BestSound™ Technology

BestSound™ Technology

Compendium for Audiologists and Hearing Care Professionals

www.usa.siemens.com/hearing

SIEMENS

Table of Contents

1 Innovations for more than 130 years

Over millions of years, nature has perfected the human auditory system for communication. The intricacy and efficiency of this system continues to amaze us today. For example, the critical bands on the basilar membrane allow us to discriminate sounds; the outer ear canal resonance naturally amplifies the frequencies most important for speech understanding; the structures of the outer ear allow for front and back localization.

Unfortunately, this system was only designed to have optimal performance for approximately 50 to 60 years. With increasing life expectancy, more and more people experience a decline in the hearing capability. Back in 1878, Werner von Siemens, the founder of Siemens, invented a special telephone receiver for his hearing-impaired wife. Since then, the engineers and scientists at Siemens have continued to develop innovative solutions to help people with hearing loss (Figure 1). As illustrated by the following timeline, Siemens has been continuously in the forefront of hearing instrument technology. In the past five years alone, Siemens introduced to the world over nine innovations, many of them copied by other manufacturers, and some of them still unrivaled in the industry. The long-standing innovative leadership of Siemens is also reflected in the number of patients we serve. In the past ten years, Siemens fitted more patients than any other hearing instrument manufacturer. In fact, just in the past two years, Siemens fitted more than twice as many patients than any other competitor.

1878
Phonophor
Siemens develops the first technical solution for hearing loss.

1910
First line of hearing instruments
Siemens begins the first serial production of hearing instruments.

1959
Auriculina 326
Siemens launches its first Behind-the-Ear hearing instrument.

1966
Siretta 336
Siemens presents the first In-the-Ear hearing instrument.

1997
Prisma
Siemens introduces the first digital hearing instrument featuring a directional microphone.

2004
Acuris™
Siemens introduces the first wireless system e2e wireless® which is also available for CICs.

2008
Siemens Tek®
Siemens introduces a revolutionary wireless enhancement system giving wearers unprecedented access to all favorite audio devices.

Figure 1: Major innovations by Siemens since 1878

1878	Werner von Siemens developed the Phonophor (amplified telephone).
1910	The first line of Siemens hearing instruments was introduced.
1959	Siemens introduced the world's first Behind-the-Ear hearing instruments.
1966	Siemens introduced the world's first In-the-Ear hearing instrument.
1997	Siemens introduced PRISMA, the first digital hearing instrument with a TwinMic® directional microphone system.
2004	Siemens introduced ACURIS with e2e wireless®, the world's first hearing instrument featuring a wireless system small enough to fit in CICs.
2008	Siemens introduced e2e wireless® 2.0., the first wireless system for hearing instruments offering solutions for TV, stereos, and telephone, allowing listening in stereo. It is the only system of its kind that is also available for CICs.

2 BestSound™ Technology–a new dimension in hearing instrument technology

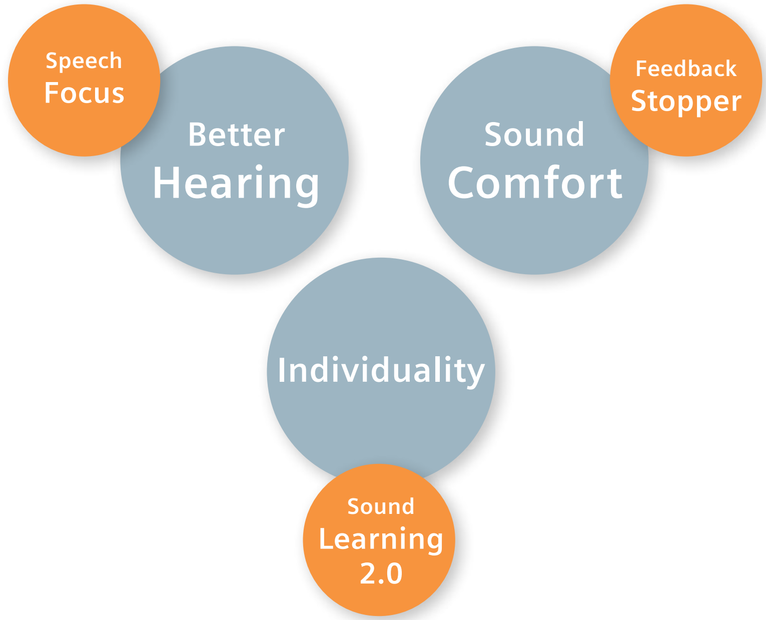

Figure 2: Needs of hearing instrument wearers and the innovative solutions offered by BestSound Technology.

Siemens is the industry leader not only in terms of the number of innovations, but also because these innovations have directly and effectively addressed the requirements of people with hearing loss. The number one requirement of the people who wear our products is **Better Hearing**. However, any means of improving hearing must come with good **Sound Comfort**. This is to ensure that patients effectively use the hearing instruments throughout the day to take the full advantage of them. Unfortunately, in many cases, Hearing (maximizing audibility) and Comfort are not always supportive of each other. In order to find the optimum balance between Better Hearing and Sound Comfort, one must take into account each patient's individual preferences and needs. The new BestSound Technology offers – in addition to a host of proven features – three new technologies designed to meet these specific requirements: **SpeechFocus™, FeedbackStopper™,** and **SoundLearning® 2.0.**

SpeechFocus was developed to foster **Better Hearing**. It continuously analyzes the environment for the most favorable speech-to-noise ratio, and automatically selects the microphone configuration that has the potential to offer the best speech intelligibility for the wearer, regardless of whether speech is coming from the front, behind, or the side.

FeedbackStopper facilitates the optimum **Sound Comfort**. This technology achieves effective feedback suppression by combining the proven Acoustic Fingerprint Technology with a new Transient Frequency Shift, eliminating feedback before it is even noticed by the wearer.

SoundLearning 2.0 is Siemens newest generation of learning technology that addresses **Individuality**. Automatically steered by an intelligent acoustic situation detector, SoundLearning 2.0 allows the wearer to teach preferred gain, compression, and frequency shape independently for various acoustic situations. This is the most sophisticated learning algorithm on the market.

2.1 Better Hearing

Better hearing, first of all, means that a hearing instrument should restore the audibility of sounds. Secondly, and more importantly for hearing instrument wearers, better hearing is not just hearing, but also the ability to understand speech, especially in noisy situations. To restore audibility, BestSound employs multi-channel automatic gain control for both a wide range of input signals, and for controlling the maximum output (AGC-I and AGC-O). To improve speech intelligibility in noisy situations, directional microphones and wireless connections (e2e wireless® 2.0, Telecoil) are used.

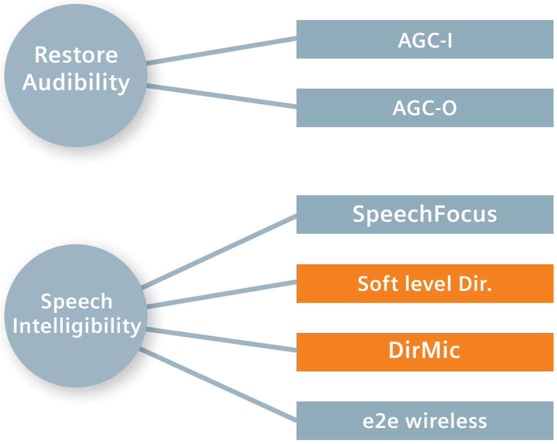

Figure 3: This figure shows the feature set for Better Hearing

2.1.1 Restoring audibility

2.1.1.1 AGC-I & AGC-O

To precisely map real-world sounds into every patient's residual dynamic range so that soft sounds are audible, average sounds are comfortable, and loud sounds are tolerable, Siemens hearing instruments offer up to 16 channels of compression (AGC-I) (Figure 4). In combination with wide dynamic range multi-channel input compression, an intelligent AGC-O is also required to guarantee that everyday sounds are not uncomfortably loud. Because UCL varies across frequency, AGC-O also needs to be multi-channel to be effective. In addition, broadband signals are perceived as louder than narrowband signals due to the loudness summation effect. So to ensure that the patient's dynamic range is utilized to the fullest without exceeding any uncomfortable levels, Siemens offers a multi-channel, bandwidth-controlled AGC-O that guarantees a comfortable level of loudness to an equal extent for narrowband and broadband sounds.

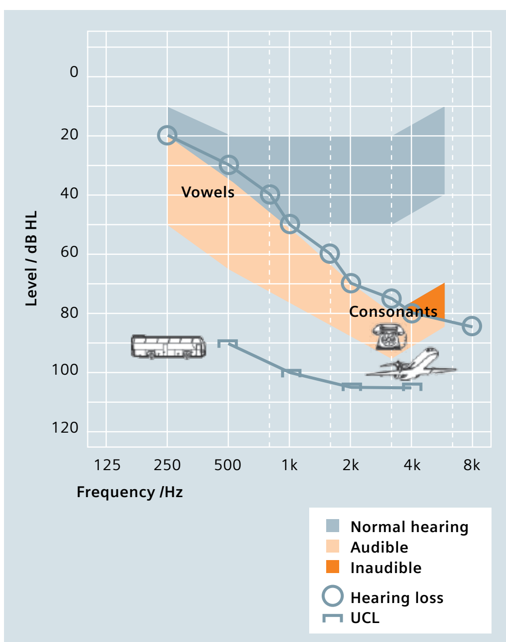

Figure 4: Note how the dynamic range becomes smaller as the hearing loss becomes more severe, illustrating the need for multi-channel compression to map speech into the patient's residual dynamic range.

2.1.1.2 Adjusting gain, compression and MPO in CONNEXX

Siemens hearing instruments offer up to 16 channels of compression. Depending on the fitting formula and acclimatization level selected, the First Fit algorithm automatically sets the master gain as well as the compression kneepoints and ratios to achieve the best possible match to the selected fitting formula for a wide range of input levels. Nevertheless, to offer the hearing care professional maximum flexibility, every compression channel has an independently adjustable compression kneepoint, ratio, and time constant. To make the fine tuning process more efficient, channels can be combined into handles (a grouping of adjacent channels), and level-dependent gain can be also adjusted separately. For instance, by clicking the "gain for soft sounds" controls, gain and compression thresholds are automatically adjusted accordingly.

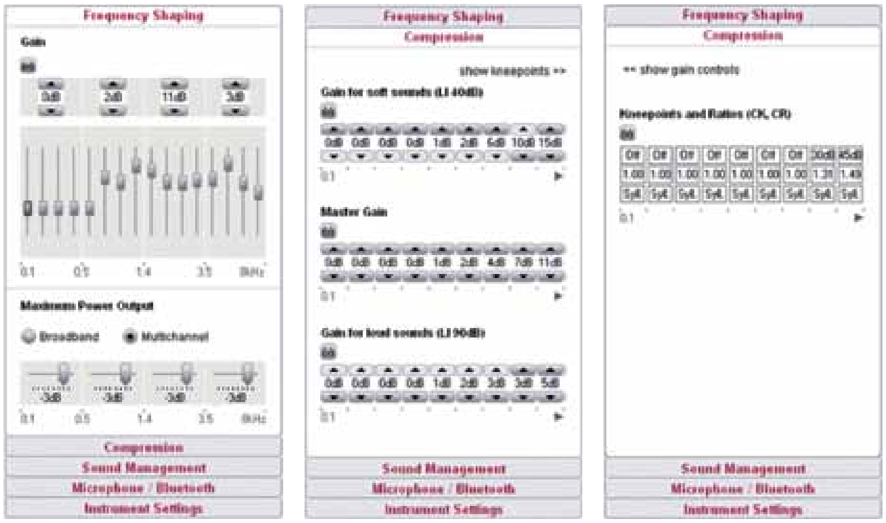

Figure 5: Compression controls in CONNEXX fitting software.

2.1.2 Improving speech intelligibility in noise

2.1.2.1 Directional Microphones

Directional microphone technology is the only feature proven to enhance speech intelligibility in difficult environments. One widely recognized limitation of directional microphones is that they can only focus directivity to the front of the wearer. In real life, however, speech comes from all directions. For instance, in a common scenario, a car driver would like to hear rear seat passengers talk. A similar situation occurs when the hearing instrument wearer is walking side by side with the speaker. In this case, speech does not originate from the front, and the hearing instrument wearer cannot turn to look at the speaker continuously. An ideal microphone system therefore, should be able to steer maximum directivity towards speech, regardless of the direction from which it originates.

2.1.2.2 SpeechFocus

SpeechFocus is the key feature in BestSound Technology that overcomes the limitations of traditional directional microphones. In addition to having all the functionalities of a four-channel adaptive directional microphone system, when necessary, SpeechFocus automatically suppresses noise that occurs from the front of the wearer, and focuses on speech coming from a different direction, for example, from behind the wearer. SpeechFocus continuously scans sounds in the listening environment for speech patterns. When speech is detected, then SpeechFocus selects the directivity pattern most effective in focusing on that speech source.

SpeechFocus works by operating three different directivity patterns simultaneously: omnidirectional, adaptive directional, and a reverse directional pattern. Unlike a typical directional microphone's cardioid pattern, which only attenuate sounds coming from the sides and the back, this backward directional microphone pattern works like an acoustic rear-view mirror and focuses on speech which originates from the back while suppressing noise from the front. SpeechFocus will only engage when the classification system identifies "Speech in Noise" and the input level is greater than 66 dB SPL. Once these criteria have been met, the signals from all directivity patterns are then analyzed for speech patterns. The microphone pattern that generates the greatest output for speech signals is then chosen as the appropriate microphone mode. This means when speech is detected from the front hemisphere, the adaptive directional microphone is employed, reducing noise from the side and the back. When speech is detected as originating from the back, the backward directional microphone pattern is selected. When speech is detected directly from the left or the right side, then the omnidirectional directivity pattern is engaged (Figure 6).

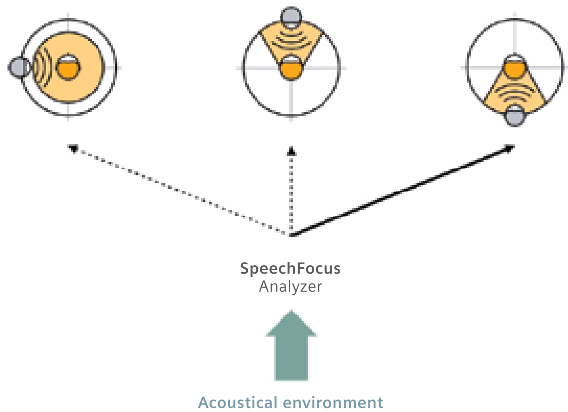

Figure 6: Illustration of how SpeechFocus analyzes the input signals from three directivity patterns for effective steering.

The effectiveness of SpeechFocus was validated in a study with 20 experienced hearing instrument wearers. Results showed an approximate 10 dB SNR advantage ($p<0.001$) for SpeechFocus over the conventional directional microphone system when speech originated from the back (Figure 7). This result was also confirmed in the real world when the subjects preferred SpeechFocus over traditional directional microphones after a home trial (Branda & Hernandez, 2010).

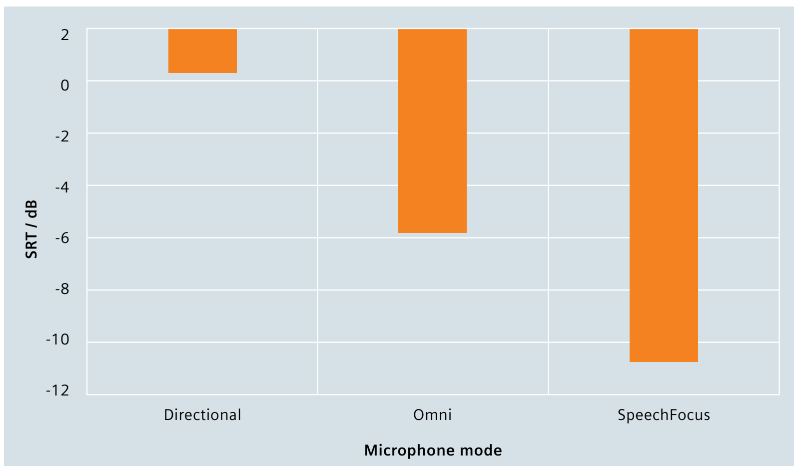

Figure 7: A recent study shows that in a situation where speech originates from behind the wearer, SpeechFocus offers an approximate 10 dB SNR advantage over traditional directional microphones.

2.1.2.3 Soft Level Directivity

Another widely recognized limitation of traditional directional microphones is the fact that while their primary function is to suppress noise, directional microphones themselves are noisy. As a result, the typical implementation is that directional microphones are engaged automatically only when the noise floor exceeds a certain loudness level, usually between 50 and 60 dB SPL. This prevents the wearer from hearing the amplified microphone noise that can be louder than the noise in the environment. Even at this setting, directional microphone noise can still be problematic for some wearers who have good low frequency hearing. Because of these higher noise levels from traditional microphones, individuals often can only take advantage of directionality when the noise floor of the listening situation is relatively high.

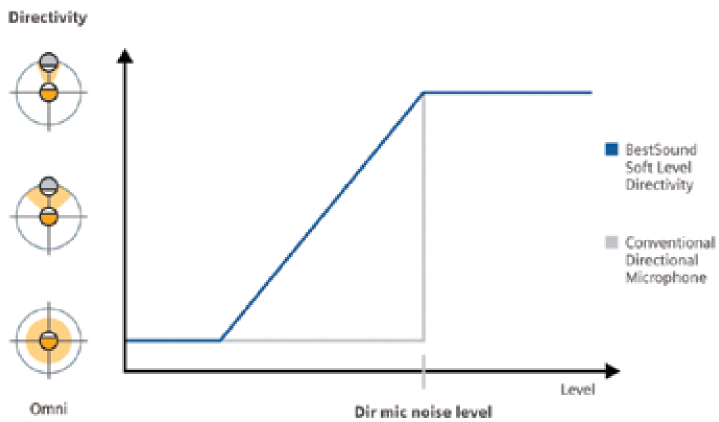

Figure 8: Soft Level directivity enhances directional benefit for soft levels

BestSound Technology's directional microphone system features a Soft Level Directivity that overcomes this limitation. With Soft Level Directivity, directional microphones can be activated at a much lower noise level. The extent of this directivity, however, is dependent upon the noise level so that the softer the noise level, the less directional the microphone is. The advantage of such a noise level-dependent directivity is that when the microphone is less directional, the microphone noise is also softer. In effect, the microphone noise is always less than the ambient noise, which allows the wearer to take advantage of directivity and achieve the best results even in low-noise situations.

2.1.2.4 Adjusting microphone settings in CONNEXX

SpeechFocus is offered as a separate directional microphone setting under the Microphone/*Bluetooth*® tab. SpeechFocus is also a canned program which can be selected in the dropdown menu in programs 2-5. We recommend it be selected when setting up programs for use when speech may come from different directions where the wearer cannot turn his or her head to face the speaker. Directional (Adaptive) should be selected for very noisy environments, and the Omnidirectional microphone should be used for music and outdoor programs, as well as in a mixed input mode, such as with DAI or Tek® Transmitter.

Figure 9: Fine Tuning options for the microphone system in CONNEXX

Automatic/TruEar mode is the default option in the Universal program and is designed to provide the most effective setting for everyday listening situations. In this mode the microphone system will automatically switch to the multi-channel adaptive directional mode when noise is detected in the rear field above the activation threshold; in quiet situations, or when the rear field noise level drops below the activation threshold the omnidirectional microphone mode will be deployed. Furthermore, when the omnidirectional mode is active in 501, or 701-level BTEs, the TruEar algorithm will adjust the microphone response to mimic the effects of the outer ear by enhancing the high frequencies as the natural ear would do; this is done to overcome the poor microphone location effects (MLE) of BTE as it sits atop the ear. To take full advantage of Soft Level Directivity, and to fine-tune its function, it is possible to adjust the Activation Threshold at which the microphone switches from the omnidirectional to directional mode. This is the level that the environmental noise must reach before the directional microphone is activated. The three threshold settings are: 48 dB SPL, 54 dB SPL, and 60 dB SPL. The default setting is 54 dB SPL, which is preferred by most wearers. A threshold of 48 dB SPL is designed for someone who requires even more directivity than in the default setting – even in low-level noise. The highest setting of 60 dB SPL would be selected for customers who prefer the omnidirectional mode in most situations and for those who are used to hearing environmental sounds from all around and are accustomed to the sense of safety this offers. To use the same Automatic/TruEar mode setting as in CONNEXX 6.3, select 54 dB as the activation threshold.

2.1.2.5 SpeechFocus – How to demonstrate

SpeechFocus can be demonstrated using the CONNEXX Real Time Display. The diagram represents an aerial view of the hearing instrument wearer facing front (0º azimuth). Depending on the direction from which speech originates, the green field indicates the focus of directivity. Have the person wear the instrument so that he or she can hear, as well as see the effects of SpeechFocus. Play the provided sound file utilizing two loudspeakers. Position the loudspeaker playing speech signal in front of the person and place the loudspeaker playing noise behind. SpeechFocus will be engaged and the green field will indicate where speech is detected and where directionality is focused. The two speakers will automatically alternate playing the speech and noise signal. When speech comes from the front, the green field will focus to the front of the wearer, indicating that traditional directional microphone is engaged. When the noise is coming from the front and the speech is coming from the back, the reverse directional microphone will be activated and the green field will appear behind the wearer (Figure 10). To demonstrate how the hearing instruments switch to omnidirectional mode when speech comes from the side, move the speakers to either side of the wearer. In this case, the green field will surround the wearer. Since the hearing instrument cannot determine exactly which side speech originates, the speaker figure will alternate on either side of the individual.

Figure 10: Demonstrating SpeechFocus with the Real Time Display.

2.1.3 Bypassing noise & reverberation with wireless connections

Even more efficient than directional microphones for improving speech intelligibility in reverberant and noisy environments, wireless connections bypass ambient noise by picking up the “clean” target signal and transmitting it directly into the hearing instruments.

BestSound Technology is equipped with e2e wireless 2.0 and is thus compatible with the Tek® Wireless Enhancement System.

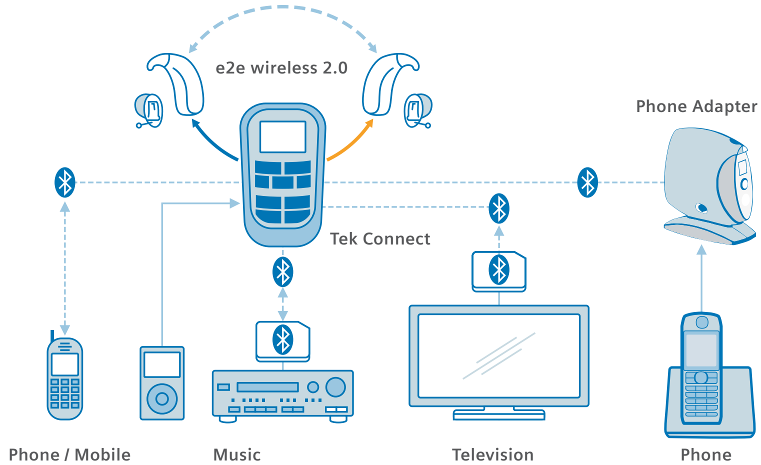

Figure 11: Connectivity options with Tek®.

Tek with e2e wireless 2.0 improves speech intelligibility on the phone for several reasons. It bypasses noise in the listener's environment and offers a better acoustic coupling than with a microphone alone. Listening with both ears allows the wearer to take advantage of binaural redundancy and central integration which can improve the SNR by 2 dB. And of course, listening with two ears assures that the better ear is always included. In fact, recent research has shown that listening on the phone with both ears improves the SNR by more than 5 dB (Ricketts & Picou 2008) (Figure 12).

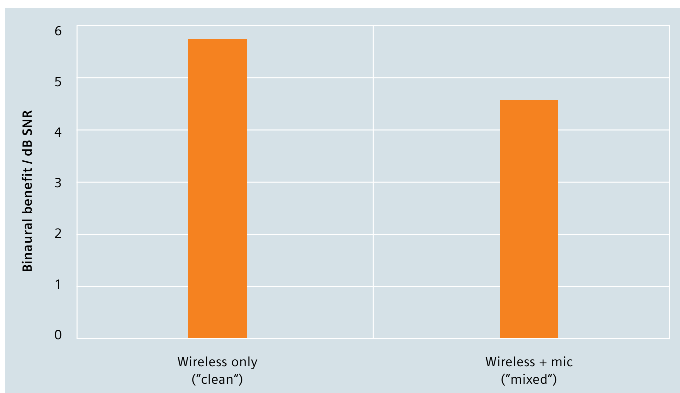

Figure 12: Study shows that in the wireless only condition where the subject is only listening to the telephone signal, binaural listening offers more than 5 dB improvement in SNR. In the wireless plus microphone mode where the telephone signal is mixed with the environmental sounds, the improvement is also more than 4 dB.

2.1.3.1 Adjusting wireless connections in CONNEXX

CONNEXX allows mixing the wireless signal with the hearing instrument's microphone signal according to the patient's needs. The levels for the wireless and the microphone signal can be adjusted independently.

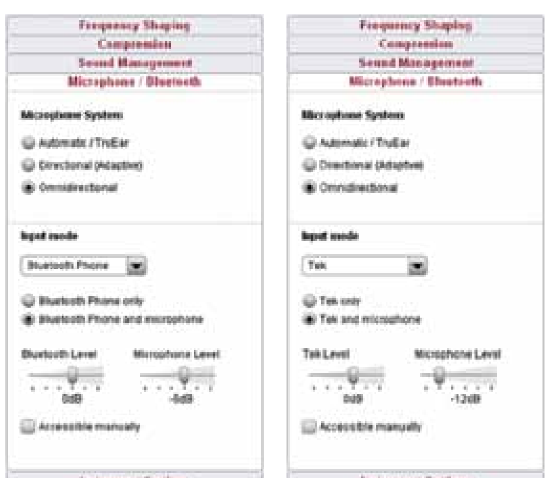

Figure 13: Fine-tuning options to set up wireless connections in CONNEXX.

2.2 Sound Comfort

The auditory deprivation caused by gradual hearing loss makes it difficult for many individuals to accept amplification immediately. This is one reason why first time wearers often complain that everything sounds too loud or too shrill. This is also one of the reasons why many hearing instruments end up "in the drawer". In order for end wearers to become acclimatized to amplification, wearing new hearing instruments must be comfortable, not only physically, but also with respect to sound quality. First time wearers typically appreciate any means to suppress unwanted noises since they experience the most difficulties in noisy situations. Experienced wearers, however, know that besides noise reduction, feedback suppression, and bandwidth extension, restored localization is also important to improve sound comfort (Kochkin 2005).

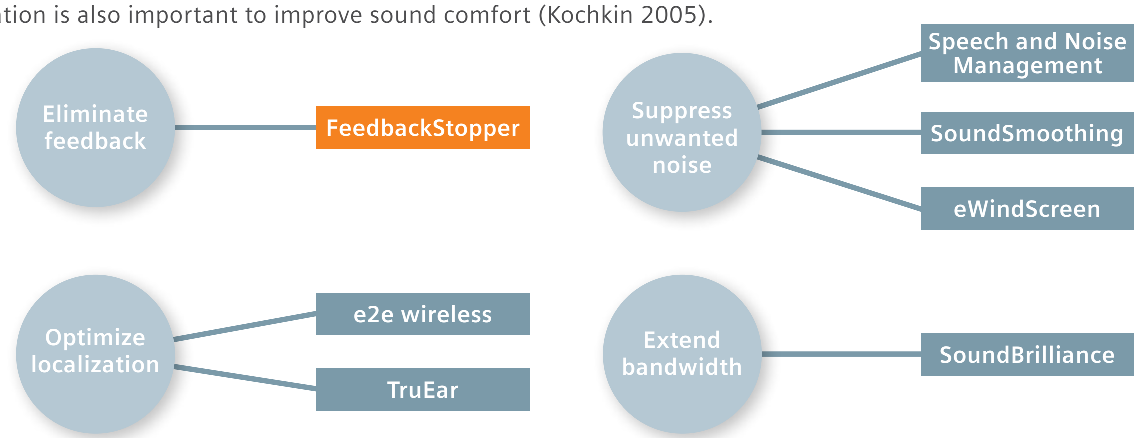

Figure 14: This figure shows the feature set for Sound Comfort.

2.2.1 Eliminating Feedback

Acoustic feedback is one of the most negative aspects associated with hearing instruments. In many cases, the annoyance caused by feedback may even outweigh an individual's perceived benefit of amplification. Feedback occurs when amplified sound from the receiver leaks out of the ear canal (e.g. through a vent), is picked up by the microphone, and in turn amplified again. The route taken by the amplified signal back to the microphone is called the feedback path. The goal, therefore, is to break this feedback path. With BestSound Technology, Siemens introduces the new FeedbackStopper which provides the most efficient protection against acoustic feedback, even in the most challenging situations. Effective feedback suppression is so fundamental and essential in a hearing instrument that the new FeedbackStopper is featured in all products equipped with BestSound Technology.

While most currently available feedback management systems are able to account for individual ear and fitting differences to counteract the static feedback path, the challenge lies in rapidly changing feedback paths. Even for the same person, no matter how well an earmold or an in-the-ear hearing instrument fits in the canal, as the person goes about in his daily life, the feedback path changes when he talks or chews, when he leans his head back against the couch, or when he hugs a loved one. Such changes in the feedback path occur very quickly. In order to eliminate feedback in these situations before it becomes disturbing to the wearer, it is essential that the feedback canceller acts within milliseconds. A potential downside, however, is that fast acting feedback cancellation systems often mistake tonal stimuli in the environment (e.g., music) for feedback, and thereby create artifacts and distortions in such acoustic situations. Therefore, a good feedback cancellation system must fulfill the following three requirements:

Figure 15: Requirements for Feedback Cancellation algorithm.

2.2.1.1 FeedbackStopper

Siemens powerful FeedbackStopper is an adaptive phase cancellation system combined with the technology to implement transient frequency shifting. The adaptive phase cancellation filter effectively cancels feedback by continuously estimating the feedback path and generating a corresponding out-of-phase signal. Thus, feedback is suppressed without reducing gain for external signals like speech, music, and environmental sounds. This adaptation process can be achieved extremely quickly due to transient frequency shifting. Briefly shifting the entire output of the amplifier by 25 Hz breaks the feedback loop and therefore helps to suppress feedback. But even more importantly, the frequency shift helps to avoid artifacts. This is because the adaptive filter is more likely to generate artifacts if external signals are similar to the feedback signal. By shifting the frequency of the feedback signal, it becomes less similar to the external signal so that the adaptive filter can react rapidly without generating artifacts. This small frequency shift may be perceived by some listeners; however, the brief shift is much less audible or annoying than the whistling of feedback.

Absolute freedom from artifacts can only be achieved if the adaptation is stopped and if the frequency shifting is switched off. To maximize listening comfort, therefore, FeedbackStopper completely halts adaptation and frequency shifting whenever there is no feedback risk. To distinguish between feedback situations and non-feedback situations, FeedbackStopper employs the Siemens proven Acoustic Fingerprint Technology (AFT). Similar technologies are used in audio coding and referred to as “digital watermarking.” With AFT, the amplified signal that leaves the hearing instrument receiver is slightly phase modulated and this “tag” or “fingerprint” is used to identify signals that have already been amplified by the hearing instrument (i.e. feedback potential). It has been demonstrated in psychoacoustic experiments that this phase modulation tag is inaudible to the human ear. It can, however, be identified by a technical modulation detector and used as a critical piece of information which tells the FeedbackStopper when to activate frequency shift and adaptation.

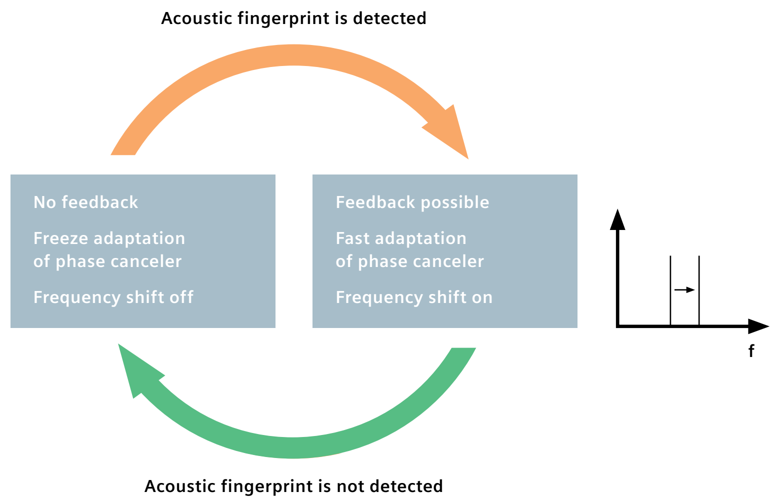

Figure 16: How FeedbackStopper works.

Figure 16 illustrates how the new FeedbackStopper works. Once the acoustic fingerprint is detected (e.g., feedback is likely to occur), FeedbackStopper briefly shifts the entire output of the amplifier by 25 Hz. As soon as feedback is suppressed and the fingerprint is no longer detected in the incoming signal, the FeedbackStopper switches off frequency shifting and completely freezes the adaptation in order to avoid artifacts.

Table 2 summarizes the technologies used by FeedbackStopper to fulfill the three main requirements of an effective feedback cancellation system.

Requirements	FeedbackStopper
Suppress feedback	Phase Cancelation
No artifacts	Acoustic Fingerprint Technology
Fast reaction	Transient Frequency Shift

Figure 17: Requirements and solutions for feedback suppression.

2.2.1.2 Evaluation of FeedbackStopper

To evaluate this new technology, FeedbackStopper was tested in a study which replicated real world conditions (Branda & Herbig, 2010). Twelve subjects with moderately sloping hearing loss listened to babble noise presented at 50 dB SPL while moving their heads, chewing, and talking. Added stable gain provided by FeedbackStopper with and without frequency shift was measured. The results showed significant ($p < 0.01$) improvements of 7 dB for the FeedbackStopper with frequency shifting activated (setting "fast" in Fig. 20) compared to a setting with the same phase cancellation setting but without frequency shifting. The "med" setting also employs frequency shifting, but with a slower adaptation speed of the phase canceller. On average, added stable gain of 24.5 dB is achieved with the FeedbackStopper in "fast" setting. This clearly exceeds the needed margin for variations of the feedback path in real-world listening for the majority of hearing instrument wearers.

Figure 18: FeedbackStopper with transient frequency shifting and fast phase cancellation provides ~25 dB added stable gain in dynamic situations.

Open Optimizer

Since open-canal fittings are the most challenging for feedback stability, the Open Optimizer featured in Pure® and Siemens Life instruments further improves the effectiveness of FeedbackStopper. This is achieved by optimizing the parameterization of the signal processing on specific properties of the acoustic feedback path of open-canal fitting. By activating the Open Optimizer, the stable gain of Pure and Siemens Life open-canal fittings can be increased by 4 dB in noisy environments and 7 dB in quiet listening situations. Hence, for fittings with closed and open domes/tips and vents larger than 2.5mm, Open Optimizer should be activated in order to achieve maximum feedback stability. If venting of 2.5 mm or larger is selected in the Acoustical Parameters portion of the First Fit Navigator, Open Optimizer will automatically be activated for Pure and Siemens Life instruments.

2.2.1.3 Fine-tuning the FeedbackStopper in CONNEXX

FeedbackStopper has three settings:

- **Slow setting:** Fast feedback suppression without frequency shift. This setting is optimized for maximum sound quality and should be chosen for individuals who are less prone to feedback and more disturbed by the roughness in sound quality introduced by frequency shift.

- **Medium setting:** Fast feedback suppression which features a less sensitive frequency shift activation so shifting occurs only when absolutely necessary. This is the default setting.

- **Fast setting:** Fast feedback suppression with a more sensitive frequency shift activation so the shift occurs more often. This setting should be used for individuals more prone to feedback and/or not bothered as much by occasional roughness in sound quality caused by frequency shift.

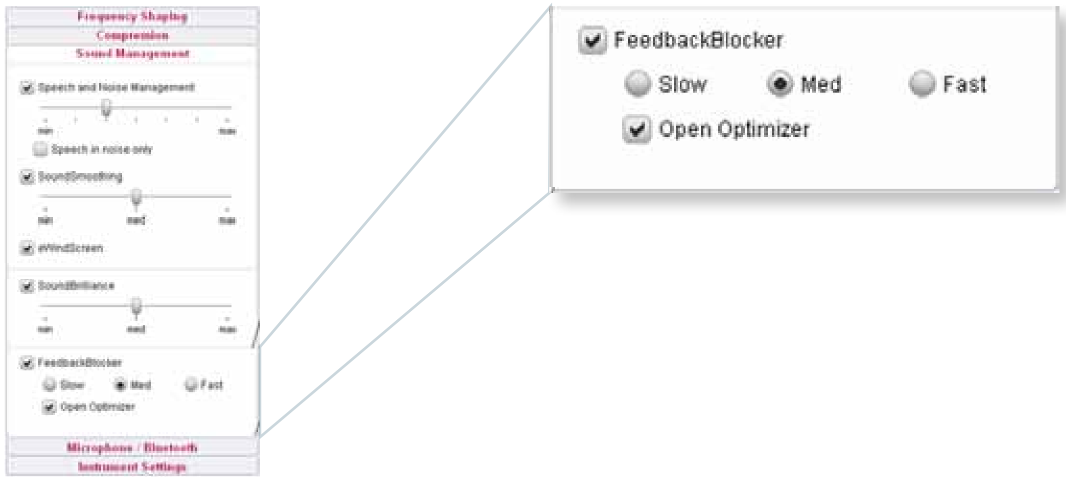

Figure 19: Open Optimizer is automatically activated by First Fit except for custom earmolds with vents smaller than 2.5 mm.

2.2.1.4 FeedbackStopper – How to demonstrate

The FeedbackStopper can be demonstrated with the hearing instrument either in the patient's ear, or on a model ear. Set the FeedbackStopper to OFF with the hearing instrument at "use gain", and slowly approach the hearing instrument with an open hand. Note the distance between the hand and the hearing instrument when feedback first occurs. Next, set the FeedbackStopper to the Fast setting. Note how much closer the hand can be to the hearing instrument before feedback occurs, indicating that the FeedbackStopper can effectively eliminate feedback even in the most critical situations.

2.2.2 Suppressing unwanted noise

One of the common needs for hearing instrument wearers is to understand speech despite different kinds of background noise. That means that hearing instruments need to be able to distinguish different kinds of noises so appropriate strategies can be applied to ensure that speech remains intelligible while noise is reduced. BestSound Technology employs dedicated signal processing schemes to suppress stationary noise, noise during speech, impulse noises (e.g. doors slamming), wind noise, and microphone noise.

2.2.2.1 Speech Enhancement & Noise Reduction

Siemens Speech and Noise Management system is comprised of a slow, modulation-based algorithm called Noise Reduction and a rapid-acting Wiener filter called Speech Enhancement. Noise Reduction works by actively detecting speech modulations in the input signal. When speech is not detected, the signal is categorized as noise and is reduced; this strategy is most effective in noise-only situations. In Speech Enhancement, the hearing instrument continuously estimates the noise level in the incoming signal; this is a very fast acting algorithm that works well to attenuate noise during speech – even between syllables. Neither of the algorithms affects speech-only situations. The fact that both strategies are multi-channel and adaptive means that different noises with different frequency content can be addressed simultaneously.

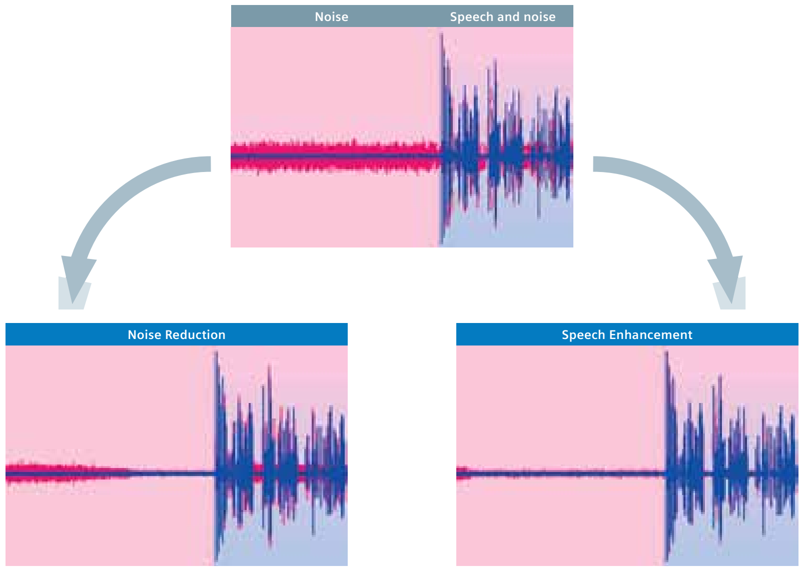

Figure 20: Two types of noise reduction working simultaneously: Modulation based noise reduction (left) and suppressing noise with Speech Enhancement (right).

2.2.2.2 SoundSmoothing®

Research has shown that 33% of background noise is transient in duration, and that for hearing instrument wearers, this type of noise is rated nearly as annoying as other noise types with longer durations (Hernandez et al 2006). Such sounds occur in many everyday activities (e.g., clinking of dishes, heels on hard flooring, newspaper crinkling, etc). SoundSmoothing is a highly sophisticated algorithm specifically designed to reduce the annoyance of such noise signals without affecting the spectrum of speech signals (Keidser, G., O'Brien, A., Latzel, M. & Convery, E. (2007); Evaluation of a noise reduction algorithm that targets non-speech transient sounds. The Hearing Journal, 60(1), 29-39). (Figure 21).

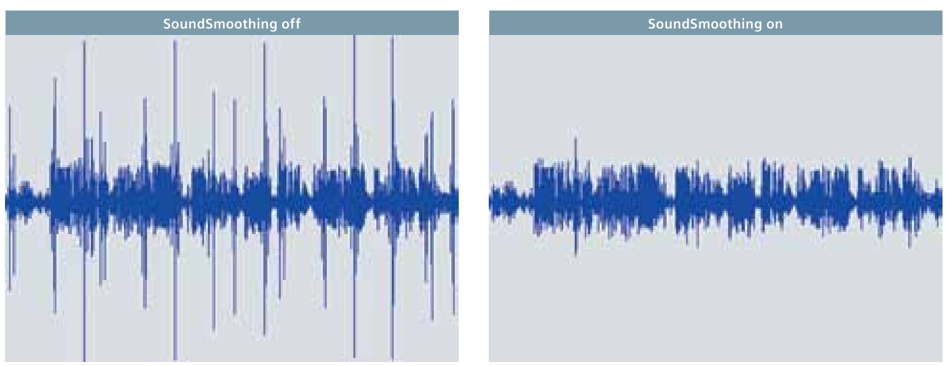

Figure 21: Speech interspersed with clattering dishes, with SoundSmoothing "off" (left panel) and "on" (right panel). With SoundSmoothing "on" (right), the sudden impulse sounds of the dishes are significantly attenuated while the speech spectrum is preserved.

2.2.2.3 eWindScreen®

For hearing instrument wearers, when wind blows across the microphones the resulting noise can be loud and distracting, preventing them from enjoying many activities like a round of golf, a cycling trip, or a day at the beach. In order to reduce wind noise, Siemens eWindScreen actively detects the presence of wind and reduces its distracting effects. This is accomplished by using two microphone signals. In a typical listening situation, when speech arrives at two microphones, the signals are correlated. However, when the wind blows, it creates turbulence at the microphone ports, producing two different and uncorrelated signals (Figure 22). When uncorrelated signals are detected, eWindScreen is activated. In the presence of wind, eWindScreen switches to the omnidirectional microphone mode and reduces the output level of frequencies below 1 kHz by up to 30 dB.

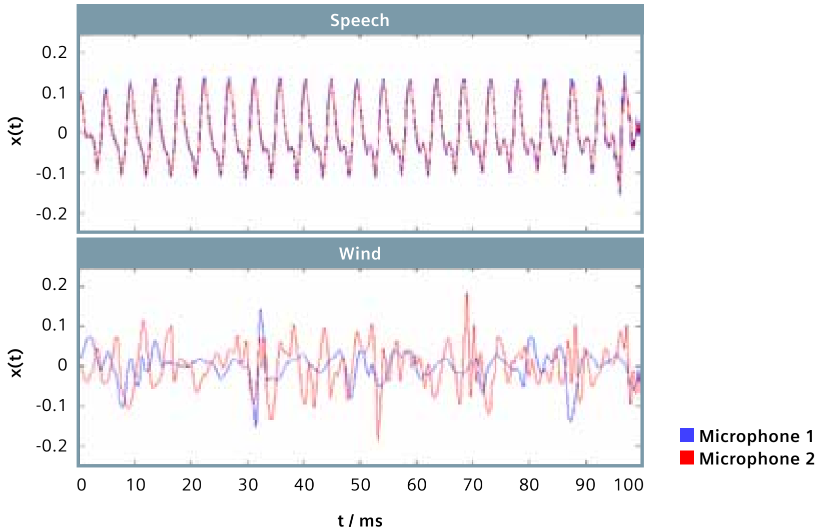

Figure 22: Speech and wind signals arriving at the two microphones. On the top graph, speech signals arriving at the two microphones are correlated. On the bottom graph, wind generates turbulence at the microphone ports and results in signals that are uncorrelated.

BestSound Technology also offers eWindScreen for single-microphone instruments such as CICs. In this configuration, the situation detector is extended to classify wind based on the typical fluctuating spectral properties of wind noise. When wind is detected, the gain in frequencies below 1 KHz is reduced. eWindScreen with two-microphone detection, however, is still more reliable and faster than single microphone wind detection.

2.2.2.4 Fine-tuning noise suppression in CONNEXX

Speech and Noise Management

The specific behavior of the different algorithms of the Speech and Noise Management can be fine-tuned in seven different steps with the "Max" setting representing the strongest noise attenuation. Additionally, by deselecting the checkbox, the feature can be deactivated. The "Tool tip" shows the range for the effectiveness of the Speech and Noise Management. The tool tip is activated by placing the cursor over the slide bar setting of the control in CONNEXX; when this occurs, a dB range is displayed in a pop-up indicating the potential gain reduction possibilities of the system at a particular setting.

After First Fit, a medium setting of the Speech and Noise Management is selected. This setting achieves high listening comfort and very good speech intelligibility for most wearers.

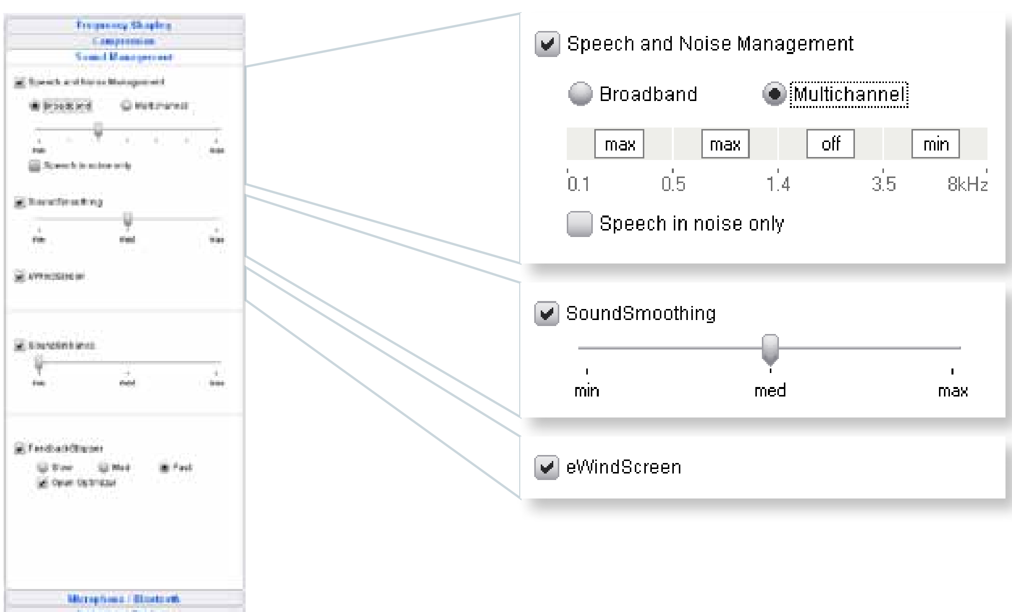

Figure 23: Fine-tuning options for BestSound Technology's noise reduction techniques.

SoundSmoothing®

There are three different settings for SoundSmoothing:

- Min: SoundSmoothing is set to -20 dB attenuation of impulsive sounds with a 60 dB SPL kneepoint.
- Med: SoundSmoothing is set to -30 dB attenuation of impulsive sounds with a 50 dB SPL kneepoint.
- Max: SoundSmoothing is set to -40 dB attenuation of impulsive sounds with a 40 dB SPL kneepoint.

"Med" is the recommended setting and also the default setting after First Fit.

eWindScreen®
Besides certain canned programs such as Music, in most programs eWindScreen is activated by default. eWindScreen can be de-activated by un-checking the corresponding box under the Sound Management Tab.

2.2.2.5 Noise suppression – How to demonstrate

The Real Time Display in the CONNEXX fitting software can be used to demonstrate the effects of noise reduction features. Under the Noise Reduction page, the fluctuating bars indicate the amount of gain reduction in each gain channel. Depending on the type of noise that is introduced to the hearing instrument, the wearer will be able to observe how appropriate gain reduction is applied. When only stationary noise is played, the hearing instrument activates a gradual channel-specific reduction in gain. This is maintained in order to ensure listening comfort while noise is present. When speech and noise are present, the bars move up and down much quicker, illustrating the effects of reducing the small pockets of noise in between words and syllables. This is to assure that speech intelligibility is maintained as well as ease of listening and listening comfort. Note that there is also some reduction in the quiet setting, as microphone noise is also attenuated in this case. To demonstrate SoundSmoothing, simply clap your hands or tap a pen against a table or glass. All the bars will instantaneously reach down to the bottom of the graph, illustrating the dramatic reduction in gain to make sure that these sounds are not too startling or annoying for the wearer. After establishing this example, now speak while tapping the pen on a hard surface; this will result in far less gain reduction which helps prove the point that SoundSmoothing does not negatively impact speech. Use a fan or blow across the microphone ports of the hearing instrument to show the effects of eWindScreen. In this case, the bars in the low frequencies will drop.

Figure 24: The Noise Reduction page of Real Time Display allows the hearing care professional to demonstrate various noise reduction algorithms. This example is for street noise, which shows the effects of both the modulation-based noise reduction and the fast-acting Wiener filter.

2.2.3 Optimizing Localization

The ability to localize where a certain sound is originating is important for many reasons, safety being one of them (e.g., approaching traffic, a car horn honking on a busy street). Knowing the location where a voice originates also assists in effective communication, as the listener is able to turn towards the speaker which will indirectly improve speech understanding through lip-reading cues. Moreover, simply locating and pairing sounds with visual stimuli in our everyday environment facilitates relaxed listening and an enhanced overall enjoyment of life.

Localization is dependent on spectral differences, interaural time differences, and interaural level differences for left versus right discrimination. Thus, synchronization of left and right hearing instruments is required to ensure good left/right discrimination (Powers and Burton 2005). For localization of elevation (above or below ear level) and for front/back differentiation, however, the determining factor is the monaural high frequency spectral cues that are shaped by the pinna. To optimize localization for all dimensions, therefore, BestSound Technology employs binaural synchronization via e2e wireless 2.0 to maintain left-right localization, and TruEar to improve front-back discrimination.

2.2.3.1 Binaural synchronization via e2e wireless® 2.0

The Siemens e2e wireless 2.0 system enables the right and left hearing instruments to work together in a harmonized way. The input obtained from both instruments is shared so that important adjustments concerning signal processing are based on this combined intelligence, allowing for symmetrical steering of functions such as digital noise reduction and directional technology. Wearer commands regarding volume control and program selection are also transmitted wirelessly between hearing instruments, maintaining symmetrical function and optimizing day-to-day individual efficiency (e.g., only one button has to be pushed to adjust both instruments). Studies with this technology have shown that synchronized microphones are required for good localization (Keidser et al. 2006) and that majority of subjects prefers linked hearing instruments in real world (Smith et al. 2008).

2.2.3.2 TruEar®

Besides binaural hearing (e.g., analysis and comparison of acoustic inputs in the brainstem), the human brain relies heavily on the acoustic characteristics of the outer ear for localization of sound, especially for elevation, and front/back localization. As a result, BTE-wearers often have more trouble correctly localizing sound sources since the microphone is positioned above the ear, rather than in the ear canal. Siemens TruEar solves this problem by replicating the unique acoustic effects of the outer ear through processing of the input signal. Figure 25 illustrates how TruEar restores pinna effects. Due to the reflections and resonant characteristics of the human outer ear, frequencies above 1.5 kHz are amplified more for sounds from the front relative to other directions. As a result, a positive directivity index is obtained for these frequencies.

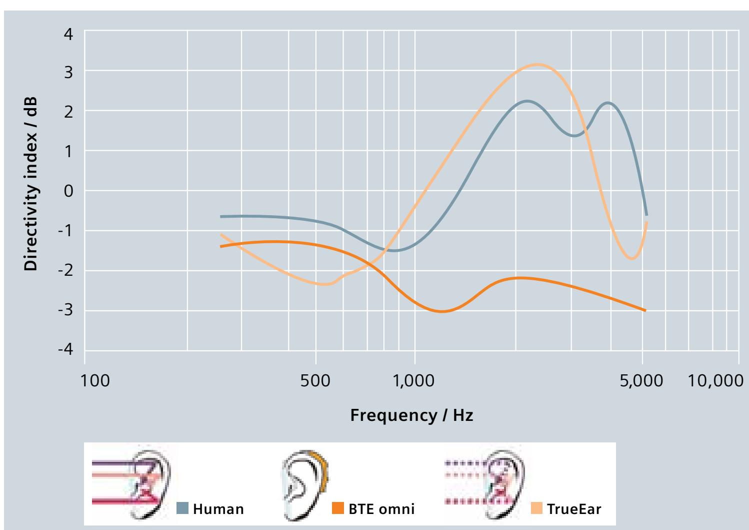

Figure 25: Directivity (as calculated by the DI) of the unaided open ear compared to a standard BTE (closed earmold) with an omnidirectional microphone and a BTE employing TruEar processing.

In order to restore aided localization with BTEs, TruEar mimics the frequency-specific directivity of the human outer ear when in the omnidirectional mode. This is accomplished by adjusting the hearing instrument's directional microphone system to match the directivity pattern of the pinna as closely as possible. As shown in Figure 25, hearing instruments equipped with TruEar provide a very similar directivity at high frequencies to the unaided ear, meaning that the brain can easily use the spectral cues available to discriminate between signals coming from the front and from the back. This benefit of TruEar has been substantiated by research. Studies conducted at the National Acoustics Laboratory in Australia (Keidser et al. 2008) show that for all signals with components above 2 kHz, TruEar significantly reduces front-back confusions.

Note that hearing instruments with a microphone located close to the ear canal entrance (such as CICs), do not require TruEar as they can use the natural pinna effects.

2.2.3.3 TruEar® & e2e wireless® in CONNEXX

The e2e wireless settings can be accessed under the Wireless Settings & Accessories page where there are options to couple the hearing instrument to remote controls, including Tek. Digital signal processing is enabled by default to allow the two instruments to synchronize in terms of digital signal processing. This coupling can be disabled if necessary.

TruEar is available in 701 and 501 series BTE instruments and is used as the default setting when the omnidirectional mode is active in the Automatic / TruEar setting.

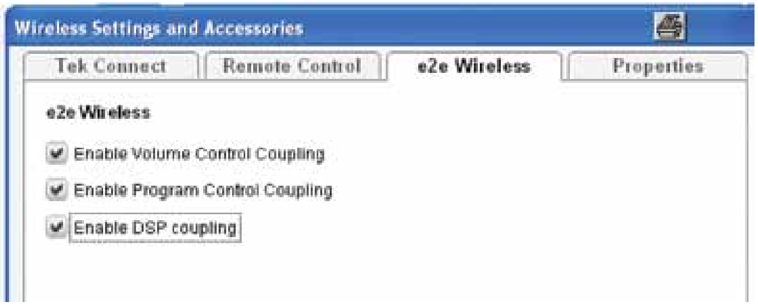

Figure 26: The Wireless Settings and Accessories page in CONNEXX allows the HCP to designate remote controls as well as configure e2e wireless 2.0 settings.

2.2.4 Extending bandwidth

2.2.4.1 SoundBrilliance™

SoundBrilliance™, another innovative algorithm from Siemens, is designed to exceed the limitations of conventional hearing instrument bandwidths to offer wearers a listening experience more similar to that of normal hearing individuals. SoundBrilliance analyzes input signals and adds artificial high frequencies up to 12 kHz to the output. The perceived effect is a more "brilliant" sound quality. SoundBrilliance is especially beneficial when listening to music, as well as when audio streaming via *Bluetooth*. This is because the bandwidth of *Bluetooth* transmission is limited to 7.5 kHz, but SoundBrilliance overcomes this limitation by adding frequencies up to 12 kHz (Figure 27).

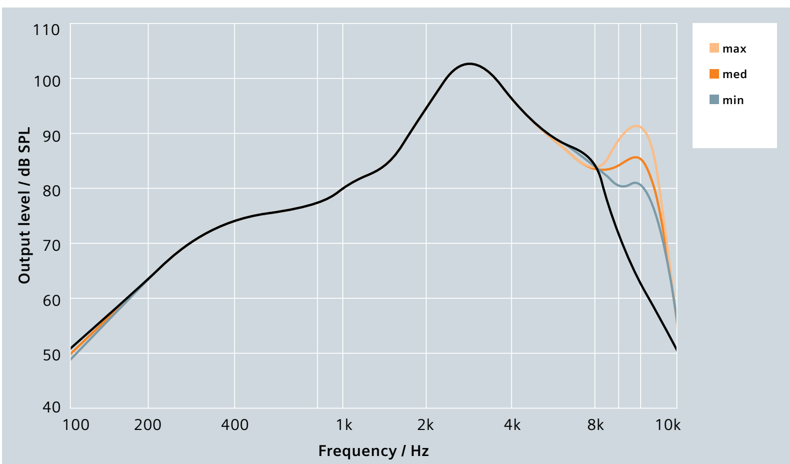

Figure 27: SoundBrilliance adds ultra-high frequency information to select hearing instruments.

Because SoundBrilliance does not operate on high-frequency components above 8 kHz from the microphone signal, but generates them from lower frequencies, these additional high frequency sounds are available without any feedback risk. With SoundBrilliance, the output in high frequencies can be increased up to 90 dB SPL in this frequency region. For individuals with hearing losses up to 85-90 dB at 8 kHz, SoundBrilliance should provide a noticeable enhancement in sound quality.

2.2.4.2 SoundBrilliance™ – Settings in CONNEXX

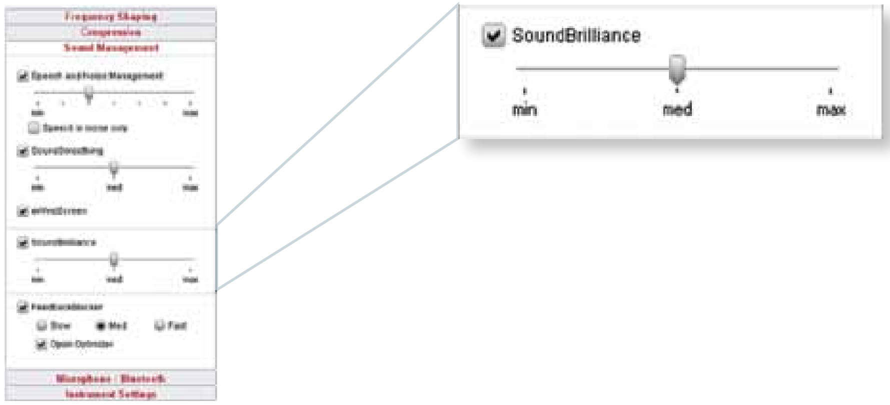

Figure 28: Fine Tuning options for SoundBrilliance.

2.2.4.3 SoundBrilliance – How to demonstrate

Set up two music programs to demonstrate SoundBrilliance, one with SoundBrilliance "on" and one with SoundBrilliance "off". Have the wearer switch between the two programs to experience the difference.

2.3 Individuality

Often times, better hearing (audibility) compromises sound comfort and vice versa. For example, increasing high frequency gain may improve speech understanding, but the wearer then complains that the sound quality is too shrill. The challenge, then, is to find the right balance between better hearing and sound comfort for every individual. This is further complicated by the fact that the best gain and output (prescriptive fitting) is only appropriate for the "average" person, and needs to be individualized as well. A learning hearing instrument that the individual can train to arrive at the preferred settings, therefore, is the ideal solution. Individual preferences, however, also vary across listening situations. A trainable hearing instrument, therefore, should learn gain, frequency response, and compression separately for different listening situations.

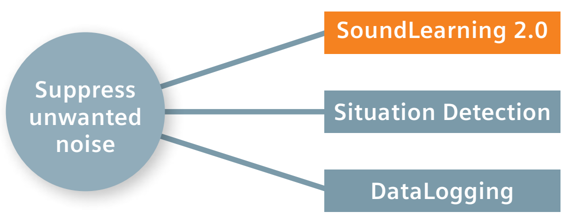

Figure 29: This figure shows the feature set for Individuality.

2.3.1.1 Situation detection

In order to learn gain and compression settings reliably for different listening situations, it is essential to be able to correctly detect such situations. Siemens introduced the first situation detector in 2002. The detected acoustic situation determined the appropriate directional microphones and noise reduction settings so the hearing instrument was always automatically adjusted to the optimal setting for each acoustic situation. In 2004, this technology was refined to become the first binaural situation detection which allowed for synchronous steering of both hearing instruments in a binaural fitting. Today, the accuracy of Siemens situation detection is still the industry benchmark. A recent study showed that Siemens excelled over the best competitor for the accurate detection of all sound categories (speech in quiet, speech in noise, noise, and music). This was most notable for the identification of music (correct identification exceeded best competitor by 44%) and for noise (correct identification exceeded best competitor by 27%). See Figure 30.

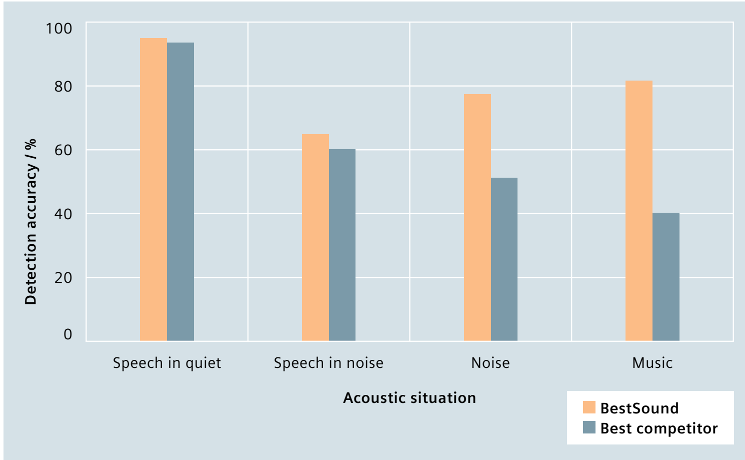

Figure 30: Compared with the best competitor, Siemens situation detection is more accurate in all sound categories, especially for the sound categories of noise and music.

2.3.1.2 Situation specific learning with SoundLearning® 2.0

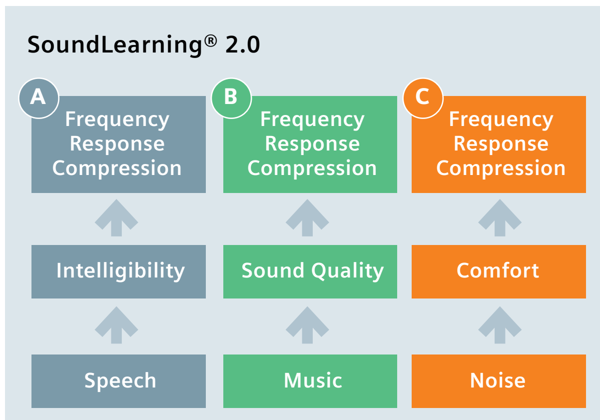

Figure 31: Situation specific listening goals require situation specific gain, frequency response, and compression.

The rationale behind situation-specific learning is that depending on the acoustical environments, wearers have individual preferences. In situations where speech is present, whether it is speech in quiet or speech in noise, the obvious priority is speech intelligibility. In noisy environments, the noise should be heard, but only at a comfortable level. For music, the sound quality is most likely the priority. It follows then, that depending whether the current listening situation is speech, noise, or music, the wearer would prefer different gain settings.

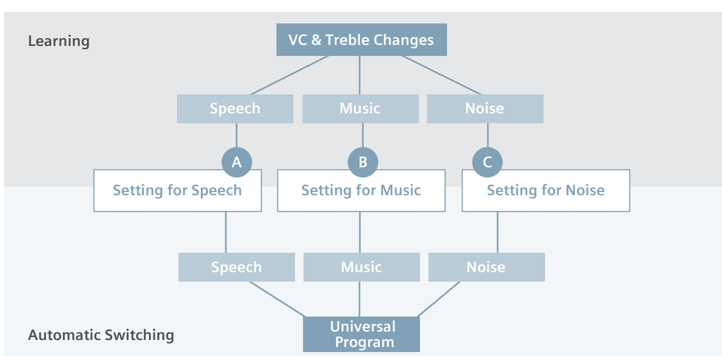

Figure 32: Situation specific learning.

In SoundLearning 2.0, every time the wearer changes the volume or SoundBalance, this adjustment is recorded along with the current input level and detected acoustic environment. This leads to a dedicated frequency response as well as compression setting for Speech, Noise, and Music. These learned new settings are then selected when the system detects these situations again. The result is a separate setting for each of these listening situations tailored to individual's preferences. To achieve the same effect with SoundLearning, three separate learning programs would be needed, one for each acoustic situation. Additionally, the individual would have to change the program manually each time the acoustic situation changes. With SoundLearning 2.0, this all occurs automatically and intuitively in program one.

In mixed situations, i.e., "music" and "speech", the activated setting is a mixture of the "music" and "speech" settings. This means that SoundLearning 2.0 is able to apply a continuum of settings rather than three single settings.

Study 1: Situation vs. time-based learning (NAL).

In any learning hearing instrument, learning can occur in two ways. In time-based learning, the hearing instrument records a data point after a specific interval, say every 1 or 5 minutes. In situation-based learning, a data point is recorded every time a specific event occurs, such as when the wearer adjusts the volume control. Results of a recent study conducted at NAL (Keidser, Convery, McLelland, 2009) indicate, that situation-based learning is more effective than time-based learning when learning is specific to different situations. SoundLearning 2.0 learns specific preferences both for different levels and situations (situation-based learning).

Based on the data from this study, it is possible to estimate the percent of situations with comfortable loudness. While former learning algorithms like DataLearning® and SoundLearning were already quite successful in increasing the percentage of situations with comfortable loudness (Mueller et al, 2008), SoundLearning 2.0 clearly outperforms the previous versions and has the potential to achieve optimum loudness in almost all listening situations.

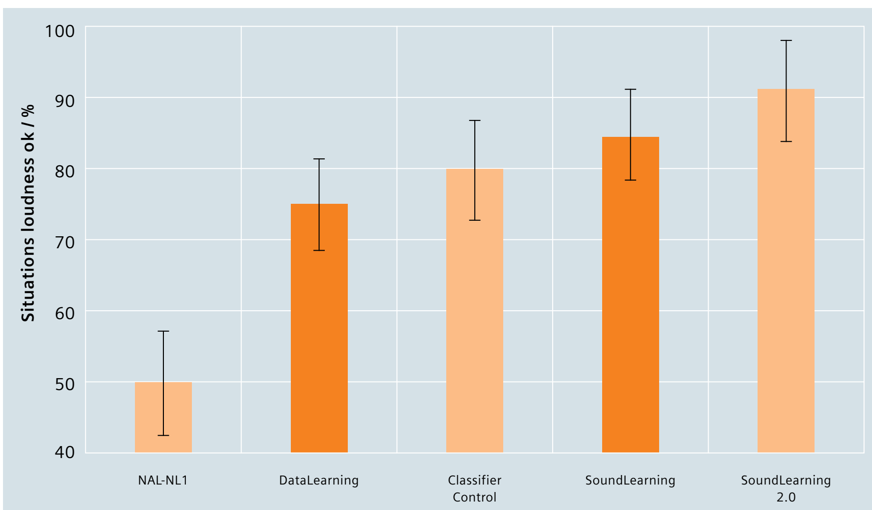

Figure 33: Percent of situations with comfortable loudness for various learning algorithms.

2.3.1.3 Managing SoundLearning 2.0 in CONNEXX

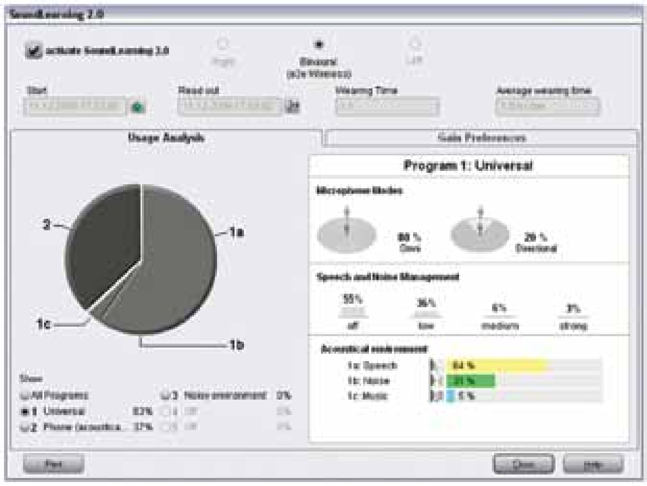

Figure 34: SoundLearning 2.0 screen under the Learning/Logging icon. 

SoundLearning 2.0 is accessible under the Learning/Logging dialog (Figure 34). Currently, SoundLearning 2.0 is only available in the Universal Program. Programs 2 and 3 support SoundLearning. When “activate SoundLearning 2.0” is checked, the Universal program automatically learns independent setting preferences for Speech, Noise, and Music. After the individual has worn the instruments for a certain time, the SoundLearning 2.0 dialog shows logged information such as wearing time, microphone, and Speech and Noise Management usage for each selected program. More interestingly, the Acoustical Environment section shows the percentage of times the wearer has been hearing speech, noise, and music. This percentage is also reflected in the Usage Analysis pie chart (Figure 34).

Click on Gain Preferences tab to see learned gain settings for each of the three situations or classes under the Universal Program. When learning has occurred for one of the three situations, an exclamation point (!) will be displayed in the respective situation on the pie chart (Figure 35). Learned values can be displayed as gain curves or as channel gain. The HCP has the option to apply learned settings automatically. If learned settings are not automatically applied, the HCP has the option to apply the learned gain to all classes, or to reset one or all three classes back to the original settings prior to the learning period. Optimal learned settings for all three classes should be arrived after approximately two weeks.

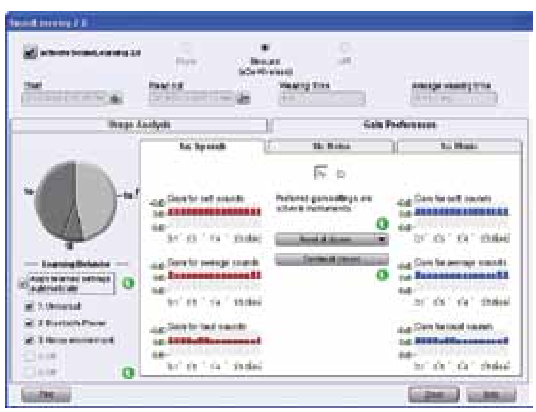

Figure 35: Gain Preferences tab in the SoundLearning 2.0 dialog. Learning has occurred for the Speech situation (1a), as indicated by the exclamation point (!). The HCP can then view the learned settings, and choose to either reset or apply them.

2.3.1.4 SoundLearning 2.0 – How to demonstrate

Since learning occurs over time, it is not a concept that can be easily demonstrated in the short amount of time allotted for most hearing instrument counseling and fitting sessions. It is possible, however, to demonstrate to customers the benefit of having three separate settings for different acoustic situations. Use the Audibility page of the Real Time Display to demonstrate the advantage of SoundLearning 2.0. Set up two programs in the hearing instrument: one as Universal, and one as Music. Have the person wear the instrument while showing the Audibility page for the Music program. Play the sound file which contains a series of speech, music, and noise. Explain that in program two, there is only one gain setting. Therefore, regardless of the sounds played, the hearing instrument responds the same way. Therefore, speech and noise will sound sub-optimal. Switch to the Universal Program and show the Audibility page again. Play the same sound file again. In the Universal program, gain settings change according to the detected acoustic situation. This is not the case in any other program. Explain to the person that depending on the sounds being played, the Universal program adjusts its settings accordingly so that speech is intelligible, music is pleasant, and noise is not uncomfortable. These adjustments are also reflected in the gain channels on the Audibility page.

Once the wearer understands the advantages of having different settings for different acoustic situations, explain that the settings currently in the instrument are presets or starting points. When the individual uses the hearing instruments on a daily basis, the hearing instruments will automatically adapt these presets to match his or her individual preferences for these different listening situations!

3 Portfolio with BestSound Technology

Figure 36 shows which technology is available in different performance levels.

			101	301	501	701
Better Hearing	Audibility	AGCi	6/6	8/8	12/12	16/16
Better Hearing	Audibility	AGCo	1/1	1/1	12/4	16/4
Better Hearing	Speech intelligibility	Directional microphone	Automatic	Auto/Adaptive	Auto/4-ch. adap.	Auto/4-ch. adap.
Better Hearing	Speech intelligibility	SpeechFocus	–	–	–	Yes
Better Hearing	Speech intelligibility	Soft Level Directivity	–	Yes	Yes	Yes
Better Hearing	Speech intelligibility	e2e wireless	–	Yes	Yes	Yes
Sound Comfort	Eliminate feedback	FeedbackStopper	Yes	Yes	Yes	Yes
Sound Comfort	Suppress unwanted noise	SoundSmoothing	–	Yes (on/off)	Yes (off + 3 steps)	Yes (off + 3 steps)
Sound Comfort	Suppress unwanted noise	S. and N. Management	Yes (off + 3 steps)	Yes (off + 3 steps)	Yes (off + 5 steps)	Yes (off + 7 steps)
Sound Comfort	Suppress unwanted noise	eWindScreen	Yes (on/off)	Yes (on/off)	Yes (on/off)	Yes (on/off)
Sound Comfort	Optimize localization	TrueEar	–	–	Yes	Yes
Sound Comfort	Optimize localization	e2e wireless	–	Yes	Yes	Yes
Sound Comfort	Extend bandwidth	SoundBrilliance	–	–	Yes	Yes
Individuality	Learning and Logging	Learning	–	DL	SL	SL
Individuality	Learning and Logging	Situation Detection	Yes	Yes	Yes	Yes
Individuality	Learning and Logging	Logging	Yes	Yes	Yes	Yes

Figure 36: The availability different technologies across four performance levels.

4 Best Fitting

BestSound Technology improves speech intelligibility and hearing comfort while accounting for individual needs and unique listening situations. Despite the sophistication of BestSound Technology, Siemens wants to ensure that fitting these hearing instruments is easier and more intuitive than ever. Therefore, Siemens also offers the Best Fitting Support with our wireless programming interface ConnexxLink, NAL-NL2 as a fitting formula, and a host of other features to facilitate every step of the fitting process.

4.1 ConnexxLink: Wireless programming interface

ConnexxLink, Siemens completely wireless programming interface, is launched together with the BestSound Technology portfolio. Simply plug in the pre-paired dongle into the USB port of your PC, and have your patient wear ConnexxLink around the neck. ConnexxLink employs *Bluetooth* technology for the PC to communicate instantaneously and wirelessly with select Siemens x01 hearing instruments, so that Hi-Pros, programming cables, and adapters are no longer needed. Not only will ConnexxLink facilitate ease of programming for hearing care professionals, it will also provide customers a greater degree of freedom and comfort during the fitting process.

Figure 38 compares the programming time of three commercially available wireless programmers for typical tasks during a fitting session. Except for two of the selected situations, ConnexxLink is the fastest system available.

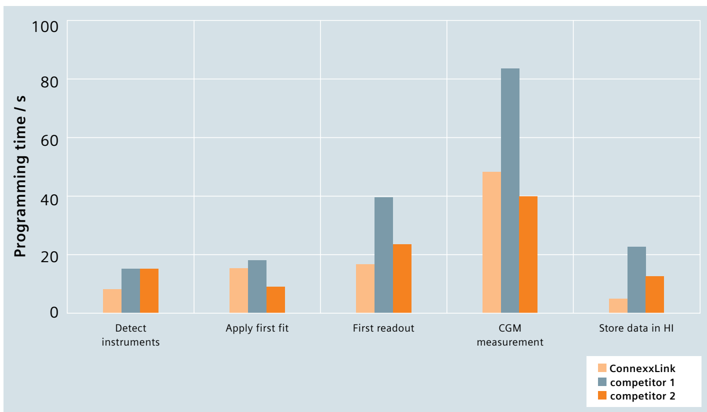

Figure 38: Programming time of three wireless programming interfaces for various fitting tasks.

ConnexxLink is compatible with all hearing instruments featuring the new BestSound Technology on 701, 501, and 301 performance levels, as well as all custom instruments with e2e wireless 2.0.

4.2 Prescriptive fitting: NAL-NL2 & ConnexxFit

Siemens CONNEXX 6.4 is the first fitting platform that supports the NAL-NL2 fitting formula. Like its renowned predecessor NAL-NL1, NAL-NL2 is an extensively researched and proven formula developed by the National Acoustic Laboratory in Australia. Unlike NAL-NL1, however, this version of the prescriptive formula takes more individual factors into account, including age, gender, hearing instrument experience, and language.

In addition to NAL-NL2 as a separate fitting option, ConnexxFit, Siemens proprietary fitting formula, is now also based on NAL-NL2. ConnexxFit makes modifications to NAL-NL2 based on instrument specific properties and individual ear acoustics to optimize spontaneous acceptance. Various studies have shown that ConnexxFit is preferred by most wearers without degradation of speech intelligibility (Chalupper, Herbig, 2009).

Achieving the optimal hearing instrument setting for each individual customer is a multi-step process. First of all, a proven fitting formula which takes as many individual factors into account as possible should be the basis of any gain and output prescription. For this step, NAL-NL2 is the ideal base formula. In the next step, ConnexxFit takes specific Siemens hearing instruments and individual ear acoustics into account to result in a First Fit that offers appropriate amplification and supports spontaneous acceptance at the same time. After the patient has worn the instruments for a few weeks, the fine tuning adjustments made by a hearing care professional together with SoundLearning 2.0 offer a personalized fit that is most appropriate for the individual. Due to the sensory deprivation caused by hearing loss, which usually occurs over the course of decades, new hearing instrument wearers often only accept gain that is not quite optimized for audibility and speech intelligibility, even after a few weeks of amplification. In order for the wearers to acclimate to amplification and to “re-train” their brains to make the most of the sounds now being heard, they need active auditory training (Taylor 2008, Hawkins 2005). For this final step in the fitting process, Siemens offers eARena®, a computer-based auditory training program. The fitting process is illustrated in Figure 39.

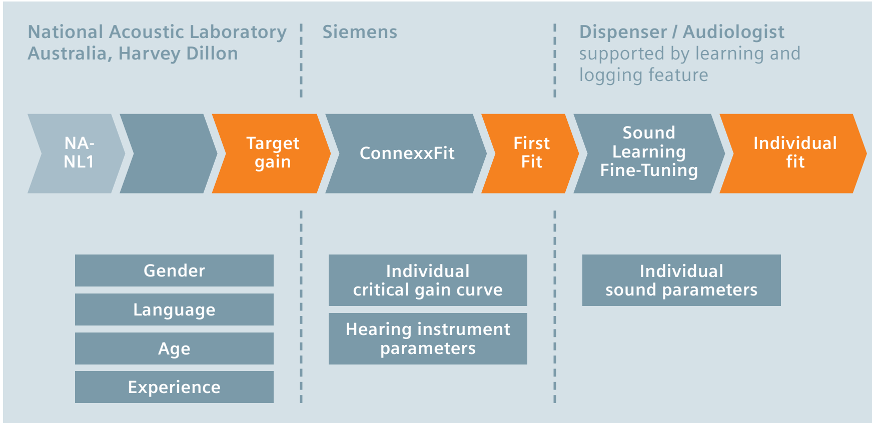

Figure 39: Illustration of how Siemens fitting and support tools help create an individual fit that is tailored to each individual.

4.3 AutoFit: Match to Target with a Click of a Button

While most hearing care professionals adjust a hearing instrument's gain and output settings to individual preferences, many still prefer, or are required to prescribe gain settings according to traditional fitting formulas such as DSL or NAL-NL1. In the latter case, the HCP typically performs real ear measurements and adjusts the hearing instrument gain until they match the targets prescribed by these formulas. This can often be a tedious and time consuming process where multiple measurements have to be made as the gain for low, medium, and high inputs are slowly adjusted to approximate targets.

Siemens AutoFit is an option within the CONNEXX in-situ feature which automatically adjusts the frequency response of a hearing instrument to match prescriptive gain targets at the click of a button (Figure 40). AutoFit works by directly utilizing the measurement systems - Siemens Unity 1, Unity 2, or Aurical - to carry out real ear measurements. This data is then used by CONNEXX to automatically adjust the hearing instrument frequency response to match specified prescriptive targets in the ear. This allows the HCP to perform in-situ measurements without initializing the measurement software separately, and thereby significantly reduces time of fitting hearing instruments to prescriptive targets.

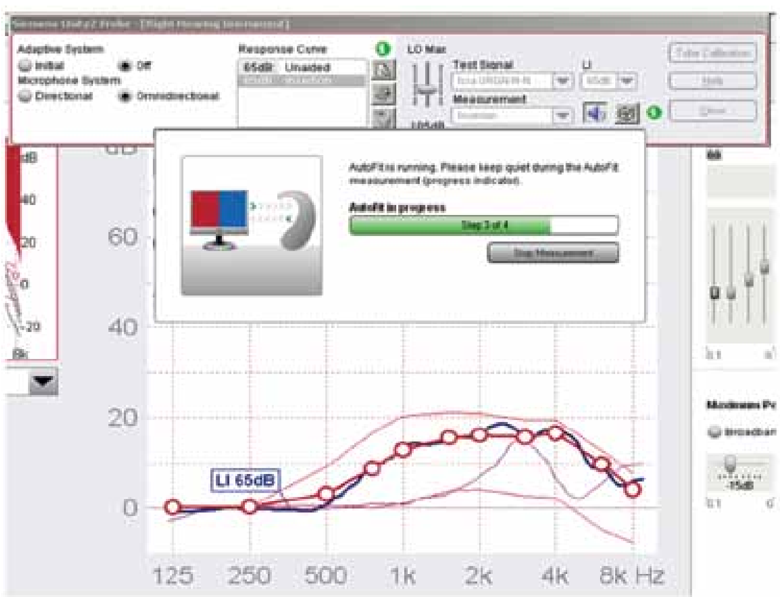

Figure 40: The AutoFit dialog in CONNEXX.

An internal study was carried out to evaluate the efficiency of AutoFit in real clinical situations. In testing of 10 ears and 5 audiograms of various configurations, the study compared the time, as well as the number of mouse-clicks it took for an experienced HCP familiar with CONNEXX to perform in-situ fittings via manual adjustments and via AutoFit. Results show that AutoFit significantly reduces the time and the number of mouse-clicks without sacrificing the accuracy of the match to prescriptive targets (Figure 41).

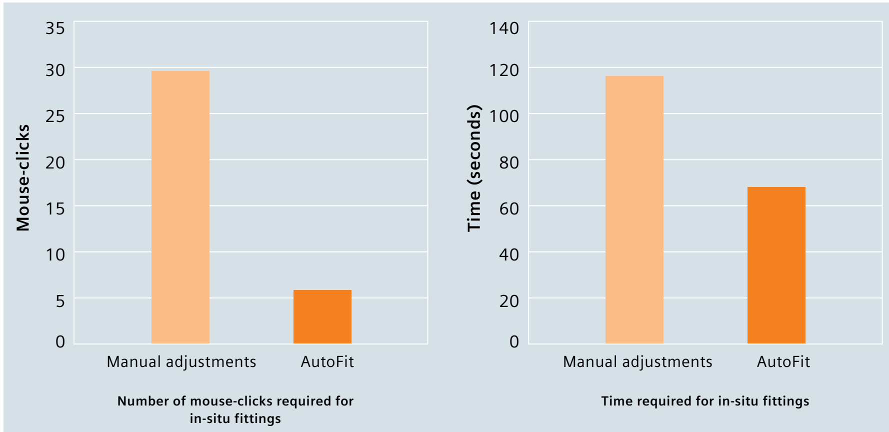

Figure 41: Number of mouse-clicks and time required for an experienced HCP to complete In-Situ Fittings via manual adjustments and via AutoFit.

4.4 Counseling & Training

4.4.1 Real Time Display

Real Time Display is a counseling tool within the CONNEXX fitting software. This feature enables HCPs to demonstrate key hearing instrument features visually and acoustically. The Real Time Display is divided into four tabs featuring Audibility, Noise Reduction, SpeechFocus, and Adaptive Directional Microphone. The Audibility page shows the instantaneous input and output of the 16 gain channels as well as the detected acoustic situation. This information is shown relative to the range of audibility so that it can be used to help individuals with hearing loss and family members understand what the hearing instrument does in terms of amplifying speech, and why amplification is necessary given the patient's hearing loss. The Noise Reduction page shows the ability of the instrument to reduce gain for noise in 16 channels. The effects of the various noise reduction features in the hearing instrument, including SoundSmoothing can be demonstrated. The SpeechFocus page shows how 701 hearing instruments can detect speech from all around the wearer and activate the appropriate microphone mode accordingly. The Adaptive Directional Microphone page shows the ability of the adaptive directional microphone to suppress moving noise sources that come from the rear hemisphere.

4.4.2 Hearing Loss Simulator

The Hearing Loss Simulator is a counseling tool within the CONNEXX fitting software designed to help families and significant others to understand the impact of the patient's hearing loss (Figure 42). By plotting the patient's hearing loss on the audiogram, the hearing care provider can explain how the hearing loss affects speech and environmental sounds. Even more poignantly, sound files that are provided allow others to hear "with" the patient's hearing loss. When family and significant others truly understand the detrimental effects of hearing loss, they can become more involved and supportive in obtaining hearing instruments for the patient.

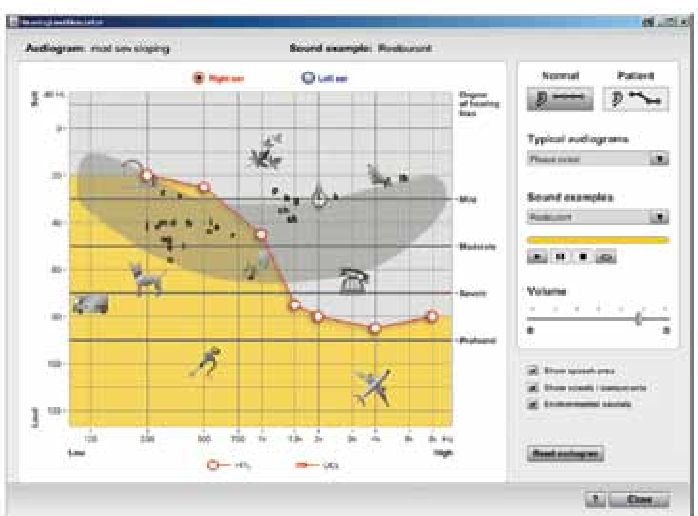

Figure 42: The Hearing Loss Simulator allows friends and family members to better experience hearing loss.

4.4.3 eARena®

For most people, hearing instruments are very effective in restoring audibility. Even the most advanced technology, however, is not able to restore listening, comprehension, cognitive, and communication skills. Since hearing loss usually occurs gradually over a long period of time, sensory deprivation often causes the brain to "forget" what it is like to hear all the sounds in the environment. This explains why the brain is not always able to efficiently make use of the restored audibility following initial amplification. Siemens eARena is a computer-based auditory training program specifically designed to help individuals with hearing loss make the most of their hearing instruments (Figure 43). eARena offers a 20-day program with 30-minute auditory exercises each day. The exercises are designed to improve auditory skills such as perception of everyday sounds, word recognition, timbre discrimination, and speech understanding in noise. In addition to being highly beneficial for the consumer, eARena also offers benefits to the HCP, as it increases patient's satisfaction with new hearing instruments and fosters loyalty.

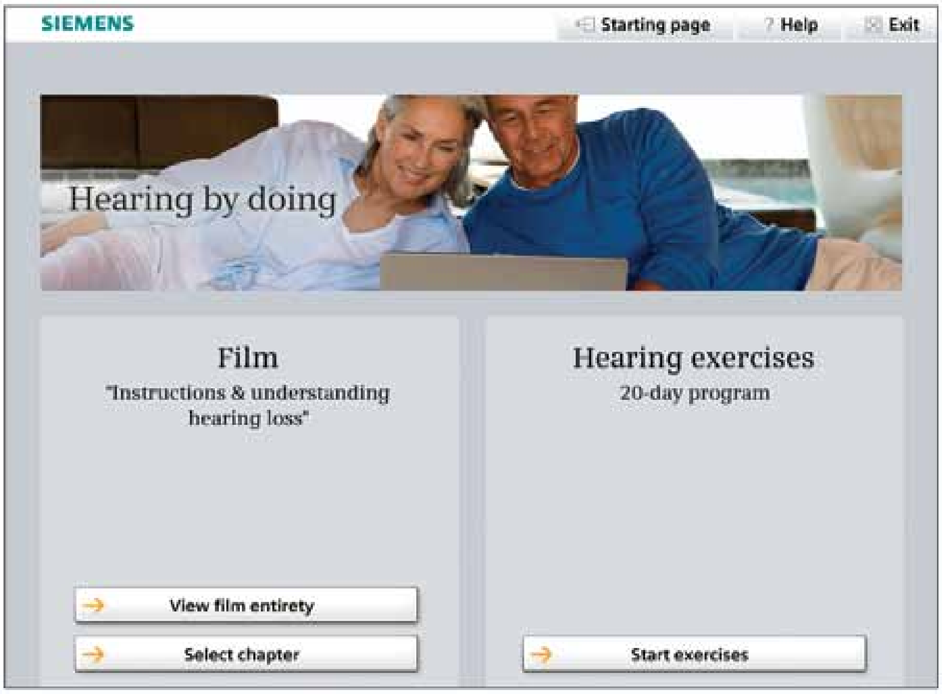

Figure 43: Siemens eARena auditory training program helps individuals to acclimatize to new hearing instruments.

5 Abbreviations

- AGC-I Automatic Gain Control Input related
- AGC-O Automatic Gain Control Output related
- SNR Signal to Noise Ratio
- UCL Uncomfortable level
- HCP Hearing Care Professional

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