INTEGRATED SIGNAL PROCESSING

A New Standard in Enhancing Hearing Aid Performance
Widex is proud to introduce INTEO, the new standard in hearing aid technology. INTEO will give Hearing Healthcare Professionals many additional options to help them serve the needs of their patients. The technical articles enclosed will give professionals insight into the many new features that were designed to work together to provide unprecedented performance.

It is also with great pride that in 2006, Widex celebrates our 50th anniversary year of serving the Hearing Healthcare Industry.

Since 1956, Widex has been one of the world’s most respected hearing aid manufacturers. Visionary research and development and the strong commitment from Widex audiologists, hearing scientists, engineers, and physicists, has led to innovations that have had significant impact on the industry for many years.

INTEO will provide a new level of performance for millions of people around the world who have a hearing impairment. And it is our privilege to be the company that provides a premier technology like this that can help change people’s lives.

Sincerely,

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Hearing aids are smarter than ever. With the improvement in the complexity of integrated circuits (IC) and the corresponding decrease in their size and current consumption, many Digital Signal Processing (DSP) algorithms that were impossible to implement previously are now possible. Recognizing such an opportunity, Widex applies this advance in hardware technology and introduces the first hearing aid using Integrated Signal Processing (ISP) technology in its Inteo hearing aid. What follows in this supplement is an explanation of this technology and the benefits it offers, along with an illustration of the use of such a technology in the Inteo hearing aid.

What is Integrated Signal Processing technology?

Quite simply, Integrated Signal Processing (ISP) technology is a technological platform that allows design engineers to integrate information from various sources to affect the outcome of signal processing. The result of ISP is processing that is more efficient, more accurate, more artifact-free, and more individualized. In other words, ISP would result in a more effective way of processing sounds so that hearing aid wearers are more satisfied with the performance and use of their hearing aids.

It is important to understand current signal processing approaches in order to appreciate the potential of ISP. Currently, there are two main approaches to signal processing that differ by how they are implemented.

Sequential Processing

Figure 1 shows a functional block diagram of sequential processing. The blocks with labels X, Y, and Z are functional components. In sequential processing, the flow of information is always in one direction. Component Y will not start processing sounds until component X has finished its task. Component Z will always be the last to process sounds. Because of this sequential nature, no two components may be working simultaneously. It also suggests that knowledge-sharing among blocks is not possible.

Parallel Processing

Figure 2 shows a functional block diagram of parallel processing. Again, X, Y, Z, and T are functional blocks each responsible for a particular function. In a parallel processing scheme, information flows into several functional components at the same time, and processing by the different components occurs at the same time. For example, while component X may be computing the level-dependent gain, component Y may be deciding if the signal that passes through is a “speech” signal or a “noise” signal by examining its level distribution function or modulation rate. Meanwhile, component Z may be estimating the feedback path of the hearing aid system. Indeed, this is the processing platform that was utilized by Widex since it introduced the Senso hearing aid 10 years ago.

Clearly, an advantage of parallel processing is that many functional components can be processing at the same time. This may increase the efficiency of the processing and the responsiveness of the system to changes in the acoustic environment.

Both sequential and parallel processing strategies are utilized in today’s DSP hearing aids. When implemented by a skilled designer, both signal processing strategies may yield equally complex and sophisticated signal processing algorithms. On the other hand, a limitation of both strategies is that information from the different functional components is not shared among each other. This results in each component working on its own (which was intended), while ignoring and not accounting for potential interactions among the different components. As a result, wearer satisfaction may not be ensured in all situations. An example of this interaction may be seen in the occurrence of feedback in a noisy place from a hearing aid that uses both adaptive directional microphone and active feedback cancellation algorithms. In this case, as the noise source changes, the polar pattern of the adaptive directional microphone also changes.
tently, this also changes the feedback path of the hearing aid. And if the hearing aid cannot re-estimate the feedback path fast enough or accurately enough, feedback occurs. Another example of this limitation may be seen in the mis-identification of musical signals as feedback by the active feedback cancellation system. An anti-feedback signal which has the opposite phase of the musical signal is produced and used to cancel the music signal. This could compromise the sound quality of the musical signals. In order to avoid these situations, some higher forms of signal processing that could identify these occurrences as soon as they occur and correct for them, or prevent their occurrences altogether would be desirable.

**Integrated Signal Processing**

Integrated Signal Processing (ISP) is the newest approach to hearing aid signal processing where not only is the input signal shared among the different functional units (so they can be processed in parallel), but the results of the processing from each functional unit are also shared amongst each other to result in a highly complex and integrated network of information flow. Figure 3 shows one of the many possible schematic block diagrams of how a system using ISP may look.

![Figure 3: A simplified functional block diagram to illustrate Integrated Signal Processing (ISP).](image)

Figure 3 shows that information from component A is processed in parallel by components X, Y, and Z while it is also processed sequentially by Y and B and T. However, a landmark of ISP is that information from X, Y, and Z is also shared amongst each other. In reality, it is not uncommon to find that each functional component is linked to every other functional component within the hearing aid. All the processes within the hearing aid function as a unit. No one process functions independently of the others; they are all controlled by the outcome of the other processes. No one process functions after the others; information from each process is shared by all other processes. Analyses are performed not only of the input signals, but also of the simultaneous outputs of each individual processing algorithm for instant updating. Processing in the hearing aid is not only affected by the acoustic conditions in the environment, but also by the individual users’ hearing loss characteristics, needs, and expectations.

The advantage of this integration is that mishaps like those mentioned in the previous paragraph may be avoided. This further improves the quality of the processed sounds. In addition, because information is shared among all components, the use of more complex algorithms which further enhance the performance of the hearing aid and wearer satisfaction are possible.

One can imagine that designing such an integrated system is a daunting task. Indeed, two main factors prevented the implementation of ISP in hearing aids earlier – knowledge and technology. To implement ISP effectively, one needs to know apriori the potential interactions among the different processing algorithms such as that between the adaptive directional microphone and the active feedback cancellation algorithm in order to contemplate designing effective solutions. In addition, one needs to know the wants and needs of the potential wearers as well as of the clinicians who may be fitting these devices to ensure that the solutions are practical. This takes time, experience, and dedication on the part of the designers.

Ultimately, human intelligence is not sufficient to realize such a processing system unless the IC technology has advanced to a stage where these extra complexities can be contained within a chip that is small enough for a completely-in-the-canal hearing aid and efficient enough to yield a reasonable current drain. From that perspective, Integrated Signal Processing (ISP) technology is a necessary evolutionary outcome that will revolutionize how hearing aids are designed in the future.

**Widex Inteo – An Example of ISP**

The Inteo hearing aid introduced by Widex is the first to use ISP technology to “link” all its functional components so that integration of information and processing is possible. In order to appreciate how this integration is realized, the Inteo signal processing can be functionally grouped into four major components/modules. They include the Dynamic Integrator (DI), the High Definition Sound Analysis (HDSA), the High Definition Sound Processing (HDSP), and the High Definition System Optimization (HDSO) modules. These four modules are highly integrated in that results of each module are shared by the other modules. In addition, wearer information such as hearing loss, hearing aid experience, etc. is stored in the Dynamic Integrator and shared among the different modules to affect processing. Using the analogy of the human body, the Dynamic Integrator is the central nervous system (CNS) of the organism. The sense organs are the HD Sound Analysis in the Inteo, the different systems of the body such as the skeletal and muscular systems are the HD Sound Processing module of the Inteo, while the Endocrine system is the HD System Optimizer that ensures each part of the body is operating optimally. Figure 4 is a simple block diagram that shows the inter-relationship and flow of information among the modules.
Wearer Information

Wearer information such as the degree of hearing loss of the wearer, years of experience with amplification, types of earmold used and the in-situ vent effect, and the individual real-ear-to-coupler difference (RECD) corrections are stored in the Dynamic Integrator so that appropriate information may be dispatched to and integrated with the processing algorithms in the other modules.

High Definition Sound Analysis (HDSA)

The HDSA module includes a number of features that sample the acoustic environments in order to identify the nature of the input signals. This can be likened to the sense organs of the body which encode the environmental conditions to the central nervous system. In the Inteo, inputs from each of the dual microphones in each of the 15 channels are sampled continuously in order to estimate the magnitude of the input signals, their azimuths of origin and their nature (i.e., speech, noise, feedback, wind, etc.). When these results are examined across the 15 channels and over time, each acoustic environment yields a unique 3-D pattern that can be used for its identification. Figure 5 shows the results of the acoustic analyses done on “car noise” and “playground with speech.” The display shows the percentage of time (Z-axis) sounds of a particular intensity level (X-axis) occurred in each of the 15 channels (Y-axis). One can see that the “car noise” is mostly a broadband noise occurring at around 50-60 dB SPL while the playground noise has a broad intensity range that is more intense in the low frequency channels (40-70 dB SPL) and becomes less intense in the higher frequency channels (10-40 dB SPL).

Sound Diary – The results of the acoustic analysis can be stored in the memory of the Inteo hearing aid as a record of the acoustic environments encountered by the hearing aid wearers. This could offer insights for better patient counseling as well as tips for trouble-shooting specific wearer complaints.

Speech and Noise Tracer – By examining the percentile level of the input (or level distribution function) at each of the 15 channels, one can determine if the input is primarily “speech” or “noise.” In addition, it allows one to determine the level of the input at each channel as well as the spectral shape of the input signal when information is examined across channels.

Speech-to-Noise Ratio (SNR) Tracer – The acoustic analysis could also yield information on the SNR of the acoustic environments by separately estimating the levels of the speech signal and the noise signal. This information could signal how much gain reduction should occur in the noise reduction systems or how much reduction in sensitivity the directional microphone should provide.

Spatial Sound Tracer – By examining inputs from both microphones, the Inteo hearing aid is able to estimate the origin of the sound source and provide the appropriate information for the adaptive directional microphone to form the most appropriate polar pattern.

Spatial Feedback Tracer – The Inteo estimates the feedback paths from the extreme polar patterns formed by both microphones of the directional microphone system so the active feedback cancellation algorithm can build a more effective feedback canceling signal.
High Definition Sound Processing (HDSP)

The analysis performed by the HDSA module provides up-to-date and input-specific (nature) information for the different processors so that each can perform its processing optimally. Because of the HDSA, the Inteo is able to treat input sounds in different ways to achieve results not possible with sequential and parallel processing alone. In addition to the extended input dynamic range (EIDR) of 107 dB SPL and the use of adaptive slow-acting dynamic range compression with a low compression threshold at 0 dB HL in each of the 15 channels, the use of ISP allows the Inteo to achieve many unique features. Some of them include: noise management with Speech Enhancer, Multi-Directional Active Feedback Cancellation, High Definition Locator, and the Audibility Extender. These features are listed for ease of demonstration of the ISP technology. They will be briefly described now and will be explained in greater detail in subsequent chapters.

Precise Input-Output Characteristics – The multi-segment input-output characteristic of the Inteo approximates closely normal loudness perception (Figure 6). The multi-segment nature of the I-O also allows discrete changes to be made to the I-O curve for discrete, level-specific adjustment during the fine-tuning process (if necessary). Furthermore, the accuracy of the gain prescription is assured by the use of in-situ threshold (or Sensogram) measurement (Ludvigsen and Tøpholm, 1997) and the use of in-situ vent effect compensation. A precise I-O is crucial for good sound quality and speech understanding.

Speech Enhancer – The Speech Enhancer on the Inteo is a patent-pending optimization algorithm that maximizes the Speech Intelligibility Index (SII - ANSI 1997) in a noisy environment. Once activated, gain settings in each of the 15 channels will be adaptively manipulated so that as much of the speech spectrum is audible without exceeding the loudness discomfort level of the wearer. For some wearers, this algorithm may improve their speech understanding in a noisy background and could be especially desirable in models where the limited space prohibits the use of directional microphones, e.g., CIC style hearing aids. This algorithm is offered in addition to the Widex classic Noise Reduction algorithm with eSIS where the main focus is to enhance listening comfort while preserving speech intelligibility.

High Definition Locator – The High Definition Locator (or HD-Locator) used in the Inteo is a 15-channel fully adaptive directional microphone system where the dual microphones are perfectly matched in sensitivity and phase characteristics at all times. The use of ISP results in an optional Speech Enhancer, Multi-Directional Active Feedback Cancellation, and Audibility Extender. These features are listed for ease of demonstration of the ISP technology. They will be briefly described now and will be explained in greater details in subsequent chapters.

Multi-Directional Active Feedback Cancellation – The Inteo active feedback cancellation algorithm takes input from the Spatial Feedback Tracer (which estimates feedback paths from the extreme polar patterns of the dual microphones) to create a signal of the opposite phase for its cancellation. This allows the Inteo feedback system to achieve higher usable gain before feedback occurs. More Inteo wearers may be “feedback-free” in more listening situations. By ensuring adequate gain before feedback, a hassle-free listening experience may result leading to improved speech intelligibility and enhanced sound quality. An even larger vent diameter may be used successfully to further reduce the occlusion effect stemming from a shell origin.

Audibility Extender – Inteo includes a unique, patent-pending linear frequency transposition algorithm that allows people with an unaidable high frequency hearing loss to hear the missing high frequencies in a lower pitch region without altering the tonal nature and transition characteristics of the transposed high frequency sounds. This algorithm improves the appreciation of high frequency music and bird songs for people with a precipitous high frequency hearing loss. With training and dedication on the part of the wearers, it may even improve the recognition of high frequency speech sounds.

High Definition System Optimizer (HDSO)

The HDSO module enhances and/or optimizes the performance of each feature in the other modules. This can be likened to the endocrine system of the human body where hormones and secretions are produced to ensure the proper functioning of the different parts of the body. The HDSO includes:

Multi-point microphone matching ensures that the dual microphones are perfectly matched in sensitivity and phase characteristics on a 24/7 basis. As many as 7 points on the frequency response curve are matched through a patent pending, proprietary optimization algorithm.

Battery monitor ensures that the demand on the current consumption is met without intermittent “blackout” or loss of amplification. Because of this vigilance, the Inteo can send
out a warning signal well before the expiration of the battery so that the wearer will have time to avoid missing important communications.

Eco-Tech II attempts to minimize current drain and maximize operational efficiency. Current drain is monitored at various stages in order to minimize unnecessary operations and maximize efficiency. Despite the complexity and the sophistication of the Inteo processing, the current drain is less than 1 mA! It can be substantially lower when certain features of the Inteo are deactivated.

DAI-Auto allows the wearer to couple any direct audio input devices (such as FM) to the Inteo without switching any program buttons. When a DAI device is connected, the Inteo will automatically and seamlessly switch into the DAI program, freeing the wearer so s/he can be fully engaged in the important act of communication.

Remote Control is available as an option for easy and discreet access to the different listening programs and volume adjustment. Especially for the wearers of the Inteo CIC hearing aid, this option increases their listening choices (for the different listening programs) in a convenient and discreet manner.

Dynamic Integrator

The key to the success of Integrated Signal Processing is the Dynamic Integrator (DI). Simply, the DI is a storage and execution unit that ensures the objectives of ISP are met. This is the component which takes (and stores) input from all the other components (such as wearer information, HDSA, HDSP, and HDSO), coordinates them, integrates them, and dispatches the necessary information to the appropriate components (or sub-components) so they can function optimally. This is also the component that takes results of the processing of the three HD modules (HDSA, HDSP, and HDSO) so that it can provide corrective actions to affect the processing in the same or other modules. This can be likened to our central nervous system (CNS) which takes its input from the 5 senses (and all body parts), coordinates and integrates them before it signals a particular part of the body to perform its function. Information on how well that body part functions is fed back to the CNS which can make further corrective actions, if needed.

Let’s consider the action of the High Definition Locator as an example of how the DI and ISP work. The DI monitors the status of the dual microphones on a continuous basis and returns that information to the Multi-point Microphone Matching algorithm of the HD System Optimizer (HDSO) so it can update its processing. This ensures that both microphones of the HD Locator system are functioning optimally. At the same time, the DI takes information from the Speech and Noise Tracer, Signal-to-Noise Ratio Tracer, and the Spatial Sound Tracer of the HD Sound Analysis (HDSA) module to decide on the correct polar pattern for each channel of the HD Locator in the HD Sound Processing (HDSP) module. Information on the momentary polar pattern in each channel is fed to the DI, which in turn adjusts (or fine-tunes) the actions of the HD Locator. One can easily see the pivotal role of the DI in integrating and dispatching information from each of the three modules.

Conclusion

The previous discussion illustrates the differences among various signal processing schemes and indicates the unique features that are realized with Integrated Signal Processing (ISP) technology in the Inteo. As discussed earlier, ISP is technology empowered, empirically derived, and human intelligence based. Through the advances in IC chip technology, we have the hardware platform to design more sophisticated processing. With the accumulation of knowledge over 50 years of hearing aid manufacturing, we have learned the key factors in designing a hearing aid system that further enhances wearer satisfaction for hearing aids. With the inputs from dispensers and consumers, we integrate the information so that the whole fitting process is simpler and more streamlined for the clinicians and more satisfactory for the consumers. What follows is a more detailed description of some of these features and illustrations of how ISP is achieved.

References


High Definition Sound Analysis - The 5 Senses of ISP

Carsten Palludén-Müller, M.Sc., Maja Bülow, M.A.

Like the five senses of the human body that gather information on our everyday surroundings and send them to the central nervous system for processing, the HD Sound Analysis (HDSA) module samples the acoustic environments of the wearer on a 24/7 basis and sends the information to the Dynamic Integrator (DI) so it may characterize the nature of the input signals for the different features of the HD Sound Processing (HDSP) module to process accordingly. In this acoustic analysis, inputs from each of the dual microphones on the Inteo are sampled continuously at 32000 times per second. By examining or analyzing the magnitude of these input signals in special ways, the Inteo is able to achieve many unparalleled features that are crucial in the application of Integrated Signal Processing (ISP) technology.

The Basis of All Analysis – Level Distribution

The Inteo uses a patented algorithm to estimate the magnitude of the input signal for signal processing. The Level Distribution function (Kuk et al., 2002) is a statistical algorithm that measures how frequently sounds of a particular intensity occur during a specific period of time. An example of the result of such an analysis is seen in Figure 1.

Figure 1: Level distribution function measured in one channel of the Inteo hearing aid over a 2 minute period. The top is that of a speech input, and the bottom is the noise generated by a car.

Figure 1 shows that for the speech input (top), sounds around 50 dB SPL occurred most frequently, followed by those around 30 dB SPL. On the other hand, sounds above 70 dB SPL rarely occurred. In fact, if one examines the level distribution function of a speech signal, one would typically see more than one peak or a broad distribution of sound intensities. This is referred to as a multi-modal distribution and generally includes sounds that vary in intensity levels over time, such as speech.

On the other hand, the bottom of Figure 1 shows a level distribution function that has only one peak around 75 dB SPL the whole time. This is described as a uni-modal distribution and it characterizes signals that do not change their amplitude over time, i.e., noise.

By continuously examining the level distribution functions within a channel, the Inteo is able to identify the nature of the input signal occurring at that channel. Indeed, a key feature of the HDSA module is the Speech and Noise Tracer which takes advantage of this level distribution property to make a 24/7 on-line analysis of the nature of the input signals. The result of the analysis is sent to the DI so that it can dispatch the information to the processing algorithms where such information would affect processing. This includes the noise reduction algorithms and the HD Locator directional microphone system with Speech Tracer.

Estimation of Signal-to-Noise Ratio

When both “speech” and “noise” are present, the relative levels of the speech signal to the noise level (or signal-to-noise ratio, SNR) is an important piece of information that instructs algorithms such as noise reduction on how much gain reduction should result. This is again achieved using data from the level-distribution functions.

As previously defined, a “noise” signal is assumed to have a constant intensity level over time; whereas a “speech” signal would have intensity fluctuations over time. Using this knowledge, one can estimate the noise level by measuring its intensity level during speech pauses, i.e., when there is only “noise.” The ratio of the instantaneous speech level to the estimated noise level would define the instantaneous “speech-to-noise ratio.” Obviously, the estimation of the “noise” level is updated regularly (along with speech) in order to ensure that the moment to moment SNR is estimated accurately. In the Inteo, the Signal-to-Noise Ratio Tracer (SNRT) performs this function on a 24/7 basis and returns information on the SNR to the Dynamic Integrator regularly. Figure 2 shows an example of how the SNR is estimated.
Advantages of Having Level Distribution Functions in Multiple Channels

One of the advantages of a multichannel hearing aid is its accuracy of capturing the nature of the input signal, and of reproducing the output is improved significantly (over a single channel system). From the acoustic analysis standpoint, the advantages of performing level distribution analyses in all 15 channels independently and simultaneously over time is that a unique visual pattern of the acoustic environment emerges for each environment. These patterns are like fingerprints – unique in their own ways and different from other environments. Figure 3 shows the results of the unprocessed acoustic analyses done on “car noise” and “playground with speech” environments. The display shows the percentage of time (Z-axis) sounds of a particular intensity level (X-axis) occurred in each of the 15 channels (Y-axis). One can easily see that the “car noise” is mostly a broadband noise occurring at around 50-60 dB SPL while the playground noise has a broad intensity range that is more intense in the low frequency channels (40-70 dB SPL) and decreases in levels with higher channels (10-40 dB SPL).

The observation that each acoustic environment has a unique display or signature is important. It suggests that one may be able to identify precisely the nature of the acoustic environments and modify the processing of the hearing aid accordingly. One such application is the automatic use of an omnidirectional microphone when the result of the acoustic analysis suggests wind noise, speech or quiet environment.

The Advantages of Sound Analysis from Two Paths

The Inteo uses two analog-to-digital (A/D) converters, one from each microphone of the dual microphone system. This opens up another opportunity for the Inteo to analyze the acoustic environments of the wearers more accurately. One feature that benefits from such a capability is the Spatial Sound Tracer (SST). The SST takes advantage of the arrival time difference for the same sound to reach the front and rear openings of a dual microphone system to estimate the location of the sound in space. This is because a sound that originates from the back of the wearer will reach the rear opening sooner than it will the front opening because of the shorter distance. The reverse is true for a sound that originates from the front. By taking advantage of that physical property, the SST is able to identify the precise location of a sound source so the appropriate polar pattern on the adaptive directional microphone may be formed.

The Spatial Feedback Tracer (SFT) provides a more realistic estimation of the feedback paths for the feedback cancellation algorithm to generate the cancellation signals. Figure 5 shows the estimation performed by the SFT.

Another advantage of analyzing the input signals from both microphones is a more accurate estimation of the feedback path(s) of a directional microphone system. On an omnidirectional microphone, the feedback path is estimated between the omnidirectional microphone opening and the receiver of the hearing aid. On the other hand, a directional microphone has minimally two openings. This suggests that there is a second pathway where feedback may occur. By simultaneously and separately estimating the feedback paths from the extreme polar patterns formed by the dual microphones, the patent-pending Spatial Feedback Tracer (SFT) provides a more realistic estimation of the feedback paths for the feedback cancellation algorithm to generate the cancellation signals. Figure 5 shows the estimation performed by the SFT.
Storage of Information Over Time - Sound Diary

The acoustic analyses of the wearers’ environments occur continuously in order that the processors are updated with the most current information as they become available. While these analyses could improve the functioning of the processors, they would not help the clinicians in understanding the wearer’s listening environments. In order to provide insights on the wearer’s acoustic environments, the results of the acoustic analyses are saved in the hearing aid so they may be reviewed later by the clinicians for better patient counseling and as tips for trouble-shooting specific wearer complaints.

In order that the wearer’s acoustic environments can be examined meaningfully, the acoustic data must be reduced in size in specific ways so that they can be stored within the EEPROM (electrical erasable programmable read only memory) of the hearing aid. Otherwise, a small segment of raw acoustic data (say, 10 s) could easily fill up the EEPROM to prevent the storage of any new data.

Thus the Inteo hearing aid has a data-logging function that analyzes each acoustic environment and returns the results of the analysis in a set of 4 values. The input signals are characterized by their (1) intensity level; (2) modulation rate; (3) overall spectral tilt; and (4) frequency of occurrence. Using these 4 parameters, Inteo is able to differentiate sounds that occur in the typical environments. Figure 6 shows the 3-D displays of two listening environments recorded for 2 minutes using three of the four parameters. One can see that these two sounds differ by their spectral tilt as well as their input level.

Figure 6 shows that for the female speech spoken at a normal conversational level, the spectral gradient is moderately flat to mildly sloping. The background noise is low, and considering that the peak occurs for “quiet” and “flat” spectrum gradient, the display would suggest that it is speech in quiet. On the other hand, a totally different display is seen with “jazz instruments.” The peak of the response occurs where the noise level was “loud” with a spectral gradient that is somewhat between steep and moderate. Clearly these two sounds are distinctly different from each other when analyzed using the criteria described earlier. Other sound situations would show different patterns.

Long-term data logging

Data logging requires that information of the acoustic environments be recorded over time in the Dynamic Integrator. Depending on the length of the recording, the utility of the recorded data differs. There are two options in the Inteo data-logging function – a long term data-logging and a short-term (or event) data-logging. In the long-term data logging, analysis and recording of the data starts when the hearing aid is initially fit until the data is downloaded into the Compass fitting software. Initially, sampling of the wearer’s acoustic environments occurs at one second intervals. With the passage of time, the sampling window increases to accommodate more acoustic data.

In the Inteo, results of the long-term data-logging are displayed as a pie-chart showing the percentage of time the wearer has been in a particular sound category. These categories are formed through extensive research and measurements. One can examine these charts and guesstimate the percentage of time the wearer spends in that category of sound environments.

Figure 6: 3-D displays of two sound environments (female speech at normal level on top and jazz instruments on bottom) based on 3 of the 4 parameters described.
Figure 7: Pie-chart showing percentage of time a wearer spent in each category of environment.

For example, Figure 7 shows the wearer spent over half his/her time in a relatively quiet environment with little background noise. Almost 30% of the time speech was recognized as the primary input to the Inteo. The patient spent less than 20% of his/her time in a noisy place such as a restaurant or in a car (i.e., transportation). This information from the long-term data logger gives the clinicians some insights into the wearer’s listening environments. That information may be helpful in counseling the patients.

On the other hand, information from the long-term data-logger would not be helpful in fine-tuning or trouble-shooting the hearing aid for stimulus-related complaints (for example, a door-slam). This is because in long-term data logging, acoustic information is averaged over time. This means any fine structure of the specific acoustic environment will be lost. In other words, while one may use the results of long-term data logging for understanding and counseling purposes, the accuracy of the sound classification decreases over time. A shorter sampling of the acoustic environment is necessary if one intends to use the results for fine-tuning and/or trouble-shooting purposes.

Short-term data-logging – Event Log

Because the Inteo hearing aid measures the acoustic inputs to the hearing aid and not the percentage of time a specific feature (such as noise program, directional microphone, etc.) is activated to characterize the environments, it provides the clinicians the tools to examine the acoustic environments of the wearer’s complaint if it keeps the recording (and analysis) short. To provide that flexibility, the Inteo also includes a short-term data-logging function called the Event Log. The wearer may activate the event log by pressing a button on an optional remote control. Once activated, the event log can sample and save one minute’s worth of the acoustic environment that is identified by the wearer. Fine-tuning is simply a matter of identifying the level and the spectral content of the troubling stimulus, and addressing the gain parameter(s) of the appropriate channels accordingly.

Conclusion

The Inteo HDSA module provides an accurate and reliable assessment of the acoustic conditions of the wearer’s listening environments. When the results of the analysis are examined instantaneously, they allow the Inteo to update its various processing units for optimal performance. When the results of the analysis are stored and examined over a longer period of time (long- and short-term data logging), they allow for a better understanding of the wearer’s listening environments and provide clues for adjustment of the hearing aid settings to improve wearer satisfaction.

References

An accurate fitting starts with an accurate determination of the hearing threshold. Through the use of the Sensogram (or in-situ threshold measurements, (Ludvigsen and Tøpholm, 1997)), Widex pioneered a practical means to ensure accurate hearing loss determination for the purpose of gain setting on its hearing aids.

With the Inteo, Widex has set another standard for accurate gain specification. By completing the simple, 2-step (sensogram and feedback test) fitting procedure, the Inteo hearing aid is able to estimate the in-situ vent effect of the hearing aid. In addition, it will automatically compensate the assigned gain for the vent effect so that people with the same audiometric thresholds or hearing loss will receive the same amount of insertion gain regardless of the openness or tightness of the earmold/hearing aids in-situ. This is achieved through the proprietary patent-pending AISA (Assessment of In-Situ Acoustics) algorithm developed only by Widex.

What is the Vent Effect?

When a hearing aid user wears a hearing aid, the sound pressure level (SPL) that is measured at the wearer’s eardrum is not only determined by the coupler output of the hearing aid, but is also influenced by the effect of any venting used on the hearing aid. This effect, termed the Vent Effect, is the difference in sound pressure level (SPL) measured at the eardrum of the hearing aid wearer consequent to the use of a vent (intentional or unintentional) on the hearing aid/earmold.

The Vent Effect is controlled by the physical dimensions of the vent. Indeed, it has been shown that the diameter of the vent and the length of the vent are two important parameters that affect the Vent Effect. On the other hand, leakage between the hearing aid and the ear-canal wall (unintentional venting), the residual volume between the hearing aid and the eardrum, as well as the compliance of the middle ear also interact with the vent dimensions to affect the magnitude of the Vent Effect. Thus, knowing the vent length and vent diameter does not always allow one to accurately predict the effect of a vent. One needs to know more about the individual and how s/he is wearing the hearing aid.

Unfortunately, the measurement of leakage and of the individual’s middle ear system is not an easy task. Conventionally, this means that in-situ vent effect cannot be easily predicted – it can only be measured directly with the hearing aid in-situ. This means if one were to estimate if sufficient gain is received by the wearer, or to examine the changes in the low frequency response of the hearing aid (Lybarger, 1985) or of the occlusion effect (Kuk et al., 2005) consequent to venting, one has to perform real-ear measurements with the hearing aids in-situ. It will be desirable if one could predict the in-situ vent effect without extensive measurements. Not only would it save time and resources, but it would also ensure the accuracy of the in-situ thresholds and that all wearers receive the intended gain regardless of the vents used (intentional and unintentional) on their hearing aids.

What is Assessment of In-Situ Acoustics (AISA)?

To explore the possibility of predicting the in-situ vent effect, we performed a series of studies where we examined the Vent Effect as we systematically changed the vent dimensions. One such possibility is the relationship between the maximum available gain before feedback on a hearing aid and the vent diameter on a hearing aid. Figure 1 shows that as the vent diameter increases, maximum gain before feedback decreases in a somewhat systematic manner. This is an important observation - if one can relate the information on the available gain before feedback to the dimensions of the vent, we may be able to estimate the equivalent vent effect simply by measuring the maximum gain before feedback when the hearing aid is worn in-situ.

![Figure 1: The relationship between max gain before feedback and vent diameter.](image-url)
Motivated by such observations, Widex engineers developed a complex model of the in-situ response of a hearing aid that relates the maximum gain before feedback to the vent effect of the individual ear insert in the wearer’s ear. Using this model, one can predict the in-situ vent effect if one knows the magnitude of the maximum gain before feedback (and vice versa). Furthermore, if one assumes that the wearer is an average individual with the average ear canal characteristics (average vent length, average hearing aid leakage, typical compliance and average residual volume), one can predict the equivalent vent diameter of the hearing aid/vent system by examining the maximum gain information. The result of such a prediction is shown in Figure 2. One can see that the estimated (equivalent) vent diameters are in close agreement with the nominal vent diameters. The “spread” in the data on the equivalent vent diameter is related to the individual variations in residual volume and middle ear compliance not accounted for by the nominal vent diameter.

AISA in Action

This new research finding is put to use in the Inteo to further improve the accuracy of the in-situ threshold determination and gain assignment. Once the AISA option is selected, the maximum gain that is measured during the routine feedback test is sent directly to the Dynamic Integrator which relates it to the HD System Optimizer where it is used to estimate the equivalent vent diameter of the wearer. The information on the equivalent vent diameter is then used to calculate the true hearing loss of the wearer by adjusting the sensogram values with the estimated vent effect. Initial in-situ gain assignment will be based on the true hearing loss or adjusted sensogram values. To compensate for the changes in in-situ gain from the use of the chosen vent, gain compensation is performed on the initial gain assignment to correct for the effect of the vent on the final output of the hearing aid. This will ensure an accurate and consistent in-situ gain use regardless of vent and individual differences. Figure 3 compares the steps in gain assignment between the original approach (blue box with filled arrows) and the new approach that has incorporated the AISA algorithm (purple boxes with unfilled arrows).

Benefits of the AISA

The immediate benefit of the AISA algorithm is a more accurate gain prescription that has considered all the individual and acoustic variables brought forth by intentional and unintentional venting. This ensures consistent gain use among hearing aid wearers using different vents, leading to consistent audibility and a consistently better sound quality. The fact that it is part of the simple fitting procedure ensures that every wearer of the Inteo enjoys this benefit even without the use of real-ear verification.

References


Individualized Noise Management - Choice of Comfort or Intelligibility

Carsten Palluden-Müller, M.Sc.

A likely consequence of a sensorineural hearing loss is the diminished ability to understand speech in noisy backgrounds. Indeed, the inability to hear in noise is still the number one reason for dissatisfaction with hearing aid use (Kochkin, 2002). Although hearing aids with a directional microphone provide a substantial improvement in the wearers’ ability to understand speech in noisy environments (Valente and Mispagel, 2004), space requirement and the uncertainty in directional characteristics when the directional hearing aid is inserted deeply into the ear canal prevent the realization of such a technology in the smallest completely-in-the-canal (CIC) hearing aids. A wearer who insists on the use of CIC style hearing aids may have to rely solely on the efficacy of the single-microphone noise management strategies available in such hearing aids.

The Inteo provides two distinct approaches to noise management in addition to its High Definition Locator directional microphone technology. The clinicians can choose between the Classic Noise Reduction algorithm with eSIS that has proven to improve listening comfort in many listening situations, or they can select the new patent-pending Speech Enhancer algorithm that is designed to maximize the speech intelligibility index (SII) in noise (default setting). Both of these algorithms are activated when the Dynamic Integrator, based on the results of the Speech and Noise Tracer analysis, suggests that the input signal includes noise in one or more of the 15 independent channels.

Classic Noise Reduction with eSIS

Details of the classic Noise Reduction algorithm were explained elsewhere (Kuk et al., 2002). Briefly, the objective of the classic Noise Reduction algorithm is to enhance listening comfort while preserving speech intelligibility. To improve listening comfort, the algorithm provides differential gain reduction in the appropriate channel when the input includes “noise.” The amount of gain reduction in each channel, and the input level at which gain reduction occurs is governed by the Dynamic Integrator with input from the HD System Optimizer.

To preserve speech intelligibility, the classic Noise Reduction algorithm remains in an inactive state at low input levels to preserve audibility of soft sounds. In addition, frequency channels that are important contributors to speech intelligibility receive less gain reduction than channels that are less important even though both channels may have the same signal-to-noise ratios. On the other hand, the amount of gain reduction is independent of the wearers’ hearing loss. Results thus far suggest an improvement in listening comfort in all wearers with some demonstrating an improvement in speech-to-noise ratio (SNR) when tested with the HINT test (Peeters, 2006).

Speech Enhancer

Whereas the Noise Reduction algorithm reduces gain in order to improve listening comfort, the Speech Enhancer improves the audibility (and hopefully intelligibility) of speech in a noisy environment through a patent-pending optimization algorithm that maximizes the Speech Intelligibility Index (SII- ANSI 1997).

**Speech Intelligibility Index** – the Speech Intelligibility Index (SII) is a complex computational approach that estimates the intelligibility of speech (or amplified speech) based on how much the assumed speech spectrum is above the hearing loss of the patient and the masking noise level. Because each region of the speech frequency carries a different weight (or contribution) to the overall speech understanding, the overall predicted speech intelligibility is the sum of each frequency region that is audible (above the threshold and noise level) scaled by a loudness distortion factor and a band importance weight.

\[
SII = \sum_{i=1}^{n} I_i A_i
\]

where \(I\) is band importance and \(A\) is band audibility

Figure 1 is a schematic diagram to show how the audibility of raised speech is determined in a car noise background. The dotted black line is the audiogram (in dB HL). The red line is the equivalent masking noise level. The green line is the average speech spectrum. The light grey area is the speech spectrum. The yellow shaded area that is above both the equivalent noise spectrum (red line, from 200 to 2000 Hz) and the hearing loss of the wearer (dotted line, from 2000 to 5000 Hz) is the audible speech spectrum. This area, when adjusted for distortion from the level of presentation, multiplied by the band importance function and summed across frequencies, yields the Speech Intelligibility Index (SII).

**Figure 1: Estimation of the audible speech area.**

Once a “noise” input is identified by the Dynamic Integrator, the Speech Enhancer begins its optimization procedure by adaptively adjusting the different gain settings in each of the 15 channels in order to achieve the maximum speech Intelligibility index for the wearer in the specific noise en-
vironment. During this process, the SII formed by millions and millions of different combinations of gain settings are compared in an adaptive manner in order to determine the setting(s) with the highest SII. Figure 2 shows the result of the optimization process where the gain settings in two channels (low and high frequency channels) are compared for maximum SII. The peak of the response surface (or the peak of the dome in red) represents the maximum SII. The gain settings on the hearing aid that yield the maximum peak SII are the recommended (or final) gain settings.

![Response surface generated from the comparisons of gain settings on a 2-channel hearing aid. The response surface of the Inteo is much more complex because there are 15 channels on the hearing aid.](image)

Although the principle of the optimization process sounds simple and intuitive, the implementation is much more complex. From the hardware standpoint, the optimization is a labor-intensive computational process. As the number of parameters (e.g., channels) increases, the number of potential settings, as well as the number of comparisons increases exponentially. The fact that the Inteo has 15 individual channels for gain adjustment means that the response surface generated by the Inteo optimization process is a 16-dimensional pattern with millions and millions of possible gain settings (15,407,021,574,586,368 combinations to be exact) for comparison. One should also remember that the current implementation of the SII model runs on-line. It constantly updates the Speech Enhancer on the characteristics of the speech and the noise spectra for a valid optimization. Indeed, the Inteo Speech Enhancer optimization algorithm is many times more complex than any other off-line or commercial implementation of the algorithm thus far. Luckily, the HD System Optimization module allows the Inteo to handle a vast amount of data in a short period of time and at a low current drain.

The results of the Classic Noise Reduction and the Speech Enhancer are different. In conventional noise reduction, gain reduction is dependent on the noise level and the SNR. It is independent of the wearer’s hearing loss. On the other hand, the action of the Speech Enhancer is dependent on the wearer’s hearing loss. Figure 3 shows the output difference between conventional noise reduction that provides a uniform 12 dB gain reduction (top) and the Speech Enhancer (bottom). The black line is the threshold of the wearer (in dB HL). The green line is the final average speech spectrum. The red line is the equivalent masking noise spectrum. The yellow shaded area is the speech spectrum that is audible to the wearer. For the uniform noise reduction (top), the area above the 2000 Hz region is not audible. On the other hand, the frequency region between 2000 Hz and 5000 Hz remains audible to the wearer after the Speech Enhancer optimization. This is one advantage of considering the wearer’s hearing loss in the Speech Enhancer algorithm – extra audibility re: conventional noise reduction.

![Difference in output between uniform 12 dB gain reduction (top) and the Speech Enhancer (bottom). Note the extra audibility with the Speech Enhancer in the high frequencies.](image)

The gain optimization process is a continuous process that is active when the Inteo is turned on. It does, however, take about 20 seconds to stabilize to a final gain setting when the wearers enter into a new “noise” environment. In this dynamic world where both the speech spectrum and the noise spectrum may change over time, an additional Fast-Acting Gain Increase mechanism is invoked as well. This mechanism constantly checks for the SII, and increases the instantaneous gain in the appropriate channels where a gain increase may be necessary to further optimize the speech intelligibility index. In its current version, as much as 6 dB gain increase in the mid and high frequencies is possible for daily speech and noise environments. Figure 4 are snapshots that show the gain reduction and gain increase across frequencies with the Speech Enhancer in 2 different noise backgrounds. The first example on the top is speech while the wearer is driving a car on the highway. The one on the bottom is speech from the radio while someone is vacuuming. Both assume the same degree of hearing loss. One can clearly see a gain increase (and decrease) for the example on the top, but only a gain decrease for the example on the bottom. This is a second major difference between the Speech Enhancer and the Classic Noise Reduction.
Enhancer and conventional noise reduction algorithms – the Speech Enhancer may also increase gain while most noise reduction algorithms only decrease gain.

Conclusions
The improvement in chip technology allows noise management strategies to be more sophisticated than ever before. The Speech Enhancer feature on the Inteo is a clear demonstration of this application. While both Classic Noise Reduction and Speech Enhancer are activated by the presence of noise in the listening environment, the objectives of the two are different – Noise Reduction has a priority of improving listening comfort; while Speech Enhancer is to maximize the speech intelligibility index first while retaining listening comfort.

References


High Definition Locator – Having Your Cake and Eating It too


It is well documented that hearing aids with a directional microphone improve the wearers’ speech understanding in noise in at least some listening environments (see Valente 1999 for a review). On the other hand, it is increasingly recognized that the very design of a directional microphone could compromise the audibility of sounds, especially soft sounds presented from the sides or the back of the wearer (Kuk et al., 2005). Indeed, as one examines the evolution of hearing aids with directional microphones, one could easily recognize that increasing attention has been paid to preserving audibility while maximizing the signal-to-noise ratio of the listening environments. Designs such as the inclusion of an omnidirectional and directional microphone on the same hearing aid that allows the wearer to switch microphone modes manually; or those that switch between the two microphone modes automatically are evidence of early attempts to preserve audibility. The pioneering use in the Widex Senso Diva of a fully adaptive directional microphone, one that automatically graduates from an omnidirectional polar pattern to other directional patterns without any program change, sets the standard for how a state-of-the-art directional hearing aid should be designed to preserve audibility and enhance signal-to-noise ratio.

High Definition Locator – A New Standard
The directional microphone system used in the Inteo is designed with an aim to set a new standard for directional microphones. This requires Widex engineers to further improve the performance of the Locator microphone in different listening situations by providing different treatments to speech and noise stimuli.

The directional microphone system used in the Inteo, the High Definition Locator (or HD-Locator), is a patent-pending, 15-channel fully adaptive directional microphone system. Each of the 15 channels has its own independent directional pattern that is capable of adapting from an omni-
directional polar pattern to a bi-directional polar pattern (and any polar patterns in between) based on the environmental conditions (i.e., the sound source). The Spatial Sound Tracer (SST) of the HDSA estimates the location of the inputs for each individual channel (or frequency). This information is provided to the Dynamic Integrator which in turn integrates information from the other modules to form the appropriate polar pattern. Figure 1 shows a possible combination of polar patterns in each of the 15 channels.

Figure 1: One possible combination of polar patterns in each of the 15 channels.

To ensure that both microphones in the HD Locator are perfectly matched in sensitivity and phase at all times for optimal performance, a highly sophisticated Multi-Point Microphone Matching algorithm in the High Definition System Optimization (HDSO) module uses environmental inputs as the stimuli to match the dual microphones on a 24/7 basis. This enables Widex to make a sophisticated multichannel fully adaptive directional microphone system in different styles of hearing aids (BTE, ITE, and ITC).

Another unique feature of the Inteo HD Locator directional microphone system is the possibility that speech and noise signals may be treated differently. Because of the use of Integrated Signal Processing, inputs from the Speech and Noise Tracer of the HDSA are made available to the Inteo HD-Locator system through the DI as an optional Speech Tracer feature. With its activation, listening environments that have only speech inputs (regardless of input levels and azimuths of presentation) will signal the HD-Locator to remain in the omnidirectional mode in order to preserve speech audibility. In addition, input signals that are identified by the HD Sound Analysis module as wind noise will also be minimized. Only channels where the dominant inputs are environmental sounds will set the HD Locator in the directional mode with the appropriate polar patterns. A directional microphone system using ISP technology would have clear advantages for adults whose communication requires hearing speech sounds from the sides and/or the back, and children whose speech and language development is contingent on consistent audibility.

A multichannel fully adaptive directional microphone may further improve speech intelligibility in listening conditions where both speech and frequency-specific noises (e.g., fan noise) originate from the same or different azimuths of the wearer. With a broadband fully adaptive directional microphone, the same polar pattern (or reduction in microphone sensitivity) results for all frequencies even though the noise only occurs at one narrow frequency region, e.g., 500 Hz. The background frequency-specific noise is reduced from the changes in the polar pattern at the noise frequency (e.g., 500 Hz); however, the audibility of the speech signal (which has a broader bandwidth) may also be reduced because the polar patterns of all frequencies are the same in a broadband directional microphone system. This result is seen in Figure 2. This is sometimes reported by hearing aid wearers as “drowning” of sounds – both speech and noise are softer as a result of the action of a directional microphone.

The advantage of a 15-channel fully adaptive directional microphone is that separate polar patterns can be assigned to the speech and noise stimuli. With the frequency-specific noise (e.g., 500 Hz), only the polar pattern of the involved channel (e.g., 500 Hz) will be affected; channels where speech is encountered will remain in the omnidirectional mode and keep the same sensitivity. This action could preserve the audibility of the speech signals. The result is shown in Figure 3. When compared to the previous figure, the multichannel system would spare the audibility of the speech signal while attenuating the narrow band noise. In principle, the more channels there are and the more discrete their bandwidths, the more likely it is for the multichannel fully adaptive directional system to be superior over the broadband fully adaptive directional system.

Figure 2: Illustration of the hypothetical “drowning” of sounds to a low frequency background noise in a broadband directional microphone. Note that the sensitivity of the microphone system is reduced across all frequencies because the same polar pattern is used for all frequencies in a broadband directional microphone.

Figure 3: Illustration of the freedom from “drowning” of sounds with the use of a 15-channel directional microphone. Note that the sensitivity of the microphone system is reduced only in the low frequency from the changes in polar pattern whereas an omnidirectional polar pattern is assumed for all other frequencies.

We have reported that the fully adaptive directional microphone system (Locator) made by Widex improves signal-
to-noise-ratios of the listening environments (Valente and Mispagel, 2004) and preserves the audibility of soft sounds spoken to the sides or back of the wearer in quiet (Kuk et al., 2005). With the realization of the HD-Locator in the Inteo which includes further refinement in microphone matching, different microphone processing strategy for speech and other signals, and the use of a 15-channel fully adaptive directional microphone system, one can only imagine that the HD-Locator will further improve signal-to-noise ratio while preserving audibility beyond what current technologies may provide.

**Multi-Directional Active Feedback Canceling - The Fruits of Successful Integration**

Kristian Klinkby M.Sc., Helge Pontoppidan Föh, M.Sc.

The term “Feedback Management” is perhaps the most non-descriptive term in explaining how a digital hearing aid handles feedback. While many would understand the general approach taken in “noise reduction” and “directional microphone,” the term “feedback management” may include strategies as simple as gain lowering and gain limitation for a specific frequency channel, to the use of fixed and variable notch filters, and the use of active phase cancellation. Even when an active phase cancellation technique is used, its success is highly dependent on the complexity of the IC chip and how well the designers anticipate and circumvent negative interactions among different processing units in their design.

For example, in the typical active feedback cancellation algorithms, inputs to the amplifier are correlated to its output to estimate if the input is a feedback signal. If a feedback signal is identified, a signal that is opposite in phase to the input will be created automatically to cancel the feedback signal at the input. This allows the wearer to use more available gain on the hearing aid (Kuk et al., 2002). In general, it works well in hearing aids with an omnidirectional microphone.

A hearing aid with an adaptive directional microphone (i.e., polar pattern that changes with noise azimuth) may moderate the success of this phase cancellation effort. There are at least two reasons for this. First, a directional microphone has at least two microphone openings. This means that there are at least two possible paths which may create feedback. Secondly, because an adaptive directional microphone changes its polar pattern depending on the noise source, the feedback paths created by the adaptive directional microphone may also change. This means unless one estimates feedback in all possible polar patterns formed by the dual microphones, the estimated feedback paths may not be appropriate for the moment and feedback may still occur. That may be one reason why some wearers of an adaptive directional microphone notice the intermittent occurrence of feedback in noise. The restrictive complexity of the IC technology limited the sophistication of the feedback cancellation algorithm in the past. The use of sequential and/or parallel processing to estimate feedback paths may have also limited the reach of current systems because the status of the directional microphone is not available to the feedback cancellation algorithm. The use of ISP technique is necessary to maximize the performance of an active feedback cancellation system. The following discussion explains how the Inteo uses ISP to achieve active feedback cancellation.

**How Does the Inteo Multi-Directional (MD) Active Feedback Canceling Algorithm Work?**

The Inteo uses state-of-the-art IC technology to ensure that it has the hardware complexity to achieve the desired rate and complexity of computation. The use of Eco-Tech II technology (part of the HDSO module) ensures that the complex computation used in the MD-active feedback canceling algorithm is achieved with the lowest current drain.

The patent-pending Spatial Feedback Tracer (SFT) in the HDSA module determines the presence of feedback and estimates the characteristics of the feedback paths regardless of the microphone mode of the Inteo (i.e., omnidirectional, fixed, or adaptive directional). While activated, the SFT correlates the input to the amplifier of the Inteo with the output from the amplifier. A high correlation is indicative of the presence of feedback while a low correlation suggests that feedback is absent. The nature of the input signal (feedback or environmental sounds) is conveyed to the Dynamic Integrator, which in turn instructs the MD-active feedback cancellation unit to generate a signal with the opposite phase as the feedback signal and route it to the input stage for its cancellation.

The patent-pending Inteo Active Feedback Canceling differs from previous advanced active feedback cancellation algorithms in many ways. Of special note is how the estimation of feedback paths is performed. The estimation of feedback paths is done in both the omnidirectional polar pattern and

**References**


the bi-directional polar pattern (hence the name multi-directional active feedback canceling). Indeed, the Inteo Spatial Feedback Tracer estimates the feedback paths by performing correlation analyses between the input from the omnidirectional polar pattern and the output from the amplifier as well as between the input to the bidirectional polar pattern and the output from the amplifier to such an input. Because the omnidirectional polar pattern and the bi-directional pattern represent the extreme polar patterns formed by a dual microphone system, the loop gain estimation that is done on these two extremes is valid for any changes in the real-life feedback path triggered by a change in polar pattern. This means less chance for feedback in more situations. This approach more than doubles the complexity of the feedback path estimation when compared to systems that estimate only one feedback path. This is possible in the Inteo because of the use of a more advanced IC chip design, the use of EcoTech II technology which optimizes the current demand, and the use of ISP technology to integrate feedback functions within the directional microphone system. A schematic of the algorithm is illustrated in Figure 1.

One advantage of ISP is the vigilance on the results of the processing. It was indicated earlier that the results of all processing are integrated by the Dynamic Integrator which then dispatches the appropriate adjustment to the appropriate processing modules. In order to be responsive to the potential changes in the feedback paths caused by sudden changes in the polar patterns, the Inteo SFT uses an adaptive rate in its feedback path estimation and cancellation. This is made possible by the action of the HD System Optimizer through the Dynamic Integrator. A slower rate is used in situations where the feedback path is relatively stable or slowly changing; whereas a faster rate is used with rapid changes in the feedback path characteristics.

Figure 2 shows the maximum available gain of the Inteo between MD-active feedback canceling algorithm “On” and “Off” and its relationship to the user gain (blue). As the polar pattern changes, the value of the maximum available gain changes as well. Despite the lowering of the maximum available gain in the directional mode, the action of the MD-active feedback canceling algorithm allows the wearer to have sufficient available gain at all frequencies regardless of the polar patterns.

The ingenious integration of the active feedback canceling system into the adaptive directional microphone system of the Inteo is an example of the successful application of ISP technology. This allows the Inteo active feedback cancellation system to yield more usable gain before feedback occurs. This additional gain allows the majority of Inteo wearers to be able to achieve their target gain without feedback in many more listening situations. The ability to use the target gain more consistently could result in better speech intelligibility, a better sound quality, and a hassle-free listening experience. In addition, an even larger vent diameter may be used successfully to further reduce the occlusion effect stemming from a shell origin (if any).

References
Audibility Extender - So the “Dead” (Region) May Hear

Henning Andersen, M.Sc.

People with a precipitous high-frequency hearing loss often miss the high frequency information even when they wear hearing aids. Sometimes it is because the high frequency gain available on the hearing aid is not sufficient to reach audibility before feedback occurs. Sometimes the severity of the hearing loss in the high frequency region is so great that it is unaidable or “dead” from the complete depletion of inner hair cells. In the former case, audibility may be achievable at the expense of a smaller vent diameter on the hearing aid. This could compromise wearer comfort because of an increase in the occlusion effect (Kuk and Ludvigsen, 2002). In the later case, acoustic stimulation of the unaidable region may decrease further the already depressed speech understanding (Moore, 2004). The loss of audibility of high-frequency sounds often compromises speech understanding and the appreciation of music and nature’s sounds (such as bird songs).

Current Situations

One of the earlier attempts to reach audibility of the high frequency sounds is the use of frequency compression or frequency transposition. In the typical scheme, the whole spectrum or a portion of the spectrum in the high frequency is compressed and transposed to a lower frequency region by a constant ratio (or frequency compression ratio). For example, if someone has no aidable hearing beyond 2000 Hz, a device with a frequency compression ratio of 2 may transpose the 2000 - 4000 Hz region to the 1000 - 2000 Hz region, thereby using the lower frequencies to decode information carried in the 2000 - 4000 Hz region. Indeed, despite the variations in approaches on frequency compression, one of the common positive findings in frequency compression is the improvement of audibility in the high frequencies as reflected by the improved sound-field aided thresholds (McDermott and Dean, 2000).

On the other hand, the improvement in aided thresholds with frequency compression has not resulted in a consistent improvement in speech understanding. In studies which showed an improvement, extensive training was often necessary (MacArdle et al., 2001) to bring about a marginal improvement. This calls into question the practical utility of such a processing algorithm.

Perhaps the most significant hindrance to the acceptance of the conventional frequency compression algorithm is the introduction of “clicks” and other audible artifacts to the processed sounds. These artifacts yield “unnatural” sound perception which many listeners find “annoying.” One possible reason for the artifacts is because after the frequency compression, the frequency ratio and the harmonic relationship of the compressed sounds are no longer the same as the original ones. Indeed, the higher the compression ratio, the more the “broad frequency range” will be squeezed into a much narrower frequency range. This severe reduction in the range of sounds gives the acoustic perception of a “click” (which is a broad range of frequencies presented in a short amount of time), making it “unnatural” and difficult for the wearers to discriminate its spectral components. The temporal waveform of the transposed signal is also very different from the original signal. Such perception would most likely be exaggerated when the transposed sounds and the original sounds do not overlap in time. Thus, many adults find it difficult to accept frequency compression, even though there may be a potential for speech intelligibility improvement with extensive training.

The Inteo Difference

Inteo includes a unique, patent-pending Linear Frequency Transposition algorithm called the Audibility Extender (AE) which allows people with an unaidable high frequency hearing loss to hear the missing high frequencies in the lower frequency region. What is unique about this algorithm is the approach Widex engineers took to move the high frequency sounds to the lower frequency. A critical design criterion of the Audibility Extender is that it does not alter the harmonic relationships between the original and transposed sounds. In addition, the temporal characteristics of the signal and the transition cues must be preserved as much as possible in order to preserve the naturalness of the transposed sounds.

The following description summarizes the action of the Audibility Extender (AE).

First, the Inteo AE receives information of the wearer’s hearing loss from the Dynamic Integrator (provided from Wearer’s Personal Information) to decide which frequency region will be transposed. Meanwhile, the Speech and Noise Tracer of the HD System Analysis module performs its spectral analysis of the environment and returns the results to the Dynamic Integrator.
The AE picks the frequency within the “to be transposed” or “source octave” region with the highest intensity, i.e., peak frequency, and locks it for transposition. In this example, 4000 Hz has the peak intensity.

Once identified, the range of frequencies starting from 2500 Hz will be shifted downward to the target frequency region. In this case, 4000 Hz will be transposed linearly by one octave to 2000 Hz.

The transposed signal will be placed on the slope of the hearing loss where the hearing loss is aidable. The 4000 Hz signal will be placed at 2000 Hz and every frequency will be shifted by 2000 Hz. For example, 3000 Hz will now be at 1000 Hz and 4500 Hz will now be at 2500 Hz.

To limit the masking effect from the transposed signal, frequencies that are outside the one octave bandwidth of 2000 Hz will be filtered out. This also keeps the frequency ratio between the original and transposed signal.

The level of the transposed signal will be automatically set by the AE so it is above the sensogram of the wearer. A manual gain adjustment of the transposed signal is also possible. The linearly transposed signal is mixed with the original signal as the final output.
The Audibility Extender is fundamentally different from other frequency compression schemes in several aspects.

1. It is active for the high frequency signals regardless of their voicing characteristics, i.e., voiced or voiceless. Thus, it is equally effective on periodic and aperiodic sounds. Systems that are active only for voiceless signals may miss high frequency periodic signals including music and bird songs.

2. It is active during all segments of speech and not at specific linguistic segments, e.g., voiced versus voiceless.

3. Typically only one octave (although two octaves may be allowed) of high frequency sounds is transposed to a lower octave. Frequencies higher and lower than the transposed region are filtered. This limits the amount of masking and avoids the need for compression.

4. Thus, it preserves the harmonic relationship of the transposed signal and the original signal. All transition cues are preserved as well. This minimizes the perception of click like artifacts that may be objectionable to some wearers.

5. The transposed signal is mixed with the original signal to give a richer, more “natural” sound perception. Systems that do not overlap the transposed sounds would risk “exaggerating” any unnaturalness of the transposed sounds.

The unique Audibility Extender is the first commercial attempt to utilize Linear Frequency Transposition to extend the audibility of high frequency sounds for people who cannot receive this information from conventional amplification. The chief advantage is the preservation of the harmonic relationship of the original and transposed sounds, yielding a more natural sound quality than otherwise possible. By extending the audibility of high frequencies, the AE has the potential to improve one’s appreciation for high frequency sounds such as high frequency music and bird songs. It may be used in open-fittings to increase the audibility of high frequency sounds while providing excellent listening comfort. Furthermore, with appropriate training, it may improve the recognition of high frequency words for those who are limited by their high frequency hearing loss. The possibilities of the Audibility Extender are endless.

References


