



THE AUDIOLOGY IN EPOQ

– A WHITEPAPER



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Introduction:

Our senses embed us into the world around us. This world is rich with stimulation: some of it designed to bring us information, some of it reflecting the layers of complexity in our surroundings and some of it simply background ambience. The sights, sounds, smells and feel of the world around us connect us to that world at many different levels. Our conscious analysis of our environment just barely scratches the surface of our experience. At a visceral level, all of this stimulation leaves us with a sense of time and place.

Interface: the hearing device is the connection between the user's remaining auditory abilities and the full world of sound.

When hearing declines, we are at risk of losing a vital connection channel to the world (Ramsdell, 1978). The surrounding world is not just the obvious (the conversations we have) but includes the ambience of a place, those important sounds we overhear that bring us information and appreciation of our setting, and, at an ever exploding pace, those sounds mediated by modern electronic sources such as mobile phones, television, radio, computers and more. This ambience is essential for the listener to feel connected to the environment. We want to have a sense of place. We want the sounds we hear to have a natural relationship to the place that we find ourselves in.

In Audiology, we often break down the world of sound into black and white: good sounds and bad sounds. We want hearing devices to amplify the good sounds and eliminate the bad ones. This viewpoint ignores the multiple purposes of sound and the reality that any given sound can change from meaningful to

ambient to disruptive depending on the moment to moment flux in daily life. By limiting our viewpoint about what sounds are important, we limit what we expect from amplification.

What is the role of a hearing device?

For years, we have seen the hearing device as performing the basic job of correcting for the user's hearing loss. We measure how much loss is present and then calculate how much amplification is needed to make up for lost sounds, especially conversational speech. The problem with this "hearing loss correction" approach is that it is driven by what we are used to measuring – hearing loss. This goal, although worthy, has underused the power of advanced digital sound processing. More importantly, it has grossly underestimated, if not disrupted, the natural ability of the auditory system and the brain, even in the presence of hearing loss, to absorb and make use of the vast variety of information in daily environments. It does not focus on the true role of the hearing device: to stand as the interface between the user and the outside world.

Hearing devices that are focused only on increasing access to speech cues (by attending to audibility and noise control) may not pay due attention to what is necessary to maintain access to the broader world of sound. Over the years, we have learned more and more about what it takes to provide access to the speech signal. It is now time to build on that knowledge and success and turn our attention to what it takes to maintain the listener's natural connection to the full world of sound.

Audiologically, we have focused on providing access to speech sounds in the immediate vicinity of the listener (let's say within 5 to 10 meters). When speech comes from more distant origins or when the sounds of interest are related more to the setting and not to



a conversation partner, we typically have not specifically targeted amplification to capture those sounds. However, the Near World of sounds – the totality of those sounds that comprise our listening environment – are what define auditory experience for those with normal hearing. In addition, sounds that come from distant sources and that are transmitted via some electronic medium comprise the Far World (Figure 1). We have paid only cursory attention to providing access to those sounds. Telecoils have only been available in some models of hearing devices and have always had their limitations. The explosion of mobile phone use has made access to telephone signals even more limited due to incompatibility with telecoils and annoying electronic interference. More and more signals are being transmitted via Bluetooth, yet a widespread, functional solution to bring Bluetooth signals into hearing devices has yet to enter the market. Our narrow focus on what the role of a hearing device should be has widened the gap between the sound experiences of those with normal hearing and those with hearing loss.

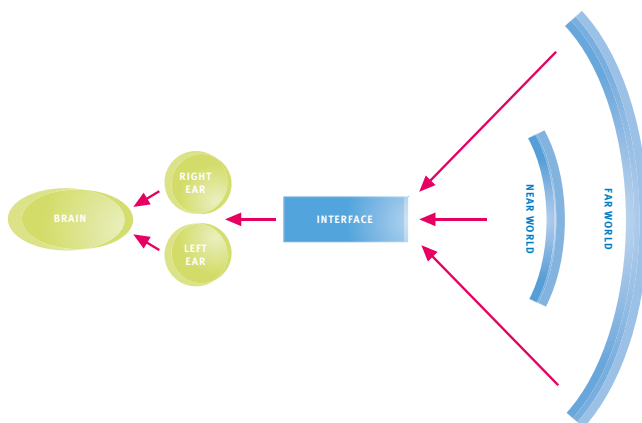


Figure 1: The hearing devices as the interface between the Near and Far Worlds and the user.

The advances in digital sound processing have been impressive, primarily focused on manipulating the sounds of the world. The very smartest processing schemes are no match for what the human brain can do when given half a chance. Many of the approaches we use in traditional amplification are designed to dramatically compensate for known changes in the loudness dynamics of the ear with sensorineural hearing loss. There is a risk, however, that these uni-dimensional corrections for one aspect of peripheral hearing loss may actually steal information from the signal, feeding the brain information that do not allow us to use binaural processing abilities optimally.

Each user comes to us with a unique set of abilities. The audiogram tells us what the patient has lost, not what skills the patient retains. Hearing devices don't change hearing – they change the sounds presented to the user. Viewing the hearing device as an interface puts the proper perspective on the challenge: to capture and enhance the full range of signals that the patient wants to hear and find the best match to the user's remaining capabilities and preferences.

Amplification, when designed as an interface, will take advantage of our ever-advancing understanding of how humans process information. As with all good interfaces, it will pay attention in both directions: capturing and processing the vast array of sounds coming from the outside world but also recognizing the best way to match that information to the powerful remaining capabilities of the user. An intelligent interface makes changes in the signal when needed but also preserves as much of the naturally occurring cues as possible, allowing the brain to do what it does best.

Near World: the sounds around us that make up the immediate, live auditory experience, including not just conversations but all acoustic sounds



Far World: those sounds that inform and entertain are sent to us via some electronic device, including mobile phones, MP3s, radios, etc.

Oticon Epoq is based on these principles of interface. It is an advanced digital signal processing system designed to take the vast array of sounds from the world around us – both those in the immediate, near world but also those transmitted from the far world – and present them to the patient's auditory system in a manner that allows the patient to best use the remaining auditory capabilities. It recognizes that the normal auditory system is designed to capitalize on inputs from two ears to connect the listener with the environment. It also recognizes that, more and more each day, those sounds that inform and entertain come not just from our immediate environment but also via an array of transmission devices. As an interface, it is designed to bridge the gap between the world of sound and the diminished yet still relevant capabilities of the patient's auditory system.

The RISE Architecture



Think about what personal computers offered us 5 to 10 years ago. They were powerful electronic storage systems, word processors and number crunchers. They were designed to increase personal productivity using the power of digital signal processing. Unfortunately, connectivity to the outside world was considered a design afterthought, requiring additional hardware and piggybacking on existing telecommunication channels. Attach a modem, dial in and hope for, at best, a low bandwidth connection.

Nowadays, connectivity is an integral part of PC design. Broadband capability is a given. New systems have on-board wireless and Bluetooth capabilities as standard features. It is hard to find a computer these days that is not connected to the web. It seems almost archaic.

This change in hardware has ushered in an era of exploding possibilities. An isolated PC is only as powerful as the data and software that is on-board. However, a computer that is connected to a network now has the capability of all the computers on that network and one that is connected to the World Wide Web taps into millions and millions of other systems. Taking inspiration from today's personal computers, Oticon's RISE architecture is the first hearing instrument architecture to be built from the ground up with connectivity as a key capability. Unlike systems in the past, connectivity is not an add-on feature. Rather, from both a hardware and functionality standpoint, connectivity between hearing instruments and from the hearing device to the outside world is at the core of the design. Epoq is the first instrument based on this new architecture.

Oticon's RISE architecture combines both hardwired and fully programmable sections to achieve both high processing speed but also the flexibility for customization. The RISE architecture provides between 10 and 50 times more computing power, than what has been available in the best products to this point. Along with the signal processing approaches that have defined advanced digital hearing devices to this point in time, the RISE architecture introduces two dramatic advances in personal amplification: Dual Path signal processing and EarStream wireless connectivity.

An advanced digital hearing device by itself can be a powerful sound processing system. Epoq offers new state-of-the-art approaches to well-know features in multi-channel, nonlinear hearing devices such as multi-band adaptive



directionality, noise reduction, feedback cancellation and device learning. In addition, Epoq introduces new concepts such as own voice detection and processing and the use of dual compression paths. However, once this device is connected at broadband speeds to the hearing device on the other side of the head, new processing capabilities are possible that could never be achieved by one hearing device acting in isolation. Once that pair of devices is connected to electronically mediated sounds, a whole new world of access is achieved.

The personal computer has recently evolved from a tool of personal productivity to a gateway to a vast universe of information and entertainment. The RISE architecture, by being designed with connectivity as the key component, changes the role of the hearing device from being an isolated correction for hearing loss to being the interface between the user's auditory system and the world of sound, both the sounds in the immediate environment (the Near World) and those transmitted from distant places (the Far World).

Dual Path Sound Processing: When digital signal processing was introduced into hearing devices over ten years ago, the first applications were designed to replicate state of the art analog processing (at the time, multi-channel, Wide Dynamic Range Compression).

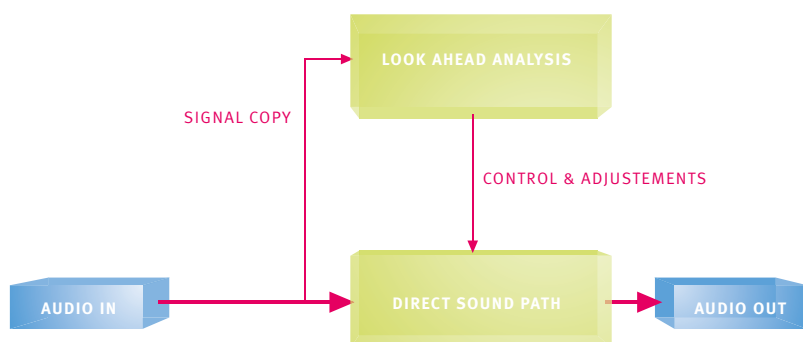


Figure 2: A diagram of the Dual Path Architecture in Oticon Epoq.

Over the past decade, more and more DSP-specific applications have been added to this core design (automatic directionality, noise reduction, feedback cancellation, etc.) In all applications up until now, single-path serial processing has been used to implement these technologies. When the number of steps was limited, this structure was certainly adequate. However, as more and more functionality is added to the signal path, the major limitations of serial processing become evident: each step requires greater bit resolution in order to maintain signal integrity. As more and more steps are implemented, more and more bits are added, requiring greater processing power and time. Given that the audio signal quality is a must as well as low current drain and processing speed, serial processing is reaching the limits of utility.

The fundamental sound processing method in the RISE Architecture is based upon the simple and powerful principle of “Dual Path” design. In contrast to the method used in traditional instruments, where the sound signal is forced through a chain of individual processing steps, the incoming signal is cloned and sent down independent analysis and processing paths. On the analysis path, the original signal is evaluated on an on-going basis (“look ahead analysis”). Signal adjustments required by all of the processing units are coordinated by a central decision unit. Epoq then instantly programs all elements in the processing path based on full knowledge of the combined effects of all settings (see Figure 2).

As a result, all the adjustments are made in one step, and as one common decision. The advantage of this approach is the optimal use of all sound processing tools in the instrument without negative interactions or competition between different systems. This high speed system guarantees clean and clear sound reproduction since the signal is not manipulated



by each separate signal processing system, enhancing the signal when needed but still preserving outstanding natural sound quality.

EarStream: The fully integrated magnetic wireless technology in the RISE architecture represents an as yet unseen combination of speed, energy efficiency and small size. Running at 120.000 bits per second, it is comparable to the speeds in entry level ADSL and cable modem connections commercially available today. This is about 100 times faster than the best in the hearing care industry up till now.

EarStream makes a wide number of applications possible. High-speed communication between the hearing instruments – what we have termed **Binaural Broadband** – makes true binaural processing possible for the first time in amplification. In addition, EarStream allows for real-time, full-bandwidth audio transmission. Yet EarStream is energy efficient and still small enough to fit nicely into small BTEs, RITEs and even small custom styles.

Binaural Broadband: high speed communication between the user's two Epoq hearings devices.

Using for instance a 312 battery, the wireless technology enabling Oticon's RISE paradigm could run continuously for more than 12,000 minutes. In comparison a standard Bluetooth device would be operational for about 540 minutes on a 312 battery – and a standard GSM mobile phone would last just 30 minutes – a difference of a factor of 400. Obviously the range of a mobile phone is far longer than

Soundscape: the total montage of the sounds of the place we are in, providing the auditory signature of our setting.

what is required for wireless communication between hearing instruments just centimetres apart or other devices close to the person, but even when the range of the wireless technology is included in the calculation, the wireless technology used in Oticon's RISE architecture is significantly more power efficient than for instance Bluetooth.

The technology breakthroughs of Dual Path processing and EarStream were essential to create this new understanding of what the role of amplification should be. These approaches have allowed us to implement signal process designed to work in concert with how the auditory system was designed: to take the full range of sounds from the world around us and present those signals in a clear, natural way to the binaural system.

Soundscape

There is a rich world of sound around the listener. It is called the Soundscape (Schaeffer, 1994). Not only are there important communication signals but also the everyday sounds that create the ambiance of day-to-day life. This ambiance is essential for the listener to feel connected to the environment (Ramsdell, 1978). We want to have a sense of place. We want the sounds we hear to have a natural relationship to the place that we find ourselves in.

Our sensory systems are naturally designed to make sense of the world around us, allowing us to automatically and effortlessly use all sources of information to identify the objects and events around us. Within the auditory world, we use two functions – Stream Segregation and Scene Analysis – to deconstruct the complex soundscapes we find ourselves in everyday (Bregman, 1990). We assign different



parts of the sound input we perceive over time into meaningful sources (Stream Segregation) and then organize these various sources in the space around us (Scene Analysis). Once we develop this organized sonic world, we can then assign attentional resources to specific sources of interest. The ability to localize sound is essential for complete Scene Analysis. Without localization, we may be able to isolate individual sound sources, but placing them appropriately in the space around us is essentially impossible. Further, even the ability to assign various inputs to different sources can be limited significantly by the loss of localization cues. In difficult listening environments, the use of localization cues can provide up to a 5 to 10 dB improvement in the functional signal-to-noise ratio (Cherry, 1953; Aborgast, Mason & Kidd, 2005).

One of the unfortunate consequences of amplification (versus unaided) is that the localization abilities of the listener can actually become poorer with amplification (Byrne, Sinclair & Noble, 1998; Van den Bogaert et al.,

2006). Closing off the ear canal will limit access to important high-frequency localization cues (Byrne, Sinclair & Noble, 1998). In addition, environmentally adaptive systems such as automatic directionality (Van den Bogaert et al., 2006) can distort naturally occurring timing and spectral cues, “muddling” the cues that assist localization. This muddling is created because these systems are designed to optimize the incoming signal for that individual ear. However, the lack of coordination between ears may provide the central auditory system with a set of cues that no longer have the between ear relationships that occur naturally.

Specifically, the presence of the head creates spectral differences between the ears when sounds do not arise from the midline (Yost, 1994) (Figure 3). These spectral differences are essential for localization. The natural tendency of compression systems is to minimize these spectral differences, as the amount of gain applied to the signal is greater for lower level input signals. A naturally occurring ear-to-ear

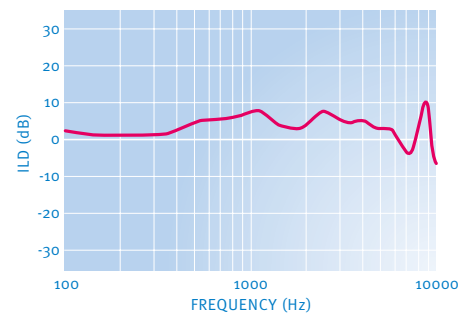
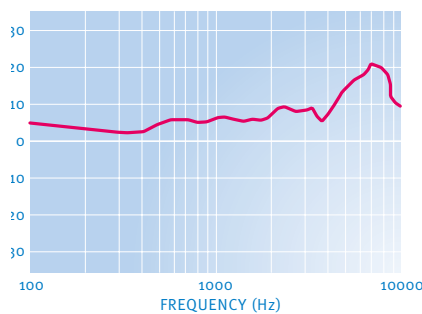


Figure 3: The inter-ear spectral differences for sounds arising from 90 and 150 degrees to the right of the midline (Sensimetrics Corp., 2003).

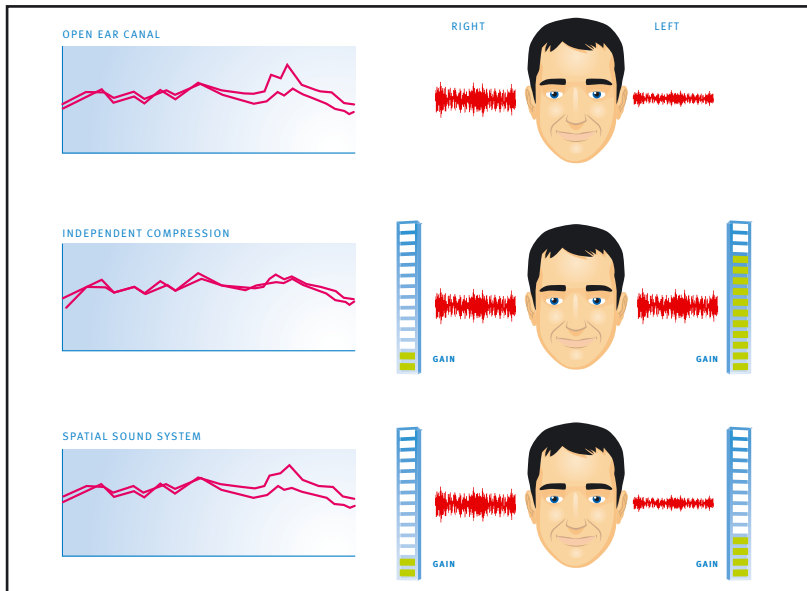


Figure 4: Effect on naturally occurring spectral differences of two hearing devices that are acting independently (middle panel) and when coordinated via Spatial Sound in Epoq (bottom panel). The upper line represents the ear closer to the sound source and the lower represents the ear further from the source.

difference of 10 dB can be reduced to only 3 to 4 dB or less with the moderate amount of compression typical of WDRC fittings for moderate hearing loss.

Spatial Sound

The Spatial Sound feature in Epoq compares the input to the two different hearing devices and, at broadband speed, adjusts the compression response to maintain precise spectral difference in the high frequencies. The effect is to provide the central auditory system a paired set of inputs that more accurately reflects the true differences between the ears: information that is vital for accurate resolution of the auditory environment. (For a more detailed discussion of the rich literature in the area of spatial perception and hearing loss, see Appendix A.)

The upper panel of Figure 4 provides the spectrum measured in KEMAR’s right and left ear canals of a broadband sound presented at 45 degrees to the right of the midline. Notice the naturally occurring level differences in the higher frequencies. For example, above 4kHz., the level difference are greater than 15 dB. The middle panel shows the spectrum levels as processed by independent nonlinear hearing aids programmed to the same hearing loss. Notice now how the spectral differences in the higher frequencies have been significantly reduced. The bottom panel shows the same conditions but now as processed by Epoq. Notice that the naturally occurring high frequency spectral differences are better maintained. The response of the compression systems when acting independently on the two different ears (middle panel) makes perfect sense when viewed one ear at a time. The input level on the left side was less than the right side, so the gain needed to be greater on the left compared to the right. However, when viewed as two input sources to a binaural system, vital intra-aural level information was lost. Spatial Sound allows that information to be better maintained when the two hearing devices are acting together (bottom panel).

This binaural processing is also applied to the action of other environmentally adaptive signal processing systems. In our devices, when they are acting independently, the AI-enabled noise reduction and directional systems are designed to constantly evaluate the environmental conditions and adjust the settings of the adaptive systems based on the immediate conditions experienced by the hearing device (Schum, 2004). Since conditions such as signal level and S/N can be different on either side of the head, these intelligent systems can make different decisions on either side of the head. Some times this makes sense but sometimes this can lead to different system responses on one ear versus the other: omni

Spatial Sound: the ability of Epoq to accurately preserve the localization cues that naturally occur on either side of the head.



with aggressive, noise-only based noise reduction on the right side but split directionality with less aggressive, speech+noise reduction on the left. This is not good for the brain. This will create the impression of something different happening on either ear when the brain's job is to create a fused, unified percept. Importantly, there are situations such as in very asymmetrical listening environments (e.g. driving in the car with the window open) where similar settings does not make sense.

Within a set of Epoq hearing devices, the Spatial Sound system will coordinate the response of these systems. This allows significantly improved sound quality as the hearing instrument will avoid situations where one hearing device lags the other in terms of decision making resulting in poor sound quality due to uncoordinated mode shifts (Keidser et al., 2006). In highly asymmetrical situations, the AI based technology in Epoq ensures that the two hearing devices work separately. The key benefit of coordinating systems is to provide improved listening stability to the listener through coordinating mode shifts and allowing increased mode stability by comparing data on both ears before changing modes. Decision making becomes more robust and avoids the perception of a system that is too active in terms of sound processing.

In summary, Spatial Sound, ensures that the listener continues to benefit from the effects of advanced processing systems without experiencing any of the reduced sound quality due to non-coordinated processing. Binaural processing (i.e. coordinated compression) maintains natural arrangement of sounds and natural contrasts and asymmetry in the sound scene. For the listener, they will experience a more stable sound picture complemented by a reduction in uncoordinated shifts in mode selection and an improved awareness of overall soundscape.

Connecting to the Far World

Despite the advances in sound processing that have been possible due to digital hearing devices, there are still some listening situations that remain inadequately addressed. These situations are consequently pointed out by hearing device user as areas where they have the least satisfaction from their hearing devices – and unfortunately, these experiences deteriorate their general perception of the hearing devices. Currently, users rate the use of mobile phones in conjunction with their hearing devices as the most troubling situation they face, with only 62% of all users reporting acceptable performance in this scenario (Kochkin 2005). Problems are usually noted in the areas of lack of compatibility and electronic interference.

For years, hearing devices have been designed with live speech as the primary signal of interest. As we all know, interesting information reaches us in so many other ways: the mobile phone, MP3 players, the television and the Internet. Until now, access to these other signals has been an afterthought of hearing device design. Epoq has been conceived to provide the hearing impaired user with full access to all the sounds in the world, no matter how they reach us.

As described previously, the RISE architecture was designed with a wireless magnetic transmitter/receiver in the hearing device. Not only is this functionality used to connect the two hearing devices in a binaural pair, it is also used to communicate with a body-worn gateway device (Streamer). The Oticon Epoq Streamer (Figure 5) can stream audio wirelessly and directly into both hearing devices from different kinds of electronic sound equipment. The EarStream magnetic wireless system is used to create a Personal Area Network (PAN) on the top half of the body, allowing the hearing devices to communicate directly with

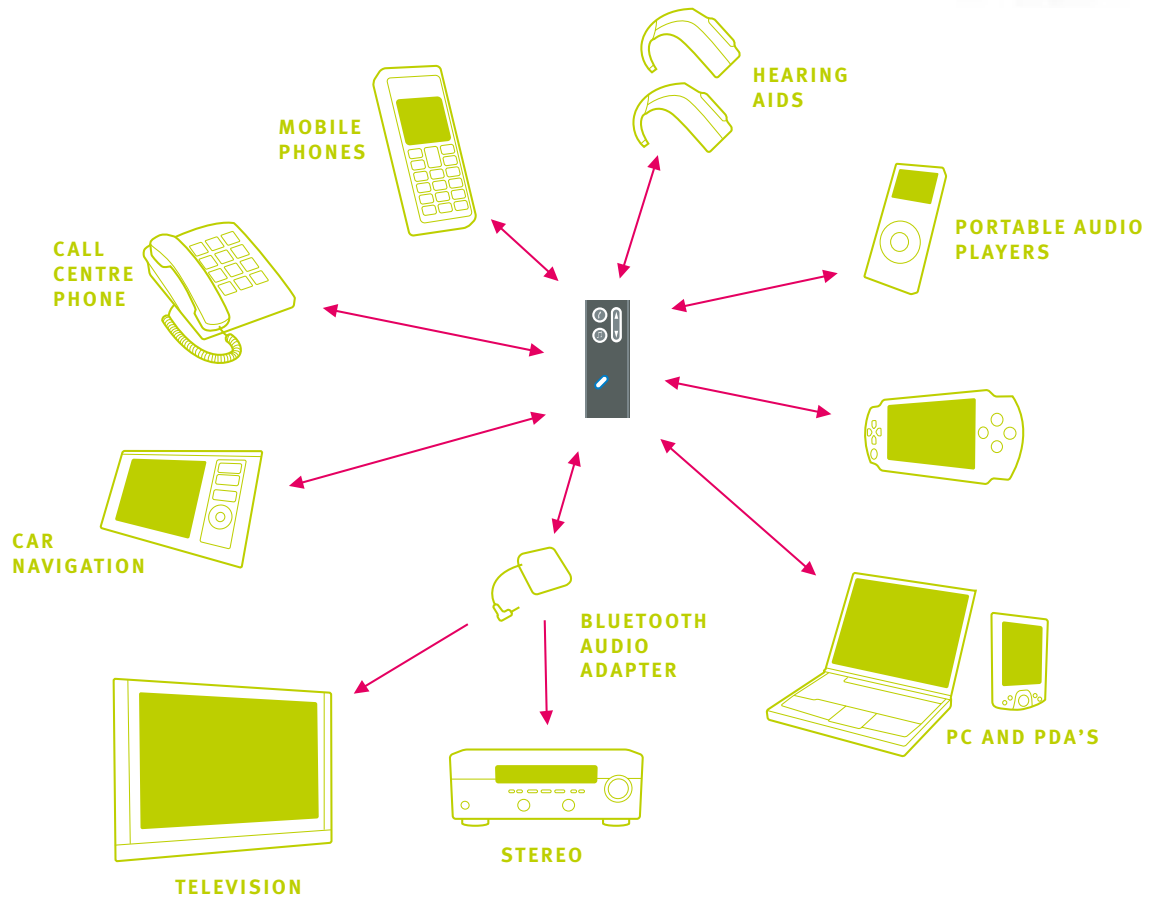


Figure 5: A diagram showing the relationship between Epoq hearing devices, Streamer and external sound sources.

Streamer. Streamer then uses Bluetooth to connect to external sound sources that transmits a signal using this increasingly prevalent transmission technique. (For a more detailed discussion of how Bluetooth is integrated into Streamer, see Appendix B.)

With Bluetooth wireless connectivity built into the Streamer, Epoq can instantly transform itself into a wireless mobile-phone headset, with clean and clear sound in both ears, eliminating the well-known compatibility problems of mobile phones.

A second important feature of the Streamer is that it connects the hearing instruments to different audio sources for entertainment or information purposes. Streamer even accepts

any audio signal that can be accessed via a direct wired connection, allowing for an audio connection to MP3 players, radios, PCs - etc. As more and more of these systems broadcast their signals via Bluetooth, wireless connectivity to Streamer will be expanded.

Realizing that great technology is worthless if it is difficult to use, the Streamer has been designed to be easy and extremely simple to use for both the professional and the patient. Using the Streamer is as simple as pressing a button.

The user controls all streaming scenarios from the Streamer itself (see Figure 6), including the volume control. By enabling the hearing



Figure 6: A close up view of Streamer.

instruments to become wireless ear phones, which are almost invisible on the ears, sound from different Bluetooth enabled devices can be relayed to both the ears with no loss of signal quality. From a built-in microphone that the user speaks into, the Epoq Streamer works as a Bluetooth interface to mobile phones and uniquely solves the problems related to the coexistence of hearing instruments and mobile phones. No more problems with buzzing and insufficient volume.

In other words, the Epoq Streamer will give its user the equivalent of a hands-free Bluetooth headset to both ears without having to attach any devices to the ear and without any cables.

All in all, Streamer uses the EarStream wireless technology to send, at broadband speed, any signals from the outside world that are transmitted via either Bluetooth or wired audio input. Since no previous hearing device has been designed from the ground up with connectivity as the key design criterion, compatibility problems have been widespread and have frustrated the hearing device user. They have felt increasingly left out of the communication boom that we are living through. Epoq and Streamer make up that lost ground.

New, Smart Signal Processing

It is clear that the most dramatic improvements introduced by Epoq are Spatial Sound and access to transmitted sound via Streamer. However, there is an important set of new signal processing techniques inherent in every Epoq that, each in and of itself, offers new benefits to patients.

True Dynamics

Compression systems in conventional hearing instruments have inherently suffered from a compromise. Either compression was relatively fast, bringing good speech understanding but at the cost of somewhat reduced sound quality and higher levels of listening fatigue. Alternatively compression was relatively slow, bringing good sound quality and little listening fatigue, but poorer speech understanding. Various authors have made suggestions as to how to optimal match the individual user to the compression system (Neuman et al., 1998; Gatehouse et al., 2003). Many of these recommendations relate to an estimate of the sorts of sounds environments the user is most likely to experience.

We recognize that different sound environments require different behaviour by digital compression systems. Through True Dynamics, Epoq is the only hearing device that offers two separate compression systems (see Figure 7) working in parallel, acting to overcome this classical compromise – one part acts relatively slowly and has 15 channels – whereas the other is relatively faster and has 4 channels. The True Dynamics system adjusts the contribution of the two compression systems constantly, depending on the character of the listening situation, to preserve speech understanding – while preserving sound quality and keeping listening fatigue to a minimum.

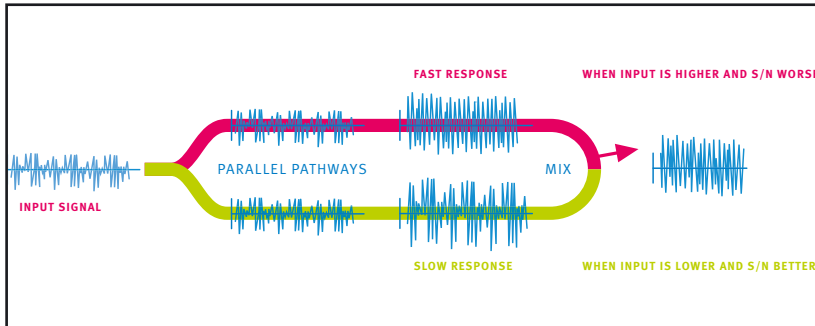


Figure 7: A diagram of the parallel processing tracks in the True Dynamics compression system.

In soft to moderate sound environments, when the signal-to-noise ratio is usually good, the system responds more slowly in order to try to maximize sound quality and minimize listener fatigue. In these softer environments, fast acting compression has the tendency to make a lot of softer, ambient-like sounds more audible than they need to be, leading to a sense that there is simply too much sound. The clicking of a keyboard, the humming of the refrigerator or tires on the pavement all tend to be louder than us natural, leaving the patient with a sense of hearing far too much and disrupting the normal relationships of sounds in quieter soundscapes. A slower response system, especially when implemented in 15 channels, will ensure that the average level of speech in the environment is fully audible, but the need to fully amplify every soft phoneme is neither needed nor appreciated by the user. The S/N tends to be good (Pearsons, Bennett & Fidell, 1977) so accurate speech understanding is typically not an issue.

As the level rises and the S/N (typically) becomes poorer (Pearsons, Bennett & Fidell, 1977) the contribution from the faster acting compression path increases. The purpose is to ensure as much speech information as possible is provided to the ear. In these more challenging situations, sound quality typically is not the issue. Rather, the patient is searching for what ever speech cues can be found

embedded in the noisy background. Faster acting compression helps to maintain audibility across the range of the device. In addition, noisy, louder situations will often have intensity spikes (laughter, door slamming, etc.) The fast acting system provides excellent protection against high level transients, yet will quickly recover, avoiding the pumping behavior that is typical of slower acting systems in louder environments.

While multiple channel hearing devices have allowed us to better match to targets, the key aspect of multiple independent channels of compression being able to better divide up the sound scene has not been met. In fact, in conventional digital signal processors, after four channels there is no real benefit of adding additional channels to the hearing device in terms of speech understanding (Moore & Glasberg, 1986; Stone, Moore, Alcantara, & Glasberg, 1999). One key reason for this is the limitation of multiple channels acting independently of each other. Multiple fast acting channels will carve up the sound scene and create spectral smearing across channels as they compress in response to signals rapidly varying in terms of both intensity and frequency. To solve this significant sound quality problem, the method of choice is to couple the channels together so that they act more like a limited channel number instrument (e.g. 2-4) than the full number of potential channels (e.g. 12-20).

Therefore, for faster more spectrally diverse signals, a limited number of channels would provide the best solution. Conversely, when the input varies over a slower time domain a greater number of channels would provide significant advantages in terms of sound quality and the ability to better map the underlying environment.



In Epoq, the faster compression path uses four independent channels to maintain spectral and temporal fidelity of the speech signal and many real-life background sounds as they change dramatically and rapidly. In the slower compression path, 15 channels are used to match the signal most appropriately to the user's remaining dynamic range. This optimal shaping can be achieved without concern for negative effects of distortions since the listening conditions are typically much better.

Extreme Bandwidth

Improvements in transducer technology has led to a steady improvement in bandwidth of hearing devices over the last two decades. However, since the Articulation Index (Kyter, 1962) suggests that there is little speech information above 5 to 6 kHz., the need for extreme bandwidth hearing devices has been

questioned. This analysis forgets an important fact about how the human auditory system works. We use information in the frequency region above the bandwidth in current hearings instruments to help us localize where sound comes from. There is speech energy up in those frequencies, just not that which carries unique phonemic information. However, this energy (Figure 8) can be used to locate the talker in space (Best et al., 2005).

Epoq offers audio bandwidth well beyond the standard among hearing instruments - extending as far as up to approximately 10 KHz. These high frequencies are essential for accurate localization . Knowing the location of a talker is the first step in understanding what is being said in a difficult listening situation (Arbogast, Mason & Kidd, 2005). The human is equipped with an amazing ability to locate the talker and focus attention on the speech message.

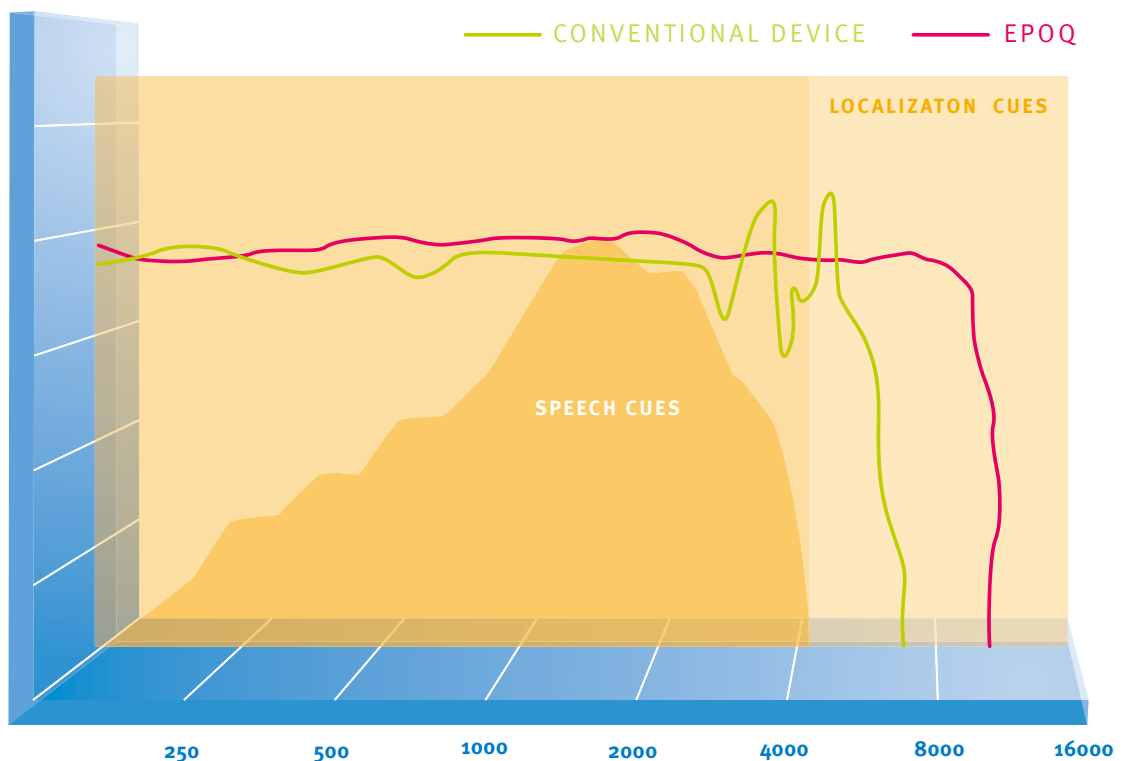


Figure 8: A comparison of the location of phonemic information versus spectral information used to help localization.



Front Focus

Our external ears are shaped the way they are for a very important reason: to improve our sensitivity for sounds arising from the front. The grooves and ridges create an acoustic “fingerprint” that the brain uses to determine the location of the sound source. When sound is picked up not in the ear canal, as happens when listening without amplification, but over the top of the ear, we lose this natural directivity (Figure 9).

Front Focus is a feature in the BTE and RITE versions of Epoq. When the hearing device is operating in the omni-directional mode, Front Focus functions to restore the natural, forward-bias of the external ear that is lost by moving

the microphones to above the ear instead of in the ear canal. By emulating this shadow effect of the outer ear – the “pinna affect” – Front Focus acts to restore the natural directionality of the ear, and thus better preserve the natural ability of the user to make correct distinction of sound coming from the front or back.

Front Focus benefits the user not only through contributing to the naturalness of the soundscape, but also by making it easier to identify the direction of sound sources immediately.

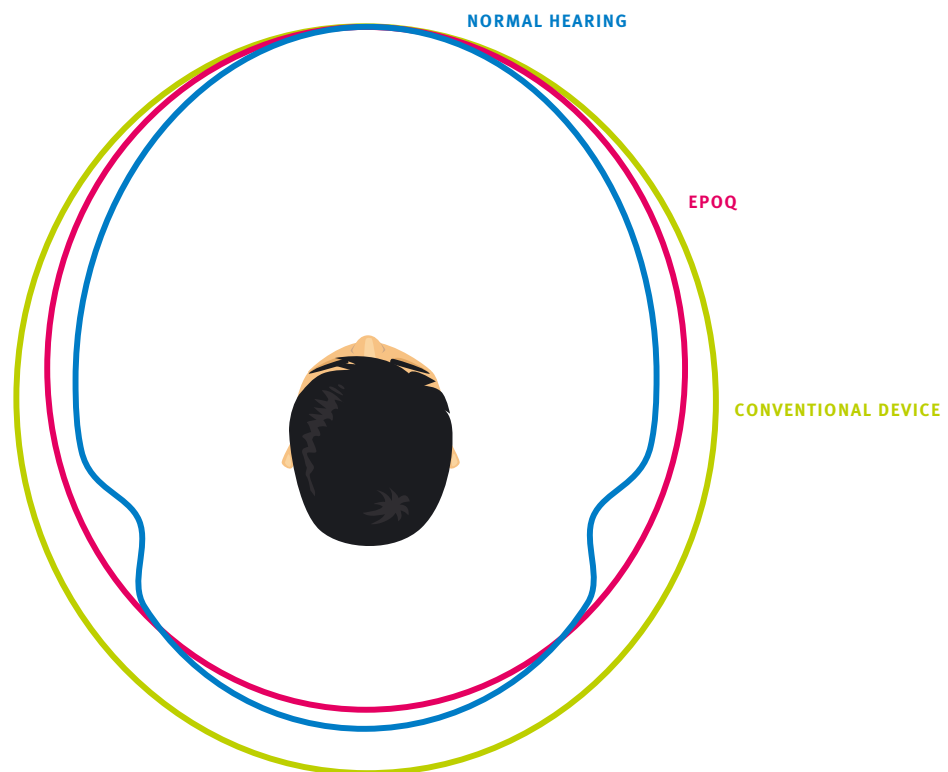


Figure 9: Sensitivity pattern of the unoccluded ear (blue line), when a standard BTE in omni is worn (green line) and with Epoq and Front Focus (red line).



Binaural Dynamic Feedback Cancellation

Dynamic Feedback Cancellation has taken a serious upgrade with Epoq. Through use of Binaural Broadband, our enhanced DFC system uses feedback detectors on both ears in combination, allowing the system to make far more precise feedback detection, avoiding “false positives” and common artifacts of conventional DFC systems. Outside, high-frequency tonal sounds such as musical instruments or alarm systems can trick a DFC system into thinking the sound is feedback. The cancellation system then creates a rogue cancelling tone that can be heard by the user and those nearby.

In Epoq, if “feedback” signals with similar spectral characteristics are detected simultaneously by both devices, the binaural system will register that signal not as feedback but as an external signal and put a hold on the unnecessary cancellation tone (see Figure 10).

This enhanced system allows users to enjoy feedback free superior sound quality, especially when for instance listening to music with many pure tones.

My Voice

With the My Voice feature in Epoq, the user will experience a more stable sound picture in noisy environments as, for the first time ever, a hearing instrument is able to distinctly identify the user’s own voice – and act upon it. It is based on the well known fact that the level of sound drops by 6 dB for every doubling of distance. The front microphone of the directional system in Epoq will be closer to the user’s mouth than the rear port. These two measurement points are close enough to the source of sound that the level drop as sound moves from the front to back port is detectable. However, sounds that originate from farther away are not changing dramatically enough in level by the time they get to the hearing device to have different levels at the front versus rear ports. Thus, Epoq can tell if the speech it is detecting is coming from the user or someone else in the vicinity. Through My Voice, Epoq will adjust its sound processing dynamically, depending on whether the user speaks or not.

When a conversation bounces back and forth between the hearing device user and others, the S/N at the hearing device changes dramatically (Figure 11). For a given background noise level, the user’s own voice will be picked



Figure 10: Compared to conventional, independently operating hearing instruments (left figure), Binaural DFC compares the potential feedback signals at both ears before initiating cancellation (right figure).

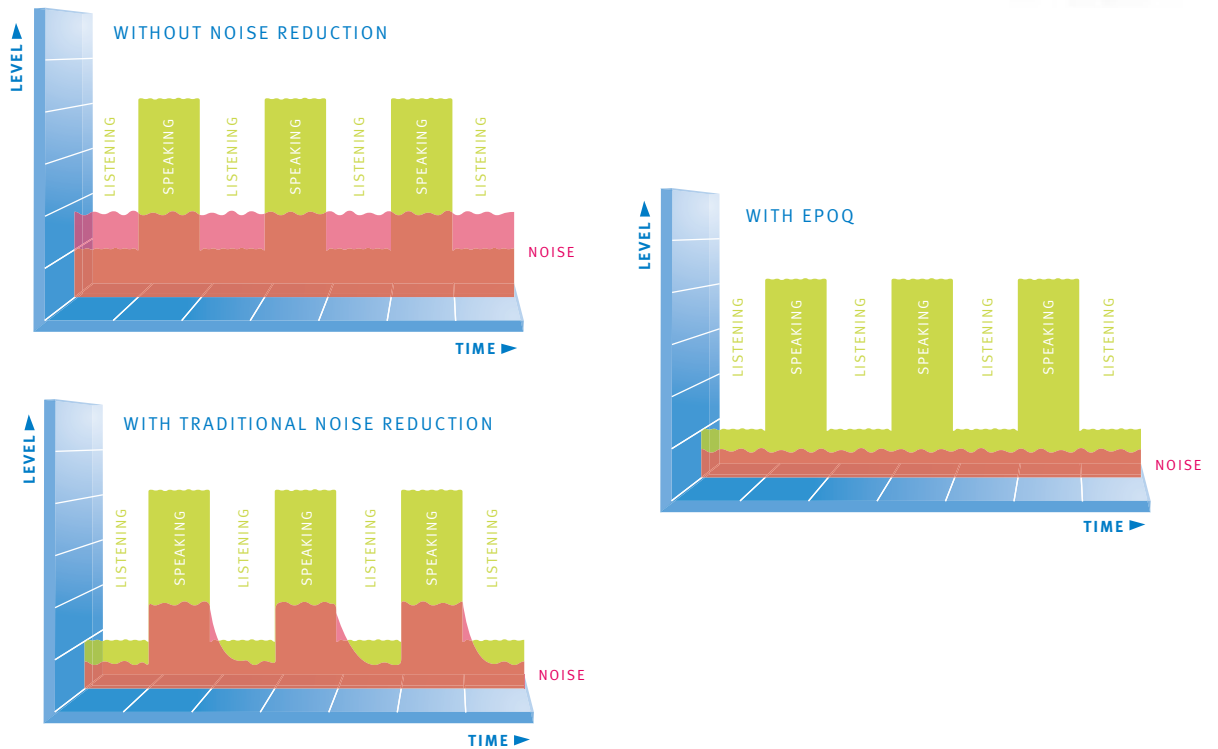


Figure 11: The signal-to-noise ratio measured by the hearing instrument is very different for the speaker's own voice than for other voices. This confuses the noise management system of traditional instruments, as the noise level jumps up and down as the conversation switches between the user and the communication partner. As Epoq knows when the user is speaking, it maintains an accurate signal-to-noise picture.

up at a much higher level than the voices of others in the conversation. The TriState Noise Reduction system, like all noise reduction systems in modern hearing devices, is constantly monitoring and reacting to the S/N seen by the hearing device. Instruments up until now are constantly reacting to these short term S/N changes even though the background noise is actually relatively stable, releasing noise reduction when the user speaks only to only to re-apply it once the conversation bounce back to one of the other participants. This makes the perception of the background and sound environment very disruptive and more uncomfortable than necessary.

My Voice makes sure that the instrument keeps the same settings, no matter who speaks, providing a more natural and smooth sound experience as the level of the back-

ground noise is stabilized. This brings the user great benefits in the most loud and noisy communication situations. The user will experience a more consistent and natural sound picture throughout the conversation – as it appears unaffected by the user's own voice.

Life Learning

Recently, the importance of the relationship between the user's auditory ecology and preferences for hearing instrument settings has been investigated (Gatehouse, Naylor, & Elberling, 2003; Jensen, Petersen, & Nielsen, 2005). Unfortunately, this interaction between the user and the environment is very individual and cannot be predicted from the audiogram. While preferences based on previous instrument selection and questions about audiology ecology provide some direction in terms of preferences, the only method to fully



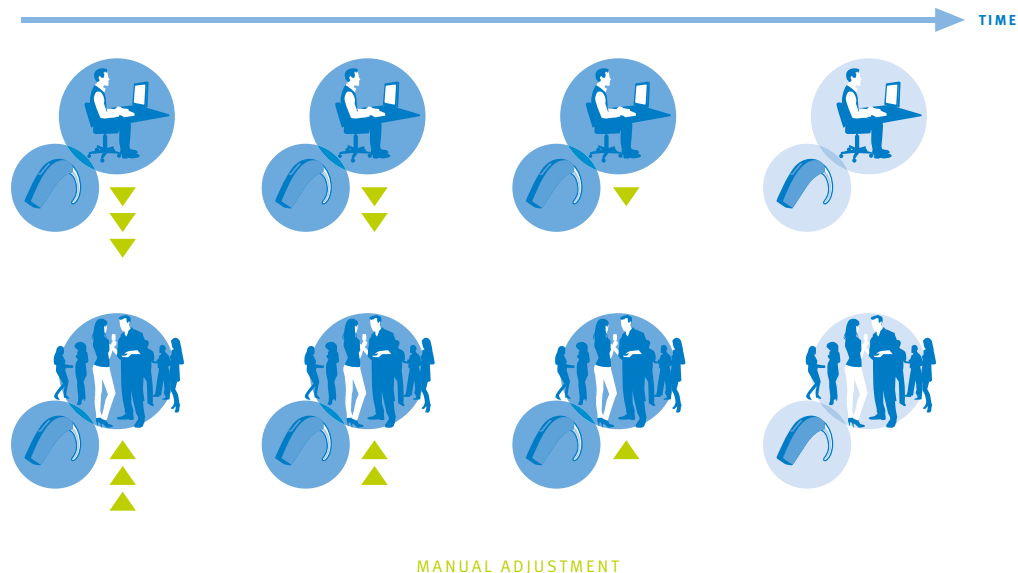
evaluate the user's environment is through the performance of the hearing instrument in the real-world. Datalogging and Envirograms provide one avenue to facilitating a better understanding of the way the hearing instrument works (Flynn, 2005). Building on this foundation, Epoq allows new flexibility for the hearing device to better match the unique listening environment of the user.

It addresses two key areas:

- VC preferences are continuously learned across different programs and a fine grained web of different listening situations (Figure 12). Epoq not only logs data but also remembers and applies volume control changes. Unlike previous systems, Epoq acknowledges that adjustments need to be interpreted carefully and adjusted over a fine scale, rather than as an overall change or a change in a limited number of sound scenes. Epoq will analyse the user's volume control changes across numerous modes depending on the presence of speech, the degree of background noise and also the level.

True Dynamics is optimized for the user's specific lifestyle, based on actual complexity of the acoustic environment. Users vary in terms of the sound environments that they experience and in terms of the effectiveness of various different compression approaches in those environments (Gatehouse et al., 2003). Based on our accumulated knowledge of the level distributions and proportion of speech and noise the typical hearing device user experiences, we have set the dynamic response of the compression system. If, over time, it becomes clear that the user experiences sound environments that are either more or less dynamic than average, adjustments are made to the mixing of the faster and slower components of the True Dynamics system.

Learning enables Epoq to build on the personalised prescription and further individualise the setting of the hearing instrument. The key is that Epoq adapts the performance of the hearing instrument across a number of automatic functions to ensure the best possible performance. Importantly, this adaptation occurs throughout the lifetime of Epoq ensuring that Epoq will continue to adjust to the client's changing lifestyle.



MANUAL ADJUSTMENT

Figure 12: Life Learning tracks volume control usage for different listening situations and adjusts both Loud and Soft Gain accordingly over time.



Enhanced Audiological Features

As discussed above, the RISE architecture is the first hearing device architecture to build in connectivity from the ground up. This connectivity allows the two hearing devices to share information at broadband speed. This ability allows true binaural sound processing. In addition, it allows us to enhance technologies such as Multiband Adaptive Directionality and TriState Noise Management, that have until now defined the state-of-the-art in advanced signal processing in hearing devices. These functions have been upgraded to work more accurately when used in a binaural system.

A key advance of coordinating the processing of two hearing devices is to be able to increase the robustness of the detector systems to reduce the likelihood of false positives. For instance, the detection of own voice is synchronized between the ears to improve reliability

(less false detections). For example, a person whispering or talking close to one of the ears will not be detected as own voice. Similarly, binaural communication is used in the control of the Dynamic Feedback Cancellation system to minimize artifacts from external pure tone sources (e.g. music).

In addition, better stability of the automatic systems can be expected. For example, when considering the Multiband Adaptive Directionality system (Figure 13), if one ear is close to the directional mode threshold then it may switch between modes with small variances in the sound scene. The information from the contralateral hearing device can confirm that directionality is indeed beneficial in this situation and therefore the ear will remain in a directional mode.

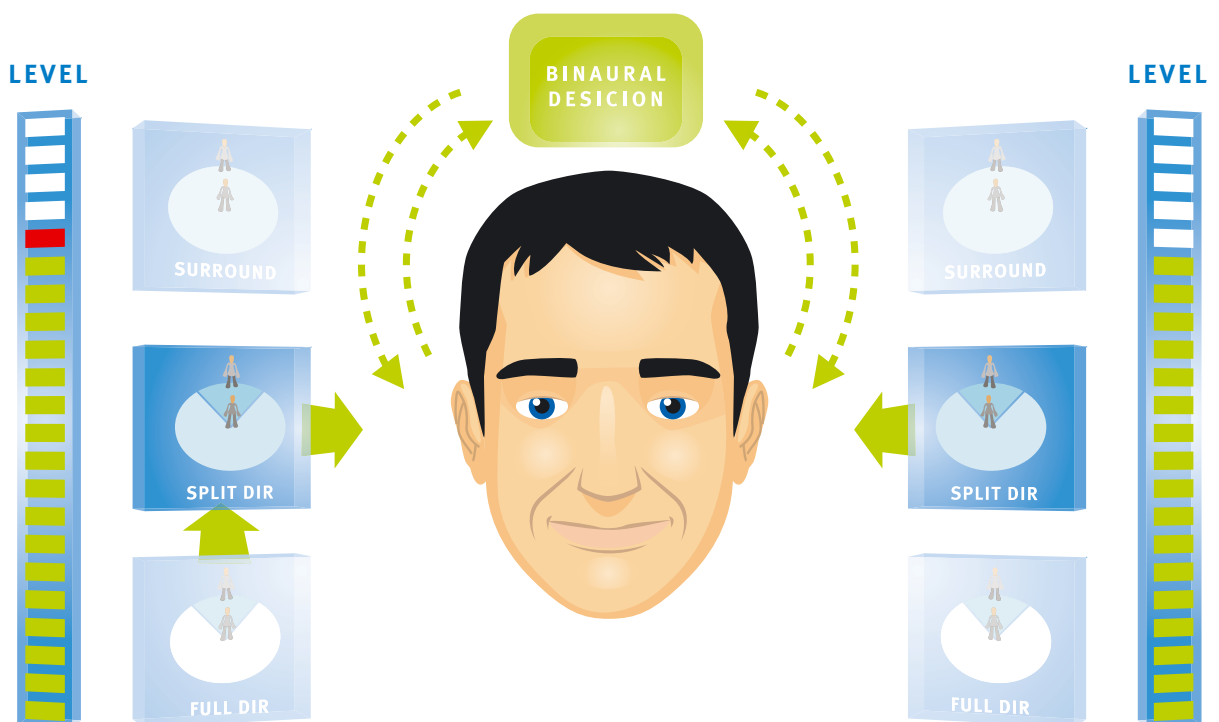


Figure 13: The mode of the Multiband Adaptive Directional system is coordinated across the two ears.



The final and most straight forward benefit of communicating between the ears is the coordination of function. Despite the development of fully automatic hearing devices, a significant percentage of hearing device users wish to exert a degree of control over their hearing devices, or alternatively, have listening needs that require the use of a dedicated or specific program. Unfortunately, one of the problems when hearing devices are worn bilaterally is that it can be an inconvenience to change the volume control or program settings on both ears. Epoq provides the user with the ability to control the action of both hearing devices simultaneously (program and volume control settings).

As seen in figure 14, a very clear correlation exists between communication speed – and the value it can provide. Relatively slow wireless communication speed in hearing instruments enables benefits from simple

synchronisation between instruments including volume control or programme shift – or shifts in slow changing hearing instrument features, such as directionality.

EarStream, the high speed wireless technology in Epoq enables the next step - bringing clients the benefits of true binaural processing, where ultra fast acting communication between instruments, including coordinating for instance compression and DFC systems. Furthermore EarStream enables real time digital audio streaming. Hereby finally the world of sound carried through electronic media – including mobile phone and music from MP-3 players, plus much more – can be made available to users – streamed directly into the ear.

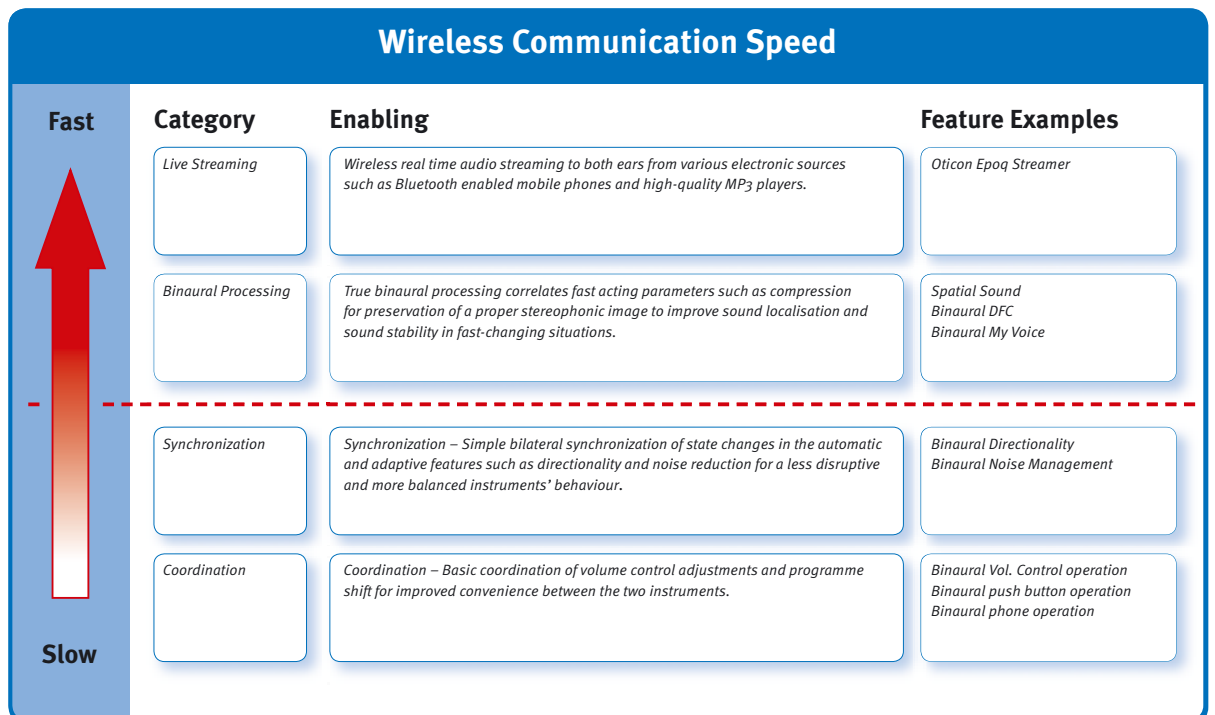


Figure 14: The relationship between wireless speed, and the functionality – and value it can provide.



The New Client Experience

We have known how to correct for hearing loss and control the effects of noise for many years. The core technology in Epoq allows us to take the next great leap in the benefits that we can provide the user.

- **Better Sharing of Information:** We benefit greatly from the duality of our senses... our experience of vision and hearing for instance, is dramatically improved when we can rely on both our eyes and ears - not just one. The Binaural Broadband technology allows for exchange of information at broadband speed. Due to limited bandwidth, previous binaural systems have only shared rudimentary setting information. By entering the high-speed world, Epoq makes the long desired dream of binaural processing a reality.
- **Better Preservation of the Rich World of Sound:** By sheer brute force, hearing devices can make sound loud enough to hear. But this aggressive sound manipulation can also distort and disrupt the appearance of the soundscape – squeezing the complexity of life into a mono-dimensional acoustic signal. Only the most sophisticated technology – found in the Epoq features of My Voice, True Dynamics and, most importantly, Spatial Sound – can enhance sound to make it available to the user yet still preserve the full range of naturally occurring subtlety so important for full appreciation of the soundscape.
- **Better Access to the World of Sound with Streamer:** Epoq has been conceived to provide the hearing impaired user with full access to all the sounds in the world, no matter how they reach us. From now on, hearing instrument users can access sounds from mobile phones and other Far World signals as effectively as can the normal hearing person.

Final Thoughts:

Getting what we expect from hearing devices

Before feedback cancellation was possible in hearing devices, we simply assumed that the ear canal had to be closed off. It became an expected part of hearing device design and the patient's experience. We discussed fine tuning and patient counseling approaches to minimize the effects of occlusion. When feedback cancellation was near to becoming a reality, the expected application was to reduce feedback in high gain fittings. It required us to step back and imagine what feedback cancellation could mean to hearing device design for us to create the concept of Open Ear Acoustics. In five short years, open fittings have become the default fitting approach across the industry, with a closed fitting only being used as a last resort.

Where are we now? We have developed a long list of signal processing techniques and fitting practices designed to optimize the hearing device one ear at a time. It takes our imagination to see how these systems can be vastly enhanced when we consider the binaural effects. Each client comes to us with amazing abilities to take in information and make sense of it. By viewing the hearing device fitting not as a correction for a peripheral loss, but as signal enhancement for a binaural system designed to take two inputs and create a single accurate percept, we can re-design the role of the hearing device. The sounds around us – the Near World – are integral in providing a sense of time and place. Amplification should be seen as a way to fully imbed that client in that world.



How do we counsel about phone use? “Get an amplified phone.” “Hold your mobile phone at this angle by your ear to hear the signal and not the distortion.” “It may be better to take your hearing device off.” These are not access solutions, they are workarounds. It takes our imagination to see clients with access to the Far World that is not limited in any way. Access to electronically mediated sounds should not be a hassle for the user. Access should be as simple as access to sound from the immediate environment. It should be a natural part of hearing device use.

Epoq redefines the role of a hearing device. It is the client’s interface to the worlds we are all a part of.



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Appendix A: What we know about Spatial Perception

Organizing the World of Sound

The human sensory systems function in such manner that we automatically and subconsciously use what ever information we can to organize the world around us into meaningful objects and events. We normally are not consciously aware of the individual auditory, visual, olfactory or proprioceptive stimuli that are constantly bombarding us. Rather, we are aware of the things and happenings all around us, especially those that are meaningful to us. Driven by Gestalt principles of organization (Moore, 2001; Neuhoff, 2004), this organization and analysis function is absolutely vital for us not to become overwhelmed by too much stimulation.

In the auditory world, the terms Stream Segregation and Scene Analysis (Bregman, 1990) have been coined to describe how we turn massive amounts of individual sounds into usable information. It is important to remember that, except in psychoacoustical labs, sounds are not created without context. Sounds are the result of activity: generated by humans, nature and machines. Importantly, sounds occur over time. Naturally occurring sounds simply do not start and stop instantaneously. Rather, they are the result of something happening. Stream Segregation refers to the cognitive activity of linking individual sounds over time to a common source.

The most common example of Stream Segregation is speech. People don't typically produce one phoneme at a time, except for example the utterance "oh?" (or, in Canada, "eh?") Speech is comprised of a series of phonemes that form words and words that form sentences. Speech does not take on meaning until these isolated speech sounds are grouped

into meaningful wholes. The listener has to be able to track these sounds over time, assigning them to the same source, so that the meaning can be uncovered.

Scene Analysis refers to the cognitive activity of deconstructing the totality of sound input into isolated sounds sources arising from specific places in space. It is the way we perceptually organize the world of sound around us into meaningful objects and events, with specific spatial relationships.

These two skills combine to allow us to understand and appreciate the world of sound around us. We use the abilities – which are normally automatic – to develop an immediate understanding of where we are in reference to the rest of the world, to recognize what is happening all around us and to focus our attention on the sound sources that we choose. When faced with the task of listening to speech in a challenging environment, Stream Segregation and Scene Analysis allow us to find and follow the speech of the talker we are interested in, suppressing the interference of the other sounds in our immediate surroundings (Cherry, 1953). The failure of current computer-base speech recognition devices is, in part, related to programmers' inability to replicate these incredibly powerful yet automatic processes of human cognition. Speech understanding is not an event of the auditory system, it is an event of the cognitive system.

Spatial Hearing

The main function of having two ears is the ability to resolve the spatial relationships between ourselves and the sounds around us. Neither core auditory sensitivity nor acuity is enhanced appreciably by two ears. However, our ability to tell where sound is coming from is almost totally dependent on a binaural system (Blauert, 1983).



Localization of sound sources is possible due to timing and level differences of sounds arriving at one ear versus the other (Yost, 1994). It has long been recognized that, when sounds originate somewhere other than precisely in front or behind us, the sound will reach the near ear earlier and at a higher level than the far ear. We use timing difference below 1500 Hz. and level differences above 1500 Hz. to determine where the sound came from. The timing differences are due to the finite speed of sound (sounds from 90 degrees to the side arrive at the near ear about .6 to .7 msec before arriving at the far ear) and level difference are due to the ability of the head to attenuate sounds with shorter wavelengths (by up to 20 dB). We are also sensitive to location of sounds that occur from somewhere other than the same elevation of the ears, using acoustical information in the extreme (above 5000 Hz.) high frequencies. This skill is related to the convoluted shape of the pinnae, where differences in source elevation will interact with the wrinkles and folds of the external ear in such a way to create significant peaks and valleys in the extreme high-frequency response.

Although experimentally we usually look at spatial perception in terms of simple planes (horizontal, vertical, mid-sagittal), in real life spatial hearing refers to our ability to locate sounds from any direction in the 3-dimensional space around us. Further, given that many if not most naturally occurring sounds have energy across a broad spectrum, we probably are constantly using a combination of timing and spectral cues to resolve auditory space. In addition, experimentally head position is usually held fixed but in real world situations, head movement is often used to help resolve sound source position. Most of us cannot move our external ears independently, but the animal world is full of examples of creatures who are skilled at using pinna movement to resolve auditory space.

Spatial resolution is typically measured in one of two ways. First, following good psychoacoustical practice, elemental signals are used under controlled conditions. Performance is usually measured in terms of accuracy of angular identification or in terms of just noticeable differences for angular differences. Secondly, the effect of precise spatial resolution is measured via signal identification in noise. Since one of the basic functions of the binaural system is to be able to separate isolated sources from a distributed background of noise, the masking level difference (MLD), sometimes referred to as “binaural squelch”, is a measure of the signal-to-noise ratio (S/N) improvement made possible by the binaural system compared to performance of the monaural system. The measure is made using tones, other elemental signals or speech.

In terms of angular resolutions, normally hearing subjects appear to be able to resolve differences as small as 20 degrees. In terms of S/N improvement using non-speech signals, the binaural system improves the function by 10 dB or more. Using speech, the S/N can be improved by up to 5 dB if the competition is something other than a competing talker. If the competition is one or two other voices, the improvement can be in the order of 5 to 10 db or greater with 90 degrees of spatial separation (Arbogast, Mason & Kidd, 2005; Hawley, Litovsky & Culling, 2004; Kidd, Mason & Gallun, 2005). The extra improvement clearly has a cognitive component, showing the power of identifying different speech streams in space and focusing attention on one versus another.



Based on the thinking of the Articulation Index (Kryter, 1962), we assume that speech energy only up to about 5000 Hz. is important for speech understanding. However, under natural conditions and consistent with the notions of Scene Analysis, it is beneficial to first locate the talker before speech information can be gleaned. Kidd et al. (2005) demonstrated significant improvements in speech understanding in noise if uncertainty about the talker location can be reduced or eliminated. Best et al. (2005) demonstrated that speech energy above 8000 Hz. can be used to effectively localize the talker in space. These results suggest a two-staged model of speech understanding in spatial contexts in which high-frequency speech energy is used to help locate the talker in the environment, then allowing for focused attention in order to glean speech information.

Sensorineural Hearing Loss and Spatial Resolution

There have been several investigations into spatial abilities of users with sensorineural hearing loss and the details of the findings are beyond the scope of this review. Some of the most comprehensive publications include those by Byrne & Nobel (1998), Goverts (2004) and Nobel, ter-Host & Byrne (1995).

As with many psychoacoustical skills, users with sensorineural hearing loss demonstrate significant user-to-user differences in performance on spatial resolution skills. Further, again consistent with studies of other core hearing abilities, some users perform nearly as well as listeners with normal hearing whereas other show dramatic decrements in performance. In some cases, even when the audibility of signals is assured, some users demonstrate no measurable ability to differentiate sound source location, as if all sounds come from the same location.

However, when measured using speech in noise, most patients with sensorineural hearing loss seem to have some residual skills. Although, on average, impaired patients show less benefit of the spatial separation of speech from competition, they still show performance above chance (e.g., Gelfand, Ross & Miller, 1998), especially when the competition is the speech of no more than a few talkers (Arbogast, Mason & Kidd, 2005).

The Effect of Amplification

First and foremost, if a sound cannot be heard, it cannot help the binaural auditory system to localize (Dubno, Ahlstrom & Horwitz, 2002). Therefore, the simple audibility improvements offered by any hearing device fitting is assumed to assist spatial hearing under some conditions. Subjective reports from users consistently indicate that hearing devices help with spatial resolution. Sometimes these reports are inconsistent with measured performance under controlled laboratory conditions. However, it is important to note that these user reports may likely be due to improvement in audibility for (especially) higher frequency information used to resolve auditory space. Since signals come to hearing devices users at all sorts of levels, improving the audibility of softer sounds may well improve general awareness of the different sounds in the environment. Although pure localization ability may or may not be improved, awareness certainly is.

Given the importance of spatial hearing to effective speech understanding in real noisy environments, there have been surprisingly few studies that have looked at the objective effects of amplification on spatial perception. There are some observations that form our working knowledge. However, given that the effects have not been studied extensively, especially as hearing device technology has advanced, these observations have to be treated as tentative.



- As noted above, most subjective surveys indicate that amplification, especially binaural amplification, improves the user's ability to tell where sound is coming from and to separate the speech of one talker from another (e.g., Noble, ter-Host & Byrne, 1995). Again, these reports likely represent the total combined effects of improved audibility.
- Objectively measured localization performance does not consistently seem to be improved through the provision of hearing devices (e.g., Koehnke & Besing, 1997), especially when test signal levels in the unaided condition are above threshold. However, there are both case examples and group data showing positive results (e.g., Rakerd, Vander Velde & Hartmann, 1998).
- Speech understanding in noise, as reflected in a S/N improvement, does seem to be better with two hearing devices compared to one (e.g., Festen & Plomp, 1986). Part of this effect seems to be due to binaural squelch (a true binaural effect). Sometimes this effect is due to improvements in audibility on one ear versus the other, depending on where the speech and competition come from.
- Localization performance seems to be worse in closed earmold configurations compared to more open conditions (Byrne, Sinclair & Noble, 1998). The more direct high-frequency sound entering the ear canal, the better localization. Improved bandwidth in hearing devices also helps to improve localization (Byrne & Noble, 1998).
- Advanced signal processing in hearing devices (multi-channel, WDRC, adaptive directionality and noise reduction) may disrupt localization ability (e.g., Keidser et al., 2006; Musa-Shufani, et al., 2006; Van den

Bogaert et al., 2006). However, the effects are highly dependent on test configuration and procedures. Given that these systems disrupt naturally occurring relationships in sounds, especially when comparing one ear to the other, it is reasonable to suspect some effects. At times, these suspicions have been confirmed.

Implications

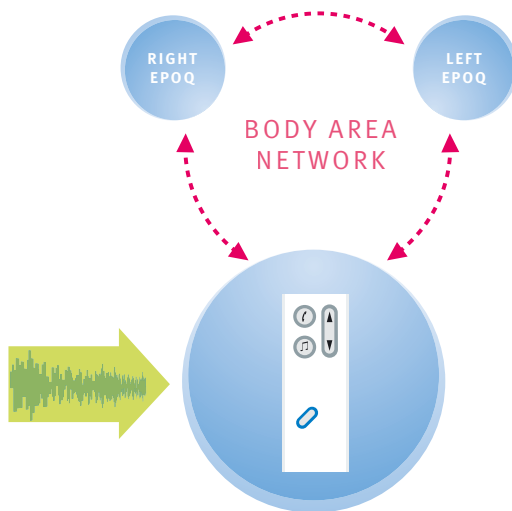
Effective spatial resolution is a highly important natural ability that listeners have to keep the chaos of the world of sound down to a workable level of information. The normal auditory system, in combination with the brain, allows us to immediately understand what is happening all around us. Spatial resolution ability is compromised to some degree for most patients with sensorineural hearing loss. However, they do usually demonstrate some residual skills.

We need to remember how amplification, especially modern signal processing techniques, interacts with the remaining capabilities of the listener. Many of our signal processing approaches are designed to radically modify sound to best match the needs of individual ears. However, this processing may disrupt natural cue sets that are used by the binaural system to resolve spatial relations between sounds in the environment. True binaural processing, as now offered in Epoq, have put us in the position of balancing the desire to optimize the signal for an individual ear with the need to maintain the natural relationships that the binaural system needs.



Appendix B: A Quick Primer on Epoq Streamer and applied wireless technologies

The Epoq Streamer is built on two wireless technologies: a wireless body area network (BAN) (between hearing devices and gateway) and a wireless personal area network (PAN) that interfaces to the surrounding applications in the near vicinity. Streamer is the physical body worn gateway device that coordinates and controls the communication of the two individual wireless technologies and ensures that the complete link is perceived as one network.



1. Body Area Network – EarStream

The technology is a dedicated magnetic wireless application – built into the RISE architecture - especially made for hearing device implementation. It enables the hearing devices to receive broadband transmission of audio and data signals at rates above 100 Kbps. The uniqueness of this wireless technology is the compactness and low current consumption that allows it to be implemented in most styles and sizes of hearing device.

The EarStream technology is well suited for a Body Area Network. The network will work within a distance of up to 1 meter for commands and control signals, and up to 0,5 meter range for continuously uninterrupted high quality audio streaming.

2. Personal Area Network – Bluetooth

Bluetooth is the technology used to extend the body area network to the peripheral audio sources in the vicinity or in home environment. Bluetooth is a global standard of digital short range wireless connectivity and is already supported by most of today's mobile phones and computers. Bluetooth is now showing up in more and more consumer devices .

Streamer offers full support to all major Bluetooth profiles, allowing all kind of relevant audio devices to connect to the hearing devices with their respective full performance and bandwidth - and as such enables the full potential of audio quality from any source to being played by the hearing devices.

In order to function and communicate, Bluetooth devices need to be connected to their correlated counterpart of the communication. Before the two devices can be connected they must once for all be prepared to this. This process is referred to as pairing. Once two Bluetooth units are paired together - they have exchanged their unique address codes - which then allow them to establish a point to point communication link. Streamer can pair to a high number of Bluetooth devices and Bluetooth enabled mobile phones.



Basics of use

When the Streamer is used to connect the hearing devices to a mobile phone, the verbal communication between the user and the target person on the line is transmitted as follows:

- From the PAN, via the BAN, the hearing devices are receiving and playing the incoming speech.
- The Streamer has a built-in microphone that captures the voice of the user and distributes this to the mobile phone via the PAN. For this reason the user must speak into the Streamer from distance of approx. up to 30 cm.

When the gateway is used to connect the hearing devices to music or information sources inside the Personal Area Network, this is a one way flow of signal why the Streamer microphone is disabled.

Good audio quality - and above all, a good link stability - is both closely related to the physical position of the Streamer when in use. Streamer's ability to provide the best possible interface between the audio devices and the

hearing devices is dependant of its placement inside the so called sweet spot. The sweet spot is an area on the upper torso of the user as depicted below.

When the streaming is discontinued by the user, commands are sent to set the hearing devices to immediately go back to the program that was active prior to the streaming scenario.

During any streaming session of audio to the hearing devices, the user interface on the hearing devices is disabled, allowing the user to concentrate the operation of all relevant controls to be carried out from the controls on Streamer.

This gives the combined advantage of always selecting the most appropriate program and full automatic operation of the hearing device with no interaction from the user necessary.

Streamer has two dedicated programs. Each program is meticulously adapted to the audio signal received to provide the optimal listening result.





The two different programs are:

- mobile phone connectivity
- music and entertainment

In both programs the user can choose to mix in the microphone signal from the hearing devices to follow cues in the surroundings or choose to mute the microphones to obtain a crystal clear listening situation.

When initiating streaming of audio, the two programs have different default settings in terms of the hearing device microphones. The music listening situation is a deliberate entered situation while the phone call on the contrary can be an unexpected situation that is entered. When shifting to a Streamer program, the mics in the hearing devices are set as follows:

| | |
|---------------------------|-----------------------------|
| Music and entertainment | MIC's are default muted |
| Mobile phone connectivity | MIC's are default turned on |

In the fitting process the microphone level relative to the Streamer signal can be configured. The standard configuration, that shows a good balance of both signals, is that hearing device microphones are damped 3 dB. This can for a specific user either be doubled or neutralized.

Priority of Services

Having a broad range of different audio sources made available to the user, it is necessary to provide a pre-determined priority of services to avoid conflicting applications. The priority chart is controlled by Streamer and is as follows (no 1 overrules no. 2 and so forth):

1. Mobile phone calls / conversations via Bluetooth
2. Audio (music) connected by wire to gateway (micro jack input)
3. Audio from Bluetooth enabled source

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People First

We believe that it takes more than technology and audiology to create the best hearing instruments. That's why we put the individual needs and wishes of people with hearing loss first in our development of new hearing care solutions.