Answering today’s bone conduction hearing challenges with new signal processing solutions and wireless capabilities

The Cochlear™ Baha® 4 Sound Processor introduces a number of sophisticated new technologies and adaptive solutions to address the challenges of a dynamic acoustic environment. Each technology has been developed and tested specifically to meet the needs of patients who use the Baha system, either through an abutment (the Baha Connect System*) or via magnetic retention (the Baha Attract System**). These technologies are designed to provide a clean signal and a smooth transition between settings, leaving the user free to enjoy a natural soundscape without having to manually adjust the processor. Combined with robust connectivity to a variety of wireless devices, it provides access to sound across the full range of listening environments. The purpose of this technology report is to provide a technical background to aid understanding of the purpose and potential benefits of the Baha 4 Sound Processor.

Introduction

The introduction of advanced signal processing and programmability in Baha sound processors have provided significant benefits in hearing performance for many users1-6. The improvements stem from enhanced fittings based on pre-defined targets combined with signal processing options such as directional microphones and noise reduction circuits. There are a number of key user needs still remaining, specifically patient access to the latest advances in signal processing and hearing device technology; designed to provide improved hearing performance in noise and easy control of the sound processor. Similarly, the hearing care professionals working with bone conduction devices, want fitting software that provides easy detection of the risk of feedback, wireless programming and simplified device fittings.

This paper describes the technical background and development considerations behind the new Baha 4 Sound Processor. The purpose of recent research and development efforts has been to address key user needs. In contrast to the Baha 3 Sound Processor family, the Baha 4 Sound Processor is based on an entirely new Digital Signal Processing (DSP) platform called Ardium™. Ardium is Cochlear’s latest innovation which provides new levels of hearing performance for bone conduction hearing solution users. It is the latest platform specifically built and adapted for bone conduction hearing. The heart of the Baha 4 signal processing is the Ardium Wireless Platform. This platform is specifically designed to address the needs of a bone conduction hearing solution. Baha PureSound iQ provides 17 channels of Wide Dynamic Range Compression compared to the 12 channels used in the Baha 3 sound processors (Figure 1). The 17 channels are logarithmically spaced to better match the organization of the cochlea, providing a more realistic adaptation of the amplified signal to the cochlea. The 17 channels still use the same 10 audiological bands in the software to ensure fitting efficiency. This benefits the patient from both the ability to better match the required gain to the hearing loss through access to more channels as well as from the improved organization of channels and increased processing resolution.

Matching technology to hearing needs

Most Baha patients have bone conduction hearing that is within or near normal limits7. This type of hearing profile is indicative of very little if any inner hair cell loss, which requires very high sound quality from the hearing solution. Some adaptive signal processing systems produce audible artefacts which affect sound quality perception and hearing performance for people with near normal cochlear function. The Baha 4 Sound Processor has a fitting range up to 45 dBHL which is ideal for many of these Baha patients.

This paper describes the advanced technological options used to ensure the features in the next generation Baha sound processors better meet the listening needs of people with conductive hearing loss, mixed hearing loss or single-sided deafness.

* The Baha 4 Connect System consists of: Cochlear Baha BI300 Implant, Cochlear Baha BA400 Abutment (DermaLock), Cochlear Baha sound processor
** The Baha 4 Attract System consists of: Cochlear Baha BI300 Implant, Cochlear Baha BIM400 Implant Magnet, Cochlear Baha SP Magnet, Cochlear Baha sound processor
Does the listener wish to actively listen or to tune out in this situation? Is the speech level quiet or loud? System will establish a relative mix of those variables. Does noise dominate the listening situation? Is the speech and/or noise, this system will establish a relative mix of those variables. Does noise dominate the listening situation? Is the speech level quiet or loud? Does the listener wish to actively listen or to tune out in this situation?

**Scene Classifier**

Improving the signal-to-noise ratio for Baha users is the goal of the underlying signal processing technology. This is effectively accomplished in a system that automatically cleans the signal, while seamlessly transitioning between different sound processor settings, to enhance the listening experience for Baha users. Previous systems set the sound processor settings utilising decision-making methods based on speech and noise characteristics with a focus on providing the best possible speech-in-noise performance at all times. A limitation to these previous approaches is a system in which the transition from one state to another can become obtrusive for the listener. To maximize signal to noise ratio, Baha users require a system that can recognize different listening situations and react accordingly to match the needs of the listener in that situation.

To meet these user needs, the Scene Classifier was developed. It precisely identifies key listening environments as shown in Figure 2. While previous systems simply focused on speech and/or noise, this system will establish a relative mix of those variables. Does noise dominate the listening situation? Is the speech level quiet or loud? Does the listener wish to actively listen or to tune out in this situation?

The new Scene Classifier uses sophisticated speech and noise detection algorithms based on a number of factors including the frequency content, spectral balance and the temporal properties of the incoming sound to categorize the input into seven different listening environments. Of course, not every listening situation can be compartmentalised into one of seven distinct scenarios. In the event that an environment does not meet the full criteria or when a listening situation shifts between two or more adjacent environments, the algorithm estimates the best combination of settings. In those cases, the algorithm gradually and seamlessly adapts the settings for each distinct listening situation. Therefore, the Scene Classifier meets the needs of the Baha user by providing the best possible set of features while at the same time ensuring that any transitions or feature selections are performed as unobtrusively as possible.

Additionally, the Baha Fitting Software 4.0 (BFS) helps the hearing care professional and patient to understand the listening situations and actions of the Scene Classifier through the Data Logging. Data from the Scene Classifier is stored in the sound processor, which provides an overview of the sound environments the patient has been in. This information combined with the program use in each environment provides useful input for counseling and fine tuning. (Figure 3).

**Noise Management II**

One of the Scene Classifier’s key benefits is its ability to control the Noise Management II system. As described by Flynn, previous generations of Baha sound processors use noise reduction systems to preserve speech intelligibility while reducing undesirable background noise. The goal of any noise management system is to provide a more comfortable listening experience in noisy or loud listening situations. However, a balance must be found, because a noise management system that is too aggressive may lead to a less natural soundscape as the listener may perceive background noise as being too soft. In addition, the unique indications for bone conduction require a different noise management system for patients with conductive/mixed hearing loss versus those with single-sided sensorineural deafness (SSD).

Typically, these previous noise reduction systems were based on modulation detection and applied a preset amount of modulation-dependent noise reduction across all environments. Unfortunately, a study by Zakis and his colleagues found that patients preferred a...
system where the degree of amplification reduction differed across listening situations rather than remaining constant. Second, they were based on modulation detection. Modulation detection analyses the level fluctuations of the input signal and assumes that as the fluctuations decrease the SNR is also poorer. Therefore, it makes sense that if the estimated SNR is poor at a given frequency then gain should be reduced in that channel. While good in theory, a study by Bentler and her colleagues9 found that while being generally beneficial, different systems reduced the gain by different degrees in the same listening situation. Although this is effective, the challenges of trying to provide suitable settings for varying situations need to be met.

Noise Management II addresses these issues through a number of technological advances. Using the input from the Scene Classifier, the amount of gain reduction applied is adjusted based on the patient’s listening situation. The degree of reduction is also adjusted by frequency depending on whether speech is present or not. Noise Management II introduces a sophisticated strategy based on spectral subtraction in combination with channel-dependent signal-to-noise ratio analysis. Spectral subtraction aims to subtract the short-term noise spectrum from the total signal, which should leave only the speech portion (Figure 4). In addition, the degree of subtraction must be tempered so that not all background noise is removed. This process places increased demands on the DSP and has only recently been made possible by the Ardium platform.

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Improving the way directional microphones work will continue to benefit Baha patients. Directional microphones have some drawbacks in terms of sound quality that need to be addressed for the hearing impaired listener. Directional microphones roll off low frequencies. Many signal processing systems compensate for this roll off by increasing gain which may increase the noise threshold and may decrease benefit12. To address this issue, a new signal processing system was implemented in the Baha 4 Sound Processor whereby the low frequency region retains omnidirectional processing while adaptive directional processing is applied to the high frequencies (Figure 5). By preserving low frequency sounds, the wearer can continue to take advantage of the natural phase differences between the ears, which aids in sound localisation13. Importantly, the so-called blending point where the low frequency omnidirectional signal blends with the high frequency directional signal is set around 1000 Hz to ensure clearest amplification of high frequency speech information in noisy listening situations. A number of studies14-15 have examined the situation where the omni and directional signals are blended and have reported significant benefits in terms of preference and sound quality with no significant loss of hearing performance.

Another improvement to the Baha 4 directional system is that the Active Balanced Directionality automatically applies different directional patterns based on the input from the environment. One criticism of conventional directional microphone systems is that they require a compromise between a narrow or wide beamwidth. Should you have a narrow beamwidth for good listening to one speaker
which comes at the expense of poorer hearing from the edges of the directional beam? Or should the beam be broadened to improve the soundscape to the possible detriment of hearing performance? In the Baha 4 Sound Processor, the Active Balanced Directionality overcomes this compromise. The beamwidth is automatically adjusted based on the speech detection algorithm and the relative level of sounds entering the two microphones (Figure 6). The beamwidth narrows with a stronger signal from the front but widens if the input level at the front microphone decreases, thus improving the audibility of surrounding sounds. In short, this provides the effect of focusing on one speaker or expanding its range to focus on multiple speakers when in a group conversation.

To compensate for the location of the Baha sound processor posterior to the ear canal, Cochlear pioneered Position Compensation functionality, which provides a slight directional tilt of the polar plot in the omnidirectional mode, enabling a more natural weighting between sounds from the front and the back of the listener. (Figure 7). This has been shown in previous tests to provide significant benefit in terms of speech understanding and sound quality. To further improve this technology, attention was focused on the directional system by better matching the intended hyper-cardioid pattern (Figures 8 & 9).

In summary, the Baha 4 Sound Processor delivers improvements in the directional system as well as position compensation. For the Baha user, blending the omnidirectional and directional signals combined with position compensation II providing a more accurate directional beam is designed to improve sound quality and hearing performance in noise. Similarly, the ability to automatically adjust the beamwidth of the directional system depending on the listening situation, also helps meet this goal.

Feedback Analyzer and Dual Track Feedback Manager

It is well understood that Baha systems are more affected by feedback than conventional hearing aids and that the solution is probably not as simple as merely applying a hearing aid solution to the Baha Sound Processor. For the most part, feedback in a hearing aid is typically focused around 3000Hz. For the Baha Sound Processor, the feedback source is generally broadband and can be of acoustic or mechanical origin. The source of the feedback is not solely related to leakage from the transducer back to the microphone and may relate to feedback originating from a mechanical or acoustic source, skull radiation, soft tissue interference, or other variables that are known to play a part in the feedback pathway for Baha sound processors.

In order to overcome these issues, the Baha 4 Sound Processor implements next-generation feedback reduction technology that is a clear improvement on the phase cancellation system that had been applied in previous applications. Sound processors are used in dynamic everyday environments, this requires a feedback system that can adapt to individual requirements (e.g. skull radiation) as well as situational challenges (e.g. raising a telephone to the sound processor microphone). Because feedback is dynamic and highly variable it is best addressed through a feedback reduction system that is flexible enough to reduce feedback without producing signal processing artefacts when the signal is erroneously interpreted as feedback.
To provide improved stable gain and reduction in sound anomalies, three important system requirements were identified for the Baha 4 Sound Processor: (a) ability to determine the individual’s unique feedback pathway, (b) ability to increase the stable gain margin and (c) ability to quickly adapt to dynamic changes in the feedback pathway. These three factors require resolution to provide a consistent reduction of feedback issues.

In the BFS 4.0, the hearing care professional can now use the Feedback Analyser to measure the individual patient’s stable gain (Figure 10). This can be done in any configuration, regardless of whether it is on an abutment, Softband or magnet-based solution. The measurement of individual stable gain (ISG) will result in each patient receiving a unique estimation of the feedback path and clear definition of the frequencies at which there may be a good or poor feedback margins. This will help ensure that the sound processor fitting has less feedback annoyance and a more precise determination of the feedback’s origin.

The individual stable gain curve offers the hearing care professional guidance as to where it is safe to provide amplification. However, the system needs to be designed to increase the limits of the stable gain area as well as cope with dynamic changes in the feedback pathway. In order to achieve these goals, a system with at least two distinct pathways is required. The Dual Track Feedback Manager uses two separate phase cancellation tracks to increase the available individual stable gain with reduced artefacts and improved overall feedback performance (Figure 11). The Slow Adaptive Track compensates for changes in the orientation of the sound processor or position of the Baha Softband. This continuous monitoring ensures good feedback performance during daily activities. The Fast Adaptive Track creates two separate inverted signals, one for each microphone. It continually adapts to cancel out feedback from sudden changes in the feedback paths, e.g. hugging or putting on a hat. The new Feedback Analyzer in the Baha Fitting Software 4.0 and the Dual Track Feedback Manager work together to minimize the risk of feedback in the new Baha 4 Sound Processor.

Outdoor programme with WindShield™

Wind noise is frequently reported as a challenge for Baha users. The previous generation (Baha 3) sound processors introduced a dedicated programme that was designed to reduce the impact of wind noise while preserving hearing performance. Based on the success of that approach research and development efforts have been focusing on the continued development and investigation of technology to provide further improvements.

The first consideration for Baha sound processors is to acknowledge the poor location of sound processor microphones in terms of wind noise. With reference to Figure 12, which shows a manikin head inside a wind tunnel, it is clear that the position of the Baha sound processor is in a challenging position due to vortices that are created behind the pinna18. Fortunately, the effect of wind noise is typically restricted to the lower frequencies with a spectral peak around 100Hz and can be measured and therefore addressed by specific signal-processing technology.
While mechanical solutions, such as covering the microphones with a fine mesh as used in the “front” of the Baha sound processor, will reduce the effect of wind noise to a certain degree, sound processing is required to address the noise that “slips” through. An option for a patient using a Baha 4 Sound Processor, is to enable the WindShield™ technology - a new, intelligent wind noise detection and reduction system that aims to reduce the effect of wind noise on the user’s hearing experience. The objective of WindShield is to reduce sufficient gain in the channels where wind is detected to increase listening comfort while at the same time preserving gain in those frequency channels that are not disrupted. Unlike the solution in the Baha 3 Sound Processors, the Windshield technology will (a) automatically apply the technology when required and (b) will adjust the degree and frequencies of reduction in relation to the degree of wind noise present. The result is improved sound quality since the annoyance of wind noise will be decreased, while preserving the important and undisturbed aspects of the signal.

WindShield detects wind and reduces wind noise in three steps (Figure 13):
1. No wind detected. No gain reduction is applied to the microphone signal. In this step, the system monitors and stores the average loudness in each frequency band when no wind noise is present, as shown in figure.
2. Wind is detected above the 70dB SPL detection point. By utilising the fact that the turbulence caused by wind noise is highly uncorrelated, the system will detect wind noise if there is a significant difference between the input levels of the processor’s two microphones. Once wind noise has been detected, the average loudness levels are stored.
3. Gain reduction in relevant bands. Once wind noise has been detected, the system reduces gain in the affected bands (typically in the low frequencies) but only to the average levels previously stored.

By limiting gain reduction in this way, wind noise reduction will not diminish the audibility of desired sounds. Thus the amount of gain reduction is intelligently adapted to the environment and level of wind noise and the user will benefit from a natural sound experience where the effect of wind noise is limited while the audibility of other sounds is preserved.

The amount of wind noise reduction is ultimately based on the wind-to-sound ratio. Taking the environment sound level from step #1 and comparing it with the level of wind noise as determined in step #2 will provide a wind-to-sound ratio in each channel. The degree of gain reduction in each channel is determined by this ratio. It is important to ensure that the system is fast enough to be able to adjust to the sudden changes in the environment (the system’s reaction time is 250ms). This is fast enough to adjust to the change, but not so fast (i.e. slower than the compression speed of the WDRC circuit) that it introduces artefacts or other perceived distortions into the overall soundscape.

The key benefit of WindShield is to reduce the annoyance of wind noise without reducing perceived loudness or hearing performance in these challenging situations. Windshield technology is available in the optional outdoor programme that can be selected by the audiologist when the Baha sound processor is fitted.

Providing patients access to Wireless capabilities

Wireless technologies have been available for air conduction hearing aids for a number of years. The new Baha 4 Sound Processor uses 2.4GHz wireless technology – a proven and established true wireless technology that has been successfully used in hearing aids as well as in other applications such as cordless telephones and Wi-Fi routers. The Cochlear™ 2.4GHz wireless technology is designed to make the Baha sound processor the centre of the system and enable it to control all connections. It offers several key advantages compared to other wireless systems that use the older Near Field Magnetic Induction (NFMI) technology.

Unlike NFMI-based systems that have a one-to-one relationship to the streaming device, a Baha user may connect to multiple wireless devices. In addition, several Baha users may share the signal transmitted from one wireless accessory. This allows, for example, several school children or family members using different Baha processors to use a single streaming device, a Baha user may connect to multiple wireless devices. Unlike NFMI-based systems that have a one-to-one relationship to the streaming device, a Baha user may connect to multiple wireless devices. In addition, several Baha users may share the signal transmitted from one wireless accessory. This allows, for example, several school children or family members using different Baha processors to use a single wireless device to listen to a speaker or a television, which would not be possible with an NFMI-based system.

Because 2.4GHz wireless technology does not rely on an intermediate device, such as a neck-loop unit, the Baha user has direct connection to the wireless signal. There is no need to wear something bulky or uncomfortable and the user retains complete freedom of movement. The Cochlear 2.4GHz wireless system can be set up either by the hearing care professional using the Baha Fitting Software 4.0 or by the user, who may pair the processor to any compatible accessory.
at any time by a simple pairing sequence. Once the 2.4GHz wireless connection has been established, the Baha user can easily access the wireless benefits directly from the Baha processor.

To ensure a robust connection without interference, the Cochlear 2.4GHz wireless system uses two innovative technologies: time division and frequency hopping (Figure 14). The system divides data into packets (small portions of digitally coded information). The packets are sent using time division; information is only transmitted for 0.4 seconds at a time in any of the 35 channels that make up the 2.4GHz band. Each time a packet is sent, it is sent in a new channel. Because the sound processor and the wireless accessory have established a unique connection in the pairing process, the two products will mutually agree as to which channel to send the next data packet in. As all devices using the 2.4GHz band will have their own individual strategy to randomly select which channels to transmit in, they will virtually always avoid interfering with each other.

Figure 14. Cochlear’s 2.4GHz wireless technology uses data distribution in packets (top) and frequency hopping to secure a robust connection.

One of the key requirements of a wireless system is its ability to transmit with good sound quality and with the smallest possible delay. Other wireless systems transmit with latency (delay) from 40 up to 125 milliseconds. This can cause lip-sync issues due to a mismatch between sound and visual information. Even when the delay is too small to be consciously perceived, there may be a significant negative impact on the television viewing experience. A key design requirement of the Cochlear 2.4GHz technology was to ensure latency of less than 35ms, which is important in reducing lip-sync or echo effects for the Baha user. Measurements made on wireless systems provided by different hearing device manufacturers clearly show a distinct advantage for the user. Measurements made on wireless systems provided by different hearing device manufacturers clearly show a distinct advantage for the user.

Whilst other wireless technologies such as Bluetooth® and NFMI may use a so-called “low density” solution for audio compression, the Cochlear 2.4GHz wireless technology uses high-density audio compression, resulting in superior audio quality.

Through the use of the Cochlear 2.4GHz technology, the Baha user is able to take advantage of a number of accessories such as:

- Cochlear Baha Remote control: the first one ever designed for Baha sound processors. With large, easy-to-use buttons, it provides all users, even those with limited dexterity, a simple means of adjusting programme and volume settings. The display gives a clear overview of the sound processor’s settings and lets users, parents and caregivers monitor the sound processor’s status and settings.
- Cochlear Wireless Mini Microphone: a small, lightweight, portable personal audio streamer that allows sounds to be transmitted directly to the Baha sound processor. The Mini Microphone may be clipped onto clothing and will transmit speech wirelessly to the Baha sound processor over a distance of up to 7 metres, thereby significantly improving the signal-to-noise ratio in challenging listening situations.
- Cochlear Wireless Phone Clip: a small clip-on accessory with a built-in microphone and Bluetooth® capability. It links the Baha sound processor to any Bluetooth-enabled telephone, thereby allowing the user to hear the telephone directly through the Baha processor, resulting in an improved signal-to-noise ratio. In addition, the PhoneClip will pick up speech from the Baha user and transmit it directly to the telephone so that the Baha user may communicate freely on the phone without needing to hold the handset, even if the handset is several metres away. The PhoneClip works with any Bluetooth enabled device such as mobile phones, audio streamers or navigation systems.
- Cochlear Wireless TV Streamer: connects to a TV, stereo, computer or other audio source. It sends stereo sound up to 7 metres directly to the Baha 4 Sound Processor without any extra equipment. For user convenience, the connection between the TV Streamer and the sound processor is automatically resumed if the user goes outside the range and returns within 5 minutes.

Improving fitting performance with Baha Fitting Software 4.0

Fitting software has been an area of continued innovation designed to ensure excellent first-time fitting of a Baha sound processor. The first version of the Baha Fitting Software introduced BC Direct – the first tool to measure a patient’s bone conduction thresholds directly through the sound processor. The importance and benefits of this tool have been demonstrated in a number of reports19,20. In the following version, Cochlear continued to simplify Baha fittings by providing BC Select, a simple and straightforward means of pre-configuring the Baha processor and improving first-fit accuracy21.

As described previously, combined with the new Baha Fitting Software 4.0, Cochlear introduces new and innovative tools to make fitting a Baha processor faster and simpler. The new Feedback Analyser allows the audiologist to quickly measure and adjust for individual variations in feedback limitations, ensuring the best possible performance with minimal feedback.

Other programming interfaces, such as NOAHlink, have limited wireless capabilities and still require the use of cables. The Baha Fitting Software 4.0 enables the use of Airlink, a tool for true wireless fitting based on 2.4GHz wireless technology. This will allow audiologists to fit Baha processors without the limitations of cables or intermediate devices such as neck-worn loops. Airlink is a 2.4GHz wireless plug and play fitting solution incorporated into a small USB device. Simply installing the Baha Fitting Software 4.0 and inserting the Airlink into a USB port on the PC enables the audiologist to wirelessly fit a Baha 4 Sound Processor at a distance of up to 3 metres.
To ensure a robust connection it’s important to keep line of sight between the Airlink and sound processor at all times. Each wireless Baha processor is factory set with a serial number stored in the device’s memory that provides a unique identification address. This allows the hearing care professional to specify which processor is fitted on the right and left side in a bilateral fitting by selecting the correct serial number for each side. Additionally, the professional can generate a tone in each of the processors, to simplify this further. This enables the professional to easily establish a secure and unique wireless connection between the fitting system and one or two (for bilateral users) sound processors. This ensures that multiple Baha processors can be fitted at the same time in hospitals and clinics where several Airlinks are used since each processor will have its own unique and secure connection.

Summary and conclusion
The Baha 4 Sound Processor marks the continued development and implementation of advanced signal processing specifically designed to meet the needs of patients with conductive, mixed or single-sided sensorineural deafness. There are two important aspects in this development. First, the Ardium platform enables a complete overhaul of the Baha sound processor’s signal processing capability. Second, wireless connectivity will provide the patient with an increased level of connectedness, ease of use and also significantly improved hearing performance in many difficult listening situations. Furthermore, fitting a Baha sound processor, particularly with the implementation of individual stable gain measurements, expanded data logging and wireless fitting should provide the hearing care professional with the tools to simplify the fitting process. The outcome of these innovations is to provide the wearer with improved sound quality and to reduce the burden of listening, even in the most challenging situations.

References