.g. Hellman, 1990) and is manifested by a steepening of the loudness function. This observation has been included in a calculation template which enables the calculation of a probable loudness function at any given hearing loss. This template is used in the calculation of SENSO's Loudness Mapping function and is illustrated in figures 9 and 10.

This loudness calculation template is likely to predict loudness for pure tones and narrow band noise, which are not representative of daily listening environments. Typically, environmental sound is comprised of speech (or music) possibly in the presence of background noise or reverberation. Such input signals are characterized by having a broad band spectrum and by fluctuating considerably over time. They are consequently neither stationary nor narrow band. For such signals, the calculation of loudness is far more complicated than when considering stationary narrow band signals. Recent research, however, has indicated that no substantial improvement is likely to be achieved by introducing more sophisticated loudness models when compensating

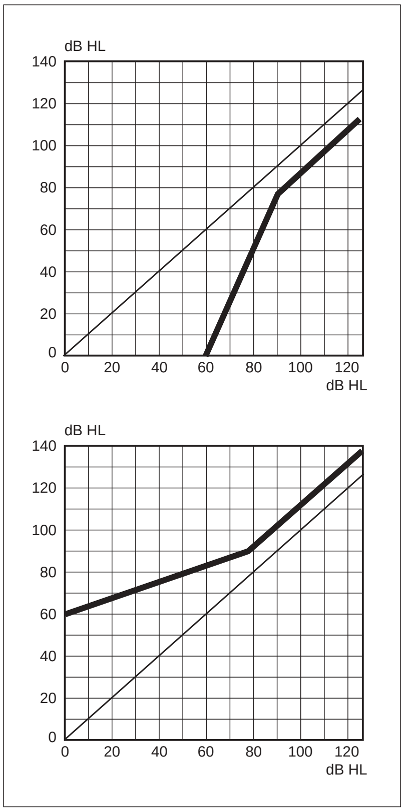

Fig. 9: Loudness template and corresponding input/output characteristic.

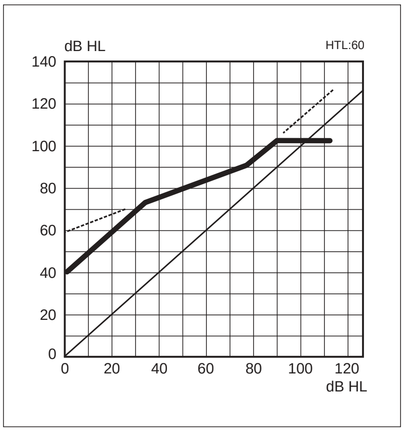

Fig. 10: I/O curve with gain and output limitation.

for loudness recruitment. Our own research has shown that the combination of the slow acting, statistically based type loudness mapping and the noise reduction approach used in the SENSO has been positively judged by practically all test persons, also when asked specifically about the loudness of the amplified signal. A special feature of the slow acting compression is that users generally accept a very low compression kneepoint, and the fitting tools allow for an individual setting of the kneepoints in all bands. The lower kneepoint, representing the level above which compression takes place, may therefore be set as low as the individual earmould or shell allows. When using a fast regulation, a kneepoint lower than approximately 45 dB SPL is rejected by a majority of the users. Note the apparent paradox, that just above threshold where the action of the outer hair cells and thereby the recruitment is at its maximum, most users do not accept a fast recruitment compensation. With the SENSO a kneepoint as low as 15-20 dB SPL has been positively received.

Temporal aspects

In most situations, especially those with a fairly continuous background noise, we have found that a slow acting regulation was preferred. This is consistent with a number of recent studies (e.g. Neuman et al. 1994). In some situations, however, where impulse noise is present, a faster regulation was generally preferred. Consequently, in the SENSO we have incorporated a complex regulation algorithm which combines both fast and slow acting regulation to address these environmental differences. As long as the listening environment is stable, the release time will be very long and the regulation will be slow. In this situation, the SENSO will act as a perfectly linear device, with corresponding low distortion. However, if the environment changes, if the noise level increases, or if a sudden, loud impulse noise arises, the regulation times will automatically become faster. In such situations SENSO's mode of operation is highly nonlinear.

Noise reduction

A second principle, denoted "Noise reduction", modifies the SENSO Loudness Mapping scheme according to a statistical analysis of the input signal, carried out independently within three separate bands. The noise reduction aims at improving the hearing impaired listener's possibilities of listening to speech in severe background noise. In such noise, speech is produced with a higher than normal effort, which gives rise to a shift in spectrum and a higher overall level. If a band contains intense noise, this will inevitably cause masking. In such a band, the speech level is typically well above threshold and a gain reduction may significantly reduce masking without corrupting the available speech information. In order to utilize this principle, it is necessary to ascertain what is speech and what is noise. This is of course a difficult task bearing in mind that background noise often consists of multi-talker speech. We approached that problem by studying the statistical properties of various signals. It appeared that the speech from a single talker differed substantially from the statistical properties of various types of noise including multi- talker babble, see figure 11.

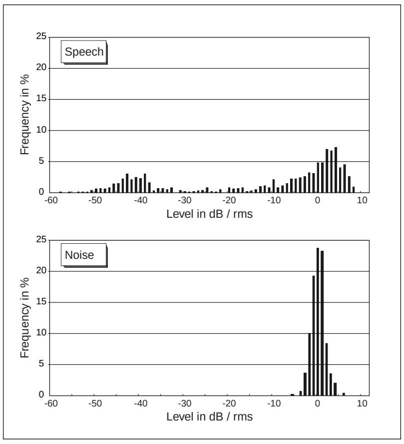

Fig. 11: Level distributions of speech and noise. Short term levels are in dB re. long term rms value.

Using digital technology it proved possible to implement a comprehensive statistical analysis of the level distribution in each band. This continuously running analysis of the incoming signal uses time windows with durations of up to 30 seconds. The analyses provide running estimates of the composition of speech and noise in each band. As an example, if the statistical analysis classifies the input signal as a speech signal in intense background noise, then the amplifier characteristic is modified to reduce the loudness of the noise (and the speech). This modification takes into account the fact that the speech signal should remain well above threshold. In "Noise reduction", therefore, the "sound image" is adjusted in order to optimize speech intelligibility by minimizing masking effects. This statistical analysis of the input signal is a continuously ongoing facet of SENSO's digital signal processing.

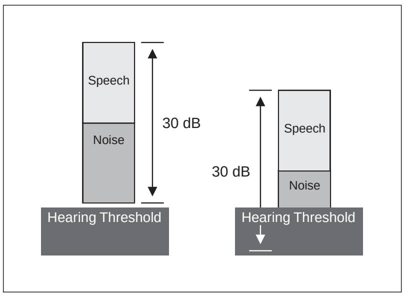

Fig. 12: Principle of the noise reduction strategy in SENSO: If no noise is detected, the speech signal is reproduced directly according to the loudness mapping template.

Throughout this statistical analysis, a running update of the parameters which control the SENSO sound reproduction takes place. This principle is illustrated in figure 12. When listening to speech in a noise-free environment, the hearing instrument characteristics (the control parameters) are calculated according to the loudness mapping template without any modifications. As such, speech will be audible in its full dynamic range across each of the three frequency bands. However, should speech be present in the context of background noise, or if noise alone is present, the calculation scheme is modified in order to minimize masking effects through gain reduction.

Aside from the audiological benefits provided by this speech enhancement/noise reduction algorithm, a secondary benefit to the user is a reduction of fatigue in the presence of severe background noise. This may explain the increased duration of hearing aid use observed in preliminary field studies.

Summary and conclusions

In summary, the SENSO sound reproduction is determined according to in-situ thresholds measured by means of the hearing aid itself. The sound reproduction is basically determined by the long term statistical properties of the different band levels, but additional fast operating functions have been included in order to handle situations with fast varying listening situations. Moreover, a noise reduction paradigm is added on top of the loudness mapping. The noise reduction is based on a continuous statistical analysis of the levels in each band which result in an estimate of the speech and noise level in this band and possibly in a relative reduction of the band amplification. In this way, the sound reproduction of SENSO not only depends on the hearing loss configuration and the spectrum of the sound environment, but also on the nature of the acoustic environments in which the instrument is utilized.

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