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Audio Zoom-Signal Processing *for Improved Communication in Noise*

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Contents

1. Introduction

Marketing strategies often make innovations in hearing instrument technology difficult to understand. Such strategies may lead to unrealistic expectations which cannot be confirmed by either clinical studies or user experience. Aggressive marketing strategies sometimes even succeed in changing the focus of hearing instrument fitting goals. For example, in the past hearing healthcare professionals generally agreed that one of the highest priorities in hearing instrument fitting is the provision of improved speech intelligibility in noise. Today, in certain markets, providing the ultimate hearing comfort in quiet is the new trend.

The necessity to communicate in noise is still a high priority, however, amongst hearing instrument users. It is also one of the major causes for users' dissatisfaction with their current fitting. In a survey held in 1991, based on a total number of 2,323 hearing instrument users, Kochkin (1993) found that while 3 out of 4 consumers report satisfaction with the hearing instruments' ability to improve their hearing, 47% express dissatisfaction with their ability to use the hearing instruments in noisy situations. In our opinion it is essential therefore, that hearing instrument systems applying new technologies, provide solutions to improve speech intelligibility under adverse noise conditions.

In view of the above target, it is important that the hearing healthcare professional be able to select hearing systems which allow, not only for maximal comfort in quiet, but also for optimal speech intelligibility in noise. In order to make a choice of the systems which may fulfill these criteria, it is necessary to have an understanding of the physical and psychoacoustical background of the respective noise suppression strategies.

In the following sections we will review and analyze the results of clinical studies which have evaluated the efficacy of currently available noise suppression systems. A description of the Phonak Audio Zoom concept will follow, along with a discussion of the estimated benefits of Audio Zoom for communication in noisy environments.

2. Background

Currently, there are two fundamentally different approaches in noise suppression techniques for hearing instruments: adaptive electronic filtering and directional microphones or beamforming. The most important difference between these two approaches is their position within the signal processing chain. The directional microphone acts on the signal before it enters the hearing instrument's signal processing path, while the adaptive filter is placed within the signal processing path. Since the signal entering the hearing instrument contains both the signal of interest and the background noise, even the most sophisticated filter can no longer separate these two signals (Dillon & Lovegrove 1993, Verschuure & Dreschler 1993). Identifying and suppressing particular frequency regions of a noise will suppress the same frequency region of the signal of interest and to the same degree. As a result, the Signal to Noise Ratio (SNR) can't be improved. Directional microphones, on the other hand, improve the SNR before the combined signal and noise enter the rest of the signal processing pathway.

3. Adaptive electronic filtering

While the term Automatic Signal Processing (ASP) has become synonymous with more recent hearing instruments with level dependent automatic filtering, hearing instruments with Automatic Gain Control (AGC) circuits, which were first introduced about 50 years ago, must also be categorized as automatic signal processing devices (Kretsinger & Young 1960). ASP is thus too general a term which does not describe sufficiently, the actual properties of the applied signal processing scheme. Killion et al. (1990) have categorized current systems according to their mode of operation. Two initial categories and terms are important to note in this context:

FFR – Fixed Frequency Response; referring to devices with linear filtering.

LDFR – Level Dependent Frequency Response; referring to devices with non-linear filtering.

Within the LDFR category they further defined three subcategories. The most important differentiating criteria being the filtering process which takes place at low input levels.

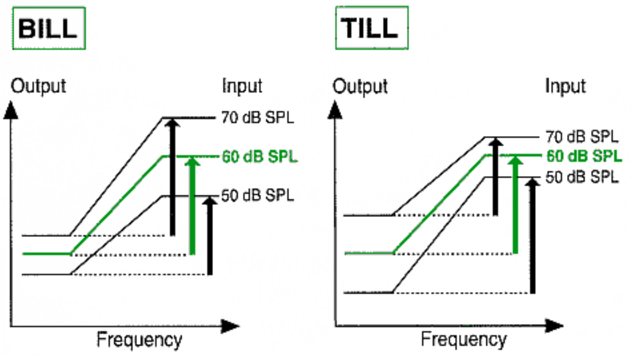
BILL – Bass Increase at Low Level:
A relatively flat frequency response occurs at low input levels, with increasing high frequency emphasis as the input level increases (Fig. 1).

TILL – Treble Increase at Low Level:
High frequency emphasis occurs at low input levels, with the response becoming flatter at higher input levels (Fig. 1).

PILL - Programmable Increase at Low Level:
The behavior of the frequency response at low and high input levels can be programmed. Thus a PILL can be programmed to act as a BILL, TILL or even a FFR.

According to the above nomenclature, hearing instruments incorporating user selectable programs could be categorized as U/PILL, User-Adjustable Programmable Increase at Low Level (Fabry, 1991).

Fig. 1: *Level Dependent Frequency Response - BILL vs. TILL: the idealized input / output responses are plotted against frequency for different input levels. As a reference, the 60 dB SPL response is identical for both processing schemes. Notice that BILL is flatter at low levels and TILL is flatter at high levels.*



3.1 Reference to psychoacoustics

Even though the BILL and TILL strategies are conceptually very different from each other, both are plausible approaches considering fundamental psychoacoustics. BILL has been developed to counter the upward spread of masking effects (Klein et al., 1990) associated with a sensorineural hearing loss. There is little evidence, however, to support a solution that offers a constant increase in filter steepness or increase in high frequency amplification at high input levels (Horwitz et al., 1991; Kates, 1993; Moore, 1991; Murnane & Turner, 1991).

The goal behind the development of the TILL strategy was to restore the loudness growth function. In the presence of normal hearing, the equal loudness contours become flatter with increasing input level. TILL instruments attempt to reproduce this phenomenon with the hearing instrument in order to provide pleasant, natural sounding amplification (Horwitz et al. 1991; Leijon 1990). On the surface, these two concepts seem to be pulling in entirely different directions. In reality however, a BILL and a TILL hearing instrument fitted with a vented mold/shell, and exposed to moderate input levels will provide similar insertion gain values. Significant differences exist only for low level input signals. The input/output characteristics of a BILL and TILL systems, measured with a composite noise instead of pure tones, show practically the same non-linear compression characteristics (Fabry 1991).

3.2 Review of the literature

The BILL system has been studied thoroughly by a large number of authors (Tyler, 1988; Van Tassell et al., 1988; Bentler, 1990; Fabry & Walden, 1990; Tyler & Kuk, 1989). Fabry and Van Tassel (1990) calculated the theoretical effectiveness of a BILL instrument based on the Articulation Index, and verified their results in an evaluative study. They found that an improvement of speech discrimination in speech-like noise could neither be theoretically predicted nor measured in practice. Horwitz et al. (1991) compared the BILL and TILL strategies with a Fixed Frequency Response (FFR) system. While their results were unable to show an average advantage for either of the level dependent filtering strategies, they did show an advantage for the conventional FFR approach. Speech discrimination results were equally distributed, with no advantage shown for either system. The FFR, however, achieved the highest ratings for sound quality.

Moore et al. (1992) evaluated a dual channel system which could be programmed either for linear or compression operation; in this case, whole dynamic range compression. They evaluated the performance of 20 experienced hearing instrument users in quiet and in noise. The results obtained for speech intelligibility in noise are of particular interest. While the average results favored the compression condition, out of the 20 subjects only 7 showed scores that could be translated into a significant improvement in SNR. An examination of the audiological characteristics these particular subjects shared, showed that all 7 had an unusually narrow high frequency dynamic range of less than 27 dB. The majority of scientific evaluations of adaptive filtering systems (BILL and TILL) have not found an improvement for speech intelligibility in noise for the most of the subjects. At the same time, individual improvements have been measured under specially defined conditions (Fabry 1991; Moore et al. 1992)

3.3 Individual improvement

Despite the fact that, on the average, no improvement was measured with adaptive filter systems, we must not conclude that these methods are useless for all individuals. First of all, it is important to remember that hearing loss cannot be quantified and classified based on pure-tone threshold data alone. Secondly, the listening requirements of different users vary significantly. It is also important to analyze several issues that can strongly influence the average results of such studies.

Frequency response, sound quality
Before it is possible to evaluate speech discrimination in noise, the hearing instrument fitting must take place in quiet. Findings indicate that when the selected frequency response of the hearing instruments makes speech signals audible, further frequency response variations within the audible range do not contribute additionally to improved speech intelligibility. Listening comfort is, however, highly dependent on the selected frequency response (Horwitz et al. 1991, Leijon 1990).

Overall loudness, volume control manipulation
The spectral changes in the signal resulting from BILL-type adaptive filtering are often associated with reduced loudness since the intense low frequencies are being suppressed. Several authors have pointed out that, in many cases, by optimally adjusting overall loudness speech discrimination in noise can be further improved (Bentler 1991, Fabry & Van Tassell 1990, Fabry 1991, Moore et al. 1992). A calculation of the Articulation Index, under the circumstances of increased gain, confirms this hypothesis. To obtain this benefit, however, the user must constantly adapt the volume of the hearing instrument or an Automatic Volume Control (AVC) must be incorporated to assume this function. At present, few of the available BILL-type products have incorporated an AVC.

Low frequency background noise
Even though, in most circumstances, no improvement in speech intelligibility can be achieved, some users can profit from BILL type filtering in the presence of narrow band, low frequency background noise. It has been clearly demonstrated that a BILL system can suppress low frequency narrow band noise more successfully than a linear scheme. This, however, does not necessarily lead to a general improvement of speech intelligibility in real life situations (Van Tassell et al. 1988, Fabry 1991). In real life, the most common background noise is more like a broadband, multi-talker babble. It has been shown that adaptive filter schemes are unable to dependably improve speech discrimination in typical environments where part of the interfering noise is also speech.

User selected programs
There have been a number of studies demonstrating that users are capable of interacting appropriately with different signal processing schemes designed for specific listening situations (Bentler, 1991, Kuk, 1992, Moore et al., 1992).

While a principle goal of amplification is clearly to make speech audible to the hearing impaired, not all of the desired signals entering a hearing instrument are speech. A prime example of a situation where an adaptive filtering scheme, especially an automatic level dependent filtering scheme, is undesirable, is in listening to music. Few of us would appreciate the resulting elimination of low frequencies and arbitrary changes in sound quality. Furthermore, in difficult listening situations, studies have shown that individual users have improved their speech discrimination scores by switching off the particular adaptive filters and returning to the only alternative within the confines of the study, a linear response. It is reasonable to conclude that systems using adaptive filter schemes should, as a minimum, include a user operated switch which allows the user to interrupt any automatic adaption of the hearing instrument when it is proving to be a disadvantage.

Most, but not all, users can profit from multi-program capabilities. They need some time however, to learn how to use the different programs designed for particular situations and how to optimize speech understanding as well as listening comfort (Kuk 1992).

4. Directional microphones

Microphones with directional characteristics have been known and used widely for many years. This includes their use in hearing instruments. They have the advantage of being able to act on the incoming signal to improve the SNR before it enters the rest of the signal processing path. The physical separation of the microphones or microphone ports, allows for selective damping of signals originating from the arc to the rear of the instrument user. Since in conversation it is usual to face the talker, a directional microphone can give priority to the signal of interest while providing relative damping to the background noise.

Digital signal processing for noise suppression is currently a popular subject for study and product development in many areas. University studies, professional publications, and patent registrations reflect the high level of activity in this field. While practical digital signal processing systems are yet to be implemented, it is interesting to note that the directional microphone is now an important part of most digital signal processing schemes providing noise suppression (Schwander & Levitt 1987, Kates

1993, Levitt 1993, Kompis & Dillier 1994). The advantages of the directional microphone in hearing instruments are too often underestimated. A review of the literature on this subject points to a clear advantage for the use of directional microphones especially for improving the SNR (Hawkins & Yacullo 1984; Soede 1990; Kates 1993).

4.1 Published results

Conventional Directional Microphones (single microphone).

The efficiency of subminiature directional microphones, which are typically applied in Behind The Ear (BTE) instruments, has been explored by Soede, 1990. Directivity measurements were made in a diffuse field with hearing instruments fitted to KEMAR. The attenuation of the diffuse noise field relative to noise presented from the front, at 0° azimuth, was measured and compared for an omnidirectional and a directional hearing instrument in a simulated cocktail-party environment. Results showed an average improvement of 2.5 dB for the directional microphone when compared to the omnidirectional microphone. Hawkins and Yacullo, (1984) measured the directional microphone effect under more simplified conditions, with signal from 0° azimuth and noise from 180° azimuth. Normal hearing and hearing impaired subjects achieved an improvement of 3-4 dB in their speech reception thresholds, i.e. an improvement of 3-4 dB in the SNR needed to achieve 50% correct on the NU-6 Monosyllable test.

An examination of the functional relationship between SNR and intelligibility shows that seemingly small improvements in SNR, can result in highly useful improvements in intelligibility.

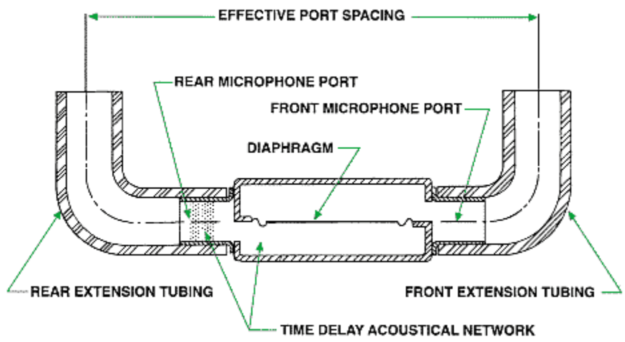
Microphone arrays – multi-microphone solutions.

The use of multiple microphones and special time delay filtering allows the creation of a system with highly directional characteristics. Soede (1990) realized and measured two types of microphone arrays. One having 5 equally spaced microphones placed across the front of the head (broadside array) and the other having 5 equally spaced microphones placed along the side of the head (endfire array). Both arrays have been measured on KEMAR in the diffuse field using a simulated cocktail-party noise, and compared to the BTE hearing instrument omnidirectional microphone and the BTE directional microphone. An overall improvement of up to 7 dB was seen when compared to the omnidirectional BTE hearing instrument and an improvement of 4-5 dB when compared to the BTE with conventional directional microphones. Schwander & Levitt, (1987) showed that similar results can be achieved by using two microphones coupled to an adaptive digital filter.

4.2 Technical background

A conventional directional microphone has two sound ports, a front and a rear port, as shown in the cross-section of Fig. 2. Each port leads to a cavity, with the two cavities separated by the diaphragm. The diaphragm measures the difference in air pressure impinging on the two surfaces. The difference in air pressure is transduced into an electrical signal.

Fig. 2: *Cross section of a conventional directional microphone used in BTE instruments, from Knowles TB21.*



A simplified explanation of how a directional microphone damps signals from the rear can be given by considering the action on a single sound wave. If we place an acoustic filter inside the rear port of the microphone, the traveling sound wave coming from the rear, enters the rear port first and is time delayed by the filter before it reaches the lower cavity. In the meantime the wave reaches the front port, entering the cavity above the diaphragm, where it arrives with a certain delay defined by the distance from the front to the rear port. If the sound wave arrives in both cavities at the same moment, because the external delay (distance) and internal delay (filter) are identical, the waves cancel each other and there is no electrical output. As a consequence, signals from the back are suppressed.

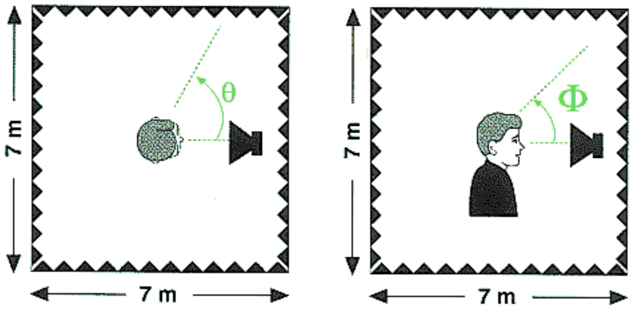
The characteristics of the directional microphone are mainly influenced by the properties of the acoustic delay filter and the distance from the front to the rear port. The spatial directivity pattern for a microphone, located in the origin (center) of a polar coordinate system with the coordinates θ,Φ, equals:

	$\mathbf{D(\theta,\Phi,\omega) = (\beta + \cos\theta)\, k\, \Delta d}$	Eq. (1)
where	$\mathbf{\beta = c\, \tau / \Delta d}$	Eq. (2)

– β is the ratio of the internal to the external delay
– c is the speed of sound in air, 344m/s
– τ is the acoustic time delay of the filter
– Δd is the distance from the front to the rear port
– ω is the frequency of the signal
– k is a constant

The measurement of the spatial equation Eq.(1) is very complex and interpretation is difficult. For simplicity's sake the directivity patterns are often measured at 0-degrees azimuth (Φ = 0), with the loudspeaker positioned exactly in front of the test-microphone or KEMAR's head, as shown by the experimental setup in Fig. 3. The two dimensional representation of Eq.(1), is called the polar response pattern and will be discussed further.

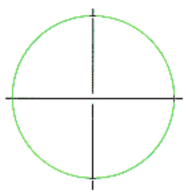
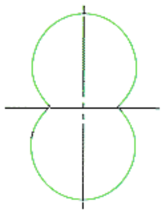
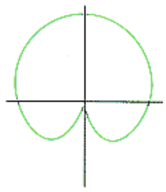
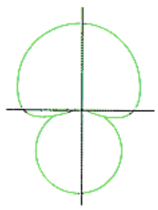
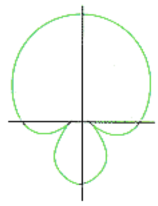
Fig. 3: *The Phonak anechoic chamber: KEMAR is mounted on a turntable, with a loudspeaker fixed at the level of the head. For the measurements in this article the following equipment was used: KEMAR (large ear -DB-065, and a single neck ring) with B&K 4157 Ear Simulators, B&K 3922 Turntable, B&K 2307 Level Recorder, Audio Precision System One.*



4.3 Directional characteristics of BTE hearing instruments

The polar sensitivity pattern of a microphone is mainly characterized by the ratio of the internal to external time delay β, shown by an overview of the idealized free field characteristics in Table 1 (from Knowles Technical Bulletin, TB21). Specific directivity properties can therefore be achieved selecting the two design variables τ and Δd.

Table 1: *Free field charateristics of different types of microphones (Knowles TB 21)*

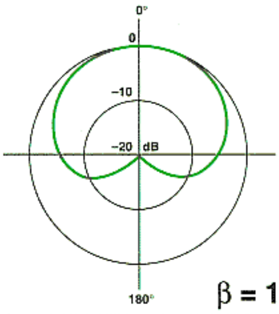
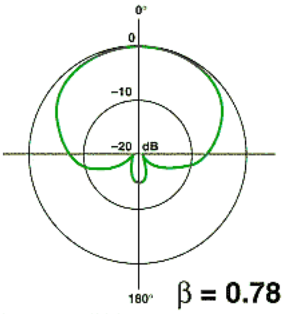
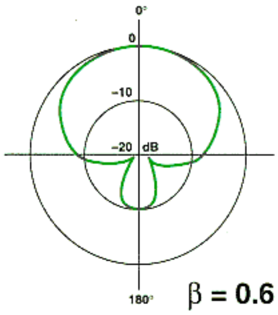
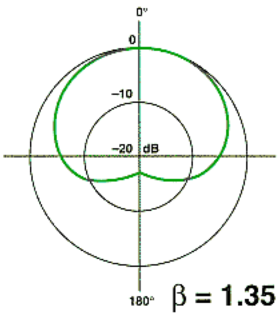
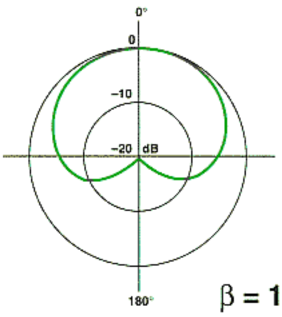
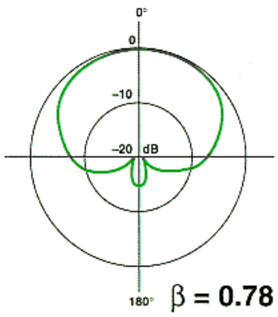
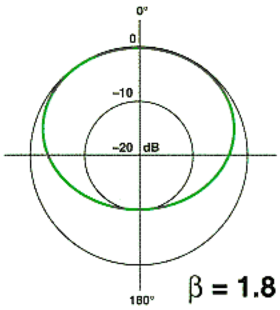
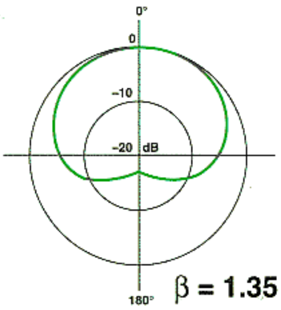
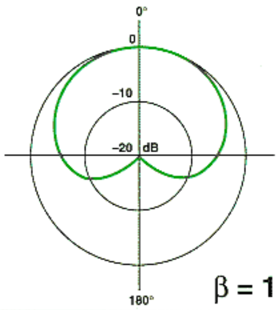
Characteristic	Omnidirectional	Bidirectional	Cardioid	Hypercardioid	Super Cardioid
Polar Response Pattern					
D (θ)	1	COS θ	½ (1 + COS θ)	¼ (1 + 3 COS θ)	$\frac{1}{(1+\sqrt{3})}(1+\sqrt{3}\ \mathrm{COS}\ \theta)$
ß	∞	0	1	0.333	0.577
Front to Back Response Ratio	0 dB	0 dB	∞	6.0 dB	11.4 dB
Directivity Index	0 dB	4.8 dB	4.8 dB	6.0 dB	5.7 dB

A matrix of typical directional hearing instrument responses, measured in the free field, is illustrated in Table 2 (adapted from Madaffari, 1983). The polar sensitivity pattern at 1.6 kHz has been plotted for 9 different configurations of the parameters τ and Δd, realized by different types of acoustical filters and extension tubing.

The diagonal of the matrix (top left to bottom right) refers to cardioids, where the internal to external delay ratio equals one ($\beta = 1$). Super Cardioids can be realized by long port spacing and short delay times in the acoustical filter path, shown in the top right corner of the matrix. The configurations below the diagonal show degrading directionality indicating that the parameters have not been adequately matched.

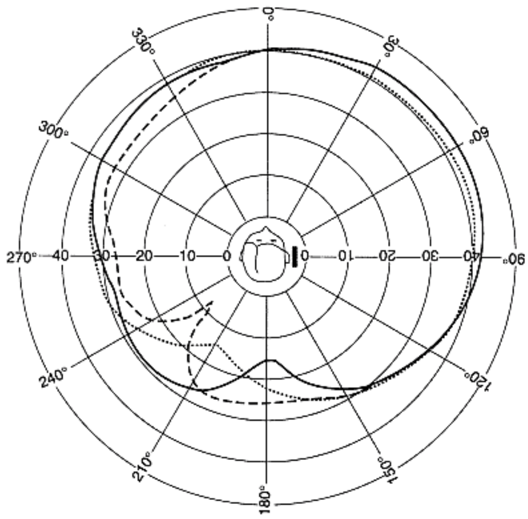
While it is not shown in Table 2, we should also note that long delay times, by means of acoustic filtering, reduce the high frequency sensitivity of the microphone. In hearing instruments, the effective directional characteristics are not specified by the microphone characteristics as such, but by the complete acoustical scheme provided by the contour of the hearing instrument body, the respective arrangement of the tubing around the microphone and the placement of the sound inlets relative to the sound source.

Table 2: *Polar sensitivity patterns measured at 1.6 kHz in free field with different internal and external time delays. (Madaffari, 1983)*

	Time Delay 43.2 µS	Time Delay 32.8 µS	Time Delay 24.9 µS
14.4 mm effective port spacing	 $\beta = 1$	 $\beta = 0.78$	 $\beta = 0.6$
11.0 mm effective port spacing	 $\beta = 1.35$	 $\beta = 1$	 $\beta = 0.78$
8.3 mm effective port spacing	 $\beta = 1.8$	 $\beta = 1.35$	 $\beta = 1$

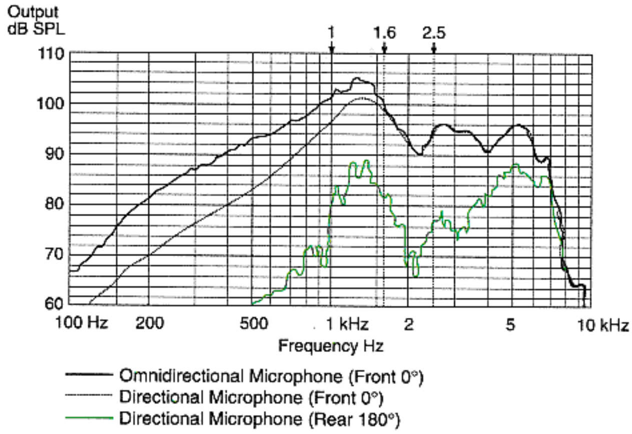
The polar sensitivity pattern of a digitally programmable BTE hearing instrument with electronically switchable microphone modes (the PiCS Piconet 232X AZ with Audio Zoom), fitted to KEMAR is plotted for three different frequencies (Fig. 4). The polar patterns are normalized to 0-dB in the front position, so that different characteristics at various frequencies can easily be compared. The ideal free field characteristics, as shown by Tables 1 & 2, can not be reproduced, in practice, by directional hearing instruments positioned over the ear since the diffraction effects caused by the human body and head disturb the sound field and the symmetry of the polar sensitivity pattern disappears.

Fig. 4: *Polar sensitivity plot of an Audio Zoom hearing instrument (Piconet 232X AZ) fitted to KEMAR's right ear, and normalized to 0 dB in the front position at 0° azimuth, for the frequencies 500 Hz, 1000 Hz and 2000 Hz.*



A fair amount of experience is required in order to interpret the results of Fig.4 since each frequency has its own pattern of maximums and minimums at different angles around KEMAR's head. In the specifications of hearing instruments it is therefore rather common to look at the frequency response from 0° and 180° sound incidence as shown by Fig.5. In Table 1 this is referred to as the " front to back response ratio".

Fig. 5: *Front to back response ratio: - The insertion gain response of an Audio Zoom hearing instrument (Piconet 232X AZ), set to the reference test position (IEC 118-8 & IEC 711) and the directional microphone mode, plotted for θ=0° and θ=180° sound incidence. Notice the large signal suppression from the back over a wide frequency range.*



The suppression of frequencies up to 3kHz is in the order of 20 dB, gradually decreasing thereafter. Frequencies above 8kHz are not canceled at all. Since most environmental noise sources are low frequency dominated, this kind of directional microphone can help improve the Signal to Noise Ratio (SNR) in many real life situations.

The polar sensitivity pattern, or the front to back ratio, gives considerable, specific information regarding the properties of a directional microphone. A more global measure, the Directivity Index, is often used in acoustics to specify the overall directional properties of a microphone.

The Directivity Index:

The directivity index (DI), is expressed in dB, and defines the amount by which a directional microphone attenuates sounds in a diffuse sound field, compared to an omnidirectional microphone. According to this definition, we can see from Table 1 that the directivity index of an omnidirectional microphone is 0 dB.

The implementation of a diffuse sound field is very complicated. The exact measurement of the directivity index is therefore rather time consuming and costly. Provided that the spatial directivity pattern is symmetrical, which is normally the case for microphones in free field conditions, and to a first order approximation this is the case with KEMAR, the measurement of the directivity index can be simplified to the measurement of the polar sensitivity pattern, as described previously in Fig.4.

The directivity index can be calculated from the polar sensitivity plot data at each frequency:

$$\mathbf{DI = 10 \log \{ \sum_{i=1}^{n} (P_i)^2 \cdot \sin (\frac{2\pi}{n} \cdot i) \bullet \frac{2\pi}{n} \}} \quad \text{Eq. (3)}$$

where – P_i is the sound pressure level at the angle of incidence $2\pi/n$ compared to the sound pressure level from 0°azimuth.
– n is the number of points measured.

Measurements and calculations according to Eq.(3) have been made initially with KEMAR's open ear in order to define a basis for comparison. It should be noted however, that KEMAR can be used with none, one or two neck rings and different ear sizes, with each setup yielding different directivity patterns, resulting in approximately +/-0.5 dB between 1 and 4 kHz.

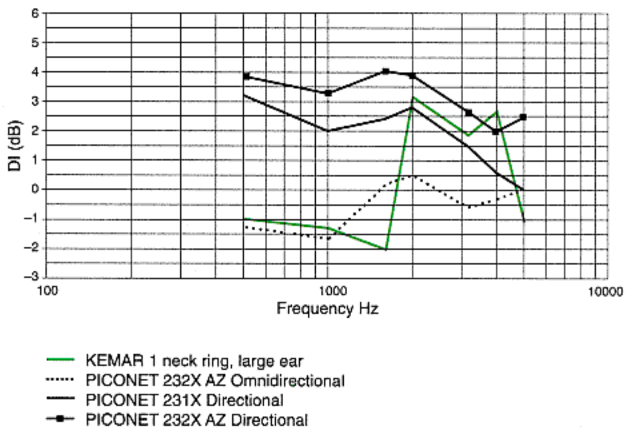
The directivity index of an open ear is very similar to the open ear frequency response, being rather flat (-1 dB) up to 1.6 kHz, where the response rises steeply to about 2.5 dB, falling steeply again above 4 kHz (see Fig. 6).

Two different types of BTEs have been fitted to KEMAR. The results are plotted in Fig. 6.

1. A BTE with a conventional directional microphone (Piconet 231X)
2. A BTE with Multi-Microphone Technology (Piconet 232X AZ) in the omnidirectional and directional mode.

Due to the head shadow effect, the directivity index of a BTE with an omnidirectional microphone worn by KEMAR does not necessarily yield exactly 0 dB, as defined for the ideal free field condition in Table 1. When compared to the open ear, on the other hand, the omnidirectional mode cancels at high frequencies about 2.5 dB of KEMAR's open ear directivity index. On the other hand, the directional mode of the Piconet 232X AZ Multi Microphone Technology (MMT) fully compensates for the loss of directivity at high frequencies. Overall performance of this sophisticated directional microphone is superior to that of the open ear, especially in the low frequency area, where environmental noises mostly have their emphasis. In such situations, even normal hearing subjects would likely profit!

Fig. 6: *The directivity index as a function of frequency has been plotted for different setups.*



NOTE: The directivity index of directional microphones is superior to that of the open ear.

Omni vs. directional microphone:
The average difference between the directivity index of the omnidirectional and directional mode of the BTE with Multi Microphone Technology (Piconet 232X AZ) is about 3.5 dB, more specifically 4.5 dB in the lows and 2.5 dB in the highs.

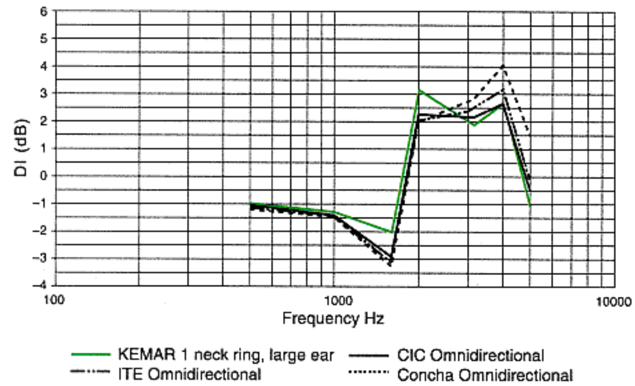
Audio Zoom technology vs. conventional directional microphone:
The directivity index of a BTE with Audio Zoom Multi-Microphone-Technology (Piconet 232X AZ) is about 1.5 dB better than that of an instrument with a conventional directional microphone (Piconet 231X).

Open ear vs. conventional microphone:
In the high frequencies the directivity index of the open ear is better than that of the conventional directional microphone. In the low frequencies, however, a significant benefit of 3.5 dB is measurable.

4.4 Directional characteristics of ITE hearing instruments

The position of the microphone for in-the-ear instruments has been described as superior for directional hearing, compared to the position for BTE microphones. In Fig.7 we have compared the directivity index of in-the-ear instruments (CIC, ITC, Full Concha), having various microphone locations, with that of the open ear.

Fig. 7: *The directivity index, as a function of frequency, for different types of In-The-Ear hearing instruments is compared with KEMAR's open ear.*



In all cases, the directivity index of ITE's is better than that of BTE's with omnidirectional microphones. Logically, we could predict that the deeper the microphone sits in the ear canal, the closer the directivity index would approach that of the open ear performance and this is consistent with the measurements shown in Fig.7. It is interesting to note however, that the microphone position of a concha hearing instrument does not degrade the directivity index at all and it is even slightly improved in the high frequencies, compared to the open ear.

In making conclusions from these measurements we would not suggest that concha instruments generally provide a better directivity index. We would conclude, however, that wide tolerances exist for the high frequencies in directivity index measurements and calculations. It is therefore unlikely that a particular style of in-the-ear instrument would reliably give better directivity index results than any other in-the-ear style. We would also conclude that ITE's, as a class of instrument, behave like the open ear in terms of directivity index. Furthermore, as shown in Fig.6, BTE instruments with directional microphones can have better directivity performance than the open ear and thus ITE's as well. Such BTE's can thus be considered as superior when directionality is an issue; which is undoubtedly the case in noisy environments.

4.5 The most favorable directional microphone characteristic

There are various opinions about what is the best choice of target parameters to use for optimizing the design of directional hearing instrument characteristics. If the goal is to maximize the directivity index as a global measure this could be maximized by optimizing the ratio of the internal to the external delay β. Alternatively, to optimize discrimination in noise with hearing instruments, we would need to achieve a beam to the front of the instrument user, which is the usual source of the signals of interest in conversation, and suppress environmental noises from the sides and the rear as much as possible. When a single directional microphone is used, we can conclude from Table 1, that cardioids, hypercardioids or super cardioids should typically be used in hearing instruments, since both the directivity index and the front to back ratio are favorable.

5. Audio-Zoom: Multi Microphone Technology allowing a combination of two microphone characteristics in the same hearing instrument.

The adaptive filter schemes currently used in hearing instruments, cannot be considered as effective noise suppression systems. There is an absence of clinical and scientific proof that they really do improve hearing ability in noise. Conversely, directional microphones have been shown to improve the Signal to Noise Ratio very effectively, both in theory and practice (Hawkins & Yacullo 1984, Soede 1990).

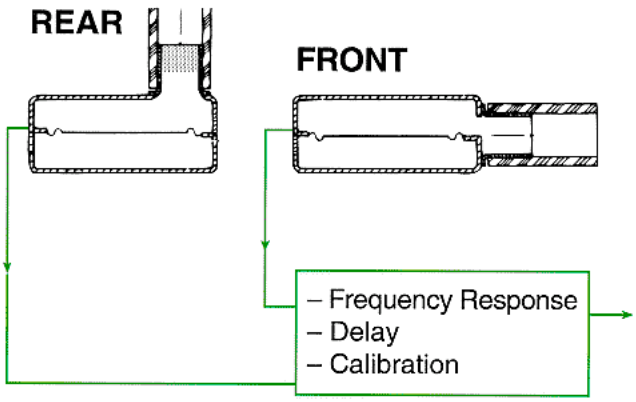
While directional microphones have advantages for listening in noise, they can also have disadvantages in certain situations. The reduction of low frequency amplification can negatively influence subjective judgments of sound quality, especially during conversation in quiet and when listening to music. An additional disadvantage is seen in situations where it may be undesirable to damp signals from the sides and the rear, such as in traffic and for the detection of warning sounds (Verschuure & Dreschler 1993).

In order to allow optimal listening conditions in both quiet and noise, a modern communication system must offer a choice of both directional and omnidirectional microphones. The introduction of the Audio Zoom system has made this possible: The design to achieve the directional microphone effect, electronically integrates two individual microphones, the hearing instrument user can easily switch from an omni to directional mode, by pressing one key on the remote control.

5.1 Technical realization

The conventional directional microphone as illustrated in Fig. 2 is basically measuring the pressure difference between two cavities. Pressure difference occurs due to an external signal delay between the two sound ports and an internal signal delay, produced by an acoustical filter built into one port. This effect has been achieved with increased efficiency in Audio Zoom, by utilizing two separate omnidirectional microphones and finely tuning their combined electrical output signals via an electronic network as shown in Fig.8. The parameters τ and Δd responsible for optimizing the directivity patterns, as discussed in Table 2, apply equally for Audio Zoom. Provided the two microphones are perfectly matched, the achieved result is close to cardioid.

Fig. 8: *Cross section of the Audio Zoom directional microphone system, as realized in the Piconet 232X AZ hearing instrument. The acoustic responses of two microphones, one mounted to the front and the second to the rear of the instrument, are combined and controlled by electronic circuitry.*



There are a number of advantages in applying the concept of multiple microphones. There is a significant degree of additional freedom for design optimization, if two independent microphones can be used to form the directional characteristic. The internal propagation delay τ, can be realized either by acoustic means or an electronic delay line, or a combination of both. Furthermore, the frequency responses of the two microphones can be shaped differently. This additional design freedom does have one, production related, drawback. Since the goal is to achieve reliable directional properties in each individual hearing instrument, it is necessary to precisely calibrate the two microphones by means of electronic tuning, in every single hearing instrument.

That the greater complexity of Audio Zoom has resulted in superior performance is clearly demonstrated in the comparative measurements shown in Fig. 6. The directivity index of Audio Zoom is typically 1.5 dB better than that of a conventional directional microphone.

5.2 Practical benefits of directional microphones – Signal to Noise Ratio improvement.

The discussion so far has focused on the physical parameters of directional microphones. The following section will concentrate on their practical benefits. The directivity index is a form factor, which describes the efficiency of the directional microphone in terms of spatial patterns, i.e. the shape of the beam or polar plot. The front to rear response ratio, on the other hand, gives more insight into the practical signal processing properties, specifically the SNR that can be expected. Both factors need to be considered for a complete impression of the overall performance of the directional microphone in a hearing instrument.

In order to discuss performance in more realistic operating conditions, while still having the ability to achieve laboratory measurements, an Audio Zoom hearing instrument (Piconet 232X AZ) was fitted to KEMAR and exposed to speech-like noise in a test environment (Fig.3). The speech-like noise was shaped according to the International Long Term Average Speech Spectrum (ILTASS) proposed by Byrne (1994). The ILTASS corresponds to an average speech spectrum and was derived from samples of a large number of languages. The dynamic range is specified to be 30 dB, with speech peaks 12 dB above the average level and the weakest components 18 dB below the average; as shown in Fig.9.

Fig. 9: *Dynamic range of speech, represented by the 1/3-Octave band levels of the International Long Term Average Speech Spectrum (ILTASS; Byrne, 1994) at an average level of 60 dB SPL, measured at 1m distance from the loudspeaker in the free sound field.*

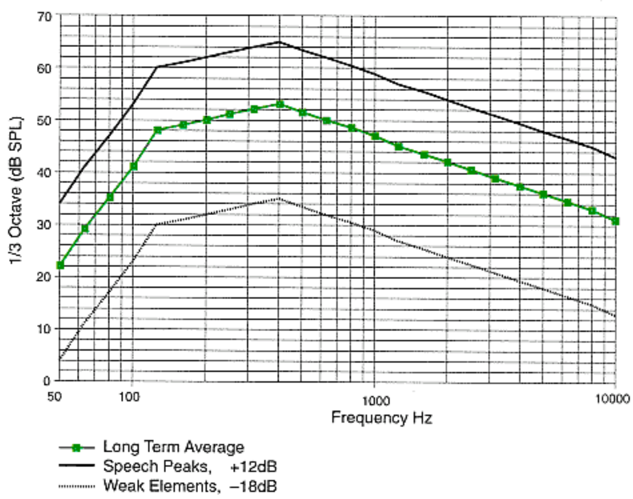
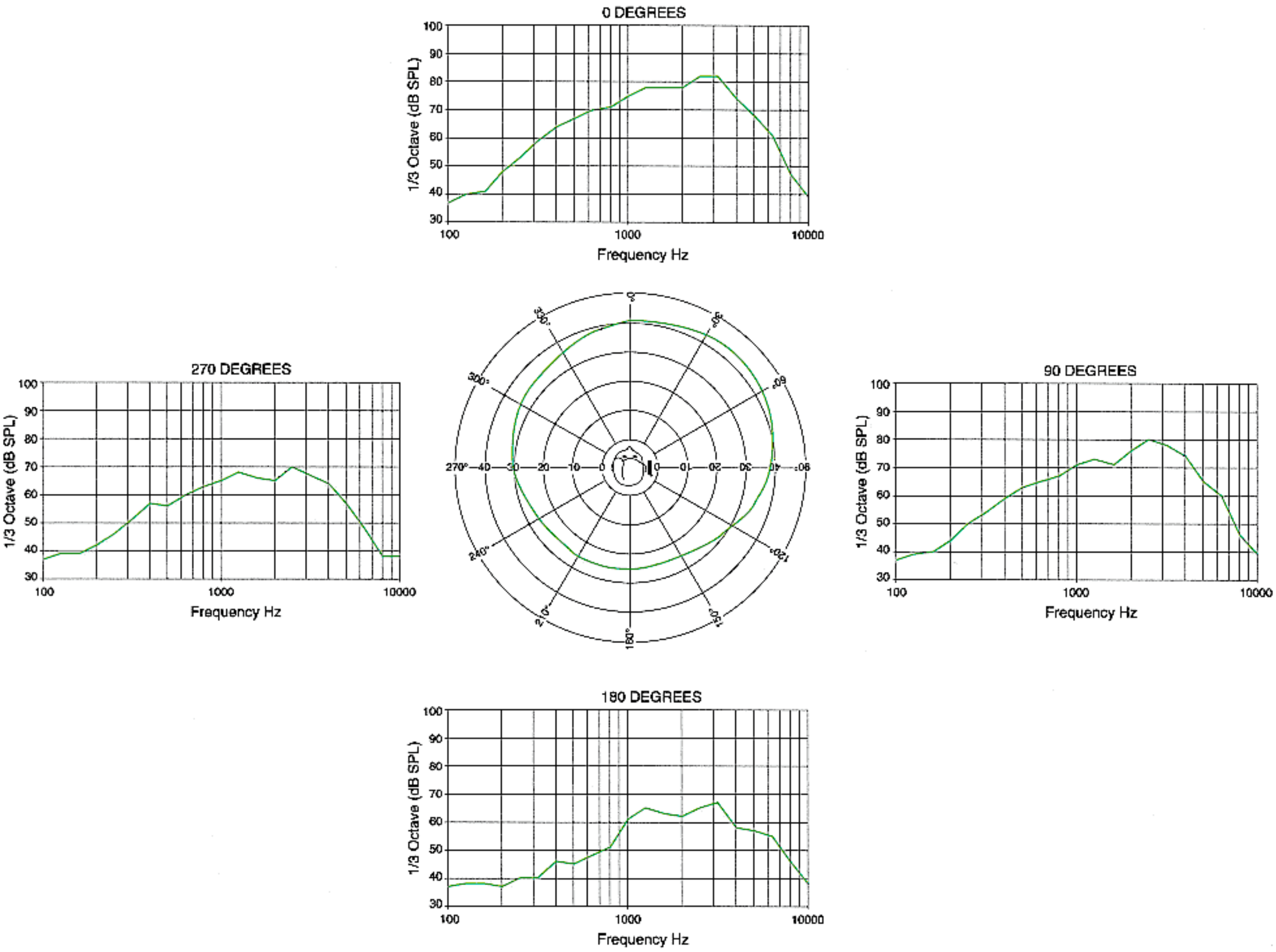


Fig. 10: *1/3-Octave band hearing instrument frequency responses (Piconet 232X AZ) from a range of sound orientations, measured on KEMAR with a speech-like signal (ILTASS). The polar sensitivity pattern of the broadband average response is plotted in the center of the figure. The average front to back response ratio is approximately -15 dB, the side suppression -3 dB to the right and -10 dB to the left.*

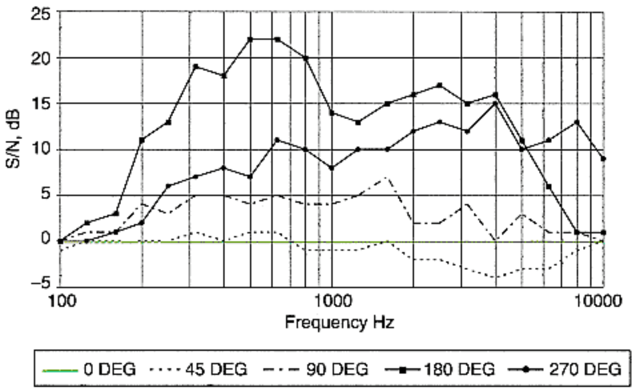


The hearing instrument output response, analyzed in 1/3-Octave bands from different orientations, and measured via ear simulator, is plotted in Fig.10. The polar sensitivity pattern of the broadband average response is plotted in the center of the figure. The average front to back response ratio is approximately -15 dB, the side suppression -3 dB to the right and -10 dB to the left.

The frequency responses around the polar plot are, in a first order approximation, identical in shape and are simply shifted horizontally (same spectrum, less gain). The directivity characteristic is therefore almost frequency independent, a fact which has already been reflected by the nearly frequency independent directivity index (Fig.6).

The most crucial and challenging question, however, is the benefit of Audio Zoom technology for speech discrimination in speech-like noise. For the following experiment we assume that speech in the form of the ILTASS is presented directly from the front. The exact same signal, matched for spectrum and power, is then presented from different angles (by turning KEMAR relative to the loudspeaker position). If we normalize the measurements to the front position of 0° azimuth, the SNR, calculated as speech coming from the front relative to speech coming from a specific angle, can be plotted as presented in Fig.11.

Fig. 11: *Signal to Noise Ratios from various sound orientations: The 1/3-Octave hearing instrument responses (Piconet 232X AZ) from different sound orientations, measured on KEMAR by applying a speech-like signal (ILTASS), are normalized to the response from the front direction at 0° azimuth.*



We can clearly see that the SNR is almost independent of the frequency and varies mainly as a function of the rotation angle. Due to the head shadow effect, the SNR is negative at an angle of 45° for frequencies above 1 kHz; the SNR is best at 180°, and easily achieves 15 dB up to 4 kHz. The same results can be seen when looking at the average power spectrum in Fig.10. The very efficient suppression of the high frequencies has been achieved by carefully optimizing the design of the directional microphone, as mentioned in the preceding section. Conventional directional microphones available for hearing instruments are not as efficient in the high frequencies.

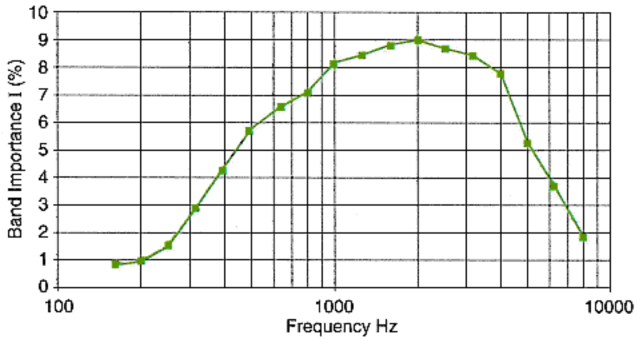
These measurements demonstrate clearly Audio Zoom's superior ability to improve the Signal to Noise Ratio, for signals arriving from between 90° and 270°. This improvement can be seen even for the very difficult situation where the signal and the noise have the same spectrum and power.

5.3 Communication benefits, Articulation Index

It is sometimes difficult to interpret the Signal to Noise Ratio (SNR) in practical terms and to estimate its real benefit for communication. One way to interpret the value of SNR improvements is via the Articulation Index (AI) which has been found to be a useful measure to calculate and quantify speech intelligibility (Pavlovic, 1987).

The Articulation Index is a weighted sum of the Signal to Noise Ratios in specific frequency bands, mostly 1/3-Octave bands, where the weights are described by the frequency importance function. The frequency importance function takes into account the fact that not all frequencies are equally important for speech discrimination. The importance of particular frequency bands is variable, across different languages and speech material used, whether it is sentences or syllables. In order to allow global, general analysis of conversational speech, Pavlovic (1987) has defined a theoretical average frequency importance function, where the frequencies around 2kHz are considered to be most important for speech discrimination (see Fig.12).

Fig. 12: *1/3-Octave Frequency Importance Function for average speech (Pavlovic 1987). The relative weights of the specific bands are expressed in % and sum to 100%.*



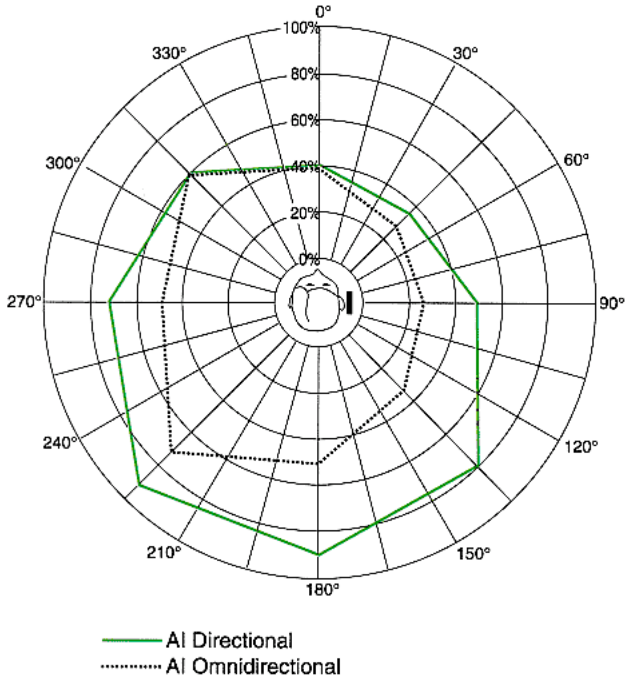
The Articulation Index can be calculated by applying the following equation:

$$\mathbf{AI = \sum I_i (SNR_i + 12) / 30} \quad \text{Eq. (4)}$$

where
- I_i describes the frequency importance of band-i, according to Fig.12
- SNR_i describes the Signal to Noise Ratio of band-i, according to Fig.11
- AI is expressed in% or from 0...1

Since we have plotted the Signal to Noise Ratio for different angles of sound incidence, we can calculate the Articulation Index (AI) for the different orientations. The results are plotted in Fig.13.

Fig. 13: *The Articulation Index for speech in noise is plotted for various positions of the noise source while the speaker remains in front (noise and speech spectra are identical). The speech discrimination benefit for the Audio Zoom is evident, when comparing a Piconet 232X AZ in the omnidirectional and in the directional mode.*



From the front, the SNR is 0 dB and yields an AI of 40%. The polar sensitivity plot indicates a maximum at around 45°(Fig. 4), where we can either locate a negative SNR, (Fig. 11), or the minimum AI of 35%, (Fig. 13). A very high Articulation Index of 92% is estimated for noise sources located in the rear position, with 52% when noise is mainly coming from the right side and 76% when noise originates from the left. While improvement for the front direction SNR is physically impossible, a higher directivity index would improve the side suppression and increase the Articulation Index accordingly.

In a situation where the signal and the interfering noise have the same power; hence SNR = 0 dB, a hearing instrument would consequently produce an Articulation Index of only 40%. This situation exists in our experiment for the front position of both signal and noise with either directional or omnidirectional hearing instruments. On the other hand, as we have already seen, the head shadow effect itself produces a directivity pattern. Therefore the Articulation Index calculated at various directions for an omnidirectional microphone worn at the head is not a constant, the results depend on the angle of noise incidence as well.

The plot in Fig. 13 provides a direct estimate of benefit for the Audio Zoom in terms of Articulation Index, when compared to an omnidirectional microphone response. The results clearly favor the Audio Zoom.

5.4 Audio Zoom - additional parameters, considerations

User selected programs, or "comfort programs", are a fundamental part of the philosophy of the Phonak PiCS system. It is clearly a logical step to include the appropriate microphone characteristic into this basic signal processing strategy. This approach allows the individual hearing instrument user to have much improved control of their particular listening needs. A prerequisite for the success of such a multi-program listening concept is a user friendly, ergonomically designed, remote control device.

The specific properties of the directional or omnidirectional microphones can be ideally combined with the adaptive filtering methods applied in the implementation of the comfort programs (Bächler & Vonlanthen, 1994). In full accordance with the environment for which each program has been designed, either suppression or emphasis of different frequency regions is provided, as a function of the assumed input spectrum and power. The most appropriate microphone position (omni or directional) is assigned to each comfort program. The selection of a particular program will activate both the adaptive filtering parameters as well as the microphone selection simultaneously and automatically.

While the microphone position assigned to the basic listening program is omnidirectional, switching to a party noise program (appropriate for a situation where the background noise is voice babble plus environmental noise) will automatically activate the directional microphone in addition to the Phonak party algorithm. As a result speech discrimination performance is enhanced, as indicated by the improved Articulation Index score (Fig.13). If a selection of a music comfort program, designed to provide full sound quality and fidelity, is made, the microphone mode will automatically switch back to the omnidirectional position.

5.5 Field studies, clinical results

Field studies have been conducted at two clinics in the United States, the results are currently pending publication (Valente & Fabry, 1995), a brief summary of the results will be given here. The subjects involved in the studies had mild to moderate cochlear hearing losses with pure tone averages from 40-70 dB HL. The Signal to Noise Ratio required for 50% correct identification of key words in sentences, was measured in speech shaped noise. The signal was presented from the front, at 0° azimuth while the noise source was located behind the subjects, at 180° azimuth. The average Signal to Noise Ratio improvement with the Audio Zoom directional microphone, compared to the omnidirectional microphone, was approximately 8 dB.

5.6 Future trends in development

The implementation of Audio Zoom is the first step in the application of multiple microphone systems in hearing instruments. Combining two directional microphones in a second order Jacobi array, as indicated by Soede (1990), would improve the directivity index by another 4-5 dB and achieve a total improvement of 8-9 dB!

Additional benefit could be obtained when combining microphone arrays with digital signal processing strategies implementing adaptive beamformers (Schwander & Levitt 1987, Peterson et al. 1987, Kompis & Dillier, 1994). At a practical level, multi microphone systems, 2nd order or higher, require very well matched microphone cells to achieve stable and reliable acoustic responses. Todays' electret microphone technology does not fulfill this requirement specifically.

6. Summary and conclusions

The design parameters for directional microphones in hearing instruments, should reflect the fact that the origin of signals of special interest to the user is generally from the front; since it is usual to face the speaker in conversation. Sound from all other directions is usually considered to be competitive and should therefore be suppressed. Directional microphones need to have both a high directivity index and a good front to back response ratio to fulfill that target. Aiming for a cardioid polar response characteristic will come close to the desired result.

The Audio Zoom system uses two separate omnidirectional microphone cells which are combined by an electronic network in order to achieve access to both a directional and an omnidirectional mode. This implementation allows very specific and detailed optimization of the microphone directional properties. As a result, directionality is superior when compared to a conventional directional microphone in the same hearing instrument. Direct user interaction with the appropriate microphone selection, by means of comfort programs in different listening environments, makes the ideal microphone characteristic easily accessible. The benefit of Audio Zoom in difficult listening situations, such as a background of competing speech, has been estimated with the Articulation Index, and calculated for different angles of noise incidence. Calculations show that speech discrimination performance is improved by 40% when switching from the omnidirectional to the directional response and the noise source is located at the rear. Laboratory tests with actual hearing impaired users seem to confirm these findings (Valente & Fabry, 1995).

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