New Solutions for Age-Old Hearing Aid Problems

More than five years ago, the Widex Senso digital instrument was introduced with the goal of improving the quality of life of people with hearing impairment. The lessons learned from the launch of this hearing instrument greatly influenced the design of the company's latest device, the Senso Diva. This new instrument has been developed to incorporate many of the existing features of Senso, along with the latest chip technology, current audiological rationales and electoacoustic principles. The new hearing system addresses the issues of listening comfort, distortion-free reproduction of music and natural sounds, audibility of soft sounds and listening in noisy environments. Traditional problems with hearing instrument acoustic feedback and the perception of occlusion are also addressed by the system.

While these new features increase the complexity of the instrument substantially, the fitting and fine tuning procedures have been simplified. A new portable programmer (SP3) and a new structure for the Compass fitting software have been adopted. This article provides the audiological background of the Senso Diva and a brief introduction to its various features.

Reproduction of Speech and Environmental Sounds

Senso Diva is a 15-channel compression hearing instrument designed to compensate for the frequency dependence of recruitment. The bandwidth of each channel approximates the width of the critical bands. Enhanced dynamic range compression (EDRC) with multistage compressors is employed in all 15 channels (Fig. 1).

At medium input levels, the instrument utilizes the loudness equalization principle and amplifies "normal" speech to the wearer's most comfortable listening level (e.g., NAL approach[1]). This strategy defines gain for typical speech levels. A loudness mapping strategy is used for soft and loud sounds. At high input levels, the objective is to ensure that the output of the hearing aid is slightly below the uncomfortable listening level of the wearer. The amplification rationale at low input levels is to amplify low-level input sounds to slightly above the threshold of the listener.

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An example of a basic I/O-curve and the corresponding input/gain curve for a moderate sensorineural loss is shown in Figs. 1 and 2. Each of the five segments has its function for accomplishing the fitting rationales described above and in the sidebar on p. 34. The five segments are also designed to provide an effective and versatile tool for fine-tuning. The position and slope of the segments are calculated in accordance with the fitting rationales. In this calculation we have included the influence of filter bandwidth, detector type, regulation speed and other hearing aid specific factors.[2]

Comfort at high input levels: Many hearing aid wearers experience discomfort when wearing their hearing aids in high-intensity noise environments.[3] In linear hearing instruments an output-controlled limiter (AGC-O) is typically used. This may relieve the problem by keeping the output of the hearing aid within comfortable levels. In nonlinear hearing instruments, various combinations of compression and limiting are used.

High Level Compression: A five-segment compressor in all channels is used by the Senso Diva (Figs. 1 and 2). Sound reproduction at high input levels is determined by the upper part of the curve (i.e., Segments 4 and 5). The intersection between Segments 3 and 4 acts like a hinge, allowing the compression ratio of Segment 4 to be adjusted independently from the lower three segments. This system is referred to as *high level compression* (HLC). In cases with mixed or conductive losses, the magnitude of the air-bone gap is considered when high-level compression ratios are calculated. Special possibilities exist for compensating for hyperacusis or hypersensitivity, since adjustment extends to zero or negative gain.

This feature was developed to allow use of the total residual dynamic range of the wearer. The fitting software automatically calculates the compression ratios using the in-situ threshold recorded during the Sensogram. The hinge or intersection between the two compressor segments (Segments 3 and 4) represents the peak levels of normal speech. Thus, the high level compressors operate at levels exceeding those of normal speech. Segment 5 of the I/O curve is determined by the *automatic output control* (AOC) system. In each channel, a fast-acting limiter-type AGC prevents overloading and distortion.

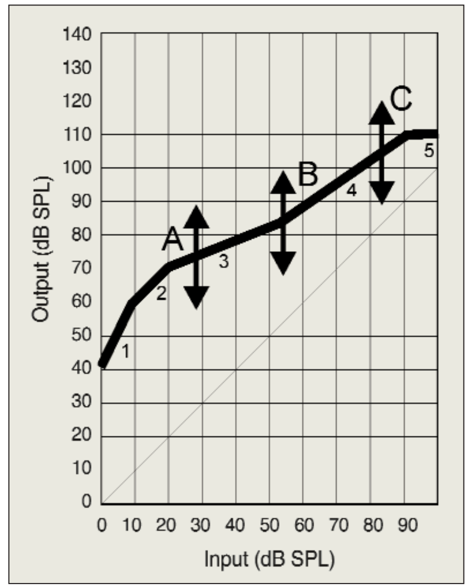

Fig. 1. *Example of a multi-segmented I/O curve for a moderate sensorineural hearing loss.*

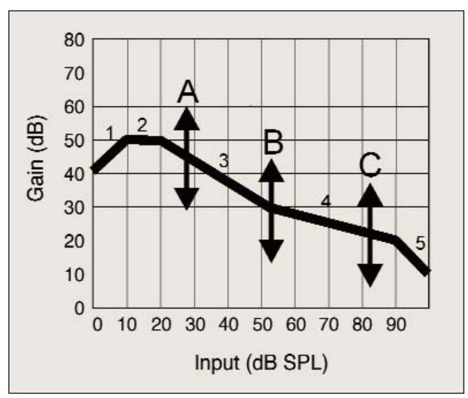

Fig. 2. *Gain vs. input (I/G) curve corresponding to I/O curve in Fig. 1.*

Audibility at low input levels: Ideally, a hearing instrument should restore the wearer's hearing threshold to a normal level. However, hearing instrument wearers using fast-acting compression hearing aids prefer less gain for low-level input than that required for establishing a normal threshold.[4] Many adult hearing instrument wearers may consider this extra audibility bothersome. This is presumably caused by the audibility of weak environmental noise when a low aided threshold is achieved. Methods to minimize the noisy perception while maximizing audibility are used in the new hearing device.

Low compression thresholds: Research conducted by Widex shows that a slow-acting compression system causes less noisiness, even though its compression threshold is below 20 dB SPL. This finding is utilized in the new hearing instrument; all 15 compression channels have compression thresholds (CT) below 20 dB HL. This is intended to produce a lower aided threshold than hearing instruments with higher CTs, enabling the wearer to detect weak sounds—like birds singing at a distance or footsteps.[5] If the wearer is dissatisfied with the extra audibility, the dispensing professional can fine tune individual gain preference for the soft sounds.

Slow-acting compression with the Sound Stabilizer™: In the new hearing system, instantaneous gain in each channel varies according to the properties of the input signal. Rather than using a fast regulation time—which has the negative effect of making background noise louder—the instrument uses long time constants whenever possible to maintain the spectral and temporal contrasts while enhancing sound quality.[8,9]

In sound environments where the intensity of sounds changes abruptly, slow-acting regulation may not be favorable because it may adapt too slowly to the rapid changes. In order to counterbalance this limitation of slow-acting regula-

Consequences of a Sensorineural Hearing Loss

A sensorineural hearing loss is associated with a number of functional deficits, which, if left untreated, have a negative impact on the communication abilities and quality of life of the people it affects.

Loudness recruitment: Loudness recruitment is closely associated with hearing loss of cochlear origin, such as presbycusis or noise trauma. It is 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 manifest itself as a loss of compression. Fig. 3 shows an input-output (I/O) curve of the basilar membrane in its normal condition and when the outer hair cells have been made temporarily inactive by injection of furosemide.[6] The figure shows how compression is lost when the outer hair cells are inactive.

A tempting approach to compensating for the loss of compression is to build a nonlinear circuit into a hearing instrument to restore the missing compressive action of the cochlea. Indeed, this is the rationale for recruitment compensation in several commercial hearing aids. Many of these instruments also apply processing algorithms modeled on the behavior of the cochlea. Such approaches may be characterized as "loudness normalization," since their objective is to restore "normal" loudness perception.

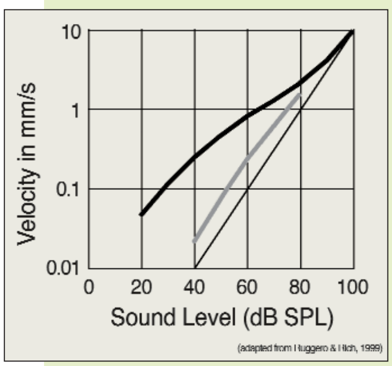

Fig. 3. *Velocity of the basilar membrane in a normal cochlea that is temporarily damaged (gray) by injection of an ototoxic agent (furosemide). The thin line indicates a linear response.*

An important consideration in designing hearing aids for loudness normalization is to realize that recruitment is frequency-dependent. This calls for a hearing aid with many independent processing channels so that each channel can be responsible for a specific frequency region. The optimum number of channels is not known, and the exact number would depend on the design rationale and hardware constraints. One suggestion is to employ enough channels to allow each to approximate the "critical bandwidth" concept discussed in auditory psychophysics. Critical bandwidths as a function of frequency are shown in Fig. 4.

One requirement in designing a processor that mimics the cochlea's behavior is that short attack times and short release times must be used. This is because outer hair cells respond quickly to external stimuli. On the other hand, two observations suggest against the use of a short time constant in compression hearing aids. Byrne[7] noted that loudness normalization may not be optimal to speech intelligibility. Indeed, he cautioned against the uni-dimensional view of auditory processing that focused solely on the spectral-intensity domain while ignoring other aspects of processing (e.g., temporal aspects). Instead, Byrne argued in favor of the loudness equalization strategy introduced by Byrne & Dillon[1] (i.e., NAL fitting rationale), which attempts to ensure that normal speech in all frequency regions is amplified to the same loudness level. A second observation contradicting the use of fast regulation time is that hearing-impaired persons found longer release times to be more pleasant and less noisy than shorter release times.[8] Indirectly, these two observations cast doubt on the exclusive use of loudness normalization .

Reduced frequency selectivity: Many people with a sensorineural hearing loss have difficulty separating sounds of different frequencies when they are presented simultaneously. This loss of frequency selectivity also manifests itself as excessive upward spread of masking. The end result is increased difficulty with speech understanding in noise, because frequency differences become more difficult to detect in environments with competing signals.

Despite various attempts to overcome the limitations of reduced frequency selectivity, it appears that no effective compensatory approach is available. One approach is to exaggerate spectral contrasts, but the results have been mixed. It seems that, if one can preserve the spectral and temporal contrasts of the input signal, one may already have provided the hearing-impaired person with the best condition for speech understanding. In that regard, a long attack time and release time (i.e., slow acting compression) with minimal amplification at high input levels may offer a good solution to the problem of preserving input contrasts.

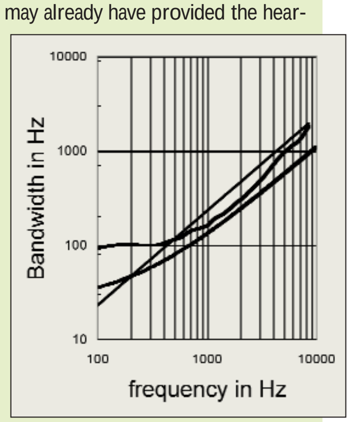

Fig. 4. *Critical bandwidths according to Zwicker[18] (black) compared to equivalent rectangular bandwidths according to Glasberg & Moore[19] (gray) and 1/3 octaves.*

Temporal resolution: Temporal resolution refers to the ability to distinguish consecutive pulses as separate events. It is level-dependent, and resolution gets poorer as the presentation level approaches the hearing threshold. People with a sensorineural hearing loss may experience poorer temporal resolution for two reasons. First, the elevated thresholds mean that speech sounds presented at a typical level may be only slightly above threshold. At these sensation levels, even normal-hearing people have poorer temporal acuity. In addition, some hearing-impaired listeners experience a further reduction of temporal resolution beyond what is expected solely from the low presentation level. The net effect is that speech features which are identifiable by temporal cues are difficult to recognize. Thus, preserving the temporal contrasts of the speech signal is an important consideration in hearing aid design.

tion, a Sound Stabilizer™ is implemented in all channels. This feature is designed to optimize the speed of regulation in a compressor by using long regulation times in stationary listening situations and adaptively vary its speed depending on the duration and intensity of the input signals. The functionality of the Sound Stabilizer™ is described in detail elsewhere.[10]

Improving Performance in Noisy Environments

For decades, 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 more persons.

Widex introduced one of the first noise reduction/speech enhancement systems in digital hearing aids with the launch of the Senso.[11] This system relies on a statistical analysis of the incoming signal by continuously monitoring the distribution of short-term intensity levels of the signal within each of its channels. Since speech is a sequence of sounds separated by brief pauses, its level distribution is bimodal, with levels falling in two separate clusters. Weak speech sounds and pauses are clustered at low levels and intense speech sounds (like vowels) are registered at high levels. In environmental noise, the level distribution is much narrower with only one mode since no silent or weak segments are present. These level distributions in each channel are used to distinguish speech from noise.

Filter bandwidth and number of channels: Although the original Senso noise reduction system may represent a technological breakthrough, its efficacy can be improved by increasing the resolution of its statistical analysis. Filters with much narrower bandwidth than those used in the original Senso could achieve this purpose. Accordingly, the filter bandwidth of Senso Diva is only one-third of an octave wide throughout the main speech frequency range. This bandwidth is close to the critical bandwidth of the auditory system. This results in 15 independent channels.

On the other hand, filters with a narrow bandwidth can result in a longer time delay than those with a broader bandwidth. Although it is generally assumed that small delays are inconsequential, recent research[12] suggests that delays as short as 5-10 ms may be rated as disturbing, especially by hearing-impaired persons with near-normal hearing in the low frequencies. Thus, minimum delay filters and a high sampling frequency (32 kHz) are used in the new instrument that effectively minimize the group delay[13] to approximately 2 ms in the speech frequency range.

Anti-smearing: A drawback of multichannel nonlinear processing is the possibility of temporal and spectral smearing, in which the intensity differences between the peaks and valleys of the input signal are diminished. This may have a negative effect on speech understanding in noise. Sound quality also may be diminished.[14] An anti-smearing circuit is designed in the new hearing system to minimize such occurrences. The function is designed to control the interaction of the various channels so that intensity differences among channels are preserved. In this way, audibility is optimized by using many filters, and speech intelligibility is not compromised from the smearing of spectral contrasts.

Speech Intensification System: The narrow filters enable the noise reduction system to be precise. As a consequence, the loudness of background noise should be substantially reduced for many wearers. However, depending on the level and spectrum of the background noise, the level of the speech signal that is present at the same time may also be reduced.

In order to minimize this effect on desirable speech signals, an enhanced *speech intensification system* (SIS) has been introduced. This system receives information about signal-to-noise ratios from all 15 channels of the hearing instrument and redistributes gain according to the speech importance and loudness function of each frequency. The net effect is an apparent reduction of noise with minimal degradation on 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. All Senso Diva BTEs, ITEs, and ITCs are equipped with a dual-mic directional system. The Locator™ is designed to be an advanced dual-microphone system that adaptively changes its directional characteristics from omni-directional to cardioid, supercardioid, hypercardioid and bi-directional patterns using DSP techniques (Fig. 5).

Microphone matching: The microphones in a two-microphone system must be identical in order to yield the desired directivity (Fig. 5). Any mismatch would result in a loss of directivity, especially in the low frequencies[15] where most of the background noise resides. Even when the microphones are carefully matched initially, drifting may occur after a period of use and the original directivity will diminish. The new hearing instrument uses a patent-pending algorithm, called OptiMic™, that continuously evaluates and corrects any mismatch in sensitivity and phase between the two omnidirectional microphones. This makes it possible to obtain the stated directivity at all frequencies at all times, even after years of use. Additionally, OptiMic™ permits a smaller distance between microphone inlets without the risk of sacrificing directivity. This allows the use of directional microphones in ITC-style fittings.

Adaptive beamforming and microphone noise reduction benefits: In real-life listening situations, noise typically reaches the hearing instrument from many directions in an unpredictable manner. Thus, it is more useful to shape the directivity pattern in such a way that the overall noise from the rear and the sides are kept at a minimum. This is the principle behind the instrument's Adaptive Beamforming. By adapting its polar pattern to the directions of the noise sources, the beamforming function reduces the noise level to close to the theoretical maximum of 6 dB in a diffuse field. In the case of a single noise source, the system's implementation results in sensitivity minima placed automatically at the direction of the noise source for its complete cancellation.

In real-life situations, the polar characteristic best suited for the listening situation is automatically selected by the DSP instrument—ranging from an omnidirectional pattern to various directional patterns without switching. When an omnidirectional characteristic is considered optimum, however, signals from both microphones are added in phase. This strategy has the net effect of reducing the effective noise of the microphone by 3 dB compared to the use of a single omnidirectional microphone. Wearers who have near-normal threshold at some frequencies may appreciate the more noise-free sound reproduction.

Wind noise cancellation: The new processor is designed to identify wind noise and quickly adjust the microphone characteristic to a wind-noise cancellation mode. The level of the wind noise can be suppressed by 25-35 dB over the directional mode.

Fitting and Fine Tuning

The functions described above provide the flexibility to adapt the instrument to an individual's audiological characteristics and personal preferences. Thus, the need for versatile programming tools is obvious. At the same time, the large number of parameters need to be controlled so that they do not complicate the programming procedure.

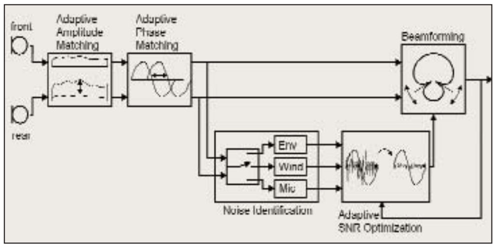

Fig. 5. *Block diagram of Locator directional system.*

Sensogram and new options: The use of in-situ thresholds has proven to be an efficient way to include the acoustic properties of individual ears in setting hearing aids.[16] This method has been further developed in the new instrument. The Sensogram now consists of thresholds at four frequency regions (500, 1000, 2000 and 4000 Hz) measured in-situ with a pulsed stimulus. In-situ thresholds between and beyond these frequencies are extrapolated based on the measured Sensogram values. In some cases of precipitously sloping losses or cookie-bite audiograms, the new instrument allows measurement of in-situ thresholds at intermediate frequencies between 250 Hz and 8000 Hz in the expanded fitting mode. However, Widex research shows that four Sensogram thresholds provide adequate information for programming the instrument for the vast majority of individuals.

Fine tuning: The Sensogram thresholds at the four standard channels (and the Sensogram obtained in the expanded fitting option) automatically determine the input-output characteristics in all 15 channels. In Senso Diva, the fine tuning procedure is now separated from the initial Sensogram-based fitting procedure. This means that the recorded Sensogram thresholds remain unaltered during fine tuning. 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 Figs. 1 and 2. Details about the fine tuning procedure are provided in the programming manual.

Documentation and verification: Often there is a need for verifying the outcome of the fitting procedure and documenting the results to third parties (e.g., insurance companies or health agencies). Currently, the traditional verification procedure in which the measured insertion gain is compared to a prescriptive gain target is not appropriate for the Diva. This is because the generic prescriptive fitting formulae have not considered the device-specific modifications. These factors include, for example, filter bandwidth, regulation speed of the compressors and the effects of the noise-reduction circuit.[5] If this is ignored, one could compromise the instrument's potential performance. However, various means to document the results of the fitting are possible in order to ensure stable performance over time. For further details on verification of these instruments, see Kuk & Ludvigsen.[17]

The new Senso Diva has been

Battling Internal Noise, Feedback and Occlusion

Circuit noise: Some hearing aid wearers have normal hearing at some frequencies, allowing them to detect thermal noise from the microphone. In order to make this noise inaudible, a microphone noise squelch is implemented in each channel as an expansive segment of the I/O characteristic (e.g., Segment 1 in Figs. 1 and 2). This is set according to the individual's in-situ threshold in each channel in order to prevent circuit noise from being audible while obtaining high gain for soft sounds.

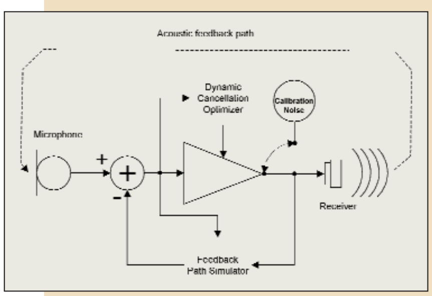

Fig. 6. *Diagram of Feedback Cancelling System.*

Active feedback cancellation: The Diva instrument utilizes an active feedback cancellation algorithm to control acoustic feedback. The DSP processor in the hearing instrument uses an adaptive process to continuously estimate the signal feeding back from the receiver. The estimated feedback signal can be subtracted from the incoming signal to eliminate feedback (Fig. 6). However, it is not a simple task to estimate the feedback signal because its characteristics may not be constant. For example, when the wearer is chewing or yawning, the ear canal may change its shape and the feedback path may change accordingly. Or, if a telephone is held close to the hearing aid, the feedback path may also change from the "no-telephone" condition. Therefore, it is important that the feedback path simulator is able to follow changes of the feedback path in all conditions.

The new hearing instrument uses a patent-pending feedback cancellation algorithm that has a slow-acting and a fast-acting component. The slow acting component, the feedback path simulator (FPS), constantly evaluates the acoustic conditions around the wearer's hearing aids in order to cancel the feedback signal. This is achieved by comparing the signal to the receiver with the signal entering the amplifier. This algorithm increases the amount of usable gain before feedback by as much as 12 dB in some frequencies. The fast-acting component, the dynamic cancellation optimizer (DCO), is activated immediately when a rapidly changing feedback path is detected. The DCO effectively prevents feedback by changing the compression characteristics in the involved channel until the FPS adapts to the new situation.

Occlusion Manager™: When a hearing aid is inserted into the ear canal the wearer may perceive his/her own voice to be "loud" and "hollow." This is especially bothersome to new hearing aid wearers with a mild or moderate hearing loss. In order to minimize the occlusion effect while maintaining audibility and sound quality, the instrument provides an occlusion manager that systematically minimizes the negative effects of occluding the ear canal. The occlusion manager allows discrete adjustment of compression characteristics in the low frequency channels without affecting gain in the nearby higher frequencies.

Enhancing music appreciation: The hearing device also provides a secondary program in which the wearer may switch from the fully adaptive mode to a "music" program in which the electroacoustic characteristics of the aid are optimized for music appreciation, especially in a surround-sound environment. This is achieved by engaging the locator microphone in an omnidirectional mode, optimizing the pre-set frequency response characteristics for music appreciation, and disabling or modifying the adaptive algorithms like noise reduction and feedback cancellation to avoid any potential artifacts. ◗

designed to fully utilize the current potential of digital technology, offering solutions to the problems that have traditionally led to lower wearer satisfaction with hearing instruments. It is hoped that these new digital solutions will encourage more potential hearing aid candidates to try and be satisfied with amplification. ◗

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