WIDEX
HIGH DEFINITION HEARING

AUDIOLOGICAL **BACKGROUND** BULLETIN

NO. 12

NEWS FROM **AUDIOLOGICAL RESEARCH** AND **COMMUNICATION**

FEBRUARY 2011

Order no.
9 502 2427 001 #02

WIDEXLINK - A NEW, INDUSTRY-LEADING TECHNOLOGY FOR WIRELESS TRANSMISSION

In recent years, we have seen a variety of new hearing aids on the market with different wireless communication abilities. Some offer the possibility of transmitting sound to the hearing aid from a mobile phone, a TV, or an MP3 player. Other hearing aids are capable of exchanging coordination data, and a small number are able to transmit sound to one another.

We believe that the key to new achievements in the hearing aid industry lies in the ability to use a sophisticated wireless transmission technology which will facilitate the coordination of dynamic settings between two hearing aids, high quality transmission of audio from external sound sources, and wireless transmission of data between two hearing aids.
We are convinced that wireless technology is the future in the hearing aid industry. However, the success of any given wireless transmission technology will depend on at least three key issues when applied to hearing aids: Sound quality, transmission robustness, and power consumption.

WidexLink is our new proprietary digital radio frequency transmission technology. The new technology has been designed to provide the highest audio quality and efficiency. WidexLink offers new possibilities for extended bandwidth audio transmission between hearing aids (for example, CROS and BiCROS applications), extended bandwidth audio transmission from external assistive listening devices (for example, DEX devices) to hearing aids, and the continuous exchange of synchronisation data between hearing aids and external devices (InterEar communication).

The unique, digital wireless link offers an unparalleled low latency (delay) of <10 ms when transmitting audio. This ensures minimum distortion and echo-free audio quality when direct acoustic sound in the room is mixed with transmitted sound.

The need for a new technology

Today, Bluetooth is probably the most popular wireless technology for transferring data and digital sound between devices. The Bluetooth technology is available in a large variety of “plug-and-play” chip solutions. From an engineering point of view, Bluetooth is therefore the fastest way to a digital wireless design. However, Bluetooth has some drawbacks which create serious problems in hearing aid applications.
First and foremost, Bluetooth is an extremely energy-demanding technology. The Bluetooth chip uses so much power that it has no practical application in the hearing aid industry. Hearing aid manufacturers are therefore forced to find ways to optimise Bluetooth transmission if the power consumption is to be kept at a reasonable level.

Another serious drawback is that standard Bluetooth has a high built-in latency (delay) of 150 ms when transferring audio through the Bluetooth codec. That is, transmitted sound will reach the ear 150 ms later than the direct acoustic sound in the room. In hearing aid applications, this is a serious issue because the time delay between sound sources comprises important psychoacoustic cues about direction and distance to sound sources.

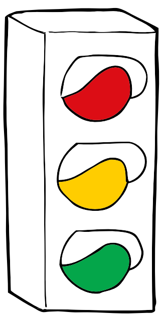
The latency can be reduced by switching off some of the technological features while transferring digital audio, but there is a limit to how much. As far as is known, the Bluetooth latency can be reduced to no further than approximately 45 ms, which is not sufficient to avoid artefacts in hearing aid applications.

Latencies and artefacts

A small delay of 1 to 10 milliseconds is unproblematic. However, when the delay between transmitted and direct sound becomes longer than 10 milliseconds, artefacts begin to occur. The first artefact to occur will be an audible comb-filter effect. It is called “comb-filter effect” because it filters away frequencies like a comb, making notches in the spectrum. The resulting sound will be hollow and metallic. Above 40 milliseconds, the streamed sound will be perceived as an echo of the direct sound. An additional problem can occur when sound is transmitted from for example a TV. When the delay reaches around 150 milliseconds, which is the default delay with Bluetooth, lip movements will begin to appear unsynchronised with the sound. Thus, hearing aid manufacturers who rely on Bluetooth or another technology which introduces a delay above 10 ms will not be able to avoid artefacts in their applications.

Also included on the list of Bluetooth drawbacks are the relatively large chip size and the high current consumption, which are obviously essential parameters in hearing aid design.

The artefacts that can occur as a result of the delay between direct sound and digital audio stream are listed in the figure below:



RED	
30-40 ms delay	Clear echo effects start to occur
> 150 ms delay	Severe echo effect. Audio and TV picture are not in sync
YELLOW	
10-30 ms delay	Sharp metallic sound due to comb-filter effect
GREEN	
0-10 ms delay	Transparent sound

Figure 1. Psychoacoustic artefacts due to latency (delay) between direct sound and transmitted sound.

The new technology from Widex has unparalleled low latency of less than 10 ms between the direct acoustic sound and the digital audio stream. And in CROS and BiCROS applications, the delay is even shorter.

Advantages of WidexLink

It is no simple matter to design a system which can simultaneously maintain high audio quality, low battery drain, and robustness against transmission errors. Widex achieved these goals by developing a very efficient audio coding method which is custom-designed for use in a hearing aid platform.

It was considered of the utmost importance that the highest possible audio quality should be maintained with the new wireless technology. In order to achieve this goal, several key components of the wireless system were carefully designed. First, an efficient and robust audio codec (short for encoding/decoding) and radio frequency (RF) transmission system were developed to ensure fast, stable, and trouble-free transmission of sound and data during normal use. It was a requirement that the digital audio codec be based on a coding principle which will ensure that the signal is perceived to be as close to the original sound as possible. Another factor in maintaining a high audio quality was a low audio delay over the wireless transmission system. To achieve this, the digital coding system was designed in such a way that no data would need to be re-transmitted and consequently slow down the transfer of the audio signal.

Ensuring battery efficiency was also a major concern in the design of WidexLink. This goal was achieved through the employment of a very efficient data compression method, and the invention of a new, highly sensitive radio receiver which permits low transmission power.

Robustness was achieved in multiple ways. In addition to the inherent robustness of the audio codec and the radio receiver mentioned above, robustness was also attained by means of a highly efficient channel coding which detects and handles errors quickly and securely.

How WidexLink addresses the key factors will be discussed in more detail below.

HIGH SOUND QUALITY

Audio bandwidth

Audio bandwidth is one of the key factors in maintaining a high sound quality. Thanks to the new technology in the CLEAR product range, we are able to offer an exceptionally broad audio bandwidth in models with a ClearBand receiver, stretching from 100 Hz to 11 kHz for transmitted sound. This is industry leading.

One area where a broad audio bandwidth makes a clear difference for hearing aid users is when they listen to music. The high frequencies provide ambience and brilliance to the sound. Thus, the sound experience will be somewhat richer with an upper bandwidth of 11 kHz when listening to the crispy sound of a hi-hat or cymbals, for instance. Similarly, an audio bandwidth stretching as far down as 100 Hz will produce a fuller bass.

Codec

Another way to maintain a high sound quality is to develop an efficient codec. Digital audio data as we know them from CDs are extremely bulky. It is therefore necessary to reduce digital audio data in some way, as the transmission bandwidth is too narrow to effectively transmit the raw audio signal. This is achieved by means of data compression (not to be confused with the dynamic compression of the audio signal in the hearing aid).

Digital audio data compression is achieved by means of a set of complex algorithms called an audio codec. An audio codec consists of two parts: encoding and decoding. The purpose of the audio encoding is to reduce the size of the digital data representing the original signal. The purpose of the decoding is to reconstruct the encoded audio signal in a manner which ensures that it is as close as possible to the original audio signal. This process is analogous to the shipping of a parcel by mail. If you wished to send an office desk from the U.S. to Europe, you could simply place the assembled desk inside a large box and send it. This shipping method would be quite expensive, though, due to the size of the parcel. A more efficient and less expensive method of shipping the desk would be to disassemble the desk into smaller pieces and package it in a much smaller box. The same principle applies to the transmission of digital audio. The cost of sending digital audio is related to its size. The larger the size of the digital audio data, the larger the transmission bandwidth has to be.

Two different compression techniques are typically used in order to squeeze the audio information into a smaller package. Both compression techniques rely on the fact that audio signals have a great amount of redundancy.

One commonly used technique is Redundancy Coding. This technique is similar to the process used when computers compress files into ZIP files. This type of compression can be demonstrated by a simple example:

If we need to send the number 1000000, we may compress it to 10^6.

The compressed number represents the same digits as the original number, but comprises fewer characters. If each character requires four bits, the uncompressed number would require 28 bits (4 bits x 7 characters), while the compressed number would require 12 bits (4 bits x 3 characters). The technique is effective with data where there is lots of redundant information as is the case with digital audio signals. However, the technique has one major drawback; namely that it is relatively time-consuming. It is therefore not very suitable for hearing aid applications, in which a minimal delay is of the utmost importance.
Another commonly used technique for achieving a low bit rate is Irrelevance Coding. It is widely used today to create MP3 and other types of digital audio files. Widex also uses this technique for analysing and compressing audio data in a special part of the codec.

It is well-known that raw, uncompressed audio contains more information than the human ear can actually detect. Our irrelevance coding algorithm removes all perceptual redundancies by extracting all of the irrelevant audio information which cannot be heard by the listener due to psychoacoustic masking effects in the cochlea. In other words, the irrelevance coding algorithm uses knowledge of masking to remove audio signal elements which are outside the limits of the human auditory system.

Irrelevance coding relies on a phenomenon known as the Simultaneous Masking Effect. When listening to a soft and a loud sound simultaneously, it is often difficult to hear the soft sound because it is drowned by the loud sound. The masking effect is largest when the soft sound is in the same frequency range as the loud sound (Moore, 2006: 66). This psychoacoustic phenomenon is very useful in relation to audio data compression. The irrelevance algorithm utilises knowledge of this psychoacoustic effect to remove softer, less dominant sounds which will be masked by louder, more dominant sounds, from the audio signal.

Basically, our irrelevance algorithm reduces the amount of audio data that needs to be transmitted wirelessly by removing sounds which would not be audible to the listener in any case. And by removing sounds in the audio signal which the listener cannot hear anyway, while preserving the sounds the user can hear, the audio signal can be reduced significantly in size without compromising the high sound quality.

SAFE DIGITAL TRANSMISSION

Channel coding

An important aspect to consider when sending any type of digital data over a wireless connection is the potential for errors induced by radio frequency interference.

Errors will occur from time to time with any kind of transmission and especially in wireless transmission. The distance between the devices may change, the orientation of the antenna in the controller might be altered, or interference from radio noise might disturb the connection. Such errors must be handled effectively to minimise the inconvenience caused to the hearing aid user in the shape of crackling, dropouts in the sound, etc. Therefore, in order to prepare the audio data for transmission and ensure the integrity of the transmission with respect to correct receiver as well as the quality of transmitted data, channel coding is introduced.

Error detection and handling

The main task of the channel coding algorithm is to provide a method for ensuring that the digital audio signal which is received is indeed correct and error free. A common and very basic way to do this is to calculate what is called a checksum on the basis of the compressed data. The checksum is enclosed in the shipment alongside the data. When the data are received at the other end, a comparison of the checksum and the data is conducted to determine if any errors have been introduced into the digital audio signal.

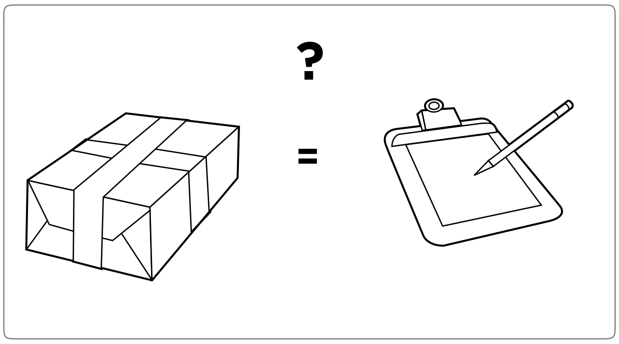


Figure 2. The data in the shipment are compared to the checksum on arrival to determine if any errors have arisen.

However, a checksum will only establish whether or not an error has occurred. It does not provide a solution to how errors should be handled. A major advantage of Widex' channel coding algorithm is that, thanks to a so-called error corrective code, it ensures that a restricted number of errors can be both detected and corrected.

In a Bluetooth transmission technology, for example, the channel coding will by default ask for a retransmission of data which did not pass the receiver's error checking algorithm (www.bluetooth.com). Since retransmission requests will cause an extra delay whenever the system has to wait for the repetitions of the data to arrive, such a method is not a very good choice in a hearing aid application.

Another method involves the removal of audio data packages which contain errors. This method is for instance used in connection with DAB (Digital Audio Broadcasting). DAB receivers cannot request a retransmission of signals vitiated by errors. Errors in audio data are simply handled by making dropouts in the sound, resulting in there being no sound playback when errors occur. An error handling method which results in clicks or dropouts in the sound is obviously not very suitable for a hearing aid application either.

Widex relies on a special channel-coding technique which is based on the principle of Graceful Degradation. This technique has none of the unfortunate by-products (long delay and dropouts) mentioned above. Instead, it provides a smooth, seamless listening experience for the hearing aid user.

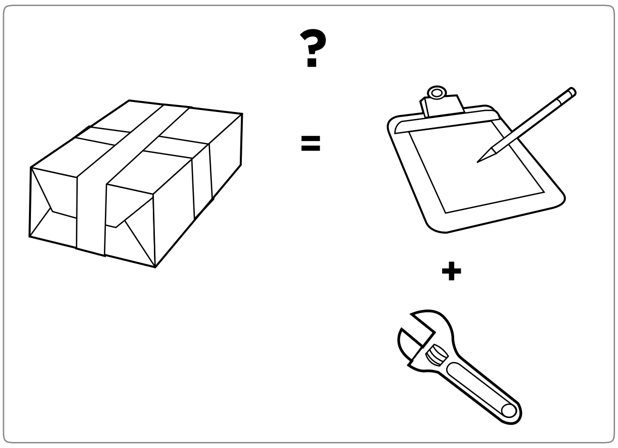


Figure 3. Widex' channel coding algorithm can both detect and handle errors.

Widex' channel coding algorithm has been designed to ensure that a small number of transmission errors can be corrected by the algorithm itself. A larger number of errors will be handled by means of the above-mentioned Graceful Degradation-based technique. This technique ensures that a large number of errors will not result in any abrupt changes in the output signal, such as dropout or crackling, heard by the hearing aid user. If the errors are too numerous for the algorithm to be able to correct them, the result will be a gradual fading of the sound. While transmission is interrupted, the HA will switch to the Master program. When the quality of the transmission channel is good enough to allow audio transmission once more, the sound will gradually fade in again to provide a nice, smooth listening experience for the hearing aid user.

Digital Audio Transmission by WidexLink

The WidexLink technology makes it possible to minimise the amount of data that needs to be transmitted in order to ensure a high quality output signal. This is essentially possible thanks to the identical audio generators in the encoder and the decoder. More specifically, because the WidexLink encoder and decoder both contain identical synthetic audio generators, it is not necessary to transmit the original signal, or even the synthetic signal. All that needs to be transmitted is information about the discrepancy between the original sound signal and the synthetically generated signal. A more detailed discussion is included in the sections below.

The WidexLink encoding procedure can be divided into five main stages. In the first stage, the original audio signal is compared with a synthetic signal generated by a Synthetic Audio Generator. A Discrepancy Analyser generates information about how close the synthetic sound is to the original. A perceptual model is then applied to determine if discrepancies are audible or not (irrelevance coding). To keep the amount of data to a minimum, only audible discrepancies are allowed to influence the sound generation process. Next, the best approximation to the discrepancy between the original sound and the synthetic signal is retrieved from a large number of synthetic sounds stored in a Sound Sample Archive. And finally, information about which sample is the best approximation to the discrepancy between the original and synthetic signal is transmitted to the decoder.

Bluetooth™	→	Resend	→	Long delay – default 150 ms
DAB (Digital Audio Broadcast)	→	No resend	→	Dropouts in sound
WidexLink	→	No resend	→	Smooth fade-in and fade-out

Figure 4. Summary of three methods for error handling in connection with digitally transmitted data.

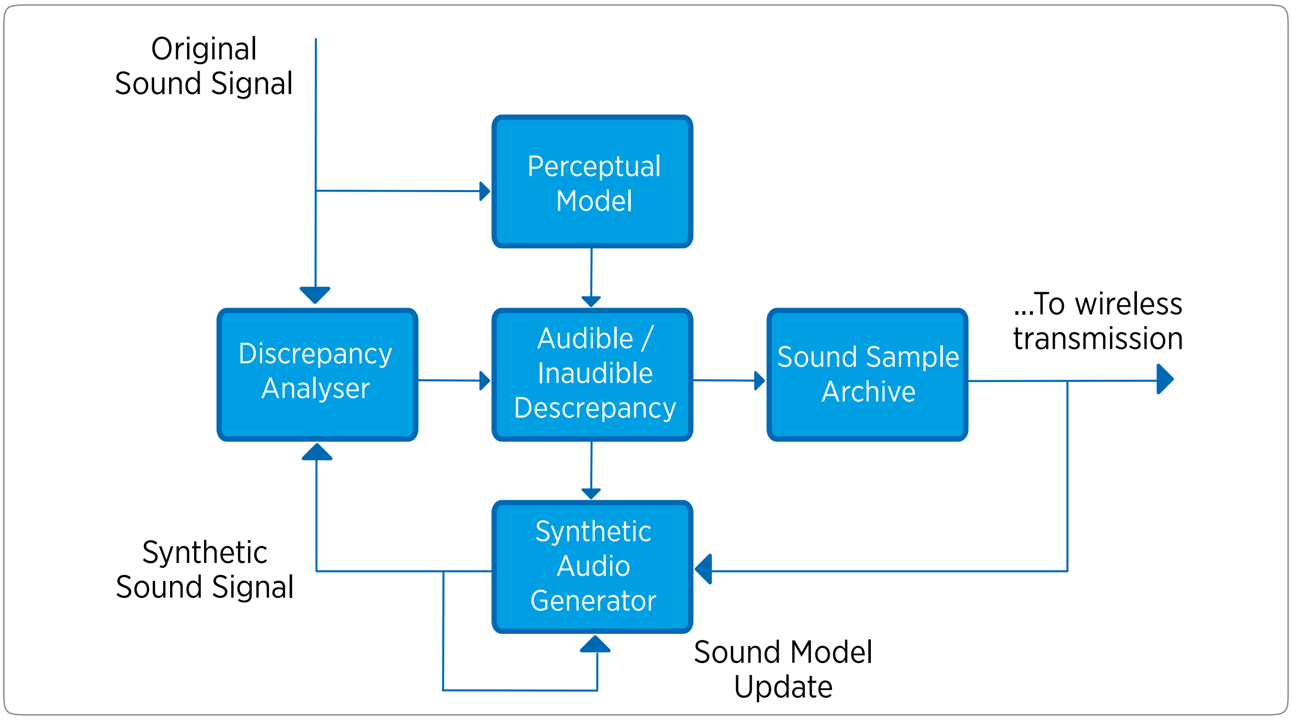


Figure 5. The WidexLink encoder principle.

With WidexLink, the sampling rate is 25.44 kHz, which means that the above procedure is repeated 25,440 times for each second of sound. This enables us to exploit the fact that there is typically not very much variation from one sound sample to the next to generate an increasingly accurate synthetic representation of the original signal.

The first time a discrepancy analysis has been conducted, information about the best approximation is used to create a Sound Model with information about the discrepancy between the original and synthetic sound. This knowledge about the discrepancy between the synthetic and original sound contained in the model is permitted to influence the generation of the next synthetic sound, whereby the difference between the new synthetic sound and the original sound sample can be reduced. By updating the model every time a sample has been processed by the encoder, the system is able to reduce the difference between the original and synthetic signal to a minimum very quickly.

The decoder in the hearing aid contains an exact replica of the synthetic audio generator module in the encoder. Thus, information about the discrepancy-based best approximation is sufficient to provide all the necessary information for the synthetic audio generator in the hearing aid to be able to generate an exact copy of the synthetic signal in the encoder. In other words, the output signal generated by the hearing aid is a 100% synthetic sound identical to the synthetic sound generated in the encoder. The decoding sequence is presented schematically below.

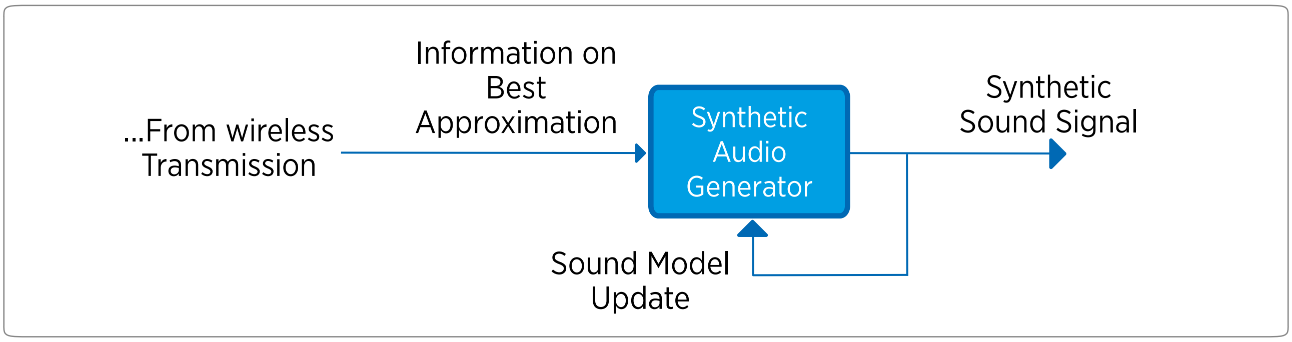


Figure 6. The WidexLink decoder principle.

Transmission robustness

A new, highly accurate and robust receiver has been developed for the WIDEX CLEAR product range to ensure safe transmission and low battery drainage. The accuracy of this new, patent-pending receiver enables us to operate with a low transmission power, which in turn contributes towards extending battery life.

Moreover, the modulation technique employed to send digital data over the wireless system also contributes to maintaing a high degree of robustness.

When sending digital data over a wireless system, a modulation scheme must be used. Modulation essentially determines how the digital information is sent through the radio frequency spectrum. One very commonly used digital modulation technique is Frequency Shift Keying (FSK). In an FSK modulation system, two frequencies are used to represent a binary “0” or a binary “1”, respectively (see figure 7 below). So once the audio has been digitised, i.e., turned into a series of 0s and 1s, it can be transmitted by means of two different transmission frequencies which represent either a 0 or a 1. The receiver has to detect which of these two frequencies is being transmitted in order to determine if the transmitter is sending a 0 or a 1. The receiver then demodulates the signal by interpreting the frequencies received as either a 0 or a 1.

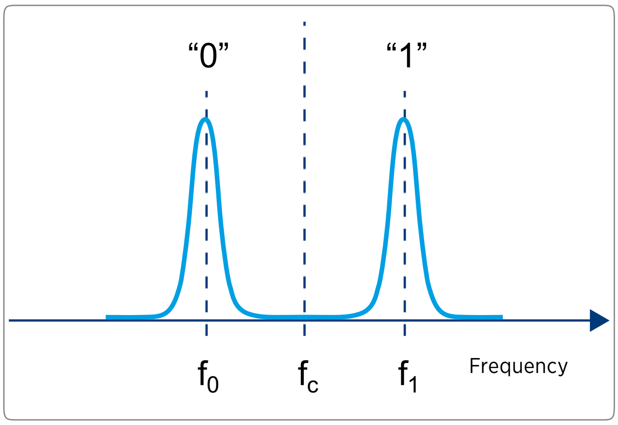


Figure 7. Illustration of the Frequency Shift Keying principle. In an FSK modulation system, two frequencies represent a binary ”0” or a binary ”1”, respectively.

Traditional methods of wireless demodulation employ a simple two point sampling of the received wireless signal. This means that, in effect, the receiver relies on only two measurering points when it has to determine if the received signal is a “1” or a “0”. Figure 8 below contains a model of the demodulation of a signal without noise.

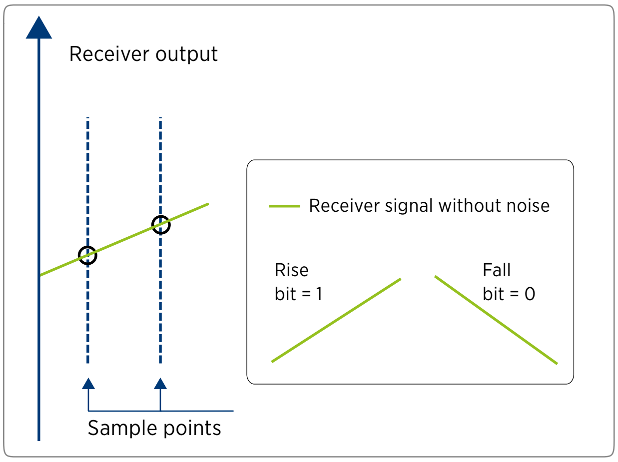


Figure 8. Model of the demodulation of a signal without noise. Traditional methods of wireless modulation employ two points of sampling. In the example, the sample points indicate a rise. The signal will therefore be interpreted as a 1.

The demodulation method described above works very well with a clear signal and no noise. However, as illustrated in figure 9 below, using only two sampling points to determine if the transmitted signal is a 1 or a 0 can result in mistakes if noise is also present in the signal.

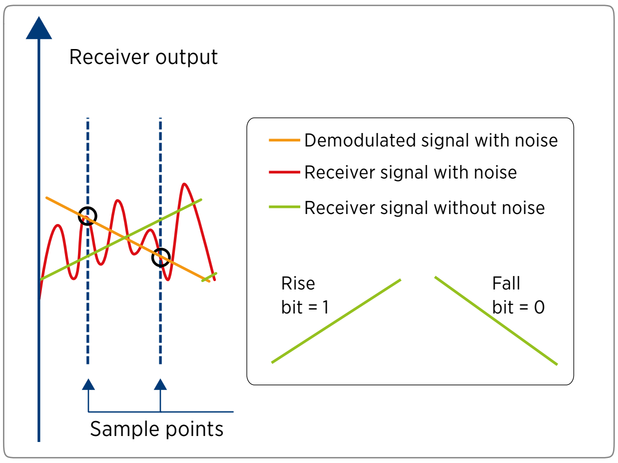


Figure 9. Model of the demodulation of a signal with noise. Signals can be wrongly identified when noise is present in the signal.

Consequently, Widex has developed a more secure variety of the FSK method for the reception of wireless transmissions with a high degree of noise in the transmitted signals. The new method introduces a larger number of measuring points than the traditional two, which means that the receiver is able to determine with a much higher degree of certainty whether a received signal should be interpreted as a 0 or a 1.

Application

The new digital transmission technology is a central element in our new WIDEX CLEAR product range. It is used for the transmission of both data and audio signals in a large number of situations.

Transmission of audio

WidexLink is used for the transmission of audio signals from external devices to the hearing aids when the user watches TV, talks on his mobile phone, or listens to music.

Transmission of audio via WidexLink

Typical situation:	Sender and recipient:
TV	TV-DEX – Hearing aids
Hi-fi	TV-DEX – Hearing aids
Mobile phone	M-DEX – Hearing aids
Personal audio device (Ipod, mp3 player, etc.)	M-DEX – Hearing aids

Transmission of data

WidexLink is also employed in the transmission of data between a remote control (RC-DEX) and hearing aid, and in the exchange of synchronisation and coordination data between hearing aids (InterEar communication).

Transmission of data via WidexLink

Sender and recipient:	Feature:
RC-DEX - hearing aid	Remote control
Hearing aid - hearing aid	HA synchronisation • Volume control • Program shift
Hearing aid - hearing aid	IE coordination • Compression • Feedback cancelling • Noise reduction
Hearing aid - hearing aid	WidexLink Surveillance • Lost partner alarm

References:

Moore, B. C. J. (2006). An introduction to the psychology of hearing. Elsevier Publishing Company

www.bluetooth.com