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Audiological background and design rationale of Senso Diva

Introduction

Improving speech understanding in noise is often stated as the main reason for acquiring and wearing a hearing aid and, indeed, better speech understanding in adverse listening conditions has been a major objective during the development of the new Senso Diva hearing aid. Nevertheless, many other important issues have also been addressed. These include the problem of preserving listening comfort under adverse listening conditions and of providing satisfactory reproduction of music and nature sounds. Another challenge is to establish audibility of weak sounds without trading off comfort. In fact, substantial efforts have been devoted to obtaining a pleasant distortion-free sound image without introducing audible artefacts from the signal processing. This has been done in order to make it more appealing to use the hearing aid also in situations where it is not necessary for the sake of speech understanding. Another main focus has been on problems related to hearing aid use in general. Within this category, two main enterprises have been: 1) to prevent whistling tones from being generated by acoustic feedback and 2) to reduce the feeling of blocked ear canals caused by the occlusion effect when using hearing aids. Thus, Widex has been committed to using today's technological possibilities extensively to provide solutions to problems or inconveniences experienced by hearing aid users. This has led to many new features and one might fear that the complexity of the many new features would result in a meticulous programming procedure. Therefore, all fitting and fine tuning procedures have been reconsidered and redesigned if appropriate. In this connection a new portable programmer and a new structure of the Compass fitting program have been worked out. This article will provide the audiological background for Senso Diva and will also present an overview of its features.

Consequences of a sensorineural hearing loss

A sensorineural hearing loss is typically associated with a number of functional deficits, some of which are extremely difficult to alleviate.

Loudness recruitment

Loudness recruitment is closely associated with hearing loss of cochlear origin, such as presbycusis or noise trauma, and it seems well documented that recruitment is a consequence of a loss of outer hair cells. Due to the compressive function of the outer hair cells this loss may be regarded as a loss of compression. Figure 1 shows an I/O curve for the basilar membrane in its normal condition and when outer hair cells have been made temporarily inactive by injection of furosemide (from Ruggero & Rich, 1991). The figure shows how compression is lost when outer hair cells are deactivated.

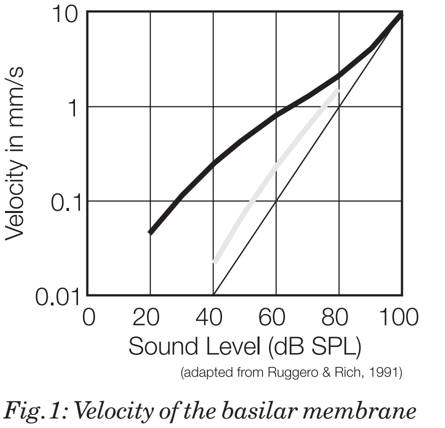

Fig. 1: Velocity of the basilar membrane in a normal cochlea (black) and in a cochlea temporarily damaged (gray) by injection of an ototoxic agent (furosemide).

An intuitively tempting method of compensating for a loss of compression would be to build a compression circuit into a hearing aid, which imitates the missing compression of the cochlea as closely as possible. Actually, this is the rationale for recruitment compensation in several other hearing aids. Cochlear models of various degrees of complexity have then been applied. Such approaches may be characterized as loudness normalization, since their objectives are to restore loudness of all possible sound stimuli to normal. In general, recruitment is frequency-dependent and it is therefore necessary to counteract its effects differently at different frequencies. The optimum number of channels is not known and the exact number is probably not critical. Since many auditory processes are related to the critical bandwidth it has been suggested to use channels with critical bandwidth to compensate for recruitment. Critical bandwidths as a function of frequency are shown in figure 2.

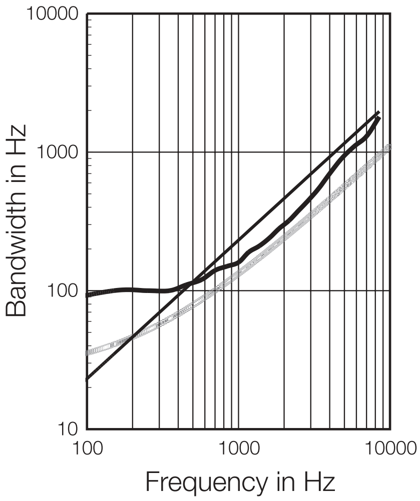

Fig. 2: Critical Bandwidths according to Zwicker (black) compared to Equivalent Rectangular Bandwidths according to Glasberg and Moore (gray) and 1/3 octaves or terts (black line).

Because outer hair cells react very fast to external stimuli, fast-acting gain regulation is required for imitating their functions. However, certain observations speak against a minute imitation of outer hair cell functions and thus against exclusive use of a loudness normalization approach. One is Byrne's 1996 observation that loudness normalization is not optimal with regard to speech intelligibility. Byrne suggests a different strategy, denoted loudness equalization. A second observation is that hearing impaired persons find slow-acting regulation more pleasant and less noisy than fast-acting regulation (Neuman et al., 1998).

Reduced frequency selectivity

Many persons suffering from a sensorineural hearing loss experience problems in separating sounds of different frequencies when sounds are presented simultaneously. This loss of frequency selectivity also manifests itself as excessive upward spread of masking. Reduced frequency selectivity raises extraordinary problems with speech understanding in noise since frequency differences become more difficult to detect. There seems to be no effective way to compensate for reduced frequency selectivity. One attempt has been to exaggerate spectral contrasts, but the results have not been convincing. Other approaches have had a more defensive character: to avoid fast multi-band regulation, which has a tendency of smearing spectral contrasts and to avoid excessive amplification, since the effect is more pronounced at higher levels.

Temporal resolution

Temporal resolution also seems afflicted by a sensorineural hearing loss. Temporal resolution indicates the ability to distinguish consecutive pulses as separate events. Temporal resolution is level-dependent and is reduced as the presentation level approaches the hearing threshold. Due to the reduced dynamic range in sensorineural hearing loss, speech sounds are often presented only slightly above threshold. And at these sensation levels, temporal acuity is reduced also in normal hearing. This appears as a reduction in temporal resolution, but has little to do with the hearing impairment. In addition to this, some hearing impaired listeners experience a further reduction of temporal resolution beyond what is to be expected solely from the low presentation level. The net effect is that speech features, which are normally identified by temporal cues, are difficult to recognize.

Basic reproduction of speech and environmental sounds in Senso Diva

Senso Diva is designed for compensating for the deficits associated with sensorineural or mixed losses. Senso Diva is basically a multichannel compression hearing aid in which the compression system is designed primarily for recruitment compensation. The frequency dependence of recruitment is coped with by using many channels with narrow bandwidths approximating the widths of the critical bands. There is a compression block in each of the channels. The compression system is of the type "enhanced dynamic range compression" with multistage compressors comprising five segments, see fig. 3. At medium input levels, Senso Diva utilizes the speech equalization principle according to which normal speech shall be amplified to match the user's MCL loudness contour (e.g. Byrne & Dillon, 1986). This strategy defines the gain at typical speech levels. In addition to this the amplification objective at high input levels is to reproduce high-level input sounds slightly below the uncomfortable loudness contour. Similarly, the amplification rationale at low input levels is to amplify low-level input sounds to slightly above the threshold of the listener. In this way a loudness mapping strategy is used for soft and loud sounds. An example of a basic I/O curve and the corresponding input/gain curve for a moderate sensorineural loss is shown in figures 3 and 4. Note the five segments, each of which has its function for accomplishing the fitting rationales described above. The five segments also provide an effective and versatile tool for fine tuning, which will be described later in this article. The position and slope of the segments are calculated in accordance with the fitting rationales outlined above. In this calculation it is necessary to include the influence of filter bandwidth, detector type, regulation speed and other hearing aid specific factors, see Kuk & Ludvigsen (1999a).

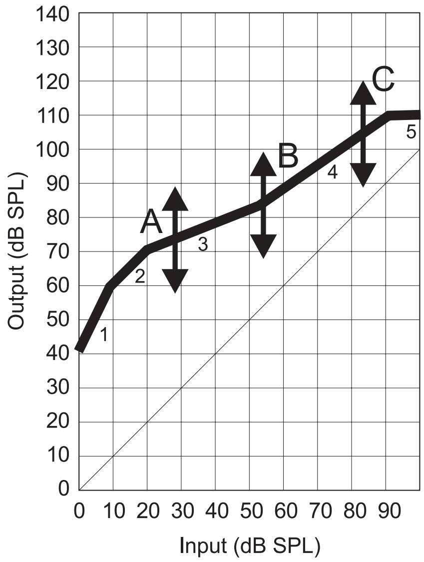

Fig. 3: Example of a multi-segmental I/O curve for a moderate sensorineural loss.

Comfort at high input levels

Many hearing aid users experience discomfort when wearing their hearing aids in high intensity noise (Kochkin, 1995). The sound environments reported to create discomfort vary from sounds with a low frequency emphasis, such as noise from heavy traffic, party noise or loud pop music over the sound of playing children to high frequency sounds like whistling and chinking of tableware. In today's hearing aids, a variety of measures have been taken to prevent discomfort. In linear hearing aids an output-controlled limiter, AGC, is typically used and this may to some extent relieve the problem by keeping the output of the hearing aid within comfortable levels. In nonlinear hearing aids, various combinations of compression and limiting are used.

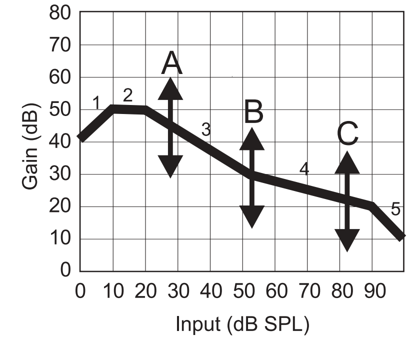

Fig. 4: Corresponding gain vs. input (I/G) curve, cf. fig. 3.

High Level Compression
Senso Diva is equipped with compressors in all channels. Each compressor characteristic comprises five segments, cf. figures 3 and 4. These multi-segmental characteristics provide powerful means for controlling the sound reproduction of Senso Diva. The sound reproduction at high input levels is determined by the upper part of the characteristic, i.e. segments 4 and 5. The connection between segment 3 and segment 4 acts like a hinge that allows the compression ratio of segment 4 to be adjusted separately from the lower three segments. In this way separate control over the sound reproduction at high input levels in all channels is provided. This system is referred to as High Level Compression, HLC.

This unique feature allows the total dynamic range of the residual hearing to be utilized. The fitting software automatically calculates the compression ratios using the threshold data recorded in-situ in the Sensogram. The "hinge" between the two compressor segments (3 and 4 in fig. 3) represents the peak levels of normal speech. Thus, the high level compressors operate at levels exceeding those of normal speech. HLC allows a detailed match of the hearing aid output to the residual dynamic range of the listener in each frequency range separately. Segment 5 of the I/O curve is determined by the automatic output control system, AOC. In each channel a fast-acting limiter-type, AGC, prevents distortion due to overloading. The AOC is mainly active when little compression is used in the HLC, which is typically preferred in conductive and mixed losses. For moderate sensorineural losses the AOCs are seldom activated because HLC keeps the output below the compression threshold of the AOC. In cases with mixed or conductive losses the air-to-bone gaps are taken into account when calculating high-level compression ratios. Special possibilities exist for compensating for hyperacusis or hypersensitivity, since adjustment possibilities extend to zero or negative gain.

Audibility at low input levels

Ideally, a hearing aid should provide the user with a normal hearing threshold. With modern hearing aids this may be accomplished in mild and moderate losses without severe technical problems. Using fast-acting compression most hearing aid users, however, prefer less gain for low-level input than required for establishing a normal threshold (Dillon et al., 1998). This is presumably caused by weak environmental noise, which becomes audible if a low aided threshold is established. Many hearing aid users may consider this "noisiness" cumbersome.

Low compression thresholds
In Senso Diva we have exploited that a slow regulation system causes less noisiness and that compression thresholds therefore can be set to levels as low as 20-30 dB SPL without bothering the majority of users. In this way the aided threshold obtained with Senso Diva will in practice come closer to normal than with hearing aids using fast compression. This enables the user to detect even very weak sounds like remote birds singing or footsteps from an adjacent room, see Kuk (1997) for further details. Since individual hearing aid users may prefer a different gain for weak sounds, the gain for weak sounds can be adjusted in the fine tuning procedure. The adjustment of gain for low-level input corresponds to a shift of the compression threshold, CT.

Slow-acting compression with Sound Stabilizer™
In Senso Diva the instantaneous gain in each channel varies according to the properties of the input signal. Senso Diva uses long time constants in order to utilize the positive effect this has on sound quality (Neuman, 1998; Hansen, 2000). In contrast, fast gain regulation has the effect that background noise sounds louder. Senso Diva uses slow-acting regulation of gain whenever feasible. In abruptly-changing sound environments slow-acting regulation may not be favorable since it may

adapt too slowly to rapidly-changing situations. In order to counterbalance these negative effects of slow-acting regulation, a Sound Stabilizer™ is implemented in all channels. The Sound Stabilizer™ optimizes the speed of regulation in a compressor by slowing down the regulation in stationary listening situations and speeding it up when the listening situation changes. The functionality of the Sound Stabilizer™ is described in detail elsewhere, see Ludvigsen & Paludan-Müller, 1999.

Improving benefit in noisy environments

Noise Reduction with Speech Intensification

For many years, research has been devoted to the problem of separating speech from noise in order to attenuate noise and enhance speech. It goes without saying that this task is very difficult – especially if the competing noise is speech produced by one or several talkers.

Widex introduced the first successful Noise Reduction/Speech Enhancement system in hearing aids with the launch of Senso (Sandlin, 1996). This system relies on a statistical analysis of the incoming signal and speech is distinguished from background noise by continuously monitoring the distribution of short-term intensity levels within each of its three channels. Due to the rapid level fluctuations of speech, its level distribution is quite different from that of typical background noise, see figures 5 and 6. Since a speech signal is, in acoustic terms, a sequence of speech sounds separated by brief pauses its level distribution is bimodal with levels falling in two separate clusters. Weak speech sounds and speech pauses are clustered at low levels and intense speech sounds like vowels are registered at high levels. In environmental noise the typical level distribution is much narrower with only one mode since no silent or weak segments are present. In Senso, this difference is used to distinguish speech from noise. In this way, the distinction between speech and noise is based on the statistical properties of the respective sound types.

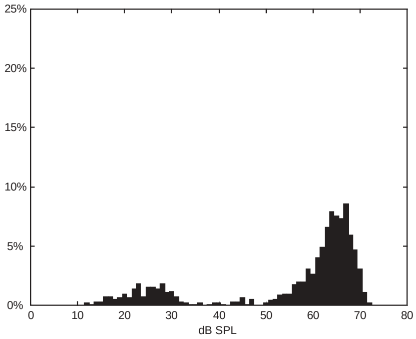

Fig. 5: Distribution of 125 ms RMS levels of female speech.

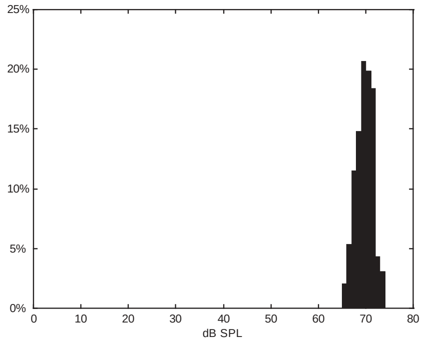

Fig. 6: Distribution of 125 ms RMS levels of background noise recorded inside an automobile.

Filter bandwidth and number of channels

Although the original Senso Noise Reduction system represented a technological breakthrough, it was evident that there was still a potential for improvements. A noise reduction system utilizes the spectral and temporal differences between speech and noise. Thus, if the potential of the algorithm should be increased, the spectral and temporal resolution of the statistical analysis, which is the basis for the Senso Noise Reduction system, should be enhanced. In this way it became clear that it would be necessary to design filters with a much narrower bandwidth than was used in the first Senso. Accordingly, the bandwidths of Senso Diva filters are only one third of an octave throughout the main speech frequency range. Filters with this bandwidth can be designed very effectively and the bandwidth is close to the critical bandwidth, cf. fig. 2. As a consequence of these narrow bandwidths, it has been necessary to increase the number of channels. Unfortunately, filters with a narrow bandwidth in general also impose a longer delay than a broader filter. Although it is generally assumed that small delays are harmless, recent research (Moore, 1999) suggests that delays as short as 5-10 ms may have a negative impact on sound quality for listeners with near normal hearing at low frequencies. In order to minimize the delay in the hearing aid, minimum delay filters and a high sampling frequency (32 kHz) have been used (Andersen et al., 1999). In this way, the group delay has been reduced to approximately 2 ms in the speech frequency range.

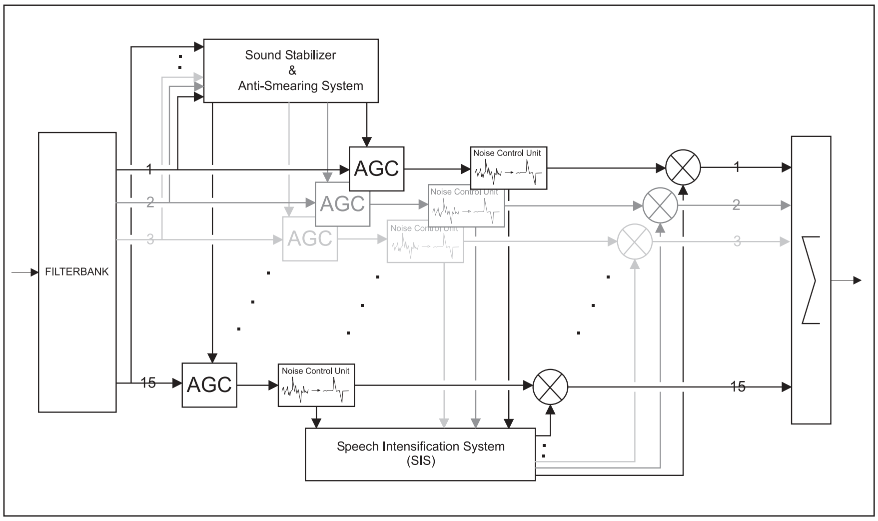

Fig. 7: Block diagram of Senso Diva noise reduction and speech intensification.

Anti-smearing
The incoming signal is analyzed in a digital filter bank consisting of 15 minimum delay filters. In each channel there is a complete analysis system essentially working according to the same principle as in the classic Senso. However, if such a large number of compressors were working independently of each other, the peaks and valleys of the speech spectrum would be smeared out and, as a consequence, the sound quality would deteriorate (Baer & Moore, 1994). In order to preserve the spectral contrasts, a special anti-smearing function has been established in Senso Diva. The anti-smearing block preserves the spectral contrast of the signal. In this way audibility can be optimized using many filters without compromising speech intelligibility due to smearing of spectral contrasts.

Loudness restoration
The narrow filters enable the Noise Reduction system to be very effective and the loudness of background noise will be substantially reduced. However, the efficiency of the Noise Reduction system has the side effect that the overall loudness of a simultaneous speech signal to some degree depends on the level and spectrum of the background noise. In order to counterbalance this tendency, a Speech Intensification system has been introduced. This unit receives information from all 15 channels of Senso Diva and counterbalances the loudness reduction caused by the Noise Reduction system. Thus, the net effect is an apparent reduction of the noise without de-gradation of the loudness of speech.

Directionality
One of the most beneficial techniques for improving intelligibility of speech in noisy environments is the use of directional microphones. Several types of directional microphones are available, but in hearing aids the possibilities are in practice limited to two types. These are 1) dedicated directional microphones which consist of single microphone cartridges with two sound inlets and 2) multiple microphones in which directivity is obtained by combining the signals from two or more omnidirectional microphones after having delayed one of the signals. Each microphone type has advantages over the other, but the most prominent difference is that the multi-microphone type can be adaptively controlled and its characteristic can be varied from omnidirectional over cardioid to supercardioid hypercardioid and bi-directional. In Senso Diva we have used a dual microphone and digital signal processing has been extensively utilized in order to eliminate some of the potential drawbacks known from earlier multimicrophone systems.

Microphone match
The two microphones in a dual microphone must be identical in order to give the best directivity and even when the microphones are carefully matched initially, drifting may corrupt the match after a period of use. In Senso Diva a special patent-pending algorithm will correct any mismatch which appears initially or after a period of use. This ensures that the hearing aid retains its performance even after years of use. A very important additional effect of digital microphone matching is that a perfect microphone match is obtained also at low frequencies. This makes it possible to obtain a high degree of directivity also at low frequencies, thereby overcoming a well-known weakness of multi-microphone systems. It should be noted that high directivity at low frequencies is important for listening in background noise since most environmental noise has a low frequency emphasis.

Wind noise
During outdoor activities many hearing aid users have noticed that even moderate wind may generate a noise in their hearing aids and – due to their two microphone inlets – this is especially annoying in hearing aids with directional microphones. In Senso Diva the processor will immediately identify the noise as wind noise and quickly adapt to omnidirectional mode whereby the level of the wind noise reproduced in the user's ear is suppressed by 10-30 dB depending on the frequency.

In-situ directivity
The directivity pattern of a directional hearing instrument when worn by a user is substantially different from its free field directivity (Killion, 1997). This is especially true at high frequencies. As a consequence, high directivity in a free field will not guarantee high directivity when the hearing aid is worn by the user. It is therefore advantageous to optimize the microphone characteristic when the user wears the hearing aid or – more practically – when the hearing aid is mounted on the head of an acoustical mannequin. The amount of directivity at a certain frequency is described by the directivity index, DI[1]. Moreover, frequency weighting has been introduced to account for the frequency dependence of the importance of speech (Peterson, 1989). Thus articulation index-weighted DI, AIDI, has been used to characterize the practical benefit of directional hearing aid microphones (Killion, 1997). This approach has the advantage that it results in a single number characterizing a directional microphone and thereby it allows a comparison between different microphones. AIDI puts little emphasis on directivity at low frequencies due to their minor importance for speech intelligibility. Thus, increasing the directivity at low frequencies has little influence on AIDI although low frequencies dominate in most environmental noise types. In the development of Senso Diva, concern has been devoted to obtaining high directivity in Senso Diva when worn by the user, and special efforts have been directed towards the low frequency range although we are aware that this may have little impact on the resulting AIDI.

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[1] DI at a specific frequency describes the signal-to-noise ratio obtained when the signal comes directly from the front while noise at the same overall level is coming with equal probability from all directions.

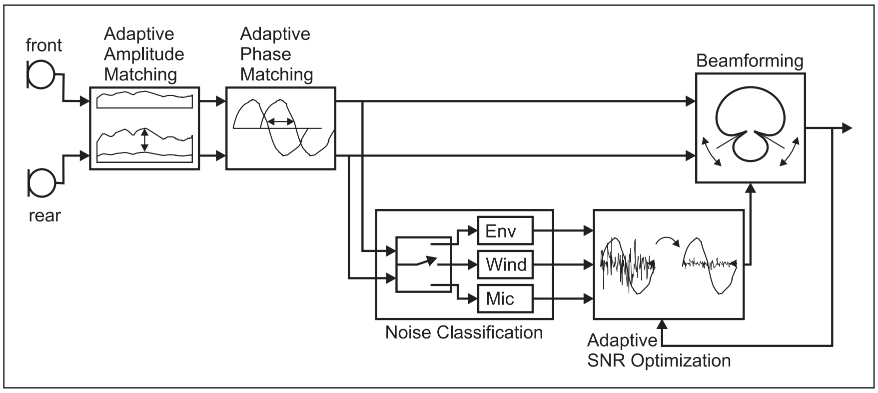

Fig. 8: Block diagram of Senso Diva's directional system.

Adaptive Beamforming
In adverse listening situations the noise will typically reach the hearing aid from many directions. This is the case in a "cocktail party" situation where a large number of "noise sources" are present or because reverberation will make sound approach the microphones from several directions. In these situations the most effective way of reducing the noise level is not to establish a sensitivity minimum in the direction of the noise source. Rather one should shape the directivity pattern in such a way that the overall noise from the rear is kept at a minimum and this is in fact the principle of Senso Diva's Adaptive Beamforming. In case of a single noise source, adaptive beamforming will of course result in a situation in which sensitivity minima are automatically placed in the direction of the noise source. In this way a single stationary or moving noise source will be virtually cancelled.

In this way the directional properties of Senso Diva are optimized for a variety of real life situations. The characteristic best suited for the listening situation is automatically selected. When an omnidirectional characteristic is considered optimum this characteristic is achieved by adding the signals from both microphones in phase. This strategy has the effect that the effective inherent noise of the microphone will be reduced by 3 dB compared with the use of a single omnidirectional microphone. This will hardly increase speech intelligibility in any way but users who at some frequencies possess a near normal threshold may appreciate the more noise free sound reproduction in Senso Diva.

Circuit noise
Some hearing aid users have almost normal hearing at some frequencies. For very soft input they may be able to detect the thermal noise from the microphone. In order to make this inaudible for the particular user, a microphone noise squelch is implemented in each channel as an expansive segment of the I/O characteristic, viz. segment 1 in figures 3 and 4. This is set according to the individual threshold in each channel in order to prevent circuit noise from being audible while still obtaining a high gain for soft sounds.

Preventing acoustic feedback

Instability caused by acoustic feedback is an annoyance for many hearing aid users – and their surroundings. Acoustic feedback occurs when sound created by the hearing aid receiver leaks back to the microphone. In some cases – when a high gain is needed or when leakage is severe – the acoustic feedback leads to instability, thus making the hearing aid "whistle". One way of preventing feedback is to reduce the leakage, e.g. by narrowing or plugging the vent. This, of course, may not always be sufficient and is certainly not always without side effects. Therefore, alternative or supplementary methods are needed.

Feedback Manager
In a nonlinear hearing aid it is possible to reduce feedback without a general lowering of the gain. This is because feedback only creates instability when the gain is high. Therefore, gain limitation can prevent instability. The effect of the limitation is restricted to low input levels, as the gain would otherwise exceed the limit of stability. This gain limit corresponds to segment 2 in figures 3 and 4. In order to measure these gain limits for the individual ear, a measurement of the acoustic leakage of the earmold (a "feedback test") should be carried out. In this test the leakage in each frequency channel is estimated. In Senso Diva this measurement is automatic and all channels are tested in one process. The total duration of the test amounts to 5-10 seconds.

Active feedback cancellation
The term "active feedback cancellation" covers a general principle, which is illustrated in figure 9. The processor in the hearing aid adaptively imitates the signal feeding back from the receiver. If the processor can imitate this signal correctly, it can be subtracted from the incoming signal whereby the feedback signal is eliminated. Evidently, it is not a simple task to imitate the feedback signal, bearing in mind that the acoustic feedback path is not constant. For example, when the user is chewing or yawning, the ear canal may change its shape and the leakage will change accordingly. Or, if a telephone is held close to the hearing aid, the feedback path will also change. Therefore, it is important that the Feedback Path Simulator is able to follow changes of the feedback path. Since the ear canal may change its shape within a fraction of a second, the algorithm has to adapt fast enough to follow these variations. On the other hand, in stable situations fast adaptation may give rise to a slight degradation of the sound quality. Widex has developed a patent-pending algorithm including fast and slow adaptation. A block diagram is shown in figure 9. During the fitting, a feedback test should be carried out in which the calibration signals are led to the receiver. The

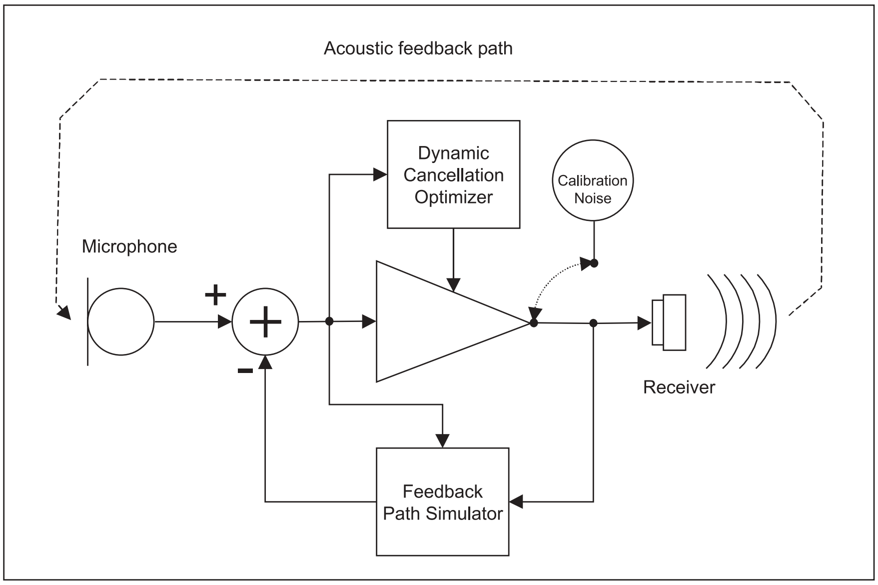

Fig. 9: Block diagram of Senso Diva's Feedback Cancelling system.

sound from the receiver leaking to the microphone through the acoustic feedback path is used to calculate the feedback margins of the earmold and to initialize the Feedback Path Simulator. During use the calibration noise generator is disconnected and the continuous adaptation of the Feedback Path Simulator is controlled by monitoring the signal entering the amplifier. In case of a rapidly changing feedback path the Dynamic Cancellation Optimizer, DCO, is activated. The DCO effectively prevents instability until a stationary situation has been reached and the adaptation to the new situation has been completed. This process will normally have a duration of a few seconds.

Occlusion

When a hearing aid is inserted into the ear canal, the user may perceive his or her own voice as hollow and at a higher-than-normal level. Chewing sounds and other tissue-conducted sounds may also be perceived unnaturally loud. This effect is called the occlusion effect and is probably due to the building up of tissue-conducted sounds in the occluded ear canal. Occlusion is especially bothering for hearing aid users with a mild or moderate hearing loss. In case of severe losses, the level of the tissue-conducted sound is below the hearing threshold and therefore hardly of any inconvenience. The occlusion effect can often be reduced by venting the earmold or by prolonging the earmold or shell into the bony part of the ear canal.

Occlusion Manager

Manipulating the compression characteristic in the low frequency channels may reduce the feeling of hollowness substantially. Normally, little compression is used in case of mild and moderate losses since little recruitment is present. Two observations, however, explain that more compression might have the potential of reducing the effects of occlusion. First, when listening to speech in quiet surroundings relatively more bass amplification is preferred compared with noisy situations (Keidser, 1996). Second, with regard to the naturalness of the user's own voice, relatively less bass amplification is preferred (Kuk, 1990; Sweetow & Valla, 1997). By introducing more compression than required for recruitment compensation both requirements can be met. This is in fact the rationale behind the recommended setting of Senso Diva. In order to facilitate this manipulation of the compression characteristic an occlusion manager has been provided and a systematic method to minimize the negative effects of occluding the ear canal has been developed.

Fitting and fine tuning

The functions described above provide a large potential for differentiated setting of the hearing aid according to individual audiological characteristics and to the user's personal preferences. Thus, the need for versatile programming tools is obvious. At the same time, the need for controlling the large number of parameters should not result in a complicated and painstaking programming procedure. Therefore, special efforts have been devoted to optimizing the programming and fine tuning procedures of Senso Diva.

Sensogram

In-situ thresholds have proven an efficient way to include the acoustic properties of individual ears in the setting of the hearing aid (Ludvigsen & Topholm, 1997). In Senso Diva this method has been further developed. The Sensogram now consists of four thresholds measured in-situ with a pulsed stimulus. Each stimulus consists of frequency modulated pulses with the maximum frequency deviation automatically set to the critical bandwidth according to Glasberg & Moore, 1990. Based on the experience obtained with earlier Senso types using two or three channels, it seems certain that the four Sensogram thresholds provide adequate information for programming Senso Diva for the vast majority of individuals.

Expanded fitting

In some cases of precipitous sloping losses or cookie bite shaped audiograms, further information about the thresholds may be useful. Therefore, it is possible to measure in-situ thresholds at intermediate frequencies in the range from 250 to 8000 Hz. The programming of intermediate channels is facilitated by an automatic pre-setting estimated from the in-situ threshold measured in the four standard channels 500, 1k, 2k and 4 kHz.

Fine tuning

When a Sensogram threshold has been recorded in the four basic channels centered at 500, 1k, 2k and 4 kHz, an initial setting of the multistage compressors in all 15 channels is automatically calculated and loaded into the hearing aid. Since most users require some time to adapt to a new sound image, it might be advantageous to postpone a possible fine tuning session until a certain acclimatization has occurred. If the hearing aid user has experienced a situation in which he or she is not satisfied with the setting of the hearing aid, a fine tuning procedure should be started. Several fine tuning tools have been provided. First and foremost, the fine tuning procedure has been totally separated from the initial Sensogram-based fitting procedure. This means, e.g., that a recorded threshold will remain unaltered during the fine tuning procedure. In this way audiological data are preserved. Instead, a special fine tuning mode has been introduced in which gain parameters can be adjusted at three input levels (soft, normal and loud) relative to the initial setting. The three levels correspond to the arrows in figures 3 and 4. Details about the fine tuning procedure are provided in the programming manual.

Documentation and verification

Often there is a need for verifying the fitting procedure and documenting the result to third parties. Such documentation can, e.g., be required by insurance companies or by local authorities. With Senso Diva the traditional verification procedure in which measured insertion gain is compared to a fitting target, is not appropriate. This is because the fitting procedure of Senso Diva already includes in-situ measurements and because the target responses, as shown in the Compass windows, deviate from generic targets by taking into account the special features included in Senso Diva. These are, for example, filter bandwidth, regulation speed of the compressors, and the effects of the noise reduction circuit, see Kuk & Ludvigsen (1999a). If this is ignored one could easily compromise Senso Diva's potential performance. It is, however, possible to document the fitting and to control that the performance is in accordance with expectations. For further details see Kuk & Ludvigsen (1999b).

Conclusion

In Senso Diva the potential of digital technology has been utilized to its utmost in order to make efficient and versatile solutions. A major goal has been to improve performance in noisy situations and this has been achieved primarily by introducing a new multichannel noise reduction system and a new adaptive directional microphone system which has better performance than previous systems in a number of situations. Furthermore, solutions to many other classical problems have been found. These include preservation of listening comfort in adverse listening conditions and provision of fine reproduction of music and nature sounds. Establishing audibility for weak sounds without trading off comfort has also been an objective. One important aim has been to obtain a pleasant distortion-free sound image without introducing audible artefacts from the signal processing. Another goal has been to prevent howling due to acoustic feedback and to reduce occlusion effects. The solutions require that a substantial number of parameters are adjustable. In order to make these adjustments practical, new fitting tools have been developed. In this connection a new portable programming system has been introduced and a new structure of the Compass fitting program has been designed.

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