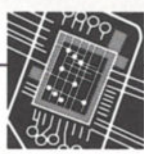
Directionality in Hearing...Revisited

By Jeremy Agnew, PhD

There is an old saying that "what goes around, comes around." What has gone around in the past is directional hearing to assist speech understanding in noise. What has come back around again is —guess what?—directional hearing. Although directional hearing instruments have not made a large impact as a fitting strategy in the past, advances in hearing instrument technology, coupled with a better understanding of the spatial separation of sounds, directional hearing and listener localization ability, offer new promise for the future by improving the signal-to-noise ratio (also called the SNR or S/N ratio) of speech in situations with competing noise present.

In the ongoing search for improved speech understanding in noise, the importance of directional hearing is often underated. This article reexamines the basic principles of directional hearing instruments and looks ahead at the use of directionality in CIC and DSP hearing instruments of the future



Importance of Directional Hearing

One of the frequent complaints of hearing instrument wearers is the inability to understand speech in backgrounds of competing noise. The similarity of desired speech and undesired background noise, when coupled with the deterioration of cochlear function that accompanies sensorineural hearing loss, makes this a very difficult communication situation for listeners with a hearing impairment.

It is now accepted that fitting strategies, such as the attenuation of low frequency amplification in background noise situations, will improve the listening comfort of a hearing instrument wearer by reducing the level of annoying low frequencies, but this type of fitting strategy will not improve intelligibility.[1,2] The key to improving speech understanding in noise is to improve the *signal-to-noise* ratio. The SNR figure indicates the number of decibels by which the signal, in this case the desired speech, exceeds the background noise level. The larger the SNR figure, the higher the level of desired speech above the undesired noise level and, thus, the easier it is for the listener to understand. Hearing-impaired individuals, due to a deterioration of cochlear function, may require an SNR of 10 dB to 14 dB greater than that required for normal-hearing listeners.[3,4]

Much research has focused on solving this problem. The expectation in signal processing schemes in other areas of communication is that the signal that the listener wishes to distinguish (whether it is speech across the table or signals from a satellite in space) is different from the competing noise in one of the three fundamental characteristics of any signal, which are its frequency, amplitude and time characteristics.

Speech understanding in noise presents an unusually difficult processing problem. The long-term frequency spectra of one talker and many talkers are usually the same. The amplitude of a single speaker in a noisy cocktail party situation is typically the same as (or much lower than) the surrounding noise. This is often what causes a speaker to raise his or her voice above normal levels in a reflex action in order to be heard clearly. Finally the speech and noise occur at the same time, or else there wouldn't be a problem. Thus, if the desired speech and the undesired background noise received at the microphone of the hearing instrument are identical in long-term frequency response, amplitude and time, then there are no fundamental signal differences that can be used to separate the speech from the noise with current ear-level hearing instrument amplifiers.

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The advantage that normal-hearing listeners have in these noisy situations is the ability to take advantage of the difference in spatial location that normally occurs between the desired speaker and the undesired noise. If spatial separation is present, the normal functioning of the central auditory system allows suppression of the undesired background noise and the ability to concentrate on the desired speech. Dirks and Wilson, for example, have shown that intelligibility scores were greater when sources of speech and noise were spatially separated.[5]

Tests for Speech Intelligibility in Noise

Some traditional speech-in-noise tests have used speech and noise presented from the same loudspeaker during the test and do not account for the advantages of spatial separation. Newer tests are:

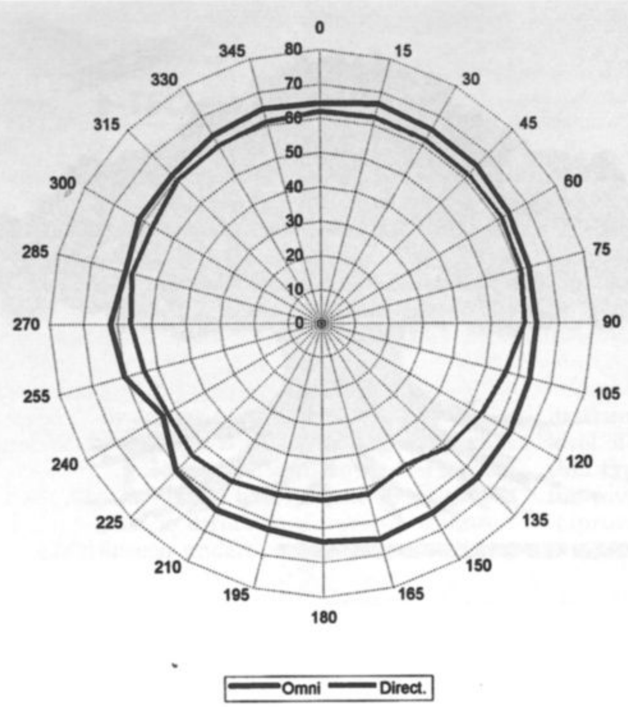

Fig. 1. *Polar patterns on KEMAR at 2000 Hz of a two-microphone directional hearing instrument, comparing the omni-directional mode and the directional mode.*

▶ *SPIN Test*: A test for speech intelligibility in noise that has been available for some years is the Speech Perception In Noise (SPIN) test, which consists of eight sets of 50 sentences against a background noise of 12-voice multi-talker babble.[6] Sentences and babble are separated, thus allowing a choice of SNR.

▶ *SIN Test*: A relatively new test is the Speech In Noise (SIN) test. The SIN test is a pre-recorded test used to compare two listening situations.[7] The target speech for each condition consists of two 40-sentence test blocks of female conversational speech with almost no contextual clues. The competing noise is a background of four-voice, multi-talker babble, consisting of two men and two women. Choices of SNR are 15 dB, 10 dB and 0 dB. The presentations are made at both 55 dB and 80 dB in order to simulate a range of listening conditions consistent with speech communication.

▶ *HINT Test*: A sophisticated test that has emerged from the laboratory and recently moved into clinical application is the HINT (Hearing In Noise Test), a new test that specifically takes advantage of binaural separation during testing.[8] This test is a practical tool for the assessment of binaural directional hearing in noise at supra-threshold levels. By manipulating competing noise at different locations in space and scoring the results of sentence comprehension, this test provides a measure of the subjects capacity to understand speech in noise. This is an adaptive test where background noise is presented at a fixed level, and then the level of the sentences is varied to obtain the Reception Threshold for Sentences (RTS), a score of the level at which the test sentences are repeated correctly 50% of the time.

Strategies for Improving Spatial Separation

Since spatial separation offers a potential for improving the SNR and hence improving the ability to understand in noise, what are some of the available and future strategies that may accomplish this task? When considering each strategy, it should be pointed out that for speech material presented as a continuous discourse, intelligibility typically improves on the order of 6% to 10% for each decibel of improvement in SNR. Thus, a 3 dB improvement in SNR could result in up to 30% improvement in speech intelligibility.

Binaural Hearing Instruments

It has been known for many years that an appropriate fitting of binaural hearing instruments will improve the ability to understand speech in noise by presenting binaural cues to the central auditory system. The improvement is typically 2-3 dB over monaural listening at threshold levels and up to 6 dB at suprathreshold levels. The effectiveness of binaural listening over monaural listening can easily be demonstrated by plugging up one ear with a finger while trying to understand speech from a nearby talker in a noisy restaurant.

Directional Hearing Instruments

To enhance the separation of speech and noise, and thus improve the SNR, one strategy is to make the hearing instrument such that it has reduced sound amplification from the rear and sides, rather than amplifying sound from all directions equally like an omni-directional hearing instrument. This is the basis for a *directional* hearing instrument, which amplifies sound from in front of the listener more than from behind. The use of directional hearing instruments also can improve the wearer's ability to localize sounds.[9]

With the continued miniaturization of microphones, improved directional hearing instruments are now available that use two microphones and offer improved SNR figures over instruments with omni-directional microphones.[10] Some of these newer directional hearing instrument designs provide directionality at all frequencies, thus improving the rejection of interfering high-frequency noise from the rear of the listener.

The directional patterns of a switchable two-microphone directional hearing instrument, comparing the omni-directional mode and the directional mode on KEMAR at 2000 Hz, are shown in Fig. 1 as a polar plot. This plot is a horizontal plot as if the viewer were looking down onto the top of KEMAR's head, with KEMAR at the center of the graph. The figures around the outside are given in degrees of rotation, thus 0° is directly in front of the listener and 180° is behind the listener. The vertical column of figures is gain in decibels. The heavy black circular plot (outer) is the omni-directional mode, which shows that the gain of the hearing instrument is essentially the same in all directions. The lighter gray line (inner) is the gain in the directional mode, showing that the gain at 0° has stayed about the same as the omni-directional mode, but that the gain at 180° has dropped by about 12 dB. This shows the amount of attenuation for sounds from the rear. The plots are not exactly circular due to the distortion of the sound field by the presence of KEMAR's head during the measurement. The effect of making the measurement on a human listener would be the same.

Directional hearing instruments are available primarily as BTE hearing instruments. The reason for this is that, in order to work correctly, directional hearing instruments require signals from the rear to enter both microphone ports with specific amplitude and time differences in order to produce maximum attenuation. If such a system is built into an ITE hearing instrument, the results will be highly dependent on the depth of the shell in the pinna. As the faceplate is placed deeper into the concha, the natural acoustic shielding effect of the pinna alters the relationship of different frequencies of sound entering from the rear, and hence degrades the directional effect. In order to consistently produce the same directional effects as a BTE hearing instrument, an ITE hearing instrument would have to be built so that it extends far enough out of the concha to be flush with the pinna. This tends to reduce the cosmetic appeal of an ITE hearing instrument.

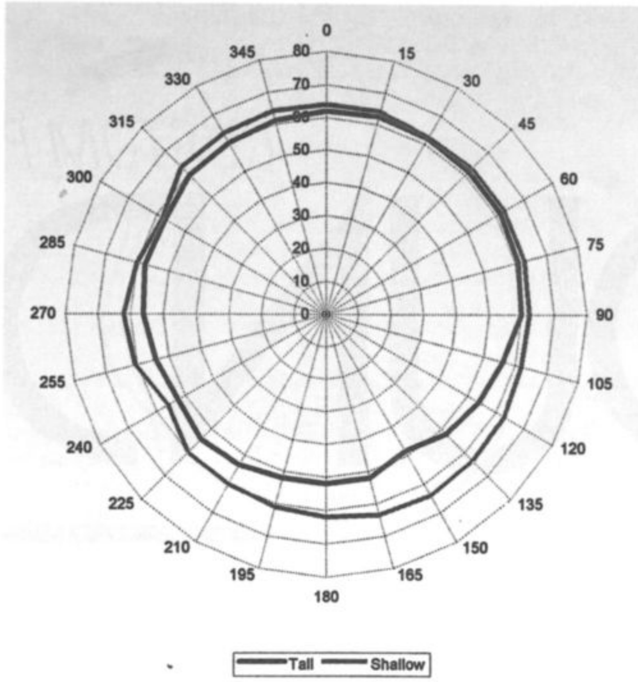

Fig. 2. *Polar patterns on KEMAR at 2000 Hz of an ITE hearing instrument, comparing shell heights with the faceplate flush with the outside of concha (tall) to the faceplate recessed into the concha (shallow).*

An example of this is shown in Fig. 2, which compares the polar patterns at 2000 Hz of an ITE hearing instrument on KEMAR that is flush with the concha and one that is recessed into the concha. The polar directional pattern of the ITE that was recessed into the concha is shown as the lighter gray line (outer) on the graph. This shows that, at 180°, the sensitivity is approximately the same as it is at 0°. When the shell is made taller, so that it is flush with the outside of the pinna in order to fully expose both microphones to sounds from the rear, the heavy black line (inner) on the graph shows that there is approximately 10 dB of attenuation from the rear direction, as compared to the sensitivity from the front.

Completely-in-the-Canal (CIC) Hearing Instruments

Another way of maintaining a degree of directional hearing ability is through the use of CIC hearing instruments. CIC hearing instruments offer many benefits, including cosmetic acceptability and a clear sound. One of the important benefits of CIC hearing instruments is that they appear to maintain hearing localization and directionality.[11, 12] Unsolicited comments from CIC hearing instrument wearers indicate that spatial perception abilities are improved over conventional ITE fittings. This is thought to be due to the placement of the microphone inside the ear canal, allowing the convolutions of the concha to remain open and provide the spectral filtering that assists in directional hearing abilities.

Directional Microphone Arrays

Though directional hearing instruments are available today, another promise for the future is the emerging development of new concepts in directional hearing instruments. The limited space inside an ear-level hearing instrument has previously limited the complexity of the circuitry that can be practically built into a hearing instrument case, however, current advances in technology are allowing multiple microphone systems to become a reality.

Researchers are developing multiple microphone arrays which provide further narrowed directionality, thus allowing a listener to acoustically focus on one speaker while suppressing surrounding noise.[13] Typically these devices use four or five microphones in an array that is mechanically and electrically tuned for optimum directionality. Examples of these experimental devices are the broadside array, a horizontal microphone array worn on a headband and the endfire array, which is built along the temple-piece of eyeglasses. The electrical outputs of the individual microphones are summed together by special electronic circuitry which results in a very narrow pickup pattern from the forward direction. Because of the requirements for wide microphone spacing, this approach has not received high cosmetic acceptance among hearing instrument wearers. However, the basic concept of the strategy is sound, and future development of sub-miniature microphones may reduce the size to a more acceptable form for use by mainstream hearing instrument wearers.

DSP Beam-forming Hearing Instrument

Sophisticated digital signal processing (DSP) techniques are being used to achieve the same sharply-focused patterns of multiple-microphone arrays with only two microphones. This technique has been adapted for acoustic use in hearing instruments from technology used in highly-directional radar systems. Such systems can provide improvements of 20 dB in SNR in rooms with low reverberation. Experimental binaural devices using these techniques are currently undergoing testing with hearing-impaired users.[14] The technology is not currently available for self-standing ear-level hearing instruments due to the complexity of the signal processing needed and the requirement that the two hearing instruments of a binaural pair communicate with each other. Continued advances in circuit size reduction, however, will make this goal an eventual reality.

Though highly-focused directionality is an advantage for concentrating on a single speaker in background noise, it can also be a disadvantage. While the speaker in front of the listener may stand out from the sea of background noise, other speakers to the side and rear will be attenuated along with the undesired background noise. Thus, some mix of sound from the sides must also be blended into the total sound to allow a full range of listening capability.

Binaural DSP Hearing Instruments

Another hearing instrument approach uses a DSP algorithm to improve hearing in noise by preserving binaural hearing cues presented to the listener. The goal of the processing is to provide correction for frequency-dependent hearing loss, while simultaneously eliminating the insertion effects of the hearing instrument and accurately balancing sensation levels for the left and right ears.[15] This effectively maintains the aided head-related transfer function (HRTF) of the hearing instrument wearer in the unaided condition, theoretically making the hearing instrument transparent to the listener for localization cues.

Conclusion

Improvements in the ability to understand speech in noise are achieved by increasing the signal-to-noise ratio. Several approaches may be taken to achieve this. One method is by taking advantage of the difference in spatial location that occurs between a desired speaker and surrounding background noise. Improvements may be achieved by maintaining locational cues presented to the listener, thus optimizing the signal to be decoded by the central auditory system. Another method is by using artificial enhancement of the polar pattern of sound reception before it reaches the inner ear through the use of one of several directional microphone strategies. Each strategy maintains, or enhances, directional hearing ability as an important, but often underrated, method of improving speech communication. ◆

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