FOCUS 26

Advantages of an Adaptive Multi-Microphone System

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It is widely accepted that hearing in noise is one of the main challenges facing those with sensorineural hearing impairment [1–4], and it has been shown that difficulties in noise are aggravated with increasing hearing loss.[5] It is also known that, currently, the most effective way to improve speech recognition in noise is to enhance the Signal-to-Noise Ratio (SNR).[6–9] A central challenge, therefore, for hearing instrument manufacturers has been to apply technologies that can increase the level of the signal of interest, while attenuating the background noise. In order to achieve this, many approaches have been used in the past (e.g. low-frequency suppression and automatic changes to the frequency response).

Currently, the most successful method for improving the SNR is the use of directional microphones and multi-microphone technology.[10–12] The success of these devices is due to the fact that noise from behind the user is prevented from entering the system resulting in a much "cleaner" signal for further processing. Other noise reduction methods use

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signal processing to emphasize speech or reduce the level of noise after it has entered the processing path.

Use of Directionality in Hearing Instruments

Directional microphones

Hearing instruments incorporating directional microphones have been available since the early 1970s. While the first products did not gain widespread popularity[13], many studies in the early-80's showed these systems to be quite effective.[14–16] The directivity pattern from these conventional directional microphones was a cardioid configuration (maximal suppression to the rear at 180°) that offered up to a 3–4 dB enhancement in SNR in a non-reverberant test environment.[16, 17] Despite the SNR advantages associated with these early directional microphones, some practical disadvantages concerned with their use in everyday listening environments were apparent: most importantly, they did not offer the option to choose an omnidirectional mode, when necessary, in appropriate situations (e.g., in the street or listening to music).

Multi-microphone technology

An important advance in directional technology took place with the introduction of multi-microphone arrays. This technology employs two separate, well-matched omnidirectional microphones designed to allow users to electronically switch between programs with either omni or directional modes. The effectiveness of multi-microphone technology has been documented by a large number of studies which have shown significant improvements in SNR and high levels of user satisfaction.[6,18–26] While multi-microphone technology has helped overcome many of the disadvantages of conventional directional microphones, these microphone arrays still have a fixed direction of maximal noise suppression (e.g. ±90° bidirectional or 180° cardioid).

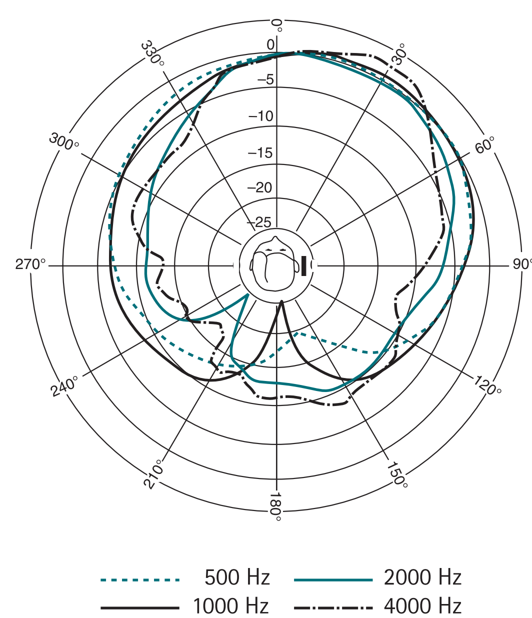
The diagram in Fig. 1 displays the directivity pattern of a multi-microphone instrument at 500, 1000, 2000 and 4000 Hz. The area of maximal suppression in this example is to the rear at around 180° and is fixed in this direction. In "real-life" use, however, noise interference can originate from any direction and, thus, the null in the directivity pattern may not always match the angle of incidence of the noise.

Digital instruments

In recognition of the limitations of a fixed directivity pattern, some digital multi-microphone systems offer the ability to switch between various fixed polar patterns. This requires the user to determine the direction of the dominant noise source in a given environment and manually select the appropriate pattern for maximal noise suppression which may be impractical or difficult for some clients and/or situations.

Figure 1

Directivity pattern of a multi-microphone instrument at differing frequencies. There is a fixed area of maximal noise suppression which, in this case, is mostly to the rear of the user.



Studies indicate that digital signal processing alone is not sufficient to significantly improve speech understanding in noise.[27–29] In addition, other studies suggest that digital hearing instruments with directional or multi-microphone technology have, to date, shown no significant advantage in noise over multi-microphones within analog instruments.[11, 30] The challenge, therefore, has been to apply digital technology to multi-microphone technology in a way that will offer increased benefits in noisy situations to the hearing instrument user.

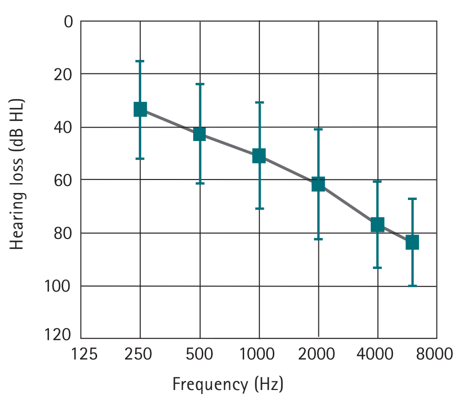


Figure 2
Average audiogram across subjects for left and right ears.

An Adaptive DSP Microphone System

A unique application of digital technology called Adaptive digital AudioZoom (dAZ) has been incorporated into a recently introduced hearing system (Phonak Claro) to improve the effectiveness of noise suppression regardless of the angle of incidence. This is achieved by automatically adapting the directivity pattern in response to the environment so that the area of minimal sensitivity (i.e., a null in the directivity pattern) is always directed towards the most dominant noise source in the rear plane (±90°).

Essentially, this digital multi-microphone system combines a real-time output from one microphone with a delayed signal from a second microphone to form a characteristic directional response. The output signals from the two spatially separated omnidirectional microphones are converted from analog signals to digital signals using separate analog-to-digital converters. The digital stream then enters the spatial processor, where the signals are combined to produce the appropriate directivity pattern for a given environment.

This pattern is continually varied to ensure that the output level from the system is kept at a minimum, and this continual process is what specifies the adaptive nature of the array and ensures that noise levels are reduced. This minimization of the output is performed under the assumption that useful sounds come from the front and noise in an arc from +90° to –90° behind the user. In effect, dAZ continually searches for the point at which the noise must be attenuated to the greatest extent and places the area of minimal sensitivity in that direction so that optimal noise reduction is achieved and maintained. In the case of a diffuse field, where noise originates from multiple directions, a hypercardioid directivity pattern is adopted, as this pattern is most effective under these conditions.

Clinical Study

Subjects and method
To test the effectiveness of dAZ in the laboratory, the SNRs required for participants to score 50% correct on a sentence test were measured with noise presented from directly behind (180°) and to the sides (±90°). A comparison was made between the omnidirectional, fixed (cardioid) and adaptive multi-microphone configurations possible in Claro 211 dAZ instruments. In addition to the hearing instrument users, unaided normal-hearing subjects were tested under the same conditions to provide a reference group.

The study took place at the University Hospital, Zurich, Switzerland, with 22 hearing-impaired and six normal-hearing subjects. The hearing-impaired group all presented with mild-to- moderate sensorineural hearing loss and had previous experience of hearing instrument use. The average Hearing Threshold Level (HTL) in the hearing-impaired group using the three frequency average (0.5 kHz, 1 kHz and 2 kHz) was 48 dB HL ±16 dB (1 SD, min: 13 dB HL, max. 93 dB HL). Fig. 2 shows the average audiogram for left and right ears for all 22 hearing-impaired subjects (44 ears).

The instruments were fitted binaurally and earmolds were fully occluding to prevent venting or feedback effects on the directional response and to ensure a homogeneous group. Testing took place in a sound-treated room with a noise floor of <30 dBA. The test stimulus was speech with an adaptive presentation level, and the speech material chosen for the experiment was the Göttinger Sentence Test.[31] This German sentence test contains highly homogeneous test material and 20 highly equivalent sentence lists with 10 short sentences each. Speech was presented from a speaker placed at 0° azimuth to the subject at a distance of 1.1 meters. Noise, fixed at 70 dB SPL for the two test conditions, was presented from speakers at 180° and ±90° again at a distance of 1.1 meters. The noise signal was a speech simulating noise developed by digitally superimposing the words of a monosyllable rhyme test, produced by the same speaker.[31] The long-term spectrum of the noise and the sentences were therefore similar. The level of speech was adapted to a point where subjects were able to score 50% correct.

Instruments were fitted to individual hearing loss according to the manufacturer's fitting procedure, and initial user acceptance was also verified according to manufacturer's recommendation before the commencement of testing. The order of test presentation was counter-balanced for the microphone setting,

Figure 3
Average SNR for 50% correct with noise source at 180°. A more negative response indicates better performance.

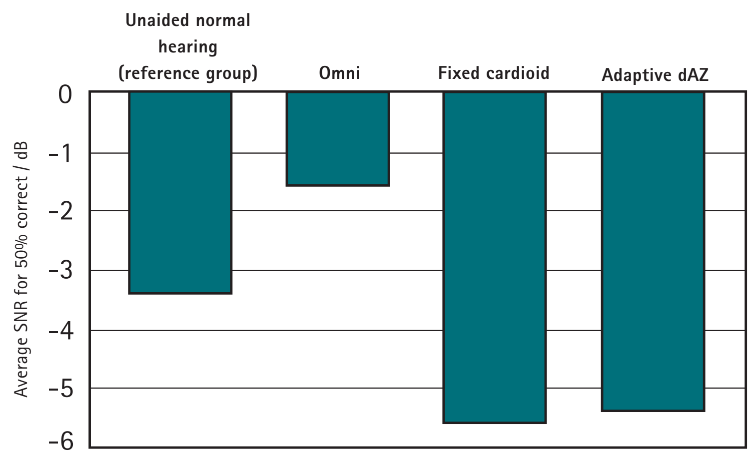
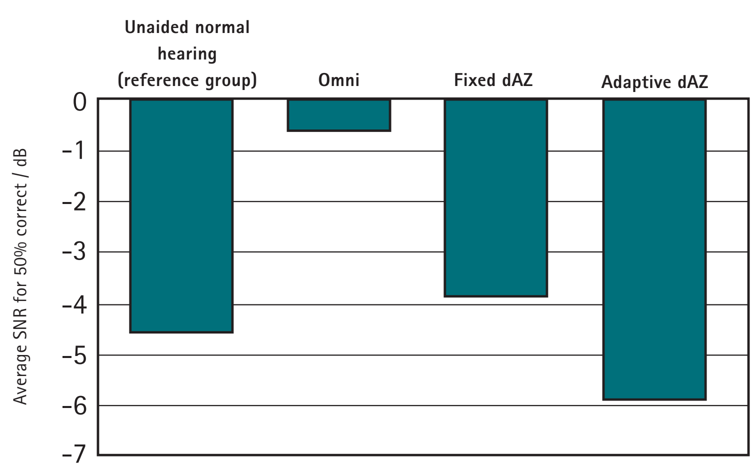


Figure 4
Average SNR for 50% correct with noise source at ±90°. A more negative response indicates better performance.



and the following microphone settings were compared for the two noise conditions (±90° and 180°):

- Program 1 + Omni;
- Program 1 + digital AudioZoom (fixed directional pattern - cardioid);
- Program 1 + Adaptive digital AudioZoom.

All instruments were programmed according to the manufacturer's algorithm which is optimized for hearing in quiet environments (QuietAdapt Program 1). This algorithm is characterized by fast time constants (attack time 5–10 ms, release time 90–100 ms, depending on signal strength and frequency) and a low compression threshold which is variable in each of the 20 critical bands as a function of frequency.

Results

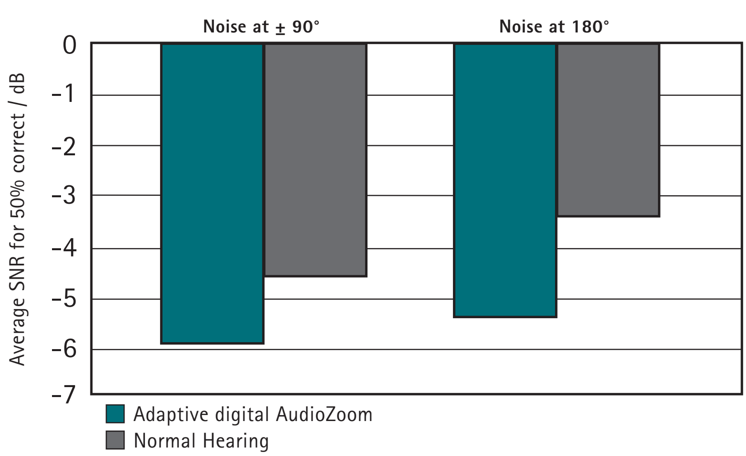
For both noise conditions, the results for all three microphone modes were compared using a paired t-test of significant differences. Fig. 3 shows the SNR necessary for 50% intelligibility with the noise source at 180°. In the omnidirectional mode, the average SNR for 50% correct was −1.6 dB.

As expected, with noise at 180°, there is no significant difference between fixed (cardioid) and adaptive because the fixed cardioid setting is optimized when noise is from directly behind the listener. Both microphone conditions (fixed and adaptive) showed significant improvement over the omnidirectional setting.

The graph in Fig. 4 displays the results when the noise source was at ±90°. Although the fixed setting is optimized for maximum suppression to the rear (180°), there is still a highly significant improvement in this situation over the omnidirectional setting. However, the adaptive array really demonstrates advantages in this situation where the difference between the cardioid setting and the adaptive setting is also highly significant. The average SNR for 50% correct in this situation was 2 dB above the fixed (cardioid) value.

The average SNRs for 50% correct for the normal-hearing subjects under the same test conditions were −4.6 dB in the ±90° condition and −3.4 dB with noise at 180° (Fig. 5). When the results from this group are compared to the hearing-impaired group with dAZ, the performance of the hearing-impaired listeners with dAZ was better in both situations. The graph in Fig. 5 compares the average SNRs for the unaided normal-hearing group and the hearing-impaired group using dAZ.

Figure 5
Average SNR for unaided normal-hearing listeners (grey bars) and hearing-impaired listeners with Adaptive digital AudioZoom. A more negative response indicates better performance.



Discussion

The adaptive microphone system discussed above is the first application of digital technology in a hearing instrument that makes automatic, adaptive directional performance available for the suppression of background noise, regardless of the angle of incidence. This overcomes the limitations associated with conventional directional hearing instruments or fixed multi-microphones, where maximal suppression is always in a fixed direction (e.g., cardioid 180°, bidirectional ±90°). The results here show that adaptive multi-microphone technology offers significant benefits in noise and is able to adaptively maintain a favorable SNR. Additionally, adaptive processing removes the need for manual switching between different polar patterns.

In this experiment, only two extreme noise conditions (±90° and 180°) were tested. It is important to remember, however, that the system has the capacity to set the maximum suppression regardless of the angle of incidence. The diagram in Fig. 6 illustrates four of the unlimited possible polar patterns in the dAZ system. Due to the adaptive nature of the array, any configuration between bidirectional or cardioid could be applied in order to attenuate noise. This means that the noise source can be pinpointed automatically, regardless of it's direction and without the restrictions imposed by one, or a limited selection of, fixed polar patterns.

If the results from the unaided reference group with normal-hearing are compared to the hearing instrument users with their microphones in the omnidirectional setting, the normal-hearing participants perform significantly better. This emphasizes the necessity to improve the SNR for those with hearing loss [6–9] and not just amplify signals above the hearing threshold using instruments with omnidirectional microphones.

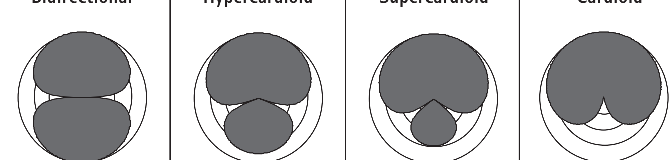
However, if the results from the hearing-impaired group with their microphones in the dAZ setting are compared to the normal-hearing group, there is a difference in performance in favor of the hearing-impaired group. Eighty-two percent of the hearing-impaired subjects with dAZ had equal or better SNR values for 50% sentences correct when individually compared to the average values for the normal-hearing group across noise conditions (–4 dB SNR for 50% correct).

Lurquin and Rafay[19] found no significant difference between a normal-hearing group and a group of hearing-impaired listeners using instruments with multi-microphones; however, they did find a significant difference between the two groups when multi-microphones were not employed. Pumford et al.[26] compared multi-microphone results to those derived from 10 normal-hearing subjects. With BTE instruments (dual-microphone + noise algorithm), the hearing-impaired group all achieved improvements in performance. Sixty-three percent of those achieved scores that fell within a 95% confidence interval of normal performance. The results of these two studies[19,26] show that, with fixed directional technology, it is possible to achieve speech intelligibility results that are within the range of normal hearing.

The laboratory results with dAZ show how it may now be possible for hearing instrument users to be given a much-needed advantage over those with normal hearing in an environment with competing background noise where communication has historically been difficult, or impossible, in some situations.

Figure 6

Four of the unlimited directivity patterns possible with dAZ. The directivity index is defined as the ratio of the output power due to the target signal from the front and the average power of the noise originating from all directions.



	Bidirectional	**Hypercardioid**	**Supercardioid**	**Cardioid**
Front to Back Ratio	0 dB	6 dB	11.4 dB	infinite
Directivity Index	4.8 dB	6 dB	5.7 dB	4.8 dB
0_{max} the angle of max. suppression	90°	110°	125°	180°

Summary
Directional technology has undergone many changes over the years but has maintained a high level of research interest due to the measurable advantages in noise. Adaptive digital AudioZoom is designed to provide maximum suppression of noise regardless of the angle of incidence, and overcome the limitations of conventional or fixed multi-microphone directional technology. The data from this study shows that dAZ is effective in various dynamic noise situations and is capable of providing significant benefits for the hearing instrument user.

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