Core

At the CORE of Innovation

Technological advances in signal processing and integration of wireless technology

The application of intelligent and automatic algorithms has significantly improved the performance and ease of use of modern digital hearing systems. However, despite all technical advances in recent years, there is still a large potential for enhancing the benefits hearing systems can give to wearers. Our understanding and knowledge about the listening needs hearing impaired people face in their daily auditory lives (Moore 2007) has significantly grown. We now can offer more solutions for the actual listening problems. In addition, microelectronic technology has made huge progress over recent years, mainly driven by the world of wireless communication applications.

CORE: The new benchmark in the hearing industry

State-of-the-art chip designs offer much more flexibility and computational power than ever before. CORE, the microelectronic heart of Phonak's latest generation of hearing systems Exélia and Naída, integrates six high-performance processors. Each has a different task such as micro-controller or digital signal processor and they collectively contain more than 8 million transistors, capable of performing over 120 million operations per second. This computational power provides the resources for introducing new solutions to enhance hearing performance and significantly improve wearer benefits. A small selection from the host of new solutions in the CORE platform will be further described below:

- SoundFlow - new and unique automatic adaptation algorithm
- WhistleBlock Technology - new revolutionary feedback management scheme
- ZoomControl - new concept for enhancing speech intelligibility in adverse listening conditions
- SoundRecover - new processing scheme for enhancing audibility of high frequency speech sounds

Additionally, the huge progress achieved in wireless communication technology made it possible to finally introduce a fully featured Body Area Network comprising an integrated system of accessories and two hearing instruments. This network solution opens the door to many new applications and will provide hearing impaired people with access to the fast developing world of wireless communication technology. Several different applications can be applied with such a new wireless network:

- Binaural signal processing - the ability to exchange broadband audio data between two hearing instruments (one of the long standing wishes in hearing instrument technology), which will enable true binaural signal processing algorithms for enhancing speech intelligibility and acoustic scene classification.
- Wireless connectivity - the ability to connect the hearing instrument wirelessly to external audio sources such as (mobile) phones, TV, and other audio sources using the iCom communication interface. In order to yield optimal sound quality, CORE wireless transmission supports streaming in two separate channels (stereo transmission) without compromising the audio bandwidth of 10 kHz.
- Remote operation - the ability to connect hearing instruments to remote controls (myPilot command center), which will provide new functionalities and modern concepts for user interaction.

- External data sources - the ability to connect the hearing instruments wirelessly to external data sources, which will enable, for example, fitting without cables using the CableFree fitting device iCube.

Integrating a wireless audio link into a hearing instrument requires fulfilling some rather challenging technical constraints:

- Transmission of audio data between the hearing instruments requires a sufficient channel capacity of 300 kBit/s.
- The power consumption has to be minimized such that the available power budget of a hearing instrument battery is met.
- The size/volume of the added microelectronic components must be very small in order to fit into housings of different styles of hearing instruments.

Comparison of some key values of the Phonak CORE platform with competitive devices

Product	CORE	Competitor A	Competitor B
Measured amount of data that can be wirelessly transmitted between HIs	300 kBit/s	120 kBit/s	0,2 kBit/s
Capability of streaming stereo signals	yes	no	no
Capability of streaming mono audio signals	yes	yes	no
Capability of exchanging control signals	yes	yes	yes

Smooth adaptation to dynamic listening conditions thanks to SoundFlow

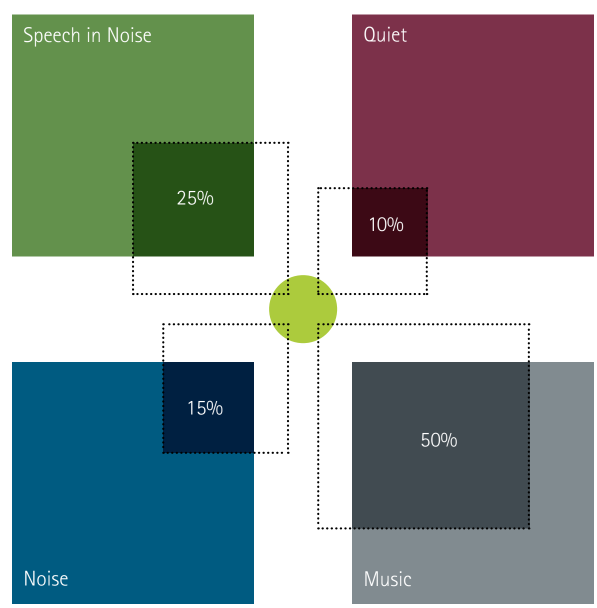
Automatic control has become a standard in modern hearing instrument technology. However, existing systems select optimal parameter settings from a very limited number of discrete program sets. Thus, in certain environments, compromises regarding the optimal processing strategy have to be made. Unfortunately, most daily life auditory scenes are not clearly defineable; many combinations of different sound sources are the rule.

In order to always provide optimal hearing performance, it is advantageous to have a system which smoothly adapts to the environment and selects the optimal parameters from a continuum of settings. In order to achieve this, the system has first to precisely assess the environment and provide a statistical analysis of the different relevant components within the scene, e.g. 25% speech in noise, 10% quiet, 15% noise, and 50% music (see figure 1). Rather than then selecting one specific hearing program (i.e. the music program), the system selects a weighted mixture of different programs. This strategy, called SoundFlow, allows the device to adapt to particular acoustic environments with unprecedented precision and provides optimal hearing performance throughout a dynamic auditory life.

Individualization is achieved by providing the hearing healthcare professional with the opportunity to optimally fit certain base programs to individual preferences. Taken together, the final hearing instrument parameter selection is defined by a precise statistical analysis of the acoustic scene combined with a weighting given by the individual listening preference of the hearing instrument wearer.

This new automatic concept has been evaluated using a paired comparison procedure for comparing switching artifacts and subjective speech intelligibility in different environments for two types of automatic systems: SoundFlow based on program composition versus discrete program switching. In all tests, more than 80% of the subjects clearly preferred SoundFlow.

Figure 1



The effective program is a weighted mixture of the available base programs.

Figure 1: Schematic representation of the functioning of SoundFlow. Based on the classification result of the environment (see box), a blended program that is specifically tailored for the actual situation is generated.

ZoomControl: Steering the listening focus towards the target of choice

Directional microphones represent the most effective method for improving speech intelligibility in difficult listening conditions. However, with existing systems, one limitation remains: The wearer has to face the target. This makes perfect sense for many communication situations as usually people are looking at each other when they communicate. However, in a significant number of conditions the target is not directly in front, but may also be to either side or even in the back.

Consider, for example, the situation in a car where the speaker is sitting to the side and the listener driving the car cannot turn his head. In social conditions, often the speaker is not sitting directly in front but to the side, e.g. at a dinner table. In these conditions, the two hearing instruments receive an asymmetric input: the one facing the target source receives the signal much louder than the contralateral device, which also receives mostly interfering sounds due to the head shadow. In these conditions it would be very helpful to pick up the signal at the side closer to the signal of interest and transmit and process it together with the contralateral signal. This requires full audio broadband transmission capabilities.

The CORE platform supports the needed broadband audio transmission between the two hearing instruments. The respective functionality in Exélia is called ZoomControl. In a binaural fitting and using the myPilot command center, the user can select the direction in which the devices should focus, allowing unprecedented speech intelligibility practically everywhere.

Feedback Management: new levels of performance

The introduction of digital technology has significantly improved the acoustic stability of hearing instruments. Modern feedback management systems enable hearing healthcare professionals to optimally use the residual dynamic range of the individual while using earmolds that either have larger vent diameters or are completely open. Despite these significant improvements, the performance of today's feedback management systems is still mainly driven by the tradeoff between the degree of feedback cancellation and the sound quality. A further disadvantage of current feedback management systems is that they incorrectly identify natural signals comprising tonal components such as music, telephone ringing or door bells as feedback, and introduce audible distortions perceived as roughness in the sound. In order to overcome this drawback and limitation, the hearing system has to precisely identify feedback and distinguish it from other tonal signals. WhistleBlock Technology eliminates feedback with much higher effectiveness and precision. It includes a state-of-the-art feedback identification and tagging module. This algorithm is able to instantly differentiate between true feedback and naturally occurring tones, such as music.

Figure 2

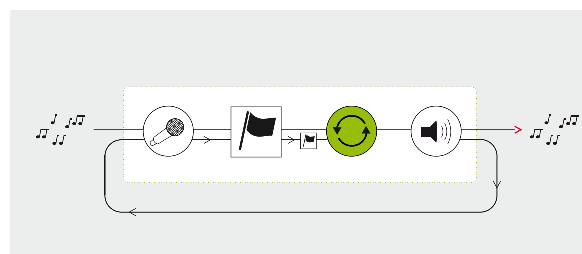


Figure 2: Schematic setup of WhistleBlock Technology including a new tagging algorithm for distinguishing feedback from non-feedback signal components. It profits from a state-of-the-art feedback identification and tagging module. This module is able to instantly differentiate between true feedback and naturally occurring tones, such as music. This accurate identification of sounds that have re-entered the system as true feedback allows for a precise feedback cancellation and significantly more stable amplification, without impacting speech clarity or sound quality.

To assess the performance of modern feedback management systems, several different aspects and quality dimensions have to be taken into account. Freed and Soli (2006) suggested looking at:
1. How effective is the algorithm at preventing feedback?
2. How effective is the algorithm at reducing sub-oscillatory peaks in the frequency response?
3. Does the algorithm sacrifice gain in any frequency bands?
4. How robust is the algorithm when presented with tonal input signals?

Figure 3 shows the added stable gain measured for six different devices. The instruments were equalized to produce the same amount of gain. It is clearly visible that WhistleBlock Technology provides significantly more stable gain. Especially in the frequency range most susceptible to feedback, between 1.5 kHz and 3 kHz, this new technology provides the largest benefit.

Figure 3

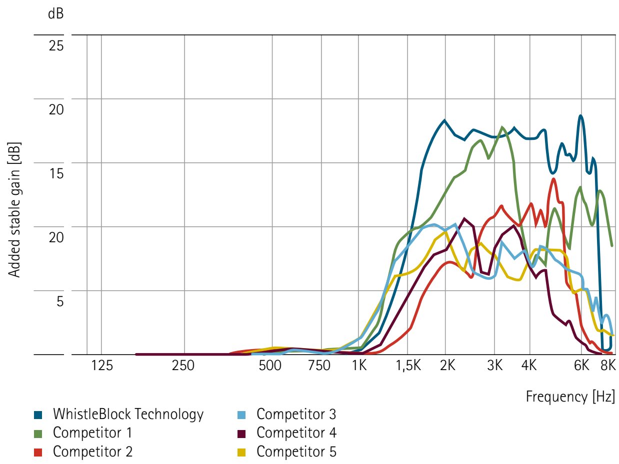


Figure 3: Added stable gain for different hearing instruments. WhistleBlock Technology shows by far the largest added stable gain especially in the most critical spectral region between 1.5 kHz and 3 kHz.

Figure 4 describes the tradeoff between performance relative to added stable gain versus sound quality. With existing feedback cancellers, increased feedback suppression results in decreased sound quality. WhistleBlock Technology eliminates this tradeoff. Higher stable gain is achieved without compromising the sound quality.

Figure 4

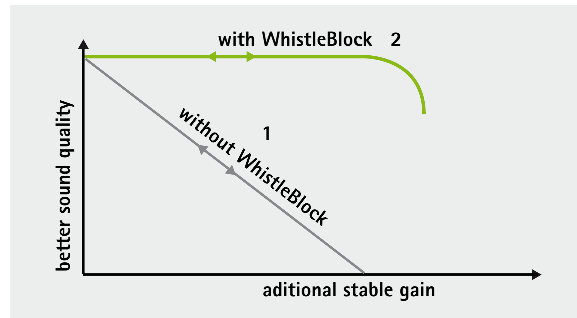


Figure 4: Qualitative representation of the tradeoff between sound quality and added stable gain. In traditional feedback management systems, there is a clear tradeoff between sound quality and the added stable gain (curve "1"). Thanks to WhistleBlock Technology, added stable gain can be increased without loss of sound quality (curve "2").

SoundRecover: The power of frequency compression

The majority of people with hearing impairment experience greater difficulty hearing sounds at high frequencies than at low frequencies. The greater the degree of hearing loss, the more challenging it is to provide sufficient high frequency gain. High frequency amplification may be a challenge due to very poor hearing sensitivity in that region. In many cases, gain is limited by acoustic feedback, discomfort or simply the physical limitations of the receiver. Not being able to hear high frequency information may result in:

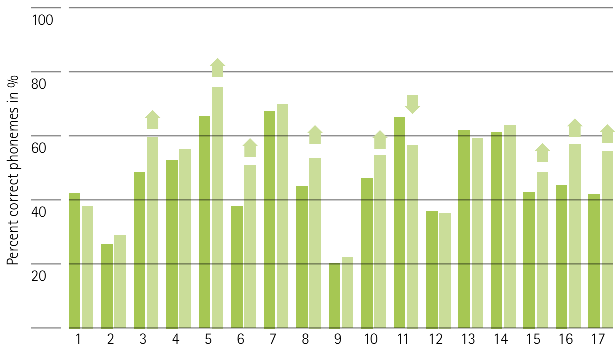
- Difficulty in recognizing important high frequency speechsounds, such as the fricative consonants /f/, /s/, and /sh/.
- Difficulty in hearing and identifying high-pitched environmental sounds, such as birdsong, alarms, and some musical sounds.
- Difficulty, particularly for young children, in learning to produce high frequency speech sounds.
- Difficulty, particularly for adults, in maintaining speech quality.

SoundRecover is a new, non-linear frequency compression algorithm that compresses selected high frequency sounds into a lower frequency range, where both hearing sensitivity and discrimination ability are better. It has been shown to significantly improve perception and recognition of high frequency sounds. SoundRecover effectively extends the audible range by providing important high frequency information without creating any annoying artifacts.

Frequencies below the SoundRecover kneepoint are not affected by the frequency compression algorithm and are amplified normally. This unique high frequency shifting scheme has been intensively studied both in the laboratory and in field trials (Simpson et al. 2005, Scollie et al. 2007, Glista et al. 2007). Several performance dimensions have been assessed such as (i) speech intelligibility in quiet and noise, (ii) perception of environmental sounds, (iii) speech production. The results of the different studies have shown a clear advantage of SoundRecover processing when compared to conventional amplification for both children and adults.

A study by Simpson et al. (2005) showed the advantages of nonlinear frequency compression with adult subjects (see figure 5). For the consonant / vowel nucleus / consonant (CNC) word tests, the mean phoneme scores obtained by each subject with their conventional hearing devices (dark green) and with the experimental device with non-linear frequency compression (light green) shows the percent of all phonemes (vowels and consonants) in the monosyllabic words recognized correctly by each subject when using each of the two types of hearing aid. Across the 17 subjects, speech recognition scores were compared with the two hearing aid processing schemes by means of a two-factor analysis of variance. A statistically significant improvement for the non-linear frequency compression scheme over the conventional hearing device was found.

Figure 5



The SoundRecover algorithm has been implemented in a new product, Naída, especially designed for wearers with a severe to profound hearing loss.

Figure 5: Speech intelligibility measured using Consonant-Vowel-Consonant with (light green) versus without (dark green) applying frequency compression (Simpson et al 2005).

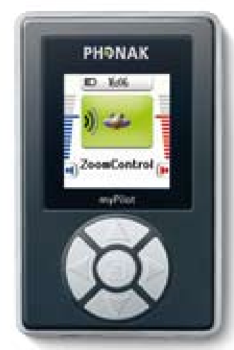
New options for user interaction with myPilot

Remote controls are an extremely useful accessory for clients who want to have full control over their hearing. Phonak has successfully introduced remote controls in digital hearing systems. With myPilot, users get a significant additional benefit. The changes that they make are displayed on a large, back-lit color display. For much more intuitive and easy use, the following information is displayed on the myPilot:

- Actual volume setting
- Fitted volume setting
- Remaing volume range (reserve)
- Actual program

Of course, all these actions are available for each hearing instrument separately (left and right). In addition, myPilot offers:

- Access to ZoomControl
- Selection of direction of ZoomControl
- Alarm and Clock
- Left/right monaural volume change
- Battery status of myPilot



Wireless Connectivity with iCom

The concept of the body area network (BAN) includes the possibility to connect hearing instruments wirelessly with external networks like mobile phones, Bluetooth, or FM systems. The iCom wireless accessory works as an interface and connects the Exélia or Naída hearing instruments with the wireless world. iCom can connect with Bluetooth enabled mobile phones and stream the phone signal to both hearing instruments, enabling the hearing impaired wearer to use the phone hands-free and binaurally. A directional microphone built-in iCom picks up the voice of the wearer and transmits it via Bluetooth to the mobile phone. When being called, the only thing the wearer has to do is accept the call by pressing the (only) button on the iCom. The rest is handled automatically by the BAN. A directional microphone is essential in such a device, as the voice of the wearer has to be picked up reliably, especially in loud environments (see figure 6). The use of omnidirectional microphones means picking up all environmental noises parallel to the speech signals, resulting in a poor SNR.

Figure 6

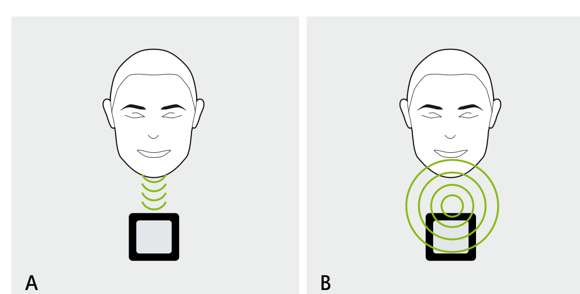


Figure 6: A directional microphone in the iCom (figure A) can pick up the voice of the wearer and provides good SNR in noisy environments. If an omni microphone is used (figure B), voice and noise are picked up simultaneously, making the conversation on the phone much more difficult.

CableFree Fitting with iCube

Fitting of digital hearing instruments requires attaching them to a programming cable specific to the manufacturer and often even the hearing instrument model. Thanks to the CORE platform, Exélia and Naída hearing instruments and all future devices based on this platform can be fitted without attaching them to any cable at all. The iCube fitting interface communicates with the PC via Bluetooth and with the hearing instruments via the wireless CORE technology. iCube is worn around the neck of the client and can easily be charged between fitting sessions. There is no need to select the correct cables anymore.

Summary

Hearing instrument technology is currently at the dawn of a new technological era similar to the introduction of digital hearing instruments 10 years ago. Microelectronics have made significant progress over the past years: the computational power has significantly increased, and it is now possible to integrate hearing instruments in a network allowing full audio communication between two instruments and also the further listening world. CORE is the new chip platform developed by Phonak that includes significant improvements in both aspects.

Additional computational power and wireless data transmission enable the introduction of more sophisticated signal processing algorithms. SoundFlow, WhistleBlock Technology, ZoomControl, and SoundRecover are just a few examples. They deliver significant user benefits and allow clients to better overcome the challenges of their daily life.

With the unique capability to transmit audio data from one hearing instrument to the other and from accessories to the hearing instruments and back, a huge world of opportunities is opening. Hearing instruments based on the CORE platform can create a Body Area Network and offer wireless access to mobile phones, steerable beamformers, cableless fitting and much more.

References

Freed, D., Soli, S. (2006), "An Objective Procedure for the Evaluation of Adaptive Anti-Feedback Algorithms in Hearing Aids", Ear and Hearing; 27(4): p. 382-398.

Glista, D., Scollie, S.D., Bagatto, M. Seewald, R., Johnson, A. (2007), "Evaluation of Nonlinear Frequency Compression II: Clinical Outcomes", submitted to Ear and Hearing.

Kochkin, S. (2005), "MarkTrak VII: Consumer Satisfaction in the Digital Age", The Hearing Journal, 58, p. 30-42.

Moore, BCJM (2007), "Cochlear Hearing Loss, Wiley and Sons".

Richars, V., Launer, S., Moore, BCJM (2006), "Potential Benefit of Across Aid Communication for Bilaterally Aided People: Listening in a Car", International Journal of Audiology, 2006. 45: p. 182-189.

Ricketts, T. and Henry, P. (2002), "Evaluation of an Adaptive Directional-Microphone Hearing Aid", International Journal of Audiology, 2006. 41: p. 100-112.

Scollie, S.D., Parsa, V., Glista, D., Bagatto, M., Wirtzfeld, M., Seewald, R. (2007), "Evaluation of Nonlinear Frequency Compression I: Fitting Rationale", Submitted to Ear and Hearing.

Simpson, A., Hersbach, A., and McDermott, H. (2005), "Improvements in speech perception with an experimental nonlinear frequency compression hearing device", International Journal of Audiology, 2005. 44(5): p. 281-92.