Siemens offers a broad range of hearing instruments for every requirement and all configurations of hearing loss. Your customer has a wide choice between many different models and performance levels, from the tiny CIC hearing instrument, to the powerful BTE system. To facilitate orientation and help selecting the right instrument, two device segments were created, “Discreet” and “Comfort”. This way, you can find out quickly which model best suits which customer.

Discreet
The user of the Discreet segments is not really comfortable wearing a hearing instrument. It is often their first contact with one. A hearing instrument is often seen as a personality impairment. It should therefore be as small as possible, almost invisible, or simply not have the appearance of a hearing instrument.

Comfort
The customers requesting the Comfort segments are more at ease with and have a positive attitude toward hearing instruments. Acceptance of a hearing instrument is a lot higher than with clients opting for discreet hearing instruments. For these individuals, a hearing instrument is often a necessary means to continue having an active social life.

Performance
Users out of both segments with a high affinity toward technology and connectivity: the needs of the “Performance” user are addressed with the range of wireless accessories.

Siemens Life micon
The much appreciated Siemens Life now comes with micon, the new platform behind BestSound Technology, to offer the best possible sound comfort in a chic design from the very start. As the smallest standard BTE in the Siemens product range, it is the ideal alternative if a RIC hearing instrument is not an option.

Ace micon
The brand new Ace micon is the most discreet RIC BTE solution in the industry. Equipped with micon, the new platform behind BestSound Technology, it skillfully offers the most discretion and best possible sound comfort from the very start.

Pure micon
An addition to the Pure™ family: Pure micon, the fully featured RIC offers numerous key features in one tiny package making it the most versatile discreet RIC in the industry. Equipped with micon, the new platform behind BestSound Technology, Pure micon offers outstanding sound comfort and comes in a great new design.

Aquaris micon
Aquaris micon is the truly water- and dustproof hearing instrument from Siemens – the first instrument of its kind tested to the IP68 standard. And now, with 65 dB of gain and micon, the new platform behind BestSound Technology, this ultra robust solution offers more power than ever.
Natural sound

The wonders of our ear

Over millions of years, nature has perfected the human auditory system for communication. The intricacy and efficiency of this marvelous system continues to amaze us today. For example, the critical bands on the basilar membrane allow us to discriminate sounds. The 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 ever-increasing life expectancy, more and more people are experiencing a decline in hearing capabilities.

“You don’t hear a croak, you hear a frog.” The French-American composer Edgar Varese is quoted as the source of this wonderful statement. It reminds us of how we listen. We do not first observe a croak coming from out there near the creek and then deduce that there must most likely be a frog there that made that croak. No, we immediately interpret the sound as evidence of a frog. So we actually hear the frog, not the croak. The interesting thing is what is out there (the message), not the registration of data we obtain from it (the medium). Good hearing means that you hear what you hear, not that you hear. Bad hearing means that you hear badly, not what you hear badly.
So what then if someone in fact has a hearing loss? What is it that they want? They want to hear, not to notice that they have bad hearing. So if they wear hearing instruments, do they want to be aware of the qualities of those hearing instruments? No. They want to be aware of what the hearing instruments allow them to hear. They want to hear with them, not hear them.

This is the challenge for hearing instrument manufacturers: if the hearing instruments function perfectly, the wearer will never notice them. However, if they do not function as well as the ears of a normal-hearing person, they will be reminded of them all the time.

When hearing instrument fail, the wearer will notice it; and in many cases the failure is because they cannot do what the ears and the brain of a person with normal hearing can do. The normal ear and brain do an outstanding job of collecting, processing and interpreting incoming data, to make this data meaningful. We experience a situation in the outside world: we select what is relevant, and we filter out what is not relevant. We process the incoming sound into intelligible language or sweet music. The criterion of success is that we do not notice this process at all. Hearing instruments must do the same, or the wearer will notice that they do not.

Back in 1878, Werner von Siemens, the founder of Siemens, invented a special telephone receiver for his hearing-impaired wife. Ever since then, we at Siemens have continued to develop innovative solutions to help the hearing-impaired. A recent major innovation in hearing solutions was the introduction of BestSound TechnologyTM in 2010. Now, Siemens introduces microm™, the new platform behind BestSound Technology. With this platform, the engineers and scientists did an outstanding job in making it easy for the wearer to forget the process and enjoy the overall result. The time has come for people to forget that they are wearing hearing instruments and to start paying attention to the world instead.

The new platform microm delivers a processing power 6,000 times greater than what was used to put the first man on the moon. This power expands Siemens hearing technology to 48 channels and extends the frequency range to 12,000 Hz. The result is unprecedented clarity and listening comfort, taking Soundability™, the perfect balance between Sound Quality and Audibility, to a whole new level.

Three key features underpin microm:

- **miSound™** provides the best amplified sound while preserving natural acoustics. Wearers will fall in love at first fit with the perfect balance between Sound Quality and Audibility, Siemens’ Soundability.

- **miFocus™** eases listening effort even in demanding environments. Improved directionality means less strain listening, and more readiness to engage and interact.

- **miGuide™** provides preferred sound in any situation and supports acclimatization to amplification. With less manual settings, wearers will stay in love with their intuitive hearing instruments.

With these breakthrough innovations and over twice the processing power of previous platforms, microm combines all industry-leading features, delivered simultaneously, without any trade-offs (Figure 1).
A new engine for BestSound Technology

micon, the new platform behind BestSound Technology, preserves all the fine and fragile acoustic cues in the ever-changing soundscape and brings wearers what they want to hear, how they want to hear it. It is designed to be forgotten, so that fine and subtle acoustic details make life sound brilliant again. micon is powered by a completely new developed digital signal processing platform. As mentioned before, our scientists and engineers are committed to continuous improvements and innovations in hearing instrument technology. Their efforts have paid off once again as we take our proven BestSound Technology to the next level.

Around the globe, Siemens BestSound Technology has proven successful; users have reported excellent sound quality and comfort. micon was designed with this success in mind and brings Soundability to a whole new level: restoring audibility while maintaining excellent sound quality to deliver the most natural listening experience for every individual wearer. The new platform allows for 18 million transistors, executing 250 million instructions per second. In addition there is room for 200 times more memory, 48 channels and a 12-kHz bandwidth, enabling fine-tuning with unrivaled flexibility. Most importantly, micon delivers a listening experience to be just that: listening to the world.

micon allows for unparalleled digital signal processing in 48 channels. These 48 channels each have a bandwidth of 250 Hz (very-high-frequency resolution) and offer unprecedented flexibility in frequency shaping, compression mapping, digital noise reduction and directional microphone computation. Compression, however, is physiologically observed in the normal ear, and therefore should be treated with as much psychophysical motivation as possible. As a result, for compression, the 48 channels coming from the signal processing unit are bundled into 20 handles (a grouping of adjacent channels), analogous to the critical bands on the basilar membrane.

micon is incorporated in a large variety of hearing instrument models to provide an optimum solution for each individual in terms of hearing loss, cosmetics and price.
We all absorb millions of bits of information through our eyes and ears without being aware of the process. All we are aware of is the scenery we see, the music we listen to and the conversation we take part in. It is no wonder then that extremely sophisticated audiologic design and advanced signal processing strategies are required to recreate audibility, while maintaining excellent sound quality and preserving fragile acoustic cues that enable spatial perception.

miSound ensures that the listening experience with hearing instruments is as natural as possible. With this system, we treat every detail individually, in order to make sure everything sounds exactly how we expect it to sound. miSound features a tremendous new amplification system to map the acoustic world into the limited dynamic range of the wearer, providing an excellent sound quality and maintaining the richness of surrounding soundscapes.

miSound offers a fully transparent hearing solution, allowing the most desired and appropriate bandwidth up to 12 kHz; and due to the enhanced feedback cancellation algorithm, acoustic feedback and distortions are avoided even in the most critical situations.
Because first impressions are so important, a great first fit must ensure optimal spontaneous acceptance. The validity and effectiveness of traditional prescriptive formulas such as NAL-NL1 for maximizing speech intelligibility and restored audibility are well established. These formulas, however, were not specifically designed for facilitating spontaneous acceptance of amplification. While they may be ideal for children and more experienced hearing instrument users, maximized speech intelligibility and restored audibility are pointless if an individual being fit with hearing instruments for the first time finds them too loud and shrill to wear comfortably.

As an alternative to the traditional fitting formulas, the goal of micon fit, Siemens’ new proprietary fitting formula, is optimum spontaneous acceptance. Maximizing spontaneous acceptance means finding the right balance between speech intelligibility and sound comfort, in a word, Soundability. micon fit does this by building upon the success of its predecessor XCEL-Fit. And by capitalizing on the advantages offered by the 48-channel filter bank, the extended bandwidth and the new noise compression system, the new micon fit offers the most natural sound quality without compromising speech intelligibility.

Following the same strategy as XCEL-Fit, micon fit is based on some of the same fundamental principles applied to NAL-NL2, and employs the same psychoacoustic models for effective audibility and sound quality. Effective audibility predicts the amount of audibility which contributes to individual speech intelligibility based on average hearing loss values (Ching et al., 1998). Because of the function of the impaired cochlea, the sensation level (SL) required for effective audibility for individuals with hearing impairment decreases with increasing hearing loss. This means, for those with hearing loss, providing gain past effective audibility may only compromise sound quality without contributing to speech intelligibility. micon fit incorporates the concept of effective audibility to generate gain targets which eliminate unnecessary gain which may not contribute to speech intelligibility, but could reduce sound quality. On the other hand, Siemens’ proprietary model for sound quality is derived primarily from two other psychoacoustic models: the Speech Intelligibility Index (SII; ANSI, 1997) and the Model of Comfort for Hearing Impaired – Speech (MCHI-S; Goenner & Haubold, 2009). By combining the results from these two models, the Siemens psychoacoustic model is able to ensure fittings which provide an excellent sound quality without undermining speech intelligibility.

Thanks to the powerful micon chip, this Siemens fitting formula is more sophisticated than ever before. Not only are gain targets provided for 48 channels, it is developed to provide amplification for up to 12 kHz to create an even more natural and brilliant sound quality. On the other hand, for those individuals whose audiometric data indicates that they may not benefit as much from traditional means of amplification, micon fit defaults to fitting targets as defined by micon frequency compression to provide audibility of important high-frequency signals (see pages 36 to 38). This is another way micon fit takes individual requirements into account and maximizes sound quality and speech intelligibility accordingly. And finally, micon fit is made possible by the new micon compression system.
micon compression system

Like its predecessor XCEL-Amp, the micon compression system's goal is to ensure superb sound quality and speech intelligibility in order to optimize spontaneous acceptance. micon, however, builds upon the proven features of XCEL-Amp and leverages the power of the micon chip to do this better than ever before. Specifically, this is achieved by preserving the dynamics of speech and the acoustic environment to sound as naturally as we have learned it over our lifetime. In other words, we need to activate compression as seldom as possible, but still map speech and environmental sounds into the reduced dynamic range of the wearer. To do this, we turned again to the functioning of a healthy cochlea for inspiration.

In a healthy cochlea, the outer hair cells have a linear response for the lower-level input, a compressed response to mid-level input, and again a linear response to higher-level input signals. The micon compression system employs two kneepoints and can mimic this behavior of the outer hair cells (Figure 2).

Figure 2
The micon compression input-output function (upper panel) is inspired by the relative response of outer hair cells in a healthy cochlea (lower panel).
While compression knee points and compression ratios strongly influence the perception of sound quality, our impression of what defines a pleasant sound is also influenced by the degree of loudness fluctuations in the incoming signal. If the fluctuations are smaller, the signal is considered more fluid and enjoyable; whereas if the loudness fluctuations are bigger and more drastic, the signal is more likely to be perceived as jarring or unpleasant. This effect is also frequency-dependent. With this in mind, we employ adaptive time constants in the micon compression system. This means in general, when the level differences in the incoming signal are smaller, the time constants controlling compression are slow and the signal is relatively uncompressed. This is because slow time constants do not significantly affect the envelope of short-term events in the input signal, and gain is adjusted according to only the long-term input level of the signal (Figure 3). As a result, a natural sound quality is preserved. In terms of speech, this implementation ensures that the smaller but rapid level changes within speech are not compressed. And in terms of music, slow compression preserves the envelope as well as the rhythmic structure of music, so that listening to music is as pleasurable an experience as it should be.

When the level differences in the incoming signal are larger, then the system reacts much faster and the attack and release time decrease. The advantage of such a fast compression system ensures that relatively loud sounds with a sudden onset, such as when someone laughs or when the wearer starts speaking in a quiet situation, are compressed in time before they are too loud for the wearer. And in between the two extremes, micon compression finds the appropriate time constants for the range of moderate level changes as well. In short, the micon compression system calculates in real time the “pleasantness” of the incoming signal in each frequency channel and intelligently combines the advantages of both fast and slow compression in order to generate a sound that is as natural and yet as comfortable as possible.
Managing gain and compression

Depending on the model selected, Siemens hearing instruments offer up to 20 channels of compression. The Siemens First Fit algorithm automatically sets the channel gains and maximum power output (MPO), as well as compression kneepoints and ratios to achieve the best possible match to the selected fitting formula for a wide range of input levels. For the most efficient and intuitive frequency shaping adjustments, hearing care professionals can adjust the gain for level input (LI) 50, 65, and 80, which are representative for soft, medium, and loud input levels (Figure 4). Such adjustments trigger the automatic recalculation of the required compression kneepoint and compression ratio settings. For reference, the hearing care professional can also click on the compression (CK, CR, CM) panel to see the exact CK and CR values when necessary. To make the fine-tuning process more efficient, channels can be combined into handles.

There was a time when feedback whistling was a constant reminder that an individual was wearing hearing instruments. That’s the past. The micon feedback cancellation, a new enhanced algorithm, is so efficient that we can offer a new brilliant quality of life without any “reminders.”

Acoustic feedback occurs when the amplified sound delivered into the ear canal leaks out, is picked up by the microphone, and is in turn amplified again. The route which the amplified signal travels back to the microphone is called the feedback path. The goal, therefore, is to break this feedback path. Phase cancellation is the only method that eliminates feedback without reducing gain. To accomplish this, the external feedback path must be continuously estimated and an adaptive filter must be adjusted accordingly. During this adaptation process, artifacts can occur if external tonal signals (e.g., microwave beeps) are mistaken for feedback. And typically, the faster the adaptation, the more likely it is to generate artifacts. Therefore, there is a general compromise between effective feedback suppression and natural artifact-free sound quality. An extraordinary feedback cancellation system, then, not only needs to be able to adapt to feedback instantaneously before it becomes noticeable to the wearer, it also needs to do so with the efficiency and accuracy to ensure an excellent artifact-free sound quality. This is precisely what the micon feedback cancellation is able to do.
The effectiveness of micon feedback cancellation is made possible by a combination of existing algorithms that have been improved by the new possibilities and flexibilities offered by the micon platform, as well as novel strategies that further enhances its performance. Namely, micon feedback cancellation still relies upon the proven acoustic fingerprint technology as well as frequency shifting to detect and eliminate feedback. Acoustic fingerprint marks the outgoing signal to differentiate it from the environmental signal and helps to improve the accuracy of feedback detection. Frequency shifting, on the other hand, moves the entire output signal by a few Hertz. This way, the frequency-shifted signal becomes less similar to the input signal and the likelihood of misadaptations is further reduced.

In addition, the new micon feedback cancellation is even more efficient in estimating the feedback path than before thanks to two new strategies. First, it is important to point out that the acoustic feedback path in hearing instruments is defined between the receiver and the microphone. However, most modern hearing instruments are equipped with two microphones, and these microphones are combined when the directional microphone system is employed. If only one feedback path is estimated to drive the phase cancellation, any change of the directivity pattern of the directional microphone system will need to be considered by the feedback cancellation algorithm as well as external changes of the feedback path, such as putting on a hat or leaning back against a couch. The new feedback cancellation therefore employs parallel processing of two separate phase cancellers to estimate the feedback path for each microphone separately, and in parallel (Figure 5). This way, the robustness of the feedback detection and cancellation is greatly improved.

A critical gain measurement (CGM) can estimate the feedback path in a hearing instrument fitting when the wearer is presumably sitting still without any objects close to the head. This is, therefore, the best basis when starting a new fitting. However, to address the changes of the feedback paths during the day, the new micon feedback cancellation includes yet another innovation. It continuously performs critical gain estimates in the background. Unlike a regular critical gain measurement, however, it does so without any measurement signals generated by the hearing instrument. In this way, the adapters of the two parallel feedback paths can make more “educated guesses” in terms of how to adjust themselves appropriately when there are sudden changes in the feedback path. This again increases the robustness of the feedback cancellation control system and thereby contributes to the fastest feedback cancellation without introducing artifacts.
Managing feedback cancellation

As shown in Figure 6, feedback cancellation can be set at two levels:

**Slow setting:** Feedback cancellation optimized for excellent sound quality in fittings where feedback is less likely, such as in a more closed fitting. This is the default setting for closed fittings, and stable open fittings.

**Turbo setting:** Feedback cancellation optimized for excellent sound quality in open fittings where feedback is more likely.

In order for users to wear hearing instruments on a regular basis, they need to be able to forget that they are wearing them. miSound ensures that the listening experience with hearing instruments is as natural as possible. micon fit allows for transparent amplification within the dynamic range of the wearer and ultrahigh-frequency listening. In addition, micon feedback cancellation provides the most efficient protection against acoustic feedback, even in the most critical situations.

Simply stated, miSound provides excellent sound quality while maintaining the richness of sounds in the world, helping the wearer to hear what they want to hear in a pleasant and natural way.
In order to follow a conversation in a crowded room or to distinguish the direction of various sound sources in rush-hour traffic, we need to be able to precisely analyze and localize incoming sounds. The micon state-of-the-art automatic microphone system offers unparalleled directivity provided by a 48-channel adaptive system. This is intelligently controlled by communication between the bilateral hearing instruments using e2e wireless™ 2.0, which also ensures optimal localization whenever directionality is not required.
With the new innovative directional speech enhancement algorithm that incorporates the advantages of both adaptive directional microphones and digital filters, micon is able to suppress all types of noise spectrums from multiple directions simultaneously, even if these are diffuse and speech-based. Furthermore, SpeechFocus allows for the best speech intelligibility in noise, even when speech is not originating from the front of the listener. For individuals whose audiograms indicate a frequency region where the hearing loss is too severe to respond to traditional amplification, micon refocuses that portion of the input signal to the regions of the cochlea which are more intact. micon frequency compression follows research-based recommendations to ensure that it is applied only to those individuals where it can potentially provide a benefit.

All these features are combined into one powerful automatic system to provide the best audibility with the least listening effort even in the most challenging listening situations, so the wearer can focus on what they hear, not how they hear.

Today’s commercially available hearing instruments use two omnidirectional microphones to generate the acoustic properties of a directional microphone. With this approach, it is possible to achieve a single notch (null) in each hemisphere by applying an internal delay to one microphone (i.e., sounds from one direction can be fully suppressed). By varying this internal delay, the direction of this notch can be adjusted and different directivity patterns can be created (Figure 7). While these directivity patterns look nearly perfect when tested in free-field conditions, as soon as these microphones are worn on the ear, their effectiveness decreases. This is due to the fact that sound waves bend around the head depending on their frequency and therefore arrive at the microphone ports at different times. As a result, the directional characteristics of the microphones in wearing position are also frequency-specific, and the directivity can be worse than that observed in the free field.

In order to maximize directivity for all frequencies, microphone systems need to be fine-tuned as frequency-specific as possible. The more channels are available, the better this effect is realized. And this is exactly what we achieve with micon. The new 48-channel adaptive TwinMic System™ incorporates a high-frequency resolution which optimizes the effectiveness of directional microphone technology in wearing conditions.
Almost every individual with hearing loss reports problems with listening to speech in background noise (Kochkin, 2010). In the presence of noise, not only does speech become harder to understand, even the act of trying to listen and concentrate becomes much more tiresome. It was shown by Bentler et al. (2008) that digital noise reduction significantly reduces listening effort. It is important, therefore, to not only ensure a high directivity index, but also to reduce listening effort in noisy situations. In fact, if the act of listening is made less strenuous, the hearing instrument wearer may become a more focused listener, and thereby indirectly improve speech intelligibility over a longer period of time (Mueller & Ricketts, 2005). To this end, hearing instruments typically rely on directional microphones to boost directivity, and digital noise reduction algorithms such as Wiener filters to improve ease of listening and reduce listening fatigue.

Figure 8 shows the directivity patterns of traditional and micon directional microphones for different frequencies when mounted on KEMAR’s right ear. The broadband noise source is positioned at 120° azimuth. As can be seen, for the traditional directional microphone, even the notch at 120° is less distinct than in free field for many frequencies. In contrast, with the micon 48-channel system, the notch is exactly at the expected direction of 120° and also is much deeper, meaning much more effective attenuation of the specific noise signal. Note that the polar patterns of neither are symmetrical. Whereas the right part of the pattern (tracings farther from the head) looks similar to the free-field pattern, the left part (closer to the head) no longer shows a distinct notch due to the head shadow effect. Bear in mind though, when hearing instruments are worn on both ears, the resulting combined pattern would resemble the intended pattern.
Most modern directional microphones attenuate noise best when it comes from only one direction. When the microphones are multi-channel adaptive, they can track and suppress spectrally different noise sources simultaneously. Wiener filter noise reduction works by continuously estimating the noise level in the incoming signal. This is a very fast-acting algorithm that works well to attenuate noise in the presence of speech. Without reducing the gain for speech, Wiener filters are able to take a signal with both speech and noise components, and filter some of the noise out from the speech, even between words and syllables. However, in situations such as crowded rooms or in restaurants where the competing noise is actually other people talking (i.e., speech), the efficiency of these traditional noise attenuation strategies is reduced.

micon’s solution to listening to speech in noisy situations is called directional speech enhancement. This innovative algorithm provides a major step in helping wearers to focus on the conversation partner in front of them. Technically, directional speech enhancement analyzes different directional patterns simultaneously to allow attenuation of any number of unwanted speakers or other fluctuating noise sources from the back. It makes listening in noise easier, so that wearers can better focus on a conversation even in the most acoustically challenging situations.

SpeechFocus

SpeechFocus is a BestSound Technology algorithm which, in addition to having all the functionalities of a 48-channel adaptive directional microphone, when necessary, can automatically suppress noise which occurs from the front of the user, and focus on speech coming from a different direction, such as from behind. SpeechFocus works by operating simultaneously three different directivity patterns: omnidirectional, adaptive directional, and a reverse directional pattern. 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. This feature is most valuable for situations where the user cannot turn to face the speaker, such as when he or she is driving a car.

When SpeechFocus is enabled in the automatic program, this algorithm would be activated if a “car situation” is detected. Because bilateral information exchange via e2e wireless 2.0 achieves the most reliable acoustic situation detection, SpeechFocus takes advantage of this feature and operates on the joint input from both instruments in a bilateral fitting. When SpeechFocus is designated as a separate program, then the assumed intention is that the wearer desires to hear speech that may originate from various directions when he or she switches to this program. In this separate program, therefore, SpeechFocus does not rely on the bilateral input and reacts to changing speech sources within milliseconds. SpeechFocus can also be enabled in monaural fittings.
Managing microphone settings

Figure 9 shows the Connexx 7 user interface for microphone mode selection. In the automatic with TruEar mode, the corresponding microphone mode would be selected depending on the detected acoustic situation. This could be either the directional mode or the spatial omnidirectional mode TruEar, which mimics directional characteristics of the pinna to optimize front/back localization. When the speech in noise only box is checked, the adaptive directional microphone will only be activated when a speech in noise situation is detected. Otherwise it is also activated in “noise only” situations.

When the SpeechFocus (car) box is checked for the automatic with TruEar mode, this algorithm would be activated as soon as a car situation is detected. SpeechFocus can also be the designated microphone mode in a separate car program, in which case it will react to changing speech sources much faster. Checking directional activates the adaptive directional microphone, which focuses on signals from the front and adjusts the directivity pattern to attenuate competing sound sources from the back or side directions. Similarly, omnidirectional can also be selected for dedicated programs. Directional speech enhancement can also be manipulated under the microphone tab in fine-tuning. In the automatic with TruEar mode, directional enhancement has a default setting of “min”, which is optimized for spatial perception in noise. The “med” setting optimizes sound quality while maintaining the highest possible level of directivity. The “max” setting is activated by default if a dedicated noise program is designated, given the assumed intention of the user to suppress background noise as much as possible.
Frequency compression

Because of the anatomical structure of the cochlea, sensorineural hearing loss (SNHL) usually affects the high frequencies more than the low. For a majority of languages, it is in these high-frequency regions where a large percentage of critical speech information is contained. For example, sounds like /s/, /f/, /th/ help us differentiate “sink”, from “fink”, or “think”. For some individuals, however, the severity of the hearing loss inhibits the interpretation of high-frequency information even with amplification. Moreover, in many instances, especially with steeply downward-sloping hearing losses, it simply is not possible to provide audibility for average speech with traditional amplification when the hearing loss reaches a profound level (e.g., >90 dB HL). A potential contributing factor is that some of these individuals may also exhibit cochlear dead regions, an area in the cochlea where the inner hair cells and/or the auditory neurons are functioning very poorly, if at all (Moore, 2001).

Frequency compression is a recent solution which has been shown to deliver improved audibility to individuals with severe high-frequency losses, or severe to profound losses in general. Simply put, frequency compression is a nonlinear function whereby higher frequencies are compressed and moved into lower-frequency regions where there is better residual hearing. The fitting goal is to restore audibility, albeit at a different frequency, of these speech signals which could not be made audible through traditional amplification.

As previously mentioned, our primary fitting goal is to provide amplification at the corresponding frequency using traditional methods. It is essential, therefore, to properly assure that this is not possible before utilizing frequency compression. For this reason, Siemens has adopted a conservative strategy for implementing frequency compression, which is designed to only be used when necessary, preserve the sound quality of the output signal, and present it to the wearer in a manner sounding as natural as possible. In implementation, this means that with frequency compression activated in the suggested individual configuration, the center frequencies of sounds such as /s/ and /sh/ are still differentiated to maximize speech intelligibility.

In the case of asymmetric hearing loss, frequency compression is set to the same setting bilaterally, and to the better ear. Additionally, since frequency compression results in an altered tonality of the output, it is not recommended for open fittings where the interaction with the direct sound may result in a suboptimal sound quality. The Siemens approach to implementing frequency compression is unique, because it is a default option only for individuals who have an audiometric configuration suggests that audibility of high-frequency information with amplification is not likely. The start frequency for compression and compression ratio are fitted individually.

Figure 10: micon frequency compression only operates on high frequencies. In the left panel, the bandwidth of the input signal is maintained after signal processing. In the right panel, the higher frequencies are compressed due to the frequency compression algorithm, resulting in a reduced output signal bandwidth.
micon BestSound Technology employs many proven as well as innovative technologies to restore audibility and improve speech intelligibility in noise. The key innovations in miFocus are a 48-channel adaptive TwinMic System and a spatial noise reduction system, directional speech enhancement. The benefits offered by proven features such as SpeechFocus and soft level directivity combined with these new innovative microphone features are automatically and adaptively applied in accordance with our excellent acoustic situation detection system (see chapter miGuide, page 40). Add to this micon frequency compression, and we have a recipe for an effective and focused hearing experience. The wearer can pay attention to what they are listening to and not to how they are hearing.

Managing frequency compression

As shown in Figure 11, Connexx 7 offers a slider control to select the appropriate fmin (the cut-off frequency) and fmax (the maximal frequency after frequency compression). The resulting compression ratio (CR) is displayed. In the curve display, the gradually shaded area (just above 2,000 Hz to above 4,000 Hz in the figure) indicates the frequency-compressed region. The two frequencies defining this area are fmin and fmax. The gray area immediately to the right indicates the region where no amplification is provided. Default settings are calculated based on criteria described in the previous section.

Summary miFocus
New things in life often require some effort before we can derive the maximum benefit from them. For example, new eyeglasses need time before they feel completely comfortable. A pair of new boots needs to be broken in before they are fit to be worn for a daylong hike in the wilderness. In order to help wearers get used to new hearing instruments, and for the hearing instruments to accommodate the individual listening preferences of the wearers, micon employs a sophisticated algorithm called miGuide.
A new hearing instrument makes sounds that the wearer has not heard for a long time audible again. Usually, these are high-frequency sounds that are necessary for the wearer to understand speech. Unfortunately, these are also the sounds that are no longer familiar to the wearer due to the effects of auditory deprivation. This is why after the initial fitting, wearers often report that everything sounds too shrill or too loud. The brain must be refamiliarized to these sounds that have not been audible for a long time. This process is called acclimatization. Additionally, each wearer has individual and situation-specific preferences regarding optimal amplification, so the hearing instruments also have to adjust gain and frequency shape accordingly in order for the wearer to be fully satisfied. Going back to the hiking boots analogy, the hiker would only get the maximum benefit from the new boots if he gets used to walking in them, and if the boots stretch to conform to the hiker’s feet.

miGuide is a powerful steering mechanism which takes the personalization of hearing instruments to a new level. It consists of three major parts that are interlinked to enable automatic individualization of the hearing instrument in daily life. First of all, the acclimatization manager supports the acclimatization process by accounting for changing amplification needs over time. Secondly, individual preferences are recorded and applied by a state-of-the-art learning algorithm. As the hearing care professional is able to “guide” the learning process by defining a learning range, this new learning technology is called “guided learning”. Finally, the sophisticated automatic situation detection enables the manual and automatic fine-tuning of situation-specific gain settings.

Acclimatization Manager

In order to encourage spontaneous acceptance at the initial fitting, hearing care professionals often have to select a lower initial gain than is actually appropriate for the hearing loss. Then, gain has to be manually increased according to the progress of acclimatization during a series of follow-up visits. This process is often tedious and time-consuming for both the wearer and the fitter.

The acclimatization manager enables micon hearing instruments to automatically and gradually increase gain until a fitter-defined gain target is reached after a fitter-defined period of time. The acclimatization manager allows the wearer to start at a soft and comfortable gain setting. As he or she becomes used to amplification, the acclimatization manager slowly and imperceptibly increases gain. Thus, this algorithm helps to achieve optimum audibility and speech intelligibility over a specified time period.

In summary, the acclimatization manager ensures that the wearer experiences a comfortable sound right from the start, and optimum audibility and speech intelligibility after the acclimatization phase. It also means less follow-up office visits, and that is good news for both the wearer and the hearing care professional.
Learning

Even though the hearing care professional has an idea of what the average wearer needs in terms of the hearing instrument gain settings based on the hearing loss and prescriptive formulas, the wearer often has different preferences regarding how the hearing instrument should sound. Moreover, individual ear differences and variances among earmold and earpiece couplings make it impossible to always correctly predict what is "best." The ideal settings for an individual, therefore, can only be achieved by satisfying both the needs and preferences of that individual. While automatic acclimatization can ensure that the wearer ultimately becomes used to a gain setting that offers maximum audibility and speech intelligibility (i.e., what he or she needs), miGuide’s learning feature allows the hearing instrument to adapt and adjust gain according to the listening preferences of the individual wearer.

When learning is activated, every time the user 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 for a given acoustic situation at a given input level. Over time, these learned settings are then selected when the user is in the same situation again. The result is separate settings for each of these listening situations tailored to individual preferences.

The hearing care professional determines the initial setting and learning range (i.e., maximum positive and negative deviation from initial gain). This guided learning ensures that the wearer cannot decrease the gain too much, which may degrade speech understanding. Additionally, learning works together with automatic acclimatization. The setting of automatic acclimatization defines the prescribed gain over time. Via learning, the wearer can fine-tune the gain within the limits of the learning range around this prescribed gain (Figure 12). Essentially this means that the fitter determines the street to go, while the wearer selects the preferred side of the street to walk. Depending on his or her preferences, the wearer can either walk on the sunny or shady side of the street, but the street ensures that the wearer always gets to the proper destination, which is optimum speech understanding and listening comfort.
Unfortunately, the same hiking boots are not optimal for every situation. You may get used to hiking in them, but you need different shoes for dancing the night away. Similarly, how you like to hear when you are relaxing on the patio with a book is different from when you are enjoying a family dinner at a favorite restaurant. But while shoes cannot change themselves, miGuide allows hearing instruments to adapt to changing listening environments with automatic situation detection.

The automatic situation detection in micon is able to identify six different acoustic environments with very high accuracy. This means that the wearers can make their instruments learn their preferences for each acoustic environment individually: quiet, speech in quiet, speech in noise, noise, music, and car. Table 1 lists examples and the requirements a hearing instrument wearer might have in those situations.

While Connexx™ provides different frequency shaping for each of these situations by default, the hearing care professional also has the opportunity to make manual adjustments to these frequency shape offsets via sound equalizer.

### Table 1

<table>
<thead>
<tr>
<th>Situation</th>
<th>Description</th>
<th>Requirement</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quiet</td>
<td>Sitting alone on the patio and reading the newspaper</td>
<td>Enough amplification to enjoy the singing of the birds in the trees</td>
</tr>
<tr>
<td>Speech in quiet</td>
<td>Sitting with someone else on the patio and having a conversation</td>
<td>Amplification which ensures optimum speech understanding with high listening comfort</td>
</tr>
<tr>
<td>Speech in noise</td>
<td>Same situation like before but the neighbor starts to mow the lawn</td>
<td>In addition to the activity of noise reduction algorithms and directional microphones, less gain at low frequencies to ensure listening comfort</td>
</tr>
<tr>
<td>Noise</td>
<td>Same situation like before but the conversation has stopped</td>
<td>Gain is further reduced and noise reduction adjusted so that the noise can still be heard, but at a comfortable loudness</td>
</tr>
<tr>
<td>Music</td>
<td>Sitting in the living room and enjoying music</td>
<td>Balanced sound impression for high and low frequencies for all input levels; noise reduction and directional microphones are disabled</td>
</tr>
<tr>
<td>Car</td>
<td>Noise environment</td>
<td>Because of the car noise, a special frequency response helps to make it sound comfortable; furthermore, special signal processing helps to improve speech intelligibility, even when someone speaks from behind (SpeechFocus is activated)</td>
</tr>
</tbody>
</table>
If we have normal hearing, no matter where we are, we are instantly able to focus on what and how we want to hear. miGuide is the powerful steering mechanism which allows the wearer to do this again effortlessly. It instantaneously detects acoustic situations and adjusts the hearing instrument’s frequency response and adaptive feature settings accordingly. Not only does miGuide offer predefined settings for these different acoustic situations after first fit, it also allows the hearing care professional to manually adjust these settings in the fitting process. It also contains the industry benchmark learning algorithm which allows the wearer to train the hearing instrument to the preferred gain settings in these different situations specifically. Additionally, miGuide also, when necessary, considers the need for auditory adaptation and automatically acclimatizes new users to more appropriate amplification over time, even when learning is activated.

Managing miGuide

Acclimatization manager
The acclimatization manager slowly increases hearing instrument gain toward a prescriptive target or an individual target that the hearing care professional can define (Figure 13). When selecting the acclimatization period, a selected duration of one month means that the target will be reached within one month with an average wearing time of eight hours per day. If the hearing instruments are worn less than eight hours a day, it will take longer to reach the target. Acclimatization can be started and stopped anytime. By clicking the reset button, the fitter can restore the original settings before automatic acclimatization was started. The prelisten button allows the wearer to listen to how the hearing instrument would sound when the acclimatization target has been reached. In the curve view, the acclimatization target will also be changed.

Using miGuide in Connexx

Figure 13
Acclimatization manager in Connexx 7. The acclimatization target and acclimatization period can be changed.
be displayed. Keep in mind that in order to fully reach the designated acclimatization target, the acoustic coupling parameters selected for the fitting should also be appropriate for the acclimatization target. For example, if the acclimatization target calls for significant low-frequency amplification, it’s important to use double domes or molds with a small vent to achieve a more closed fitting.

Learning
Since the learning function is intended for the user to fine-tune the hearing instrument settings after the hearing care professional has already done most, if not all of the initial gain and output adjustment, it should be only activated at this point. Learning can be enabled for all programs together, or for individual programs (Figure 14). The learning range defines how many dB the hearing instrument can be trained to deviate from the current settings and therefore allows the fitter to ensure that the wearer still receives the amplification he or she needs. In the curve view, the initial as well as the learned settings are displayed for each program.

From the situations tab, trained settings for each of the six situations can be reviewed. Bear in mind that trained settings will only be available after the read-out from a hearing instrument that has been through an initial fitting. This implies that it has been used for a sufficiently long time and that there have been enough adjustments of the user controls. Residual range indicates the remaining learning range after learning has already started. For example, if the maximum learning range for increasing gain is 9 dB, and the wearer has already learned gain up by 2 dB, then the residual maximum up will show 7 dB.

Sound equalizer
The automatic situation detection differentiates among six different acoustic situations in the universal program. Each acoustic situation has a dedicated frequency response which is defined as offsets against the general settings in the universal program. These offsets are individually adjustable under the sound equalizer tab in miGuide (Figure 15). The settings for the situation speech in quiet are identical to the general settings in the universal program. This is why the offsets here are set to zero and cannot be changed. The general settings as well as the offset defined by the fitter can be seen in the curve view.
Data logging

Under the Data logging tab in miGuide, the hearing care professional can read out all changes from volume control and SoundBalance, wearing time, and the detected situations where the hearing instruments were used (Figure 16). All this information helps to understand the comments of the hearing instrument wearer and enhances counseling.

Using miGuide in Connexx

Data logging in Connexx 7 displays information such as wearing time and manual changes made by the wearer.

Summary miGuide

Our ultimate goal is to allow our hearing instrument wearers to pay attention to what they hear, not how they hear. Listening to a concert should allow the listener to enjoy the emotions of the music, not pay attention to the performance of the hearing instruments; and listening to a loved one talking even in a noisy restaurant should be made easy for the individual by clever and automatic signal processing, rather than cumbersome manual adjustment or program selection.

miGuide is the new powerful steering mechanism which reliably detects acoustic situations and smoothly adapts the hearing instruments’ frequency response and adaptive feature settings to how the wearer prefers to hear. Not only does miGuide offer predefined settings for these different acoustic situations and allows the hearing care professional to manually adjust these settings, it also contains the industry benchmark learning algorithm which allows the wearer to train the hearing instrument to the preferred gain and compression settings in these different situations specifically. miGuide also considers the fact that many wearers need to gradually adapt to sounds that they have not heard for many years. Automatic acclimatization can provide new users more appropriate and advantageous amplification over time, without compromising loudness perceptions or listening comfort. In other words, micon adapts, optimizes and steers according to the wearer’s needs and preferences. Now is the time for wearers to forget that they are wearing hearing instruments and to just listen!
Appendix
Tinnitus noiser

It is known that the use of external sound(s) provides relief from tinnitus by inducing neuro-physiological functional changes in the different auditory pathways. The aim of sound therapy is to decrease prominence of the tinnitus and facilitate tinnitus habituation. To transform the individual’s reaction to tinnitus, counseling is required. To transform the perception of tinnitus, sound therapy is recommended to decrease the prominence of tinnitus and facilitate tinnitus habituation. This sound therapy is now available in micon hearing instruments and can support a wide variety of acknowledged tinnitus treatment programs. In the new Connexx 7, the frequency shape of the masking noise can be adjusted precisely for each individual wearer, and can be used alone or combined with individual amplification where a hearing loss is also present. While the level of the noise can be set in the software, it is still possible to adjust this manually via hearing instrument controls or a remote control.

If the individual also has a hearing loss, the setup for the tinnitus noise program will be slightly different. Figure 17 shows how different noises can be selected according to the individual’s preference (pink, white, high tone, or speech noise) and how each frequency can be individually set. Switch to the output mode in the curve view to see the exact frequency shape.

Mixed mode can be selected when amplification alone does not relieve the tinnitus for the wearer. In this case, amplification can be combined with individually shaped masking noise. Once set, the wearer may also be allowed to alter the hearing instrument gain or the noise via the onboard volume control or remote control unit.

For further information on tinnitus and the tinnitus noiser feature in Connexx 7, refer to the Siemens Tinnitus Whitepaper.
Adaptive TwinMic System
The adaptive TwinMic System is a part of miFocus and takes automatic adaptive directional microphone systems to a new level. It incorporates a 48-channel high-frequency resolution which optimizes the effectiveness of directional microphones in wearing conditions.

Acclimatization Manager
A part of miGuide which considers the need for auditory adaptation and automatically acclimates new wearers to more appropriate amplification over time. This feature saves valuable time for hearing care professionals and wearers alike by eliminating the need for follow-up appointments for the purpose of increasing gain.

Automatic situation detection
Algorithm which detects, analyzes, and classifies the current acoustic situation such as quiet, noise, music, etc. The resulting decision dictates the configuration of numerous hearing instrument features such as frequency response, microphone mode, speech and noise management, and learning. In bilateral fittings, input from both hearing instruments is taken into account in order to synchronize the steering of both instruments. Siemens automatic situation detection is another industry benchmark feature and is one of the key components of miGuide.

Data logging
Records data such as program usage and time spent in specific acoustic situations to be displayed in Connexx. This information helps hearing care professionals to make more informed decisions for counseling and when making fine-tuning adjustments.

Directional speech enhancement
Analyzes different directional patterns simultaneously to allow attenuation of any number of unwanted speakers or other fluctuating noise sources from the back. It makes listening in noise easier, so that wearers can better focus on a conversation even in the most acoustically challenging situations.

e2e wireless 2.0
Siemens’ technology for coupling and synchronizing two hearing instruments into one holistic, binaural hearing system. It is used for communication between the remote control and hearing instruments, as well as between bilateral hearing instruments. In a bilateral fitting, e2e wireless 2.0 allows the onboard controls of the two hearing instruments to have different functionalities so that, for example, a rocker switch on the left could adjust volume bilaterally, and a push button on the right could change programs bilaterally.

micon feature summary
Besides the key features described in the previous sections, hearing instruments powered by micon are also equipped with a host of proven BestSound Technology features. With the extended bandwidth and higher channel resolution offered by the powerful micon chip, many of these features, some of which are already industry benchmarks, are now better than ever.
Soft level directivity
Dictates the noise level at which directional microphones are activated automatically. In hearing instruments powered by micon, the effectiveness of soft level directivity is improved because of the superior frequency resolution offered by 48 channels.

SoundBrilliance
Exceeds the limitations of conventional hearing instrument bandwidths and adds artificial high frequencies to the output up to 12 kHz without added feedback risk. The result is a more “brilliant” sound quality which is especially helpful when listening to music or using audio streaming via Bluetooth.

SleepFocus
Continuously analyzes the environment for the most favorable speech-to-noise ratio, and automatically selects the microphone configuration which has the potential to offer the best speech intelligibility for the wearer, regardless of whether speech is coming from the front, the rear, or the side. In hearing instruments powered by micon, SleepFocus can also be a part of the automatic microphone mode.

eWindScreen™
Automatic algorithm which detects the presence of wind and suppresses it. It works virtually instantaneously and even when directional microphone technology is applied, so that speech understanding in outdoor situations can be improved even in windy situations.

Feedback cancellation
A part of miSound which combines parallel processing of two phase cancellers, as well as continuous critical gain estimation to offer better feedback suppression than ever before.

Frequency compression
An alternative fitting strategy for individuals with profound high-frequency hearing loss or whose hearing loss inhibits the interpretation of high-frequency information even with amplification. Preset based on the individual hearing loss and research-driven formulas, this algorithm compresses and shifts high frequencies otherwise beyond the audibility of the wearer to a lower frequency range where there is better residual hearing.

micon compression system
new compression system incorporating two kneepoints, which allows independent adjustment of level-dependent gain, and adaptive time constants, which combine the advantages of both slow and fast compression systems.

micon fit
Siemen’s latest proprietary fitting formula which leverages the 48 channels and 12 kHz bandwidth made possible by the micon chip. It is powered by the new micon compression system with two kneepoints and adaptive time constants to deliver the most natural sound quality without compromising audibility.

SoundSmoothing
Industry benchmark noise reduction algorithm that targets transient and impulsive noises such as rustling paper or clanging dishes without affecting speech signals. In hearing instruments powered by micon, the effectiveness of SoundSmoothing™ is improved because of the superior frequency resolution offered by 48 channels.

SpeechFocus
Continuous analysis of the environment for the most favorable speech-to-noise ratio, and automatically selects the microphone configuration which has the potential to offer the best speech intelligibility for the wearer, regardless of whether speech is coming from the front, the rear, or the side. In hearing instruments powered by micon, SpeechFocus can also be a part of the automatic microphone mode.
Speech and noise management

Noise reduction scheme comprised of a slow, modulation-based algorithm which works optimally in noise only situations, and a fast-acting Wiener filter which works well in speech in noise situations. Together, these two multi-channel and adaptive algorithms provide optimal listening comfort for the wearer in a variety of listening situations. In hearing instruments powered by micon, the effectiveness of speech and noise management is improved because of the superior frequency resolution offered by 48 channels.

Tinnitus noiser

Allows the hearing instrument to deliver a programmable tinnitus-masking noise, either by itself or in addition to the amplified environmental signal. This function is now programmable in 20 handles and is more flexible to adjust than ever.

TrueEar

Simulates the acoustics of the natural outer ear to allow for better front/back localization in BTE, RIC, and open fittings. In hearing instruments powered by micon, the effectiveness of TrueEar™ is improved because of the superior frequency resolution offered by 48 channels.

References


The information in this document contains general descriptions of the technical options available, which do not always have to be present in individual cases and are subject to change without prior notice.

The required features should therefore be specified in each individual case at the time of conclusion of the respective contract.

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