

## Intelligent binaural algorithms to improve speech understanding

With Binaural VoiceStream Technology® it is possible now to realize algorithms supporting binaural hearing providing a solution to improve speech intelligibility also in complex listening situations for the hearing instrument user.

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### Introduction

The technical possibilities of hearing instrument technology have continuously advanced over the last 20 years. Three major steps can be identified:

- Channel-independent compression was introduced in the late 1980s
- Digital signal processing started in the mid-1990s
- The implementation of real-time binaural signal processing began in 2008

These advances in hearing instrument development have led not only to the increasing miniaturization and bigger complexity of the hearing instruments, but have also ensured that end users are more satisfied with their instruments. This can be chronologically illustrated by the well-known MarkeTrak market analyses (Kochkin 2010). The number of binaural supply systems is also continuously rising (Kochkin 2009), so that the focus in hearing instrument research is moving towards binaural signal processing

technologies. New approaches are to be expected. The introduction of the new Quest platform by Phonak extends the range of binaural algorithms. These algorithms support binaural hearing for hearing instrument users. The binaural algorithms offer the potential for the hearing instruments to solve the problems reported by hearing instruments (Kochkin 2010):

- Speech understanding in large groups when facing the speaker
- Speech understanding in noise, when the listener is not facing the speaker
- Making telephone calls in noisy situations
- Speech understanding in windy situations

This article provides an overview of the binaural approaches implemented on the Phonak Quest platform and shows to what extent the new platform offers solutions for the difficult acoustic situations described.

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## Binaural VoiceStream Technology®

The exchange of audio data between hearing instruments is an important step towards the use of binaural signal processing technologies. It makes it possible to support not only the peripheral, but also the binaural hearing properties of the impaired user. However, the wireless system required for the exchange of data between the hearing instruments needs to comply with important peripheral conditions. It is essential that the transmission delays are short and that power consumption is kept to a minimum. The system is therefore based on an inductive principle that is known for its low power consumption. However, unlike a telecoil, it is designed to transmit digital data. This ensures stable data transmission, which is essential for processing binaural algorithms. The wireless system works at a transmission frequency of 10.6 MHz. This relatively low frequency (compared to Bluetooth, which operates at 2.4 GHz) was chosen to reflect the need for low power consumption. It is also advantageous for inductive coupling. The bandwidth for the acoustic signal is 8 kHz, which is more than

sufficient for binaural algorithms. Transmission is bidirectional, so that the required data rate is thus doubled. The system provides an overall data rate of 300 kbit/s for audio and control data.

Bidirectional transmission of the audio signal requires a data rate of 640 kbit/s when a scanning rate of 20 kHz and a resolution of 16 bits with linear encoding are assumed. The available system is thus unable to transmit the full signal required. The audio data is thus encoded before transmission (as in MP3 encoding). The codec used has a data rate of 300 kbit/s and a delay of approximately 2 ms. It is therefore optimized for use in the binaural hearing instrument. This data rate can be transmitted by the wireless system described above and provides sufficient buffer capacity to exchange the control data required between the two hearing instruments. The extremely short delay allows binaural processing and preserves the natural information about spatial objects.

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## auto StereoZoom

The multi-microphone technology in hearing instruments is currently the key element for separating the effective signal from the noise signal and improving speech understanding in a measurable and subjectively perceptible way (Ricketts and Mueller 1999; Chung 2004). The two signals should differ in terms of their spatial properties. It is also assumed that the effective signal arrives from the front. The sensitivity of the system is directed forwards, towards the effective sound source, by using a simple difference calculation and the delay between the two microphone signals. This beam format can either have a fixed or variable directional characteristic that adapts to suppressing changing or moving sources of noise by evaluating a number of the parameters of the input signal.

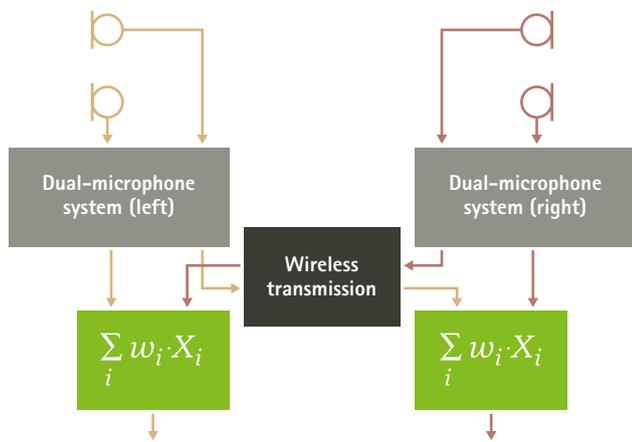
A microphone system that processes the information from two omnidirectional microphone signals is called a directional microphone of the first order. Effective noise suppression by a microphone system of the first order requires the direction of the source of noise to be offset by more than 60° from the direction of the source of the effective signal, so that the effective and noise sources are not within the main lobe of the directional characteristic. The system described thus has limits that affect everyday life, for example when the source of noise is within an angle of 60° of the effective source or when the effective source is outside the reverberation radius.

These limits can be exceeded by narrowing the main lobe of the directional characteristic of the microphone system and by making the system more sensitive. This can be achieved by increasing the order of the microphone system. A two-dimensional approach is sufficient, as the acoustic objects are usually in the same hemisphere as the head. Approaches used in sound studio technology, where several microphones are arranged behind one another, cannot be implemented, due to the small design size of the hearing instruments.

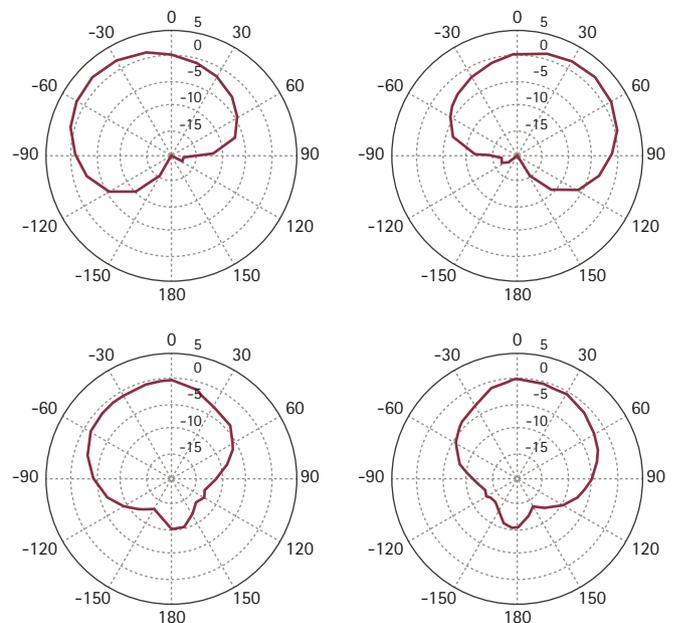
A hearing instrument system has a total of four microphones when the binaural supply is not considered as two individual systems but as one single system. This corresponds to the natural binaural hearing instrument of humans, which also has an overall characteristic consisting of the two individual directional characteristics of the two outer ears. StereoZoom combines the signals of the two dual-microphone systems in the right and left hearing instruments via a wireless link and then processes them. Figure 1 shows a schematic drawing of this link. In the first stage, the input signals of both the microphones of both hearing instruments are used to calculate a standard dual-microphone system of the first order. The respective output signal is sent to the contralateral side using wireless transmission to cover the full bandwidth of the audio data. There it is processed together with the output signal of the ipsilateral dual-microphone system, using

a predefined weighting function. This results in a microphone array consisting of 4 microphones, together forming a microphone system of the third order. The phase and amplitude characteristics of the microphones must be closely matched for this purpose. The microphones are already matched before they are installed in the instruments. It can also be done automatically while the instruments are being worn, using an appropriate algorithm that compensates for possible differences in sensitivity or phase characteristics. Figure 2 shows the polar diagram for such a microphone system in a free field, which was measured in an anechoic chamber. A clearly narrower main lobe of the directional characteristic with zero points at approx.  $\pm 45^\circ$  can be seen. This implies that even when a noise speaker is arranged more closely to the desired speaker, the microphone system can still recognize these sources as being different objects. There is a clearly increased

attenuation of all signals that do not come from the front and are interpreted and suppressed as noise signals. The signal-to-noise ratio (SNR) for non-coherent signals is further improved by approximately 3 dB compared to two separate monaural dual-microphone systems. The "virtual" increase in the distance between the microphones clearly boosts the differences between the individual microphone signals from different points in the room, so that this system can also provide a more effective spatial effect at lower frequencies. Figure 2 further illustrates that the right and left hearing instruments do not have an identical polar diagram, rather their spatial separation is retained. This is important in terms of spatial perception and localization. Identical directional characteristics on the right and left would result in a diotic perception of sound that lacks any directional information for lateralization.



**Figure 1**  
Block diagram of the technical implementation of StereoZoom.



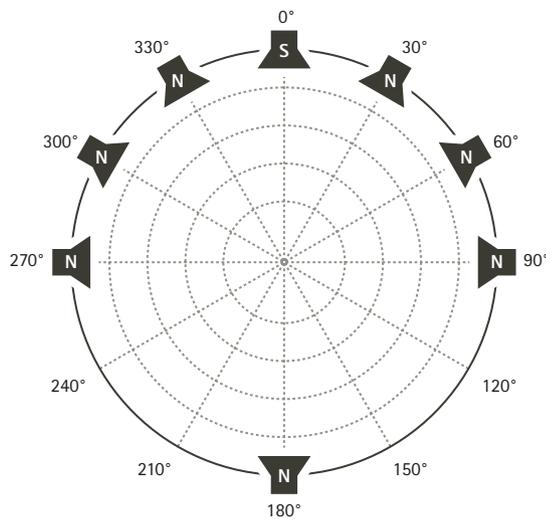
**Figure 2 top**  
Polar diagram of a static beamformer (directional microphone of the first order) for the left and right KEMAR ear with broad-band stimulation. Conditions: Free-field, low-reflection room.

**Figure 2 bottom**  
Polar diagram of StereoZoom (directional microphone of the third order) for the left and right KEMAR ear with broad-band stimulation. Conditions: Free-field, low-reflection room.

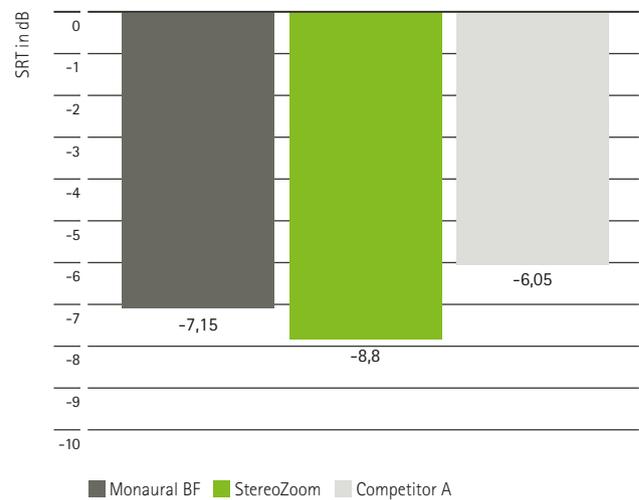
StereoZoom has already been available in premium products since the introduction of the Phonak Spice platform, and the effectiveness of the algorithm has been proved by a number of studies (Nyffeler 2010a, Nyffeler 2010b, Nyffeler 2010c, Timmer 2010).

StereoZoom was compared with a monaural beamformer and another system available on the market in a study carried out at the hearing center in Oldenburg. The Oldenburg Sentence Test (Wagener et al. 1999) was used in a defined noise environment, in which seven loudspeakers presented cafeteria noise from different directions (see Figure 3). The noise signal was therefore considered to be diffuse. The two front loudspeakers were located at 30° and 330°, so that the main lobe of the directional microphone system had to be very narrow in order to allow for the separation of the speech and noise signals. Fifteen test subjects with moderate

hearing loss participated in the study. Figure 4 shows the median of the speech reception threshold (SRT) for the different directional microphone approaches. A negative SRT indicates a better result. StereoZoom was clearly better than the two other directional microphone systems of the first order. The test subjects with StereoZoom could understand 80% of the words of a sentence at 1.65 dB lower than the equivalent threshold for the monaural beamformer and 2.75 dB lower than that of the competing system. The results are statistically significant. They show that only a microphone system of a higher order is able to discriminate between effective and noise sources when their directional distance is less than 30°. The study further confirmed that StereoZoom clearly improves speech understanding not only under ideal conditions but also in diffuse noise, which is similar to a real life situation.



**Figure 3** Test arrangement for the Oldenburg Sentence Test measurements to investigate the StereoZoom. Cafeteria noise was presented from loudspeakers located at 30°, 60°, 90°, 180°, 270°, 300° and 330° and their signals were shifted in time so that a diffuse noise signal could be assumed for this purpose. The speech signal was presented from 0°.



**Figure 4** Results of the Oldenburg Sentence Test speech recognition measurement under the conditions described in Figure 3. The results show the median of the SRT at which 80% of the speech is understood for the three applications: monaural beamformer, StereoZoom and the competitor A's directional system.

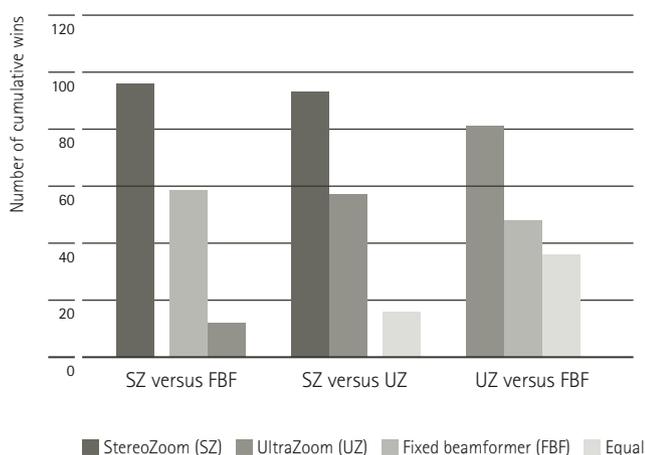
Kreikemeier et al. (2012) investigated the effectiveness of the binaural beamformer in comparison with the monaural approach for people with severe hearing loss (14 test subjects) using a pairwise comparison. Three different beamformer approaches were compared by presenting spoken words with noise:

- (monaural) beamformer with fixed directional characteristic (FBF)
- (monaural) beamformer with adaptive directional characteristic (UltraZoom – UZ)
- (binaural) beamformer with fixed directional characteristic (StereoZoom – SZ)

The results in Figure 5 show that the binaural beamformer (SZ) prevails in most direct, subjective, pairwise comparisons with the monaural approach which also has a fixed directional characteristic (see "SZ versus FBF"). The direct comparison "SZ versus UZ" – a

binaural system with fixed directional characteristics compared to a monaural system with adaptive directional characteristics – showed StereoZoom performed better more often, although an adaptive directional microphone works better in diffuse noise than a fixed approach. The results of the direct comparison of the two monaural approaches prove that this assumption is correct: In the comparison "UZ versus FBF", UltraZoom is clearly preferred more often, as the adaptive directional microphone can better adjust to the specific noise conditions. Kreikemeier et al. (2012) concluded from these results that the "more complex directional microphone technologies and in particular narrow directional beams lead to subjective improvements in difficult hearing situations".

It can be concluded from all the studies presented that the StereoZoom is superior to monaural dual-microphone systems in objective as well as subjective tests.



**Figure 5**  
Presentation of the absolute number of cumulative wins in all pairwise comparisons between the three beamformer approaches. The test/retest and forwards/backwards results in all dimensions (speech understanding, noise suppression and sound quality) were accumulated.

In order to utilize StereoZoom in the Spice platform, the end user had to manually activate this program. StereoZoom could, therefore, only be activated with certain types of hearing instruments or by remote control. The hearing instrument end users could also decide when to activate StereoZoom. However, it is often difficult for hearing instrument end users to select the correct program for the given situation. It can therefore happen that hearing instrument end users use StereoZoom either too rarely or too often. This may increase power consumption and lead to the activation of StereoZoom in situations where it offers no additional benefit or is even counterproductive.

This led to the development of the automated activation for StereoZoom, so that beamforming can also be used with the SoundFlow automatic system. The automatic system was developed to ensure that the StereoZoom is only switched on in relevant situations to prevent the disadvantages mentioned above. The algorithm analyzes the input signal of both hearing instruments according to a multitude of parameters (e.g. the input level and classification of the current acoustic situation), which are then compared by the two hearing instruments, using the wireless connection. The decision from both hearing instruments to activate StereoZoom is only made when a specific combination of these parameters occurs. The activation is not

suddenly introduced but follows a hysteresis curve, so that StereoZoom is only switched on in situations in which the parameters measured remain stable for a specific period. The sensitivity of the switch can be fine-tuned in Phonak Target 3.0. The time constants of SoundFlow and auto StereoZoom are linked to each other. This ensures that the narrow directional beamformer is only switched on or off when the parameters controlled by the SoundFlow are set accordingly. This affects the response time and the delays of the system in changing situations.

These technical measures ensure that auto StereoZoom is only activated in relevant situations and is always available when the use of a binaural microphone system provides actual benefits. The additional power consumption used by StereoZoom is limited to situations in which StereoZoom provides additional benefits. The automated activation further ensures that the users of hearing instruments are no longer restricted to specific models when they want to use the StereoZoom feature.

**Therefore, Phonak offers an effective solution for the complex listening situation "Speech understanding in large groups when facing the speaker" based on Binaural VoiceStream Technology®.**

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## auto ZoomControl

The previously described directional microphone systems are based on the assumption that the effective sound source is in front of the hearing instrument end user, i.e. at an angle of approx. 0°. However, there are situations in which the speech signal arrives from a different direction than the front and the hearing instrument end user cannot (or does not wish to) turn his head to look at the speaker. A typical situation is when an end user drives a car and has a conversation with the passenger but has to focus on the front (traffic). It is desirable to focus the directional microphone to the side of the passenger.

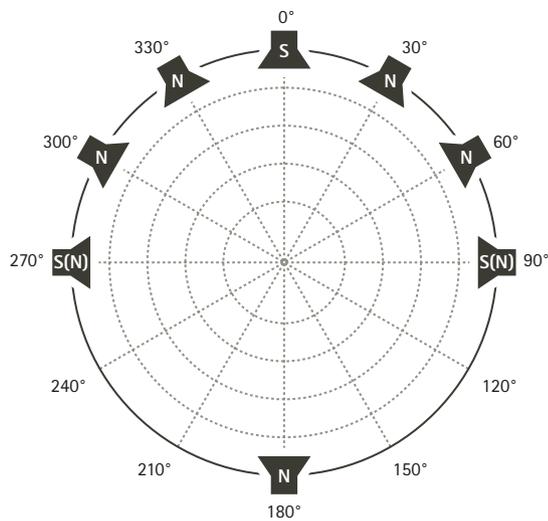
The auto ZoomControl algorithm was made for such applications. It has been implemented in all Phonak premium wireless devices since the introduction of the Spice platform. This function also makes use of the four microphone network to detect the appropriate situation quickly and reliably and automatically set the most effective directional selection. The hearing instrument end user must switch both hearing instruments to a separate, manual program to utilize auto ZoomControl. The wireless link must remain permanently active, as it facilitates the exchange of data for the classification of the acoustic scene and the transmission of the audio data.

The algorithm first detects which direction the speech signal with the highest SNR and/or level originates from. It then switches to the appropriate focus direction. Switching takes a few seconds, as with auto StereoZoom, in order to prevent uneven system behavior. The specific switching time further depends on the noise level and the situation that the classifier has detected.

When the speaker is in front of or behind the hearing instrument end user, both hearing instruments are set to a fixed directional effect towards the front (cardioid) or towards the back (anti-cardioid) to ensure optimal speech intelligibility. In this case, the wireless link is only used to exchange control parameters between the hearing instruments and to monitor whether the acoustic conditions are stable. When the speaker is on the right or left side of the hearing instrument end user, the hearing instrument closest to the speaker (ipsilateral), which has a better SNR due to the proximity to the speaker and the absence of the head shadow effect, is switched to omnidirectional. The microphone signals of the hearing instrument on the other side (contralateral) are attenuated. The wireless link transfers the audio signal from the ipsilateral to the contralateral side. This is done taking into account the different transmission functions such as the microphone

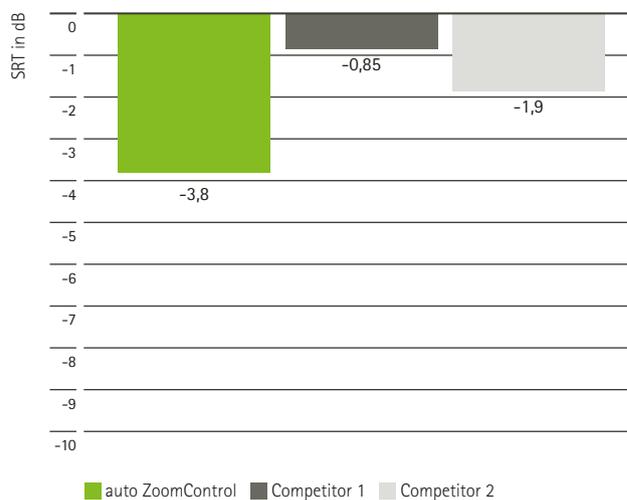
location effect (MLE), so that switching takes place without noticeable changes in loudness or tone. The MLE for the sound arriving from the front is used in each case. The amplification of the signals in both hearing instruments is based on the amplification/compression for the respective ear to ensure that the system also works in case of asymmetric hearing loss.

Improved speech intelligibility whilst retaining the ability to localize has already been proven in numerous studies (Nyffeler 2010d, Nyffeler 2010e). A later study, which is being performed in Oldenburg and was already mentioned above in the "auto StereoZoom" section, investigates auto ZoomControl in comparison to two competitor systems. Objective speech intelligibility was determined by using the Oldenburg Sentence Test. The noise signal was presented at different loudspeakers simultaneously in order to generate a diffuse noise signal.



**Figure 6**  
Loudspeaker setup for the Oldenburg Sentence Test measurements to investigate auto ZoomControl. Cafeteria noise was presented from loudspeakers located at 0°, 30°, 60°, 180°, 300° and 330° and their signals were shifted in time so that a diffuse noise signal could be assumed for this purpose. The speech signal is presented from an angle of 90° or 270°.

However, the speech signal is, in this case, not presented from the front, but randomly from either 90° or 270° (compare Figure 6). The test subject did not know from which of the two loudspeakers the Oldenburg Sentence Test would be presented through, thus this test also measures attention capabilities. A notification sentence was presented to allow the system sufficient time to focus towards the speech sound source. Figure 7 shows the median of the SRT (combination of the measurements for both angles, 90° and 270°) for the various measuring conditions. It is obvious that auto ZoomControl is clearly better than the other two systems. The auto ZoomControl reaches an SRT that is 1.9 dB better than Competitor 2 and nearly 3 dB better than Competitor 1. Both values are clearly above the detection level of the Oldenburg Sentence test, thus the results can be considered significant.



**Figure 7**  
The SRT median for the OLSA measurements are combined results from measurements made at 90° and 270° for the three test conditions: auto ZoomControl and two competitor's systems.

Wu et al. (2012) compared three different options for boosting communication between a driver of a vehicle wearing a hearing instrument and their conversation partner sitting in the passenger seat or rear seat. The three options used were as follows:

- a directional microphone focusing backwards (HA1)
- a system that used the signal from the hearing instrument (ipsilateral) side with the highest SNR which was transferred to the hearing instrument on the other (contralateral) side via wireless streaming (HA2 corresponds to auto ZoomControl)
- a system that suppressed the interference on the side with the lower SNR (HA3).

To prepare the perceptive measurements, the speech or interference signal in a driving vehicle was presented via speakers, either from the side (side-talker position) or from behind (back-talker position) (see Figure 8). At the same time, the output signal of both hearing instruments, which were mounted on the KEMAR in the passenger seat, was recorded. These recordings were subsequently aurally corrected and presented to 25 test subjects suffering from symmetrical hearing loss via in-ear headsets, with objective speech understanding being evaluated. The Connected Speech Test (CST) (Cox et al. 1987) results for the HA2, i.e. the auto ZoomControl system used in Phonak Quest, are shown in Figure 9. The directionality is always ideally adapted to the relevant situation, in which the most useful directionality is used

automatically. This improved the SNR to the extent that the objective use, which previously had been theoretically proven by means of KEMAR measurements, could also be confirmed by the results of the speech test. Thus, the test subjects averaged a 10.6% improvement in their speech understanding with auto ZoomControl "NewTech" compared with an omnidirectional setting. When compared to the directional setting, which was of course counterproductive in this situation, auto ZoomControl showed an improvement in speech understanding of 37.6%. In the "speech from the side" situation, auto ZoomControl improved speech understanding by 18.5% (in comparison with the omnidirectional setting) and by 33.8% (in comparison with fixed directionality to the front). All these results are statistically significant, with an additional ANOVA producing a clear link between the CST results measured and the auto ZoomControl function.

A general comparison of the solutions with each other at statistical level was not included in the cited work, but some advantages of the auto ZoomControl function over the other two processes shown above were recorded on average.

**Therefore, Phonak offers an effective solution for the complex listening situation "Speech understanding in noise, when the listener is not facing the speaker" based on Binaural VoiceStream Technology®.**

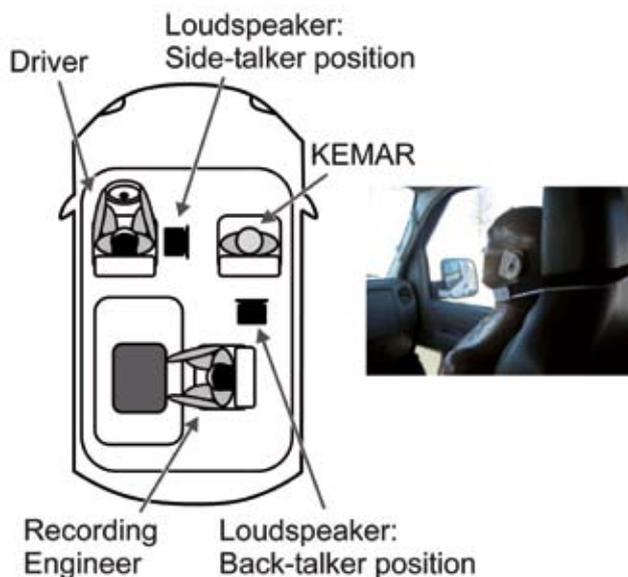


Figure 8 The test setup for recording sound samples for the CST (from Wu et al. 2012).

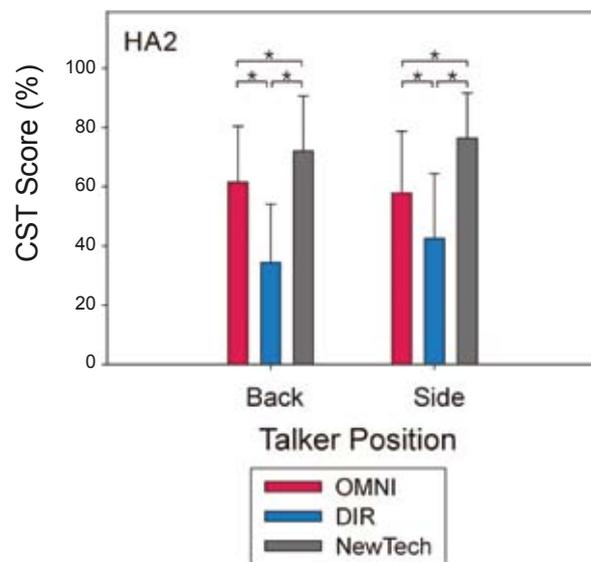


Figure 9 Average results of the CST for the three approaches tested, separated according to the test situations, i.e. speech from one side or speech from behind. The results are statistically significant (significance level: 0.001) (from Wu et al. 2012).

## DuoPhone

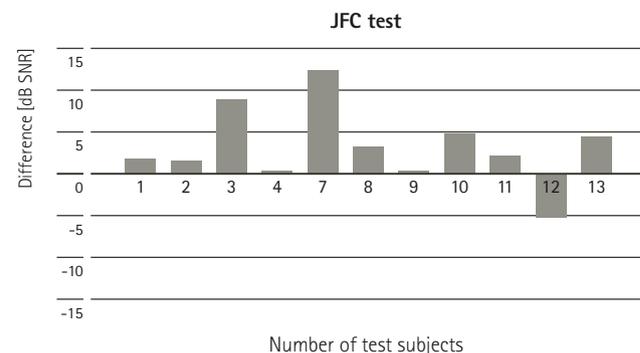
Understanding properly when speaking on the telephone poses a great challenge for many people who are hard of hearing (Latzel 2001). This is particularly difficult in an environment with loud background noise. This is why almost all hearing instruments have a telecoil which picks up the magnetic signal transmitted by a magnetic telephone receiver system. In addition, the microphone is often switched off, to ensure that the acoustic background noise is not processed. The SNR is then correspondingly high. Modern telephones very rarely have magnetic receiver systems, which means that the telecoil is of limited use. The hearing instrument signal also interferes with the contralateral ear, i.e. the ear not used for the telephone receiver, or it may be blocked out. This means that the advantages of binaural hearing cannot be enjoyed on the telephone. This is why there is interest in alternative solutions to facilitate what may often be a difficult acoustic situation for hearing instrument wearers, especially those who may be professionally dependent on using a telephone.

The DuoPhone function was developed to reflect these considerations. DuoPhone makes use of audio data transmission between the hearing instruments to transmit the telephone signal picked up via the microphone of the ipsilateral hearing instrument to the contralateral hearing instrument. The input signal on the contralateral side is simultaneously attenuated by 6 dB, to ensure that the signal received by the ipsilateral hearing instrument is dominant. Thus the more favorable SNR on the ipsilateral hearing instrument is transmitted to the contralateral ear. If the EasyPhone function is also activated, the hearing instrument is adapted automatically to the new hearing situation, so that the hearing instrument wearer only has to place the telephone receiver on to the hearing instrument. The telephone conversation can then start with a clearly improved SNR and make use of the even more effective binaural functions.

The use of DuoPhone in improving speech understanding when making a telephone call in a noisy environment was tested at the university of applied science in Lübeck, using 15 test subjects with moderate to severe hearing loss. In order to better take into

account the individual requirements of the various test subjects, a "Just Follow Conversation Test" (JFC test) (Hygge et al., 1992) was carried out. In this test, each participant decides what the level of the speech signal should be in comparison with the background noise (55 dB), so that the criterion "I can just follow the story presentation" is met. Figure 10 shows the individual results for the JFC test. Positive values result in a better SNR when DuoPhone is used, while negative values are an advantage when the telephone signal is used in only one ear. Subjective speech understanding, which is determined via the JFC test, failed to improve in just three test subjects. In all other cases, the use of DuoPhone improved the SNR by up to 12 dB. This means that, for this test subject, the telephone signal may be 12 dB softer than the background signal when using DuoPhone than when the telephone signal is only transmitted into one ear.

**Therefore, Phonak offers an effective solution for the complex listening situation "Making telephone calls in noisy situations" based on Binaural VoiceStream Technology®.**



**Figure 10** Presentation of the individual results for the JFC test. The speech signal was presented via a telephone receiver, while the background noise signal came from the free-field speakers. The test subject varied the speech signal level until he was "just able to follow what was being said" (JFC threshold). The JFC thresholds shown concern the monaural condition, in which the speech signal is only offered to the "telephone" ear.

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## Speech in Wind

The deterioration of the SNR in wind plays a major role in many everyday situations (Chung 2012a, Chung 2012b). This applies particularly when a hearing impaired person wears a hearing instrument, as they can be exposed to background noise caused by the wind around the hearing instrument microphones, which are often in an unfavorable position. Many innovations have been developed in this field to reduce the influence of wind either mechanically or by means of digital signal processing. Although purely mechanical solutions are effective, they are often not sufficiently acoustically transparent. For example, when the microphone filters get dirty during everyday use, algorithmic approaches to separate the desired signal from the wind signal, for example, by using adaptive filters, often only manage to achieve a reduction in amplification in certain frequency ranges. Although this attenuates the background noise, it also reduces the speech signal, so that these solutions are insufficient in many windy situations to ensure good speech understanding.

Apart from inadequate wind suppression, the detection mechanism may also be faulty, which can result in an undefined function of the hearing instrument. For example when wind is "detected" although there is no wind present (false positive: specificity) or when wind is not detected although it is windy, because the signal may still be too weak ( false negative: sensitivity).

The new Speech In Wind function specifically aims to solve these two problems. On the one hand, the sensitivity and specificity of detection are increased, while on the other hand the filtration of the wind noise is more intelligent, thus improving speech intelligibility too.

### Detection

Based on the existing Phonak technology, the level of the low-frequency portion of the input signal serves as an indicator for detecting wind. In addition to this Phonak Quest analyzes the microphone signals of the dual-microphone system. In the case that there is no direct correlation between the level and the phase of both microphone signals in certain frequency ranges over a defined period (the time constant of the system), it is assumed that there is wind. In addition, the ratio between certain frequency components is analyzed to find out how strong the wind is. Depending on the wind noise level, these two results together will initiate the corresponding suppression within only a few seconds, with this time constant being related to the adjusted sensitivity of the automatic SoundFlow system.

This new algorithm requires the hearing instrument to have two separate microphone signals. Where a hearing instrument has only one microphone or has been adjusted so that only one of the two analogue/digital transducers is available in the pre-

amplification system (the second analogue/digital transducer may, for example, be used for the signal of the audio shoe), detection works according to the principle already used in Spice+.

### Suppression

In addition to the wind noise suppression feature already available, the system operating on the new Quest platform calculates the suppression which is dependent on the signal-to-noise ratio, calculated from the difference between the estimated wind noise level and the level of the speech signal. When the level of the speech signal of Speech In Wind is higher than the level of the wind noise, suppression will be reduced. This ensures that the speech portions of the input signal are not adversely affected, so that the feature Speech in Wind not only increases the comfort, but also the intelligibility in windy situations.

The new wind noise suppression system also operates depending on the wind noise levels measured. This makes use of the frequency dependency of the estimation parameters, which then provides information about the strength of the wind noise. Depending on the value determined, suppression is controlled, so that attenuation is reduced when the wind noise is low or increased when the wind noise is high.

To ensure adequate suppression of the wind noise during more open fittings, the threshold frequency below which suppression takes place depends on the adjusted vent size, which means that frequencies up to 3.5 kHz can be influenced in open systems, while this value is reduced to about 1.6 kHz for completely closed systems.

### Improving speech intelligibility

Suppressing wind noise makes it more pleasant to wear the hearing instrument in a windy environment, but generally does not promote communication in such an environment. This is apparently one reason why the latest study by Kochkin (2010) still mentions the wind situation as a major problem. It is thus important not only to create a more pleasant situation, but also to increase speech understanding in a windy environment. This is why the new feature Speech in Wind makes use of wireless communication between the hearing instruments, thus facilitating binaural processing. The newly-developed binaural algorithm transmits the wind noise level independently detected in each hearing instrument to the other side via the wireless link and analyzes the parameters with each other. In the case that the parameters exchanged have approximately the same values, the system remains inactive. The system assumes that the wind situation is symmetrical and only the monaural WindBlock function operates separately in each hearing instrument. If, on the other hand, different values are detected, the system assumes an

asymmetrical wind situation. In this situation, the signal is transferred from the side with the least amount of wind noise (ipsilateral) to the other side (contralateral). On the contralateral side, both signals – i.e. the ipsilateral signal with less wind noise and the contralateral microphone signal – are filtered and subsequently overlaid. Signal components contributing to binaural hearing and localization are thus retained in the best possible way. On the contralateral side, only these frequency ranges of the microphone signal are suppressed, while the remaining signal is overlaid with the signal transmitted from the ipsilateral hearing instrument. This combined signal is then processed by the contralateral hearing instrument, using its signal processing technology and setting. Figure 11 shows an example of an asymmetrical wind situation and the behavior of the different systems. The Diagram 11a shows the input signal for the right and

left hearing instrument. The input signal at the right ear is distorted by wind noise. The wind noise signal is not suppressed as Speech in Wind is not activated. (Figure 11a). In Figure 11b Speech in Wind is activated. The system detects wind noise in the right hearing instrument so that the level of the input signal is **reduced** and the listening comfort is improved. In Figure 11c Speech in Wind is activated as well. As less wind noise is detected in the left hearing instrument and the audio signal is transmitted from the left to the right hearing instrument. This means that the undisturbed signal is processed in both hearing instruments and consequently **speech intelligibility** is significantly increased.

**Therefore, Phonak offers an effective solution for the complex listening situation "Speech understanding in windy situations" based on Binaural VoiceStream Technology®.**



**Figure 11**

Demonstration of the effect of the Speech in Wind. The time signals of the right and left microphone signals are shown.

In **a** Speech in Wind is deactivated. The right microphone signal is disturbed by wind noise, while the left microphone signal is an undisturbed speech signal.

In **b** Speech in Wind is activated. The left microphone signal remains unprocessed, as no wind noise interference is detected. Speech in Wind **suppresses** the right microphone signal significantly.

In **c** Speech in Wind is activated. The systems recognizes that the wind noise on the left side is weaker (or non-existent) than on the right side, so that the speech signal is transmitted from the left to the right side via the wireless link.

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## Conclusions

All the algorithms presented in this article are available on the new Quest platform. To ensure that the algorithms can be as effective as possible, the hearing instruments must be optimally adjusted to the individual needs of the end user. This means, for example, that the individual acoustic characteristics of each hearing instrument user needs to be taken into account and the fittings made that they are "as closed as possible and as open as necessary". In addition, it is necessary to check the adjustments and technology at regular intervals to see whether, for example, the user's hearing loss has changed or the microphone system has been affected by environmental influences.

Where these conditions have been met, this article shows how the binaural algorithms used in the new Phonak Quest platform provide effective solutions for the major problems experienced by hearing instrument users in everyday life situations, thus significantly improving their satisfaction with their hearing instruments.

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