Savia™

Digital Bionics

Natural wisdom
captured in technology

The anatomy of Savia – Summary

Savia, Phonak's first class hearing instrument family is consistently focused on natural, effortless hearing in all listening situations.

To achieve this goal, a number of innovations, including novel signal processing strategies, unique fitting tools and breakthrough technological features have been developed. These allow Savia to precisely navigate through all the listening situations of daily life and to continuously adapt to the acoustic environment. With Savia speech intelligibility and effortless hearing even in difficult environments are ensured leading to an exceptional level of user satisfaction.

PHONAK
hearing systems

Introduction

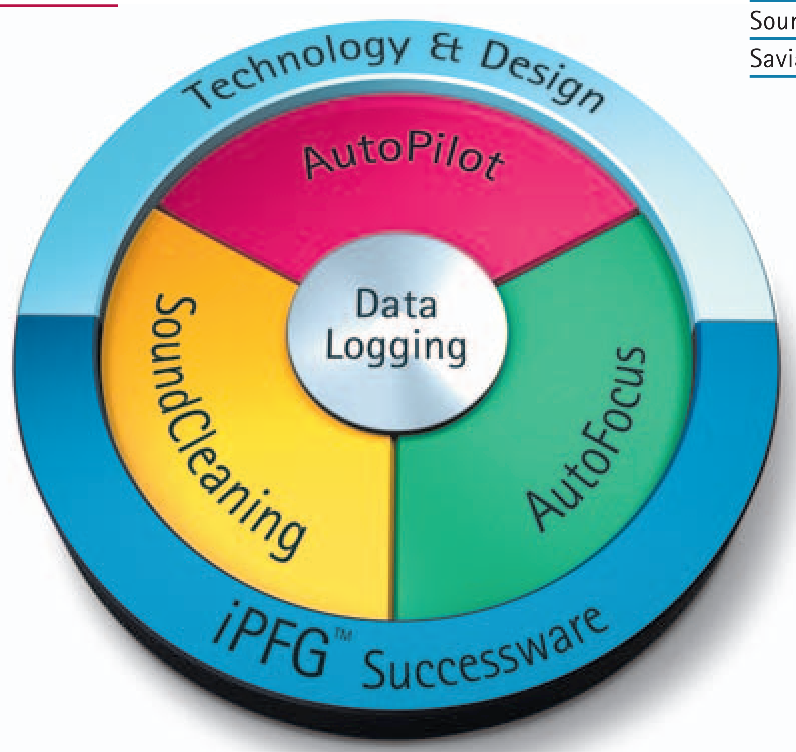
The needs of hearing impaired clients encompass much more than just amplification. Besides effortless understanding, especially in the presence of multiple noise sources, and natural localization abilities, they require feedback-free, unoccluded performance and comfortable communication even in reverberant environments. Accordingly, the hearing instruments have to react in an appropriate way and precisely “navigate” through the acoustic environment. In addition, the fine tuning tools need to be client oriented and easy to manage, so that the optimal settings will result in maximum satisfaction in a fast and efficient manner.

To achieve this goal, a series of world first innovations has been developed and implemented in a hearing instrument. These components compose the anatomy of Savia (see Fig. 1). They include core functionalities, fitting tools and technological features which all directly offer improved performance to match individual needs.

Savia conceptual anatomy

AutoPilot
- SoundNavigation
- EasyPhone
- EasyFM

iPFG Successware
- DataLogging with User Preference Tuning
- iCOSI
- Sound Type Tuning
- Savia Insight



SoundCleaning
- EchoBlock
- Feedback Phase Inverter
- High Resolution Noise Canceler
- Wind Noise Management

AutoFocus
- digital SurroundZoom
- Real Ear Sound

Figure 1: Core functionalities, fitting tools and technology components of Savia hearing systems.

AutoPilot

Client's need:

The hearing expectation should be met in every relevant listening situation. Hearing impaired listeners experience different types of acoustic environments every day, with different frequency of occurrence and importance to the individual. In each of these listening situations, they have different needs and hearing expectations which should be met by the hearing instruments (e.g. maximum and effortless understanding in quiet conversations but comfort in noisy situations).

The Savia solution: AutoPilot with SoundNavigation

The acoustic environment is multi-faceted and individual, but research shows that every-day situations can be grouped into four main, relevant clusters which cover virtually every listening situation: calm situations, speech in noise, noise only, and music (see Fig. 2).

In each of these four basic clusters of listening situations, the hearing instrument has to meet specific hearing expectations. In quiet, for example, the user expects overall sound awareness and effortless hearing. Listening to speech in noisy environments requires effective noise suppression with focus on speech intelligibility. Listening to music, on the other hand, the user expects rich, full, and undistorted sound. To effectively meet these different expectations, Savia offers four base programs. Each base program is made up of the signal processing features and settings which are appropriate for the respective listing expectations.

There is no need for the client to manually switch between programs. SoundNavigation constantly analyses the acoustic situation, classifies it into one of these four categories and activates the appropriate base program (see Fig. 3).

In the feature extraction stage, a range of acoustic parameters are constantly computed from the input signal. They characterize the signal and allow for precise classification. The extracted features describe intensity, as well as spectral and temporal properties of the signal. It is known from Auditory Scene Analysis (Bregman, 1990) that these features are also utilized by humans to segregate distinct "acoustic objects" in the environment. This is the basis for understanding in a noisy environment or identifying a specific instrument in an orchestra. Generally, these features allow for classification of the input signal. This is achieved in a

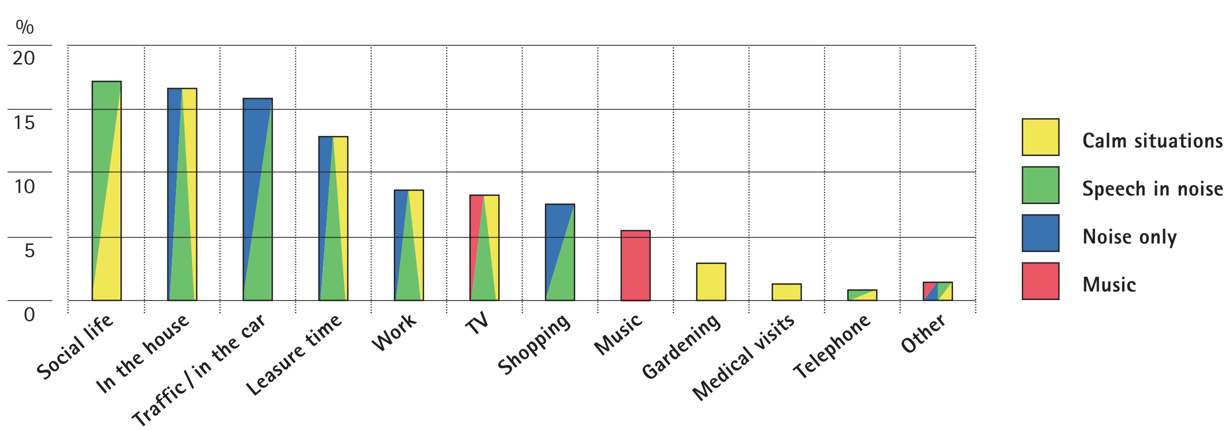


Figure 2: Listening situations which have been described as relevant (Gabriel, 2003). The data were extracted from diaries held by 20 adult hearing impaired subjects over a period of 4 weeks. In total, more than 750 individual situations were described. Most situations are not covered by one of the four categories alone but include e.g. calm situations and speech in noise, which is illustrated qualitatively.

Figure 3: Categorizing the acoustic environment: Choice of base program.

subsequent processing stage which includes statistical analysis of the extracted features. Post-processing includes correction of classification outliers. For categorizing the input signal into one of the four classes, an observation time of less than 10 sec is considered. The smooth transition into the appropriate base program takes up to 5 sec, depending on the user preferences. Figure 4 illustrates how SoundNavigation automatically selects the appropriate base program.

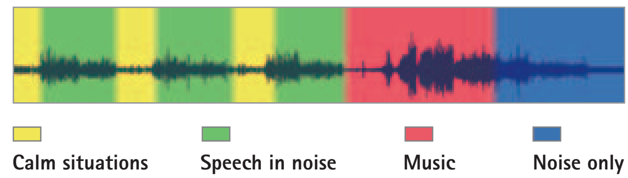


Figure 4: Illustration of SoundNavigation. As soon as the acoustic environment changes, SoundNavigation detects it and chooses the appropriate hearing program.

Client's need:

Hassle-free usage of telephone and FM equipment. The telephone is an essential part of today's communication. FM systems are of great importance for many hearing impaired clients as they provide tremendous benefit, especially in acoustically demanding situations. Using these communication channels should be as easy and effortless as possible.

The Savia solution: EasyPhone and EasyFM

Savia BTE hearing instruments with an attached FM receiver, automatically detect the presence of an FM signal. For example, when a speaker (with FM transmitter) starts a lecture, Savia automatically switches to the "FM+M" mode. There is no need for the user to manually switch to an FM program. When there is no audio input anymore, Savia automatically switches back to the appropriate base program chosen by SoundNavigation. Accordingly, when the client picks up a telephone receiver (with attached magnet) Savia automatically switches to the designated telephone program (T-Coil or acoustic telephone) for optimal speech understanding, which makes using the telephone for hearing impaired listeners as natural and effortless as for those with normal hearing.

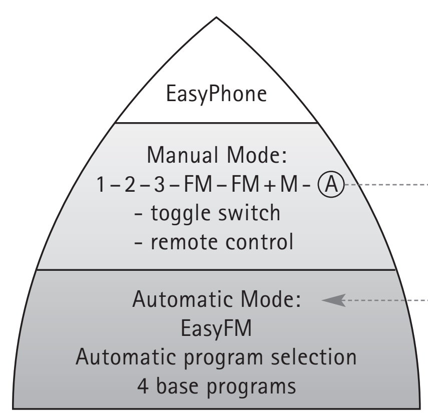
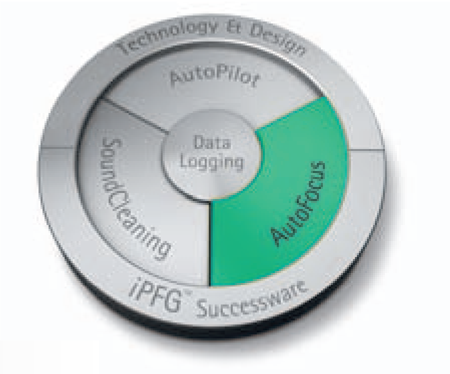


Figure 5: Savia's program switching hierarchy levels.

Figure 5 shows Savia's program switching hierarchy levels. At the top, there is the EasyPhone detector. Whenever the user holds a telephone receiver against the ear, the intention is unambiguous: the user wants to phone. The telephone program is automatically activated, no matter in which mode the hearing instrument was operating in before.

The manual mode is in the second hierarchy level, a manual selection is a clear user choice. The automatic mode autonomously chooses the appropriate base program. When in automatic mode, FM+M will be activated whenever a signal is received from the audio input.

AutoFocus



Client's need:

Ability to precisely locate sounds. Accurate localization of sounds is obviously beneficial in daily life. A precise localization of sounds is the prerequisite for "mapping" the environment and feeling comfortable and secure. Besides that, the intelligibility as well as natural sound quality of a speech signal is enhanced when its spatial position is perceived separately from competing noise sources (Plomp, 1976).

The Savia solution: Real Ear Sound

Our ability to localize sounds is based on acoustic cues such as interaural differences in time and intensity, as well as spectral shape cues provided by the pinna. While interaural time and intensity differences provide information about the horizontal angle of the sound source, pinna cues are mostly important for vertical localization and front/back discrimination (see e.g. Blauert, 1997 for a review). It would be expected that hearing instruments which degrade pinna cues impair localization performance. In fact, aided localization tends to be poorer than unaided localization of audible sounds. This was found for both vertical (Noble and Byrne, 1990) and horizontal localization (Orton and Preves, 1979; Noble and Byrne, 1990). Degradation of localization performance with BTE hearing instruments, compared to the unaided condition, is due to the microphone position. The microphones are located above the ear, outside the pinna and thus do not pick up pinna cues.

Savia with Real Ear Sound is the first hearing instrument to actually simulate spectral shaping effects of the Pinna. Figure 6 shows the directivity index across frequency in three conditions: 1) inside the unaided ear (i.e., the natural directivity pattern as measured in KEMAR), 2) output from a conventional BTE in omnidirectional mode, and 3) with Real Ear Sound. It can be seen that the natural directivity in the high frequencies above 1.5 kHz, which is lost in conventional BTEs, is restored with Real Ear Sound. Thus, Savia meets the demand for precise localization and natural sound impression.

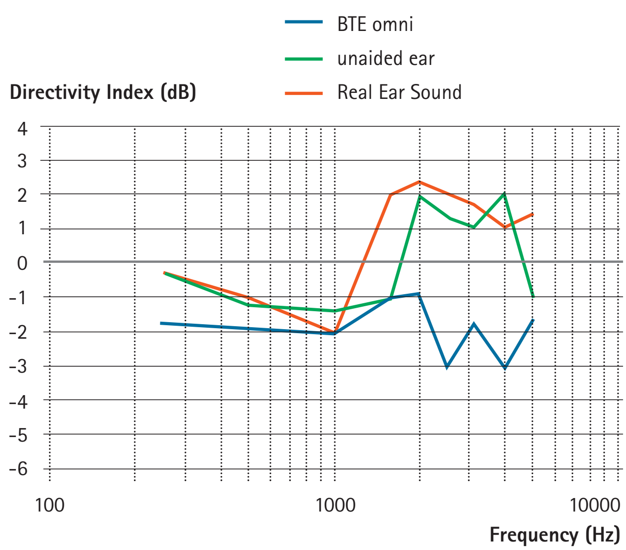


Figure 6: Real Ear Sound restores the natural directivity pattern of the human ear.

Client's need:

Maximum directional benefit. Good speech understanding in noise is the highest priority for the hearing impaired (Kochkin, 1993). Today, use of directional microphones is the most efficient technology in hearing instruments to improve speech understanding in noise (e.g. Killion, 2004). This is achieved by suppressing the noise source and thus increasing the signal-to-noise ratio in adverse situations. Providing directivity in hearing instruments has a strong impact on satisfaction.

The Savia solution: digital SurroundZoom

In recent years, directional technology in hearing instruments has been improved substantially. In 1999, Phonak introduced adaptive directionality which further enhances the directional benefit in many real-life situations (Ricketts and Henry, 2002; Kühnel and Checkley, 2002). With adaptive directionality, the polar pattern is continuously adjusted so that the strongest noise source is suppressed the most, while the speech signal from the front is amplified. While a cardioid pattern is chosen for noise sources from behind, a bi-directional polar pattern is selected to optimally suppress noise from the sides. However, real life includes many situations with a multitude of noise sources which are spatially and spectrally separated.

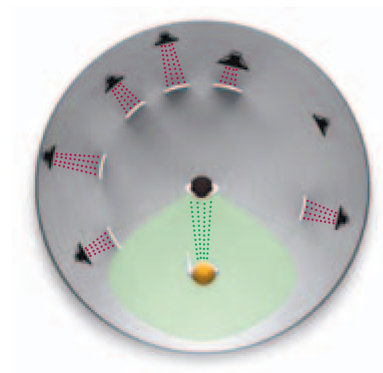


Figure 7: In many real life situations, noise sources are spatially and spectrally separated. Savia digital SurroundZoom allows for selective suppression of these noise sources.

Savia digital SurroundZoom accounts for this and takes directional technology one big step further. The adjustment of the directional characteristic is performed in a frequency specific manner. The polar pattern is not identical across all frequencies, but 20 independent beam formers are active to provide selective suppression of different noise sources (Fig. 7).

For each frequency channel, the optimal polar pattern is computed and activated in a few milliseconds (Fig. 8). This allows for the most favorable directional adaptation to the acoustic environment and meets the client's most prominent need: support in demanding listening situations.

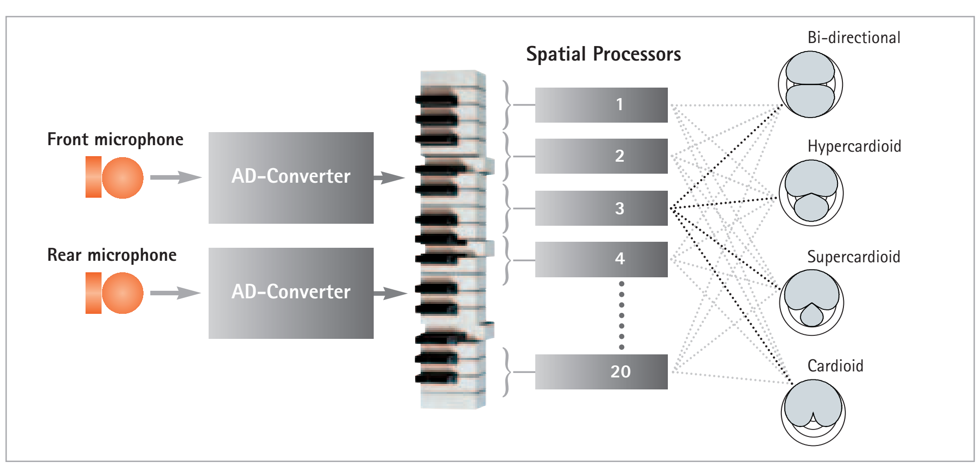


Figure 8: Digital SurroundZoom computes optimal polar patterns in 20 independent frequency channels.

SoundCleaning

Client's need:

Effortless listening and comfort in demanding situations, without loss of either sound quality in speech understanding.

The Savia solution: SoundCleaning

Savia's SoundCleaning features a number of novel signal processing strategies to increase the ease of listening and comfort in difficult acoustical situations. Reverberation and wind noise, for example, degrade speech understanding and listening comfort. Acoustic feedback, to give another example, is not only annoying but also limits the applicable gain. SoundCleaning means state-of-the-art digital signal processing to eliminate these difficulties and to further increase the benefit for the client.

EchoBlock

Reverberation is caused by reflections of sounds from walls, ceiling, or windows. These reflections generate delayed, slightly spectrally modified and attenuated copies of the original source signal. At the ear of the listener, a superposition of the direct sound from the source and the reflections is perceived (see Fig. 9). In effect, the original signal is temporally smeared. Reverberation is characterized by the reverberation time, which indicates how long reflections are present. Typical reverberation times range from about 0.4 sec in offices and small lecture rooms to up to 2 sec or more in concert halls and churches. Reverberation reduces and further degrades speech intelligibility in quiet and noisy situations, respectively (Johnson, 2000). In addition, the benefit from directional microphones is reduced in reverberant environments (Ricketts and Hornsby, 2003).

EchoBlock effectively detects and suppresses reverberations (see Fig 10). Now, for the first time, a hearing system is able to efficiently attenuate reverberation due to EchoBlock technology.

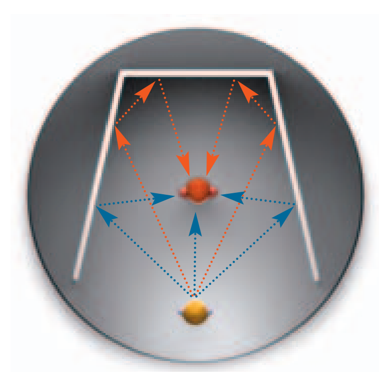


Figure 9: Reverberation – superposition of the direct sound from the source with reflections.

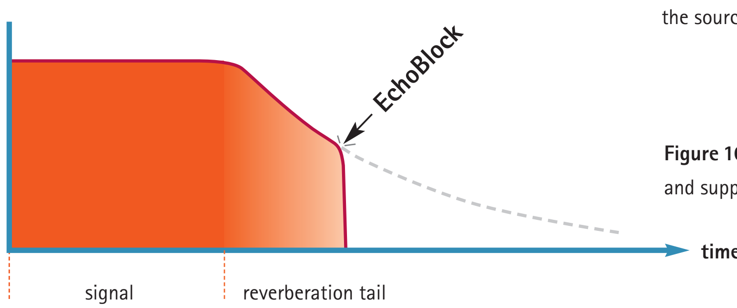


Figure 10: The principle of EchoBlock. The reverberation "tail" is detected and suppressed.

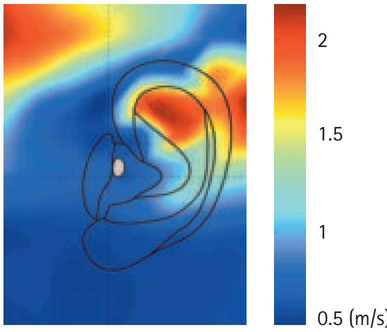


Figure 11: Turbulences at the ear in a light wind breeze (5 m/s).

Wind Noise Management

Wind noise originates from turbulences at head, torso, pinna and hearing instrument (see Figure 11).

Wind noise has a low frequency characteristic and can reach very high sound pressure levels. It is dynamic and depends on the direction and speed of the wind. Wind noise in hearing instruments can cause masking effects and microphone saturation. Due to the high level of wind noise, compression algorithms will react with gain reduction.

Wind Noise Management in Savia is based on two aspects: mechanical protection and electronic canceling. Mechanically, the wind and weather protection in BTE instruments attenuates turbulences before they reach the microphones. Electronically, wind noise is detected based on an input signal analysis. Subsequently, the wind noise portions are suppressed from the input signal. The unique combination of mechanical and electronic Wind Noise Management allows Savia to maintain directional microphone settings even in windy environments.

High Resolution Noise Canceler

Noise cancelation systems suppress steady-state background noise such as sound from engines or ventilators. In Savia, noise cancelation is performed in 20 independent and high resolution frequency channels which allows for efficient suppression of background noise irrespective of spectral shape. Improved analysis methods allow for better distinction between noise and speech signals.

Feedback Phase Inverter

Today, state-of-the-art microprocessors allow for the implementation of powerful signal processing strategies for effective cancelation of acoustic feedback. Savia's feedback canceler is based on an inverted phase approach which operates in the frequency domain. It follows the principle of phase inversion, where sound waves are canceled out by their own 180° phase inversion (see Fig. 12). This is the only technology that removes feedback without reduction of gain.

The algorithm comprises three steps: feedback detection, modeling of the feedback path and feedback erasure. For detection, a high resolution correlation analysis between hearing instrument input and output is performed. Acoustic feedback has a characteristic correlation pattern which is utilized for detection. For cancelation, a phase inverted signal with the same frequency as the feedback signal is generated. Due to destructive interference, the feedback signal is efficiently eliminated without gain reduction.

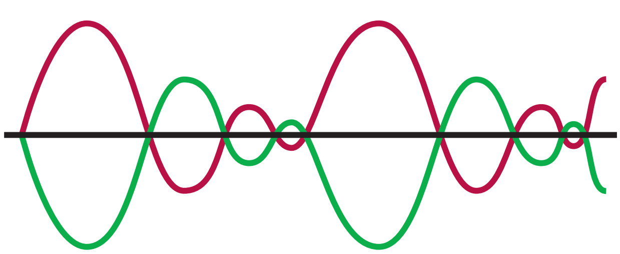


Figure 12: Principle of phase invertion. The feedback signal (red) and the phase inverted signal (green) cancel each other out (flat line).

iPFG™

Client's need:

Direct translation of individual demands into hearing instrument settings. Clients have individual hearing priorities, and situations in which they need the most support from the hearing instrument. These demands must be directly considered in the hearing instrument fitting to achieve maximum satisfaction.

The Savia solution: iCOSI

The **C**lient **O**riented **S**cale of **I**mprovement (COSI, Dillon, 1997) is a self assessment survey that identifies key hearing objectives. In contrast to questionnaires with fixed items (which may not be relevant for the individual), the client indicates those hearing situations which are the most important for him/her. This allows for efficient focusing of the fitting process on most central issues. In iPFG, Savia's fitting software, iCOSI is not just an "appendix" for the evaluation of the fitting success, but is actually embedded and fully integrated in the fitting process (see Fig. 13).

The use of iCOSI ensures that the client's most important hearing needs are integrated into the fitting and fine tuning process. These needs are not only addressed at the start and end of the fitting, but also throughout the whole fitting process. The individual needs of the client are uncompromisingly put at the center of the fitting process.

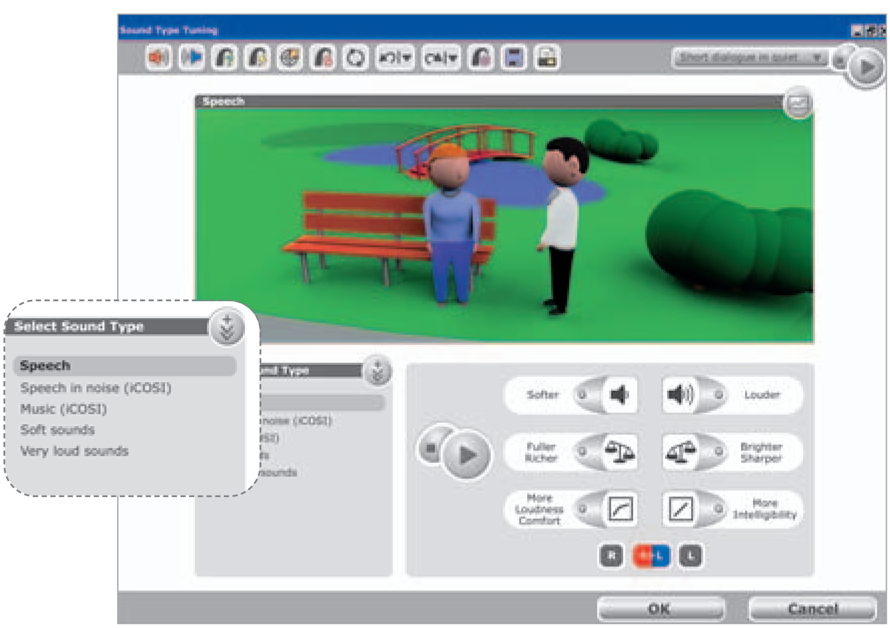


Figure 13: Integration of iCOSI into the fitting process. The individual iCOSI needs appear in the fine tuning tools.

Client's need:

The hearing instrument fine tuning should reduce the need for own volume changes. In different listening situations, the client increases or decreases the volume. Sometimes, these changes tend to be systematic, e.g., the client almost always decreases the volume in noisy situations, indicating a need for fine tuning. Ideally, the hearing care professional should know about these changes to be able to easily meet the client's needs. In reality, fine tuning has mainly been driven by unspecific client input. The hearing care professional could only roughly estimate the amount of fine tuning which is needed.

The Savia solution: DataLogging with User Preference Tuning

Savia hearing instruments include an innovation which opens up completely new possibilities of hearing instrument fitting and counseling: DataLogging. Quantitative information on the individual hearing instrument usage is now available to the professional. In addition to subjective comments from the client, the professional now has objective data at hand to further optimize the fitting process in a very efficient way. These objective data include:

- Total wearing time
- Average wearing time per day
- Proportion of time in automatic mode and for each manually chosen program
- Volume changes made by the client

A running average is computed from the volume changes made by the client in daily life. More recent volume changes have a stronger influence than changes conducted relatively long ago. Possible acclimatization effects are thus considered appropriately. Based on the running average values, Savia User Preference Tuning computes a fine tuning suggestion which can be activated by the hearing care professional (see Fig. 14) at the next appointment. The hearing instruments do not change the settings autonomously, the hearing care professional has full control over the fine tuning actions, which can now be conducted based on additional, objective data.

Further information provided by DataLogging such as the average wearing time per day or the frequency of chosen programs help to counsel the client in a tailor-made way and thus to effectively meet the individual needs of the client for optimal long-term benefit and satisfaction.

Figure 14: User Preference Tuning. Fine tuning suggestion based on actual volume changes done by the client in daily life.

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