Using digital signal processing to enhance the performance of dual microphones

By Francis K. Kuk, Lars Baekgaard, and Carl Ludvigsen

In a previous paper, we considered the design issues between a single cartridge and dual-microphone directional designs implemented in an analog manner.[1] The application of digital technology to microphone designs may overcome some of the limitations of directional microphones if the technology addresses the wearer's expectations for a directional microphone. In this paper, we will discuss these wearer expectations and explain how a new directional-microphone system, the Locator™ system used in the Senso Diva, may meet these expectations.

WEARER EXPECTATIONS FOR A DIRECTIONAL HEARING AID

(1) Improvement in speech intelligibility in noise—everywhere and always

The main reason that hearing-impaired people choose hearing aids with directional microphones is their proven record in improving speech understanding in noise.

Valente provided an excellent summary of studies conducted on the efficacy of directional microphones.[2] In the laboratory, it has been demonstrated that directional microphones improved speech understanding in noise for both children[3,4] and adults[5,6] when the noise was presented at 180˚. Others have reported signal-to-noise ratio (SNR) advantage even when the noise was presented to the sides[7] or surrounding the wearers.[8]

However, the theoretical SNR advantages seen with directional microphones in the laboratory may not be fully experienced in real life. There are several circumstances that make this cancellation less effective. First, the sensitivity minima in the polar pattern are determined by the physical distance between the microphone openings and the acoustic delay in the second microphone opening. Unfortunately, it is not possible with traditional analog techniques to achieve the same delay at all frequencies. The result is that the direction of the sensitivity minima varies with frequencies.

Figure 1 shows the polar pattern of a directional microphone at two frequencies. Note that the sensitivity minima occur at approximately 140˚ for 1000 Hz but 160˚ for 2000 Hz. As the difference between frequencies increases, the difference in the positions of the minima increases. For a broad-band noise source (e.g., babble noise), the minima at different frequencies will not be in the same direction. Thus, the theoretical SNR advantage of the directional microphone could be reduced.

On the other hand, it is possible to use digital techniques to obtain polar patterns with sensitivity minima that are virtually identical at all frequencies. This would suggest that for the same reported directional characteristics, digital implementation of directional microphones may be more effective than analog implementation in real life.

A fixed polar pattern yields acceptable signal-to-noise ratio (SNR) improvements in laboratory (and real-life) evaluations where the noise is fixed in one direction. In real life, however, noise sources are unpredictable in their locations, intensity, and/or frequency content.

A directional microphone with fixed polar patterns cannot accommodate this randomness in noise characteristics. As a consequence, the wearer's expectations for an improved SNR may not be met unless the wearer is taught to physically adjust the hearing aids or to control the listening environment. Since it may be physically impractical or impossible to exercise this control all the time, mechanisms to automatically adjust the polar patterns may be helpful in ensuring consistent speech intelligibility in most noisy locations. This is achievable using adaptive DSP algorithms.

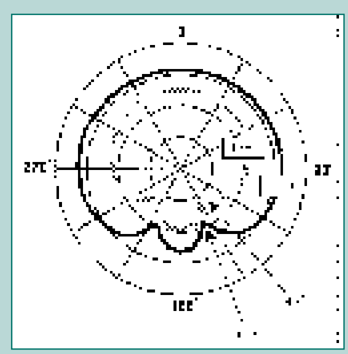

Figure 1. *Polar patterns showing sensitivity minima at two different frequencies (1000 Hz and 2000 Hz) at the same microphone distance.*

The number of noise sources can impact the effectiveness of a directional microphone. The more noise sources there are, the more likely it is that some will not be canceled because they are outside the sensitivity minima. For example, a single sound source in a reverberant room will be reflected from the surfaces of the room and every reflection represents an additional noise source. This results in a diffuse field and reduces the effectiveness of a directional microphone.[9,10]

Given these real-life situations, the immediate task is to design a directional hearing aid that provides the maximum improvement in signal-to-noise ratios, regardless of how speech intelligibility in noise is measured and where the noise signals originate. Achieving this requires that the directional microphone have a high directivity index (DI) at all frequencies. This is especially important in the low frequencies where the predominant noise frequencies reside. Furthermore, the microphone must be able to change its polar pattern according to the direction of the noise.

(2) No loss of audibility of soft speech sounds in quiet

Directional microphones achieve their intended functions by reducing their sensitivity to sounds from directions other than the front.[1] This can result in a loss of audibility for soft sounds when they originate from those directions. Lee et al. reported a 24% decrease in speech intelligibility between omnidirectional and directional modes when soft speech was presented from the back of wearers of a programmable directional hearing aid.[11]

Under the same conditions, wearers of a digital directional hearing aid that had a compression threshold (CT) at 20 dB HL achieved better speech-recognition scores in its directional mode than they did wearing the programmable hearing aid in its omnidirectional mode. It was suggested that a low CT would ensure minimum loss of audibility.

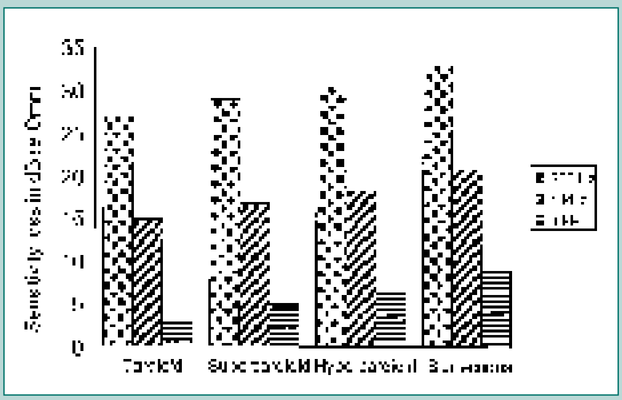

Figure 2. *Loss of microphone sensitivity for different microphone polar patterns re: omnidirectional microphone at 250 Hz, 1000 Hz, and 4000 Hz for frontal sounds.*

(3) Preservation of low-frequency cues

One difference between a directional microphone and an omnidirectional microphone is reduced low-frequency sensitivity in the directional mode. This reduction increases as the distance between microphone openings decreases.[1]

Furthermore, Figure 2 shows the reduction in sensitivity at various frequencies as the polar pattern is changed from omnidirectional to various directional patterns. The decrease in sensitivity is more noticeable in the low frequency than at higher frequencies. Also, the greatest decrease in sensitivity is seen for the bi-directional pattern and the smallest decrease is seen for the cardioid pattern.

The loss in low-frequency sensitivity would lead to an immediate decrease in perceived loudness because low-frequency sounds contribute significantly to the perception of loudness. This is especially noticeable if the change in polar patterns occurs abruptly. The task is to design systems that change the polar patterns in an unobtrusive manner. This requires proper compensation for the low-frequency sensitivity and a careful consideration of the timing parameters that control the changes in polar pattern.

In addition, Noble et al. have shown that low-frequency hearing in both ears is an important contributor to localization.[12] Consequently, auditory localization may be degraded if one does not compensate for the reduced low-frequency sensitivity of directional microphones. Thus, the task is to design a directional microphone system that compensates for the loss in low-frequency sensitivity in all polar patterns. This would require an intelligent system that can detect the resultant polar pattern and apply the appropriate amount of low-frequency compensation so that both the loudness and localization cues (in the lows) are preserved.

(4) Preservation of binaural cues

One of nature's noise-reduction systems is the ability of the two ears to provide separate inputs to the brain in order to "squelch" background noise and improve the perceived signal-to-noise ratio.

Such ability (binaural masking level difference, BMLD) is highly dependent on small time differences (i.e., phase relationship) between the signals reaching the two ears. However, disruption in the relative phase difference between the signals reaching the two ears becomes more likely when multiple microphones and adaptive strategies are used in directional microphone design. Drifting of the phase and amplitude responses of the two microphones within a dual-mic assembly, and differences in phase characteristics between the two dual-mic assemblies on the two ears could modify and reduce any binaural cues available to the wearers. It is desirable, therefore, to design a system that preserves the amplitude and phase characteristics of the binaural cues.

(5) No increase in wind and circuit noise

One limitation of directional microphones is their susceptibility to wind noise and circuit noise.[13] As a result, the target signal may be masked by the wind noise in windy environments and the wearers have to tolerate circuit noise in quiet. This limits the usefulness of directional hearing aids to indoor, mild-to-moderately noisy environments. A directional system that minimizes the annoyance from these noise sources could increase wearer satisfaction.

(6) No need for wearer manipulation

Some of the limitations of a fixed directional microphone can be minimized if the wearer has the cognitive and physical ability to re-position himself and/or adjust the microphone mode to optimize the SNR in different real-life situations. In practice, however, some wearers may not know that they need to make the adjustment and such adjustment may not be possible for young children and some elderly individuals. In addition, successful adjustment requires that the clinician adequately counsel the wearer on the appropriate strategy to maximize directional microphone use.

A hearing instrument with a low CT minimizes the need to switch,[11] but it cannot fully compensate for the loss of sensitivity. To ensure consistent maximum SNR improvement for the wearer, an adaptive system that requires minimal participation from the wearers would be necessary.

(7) The aid should be invisible (or nearly invisible)

Directional microphones are based on

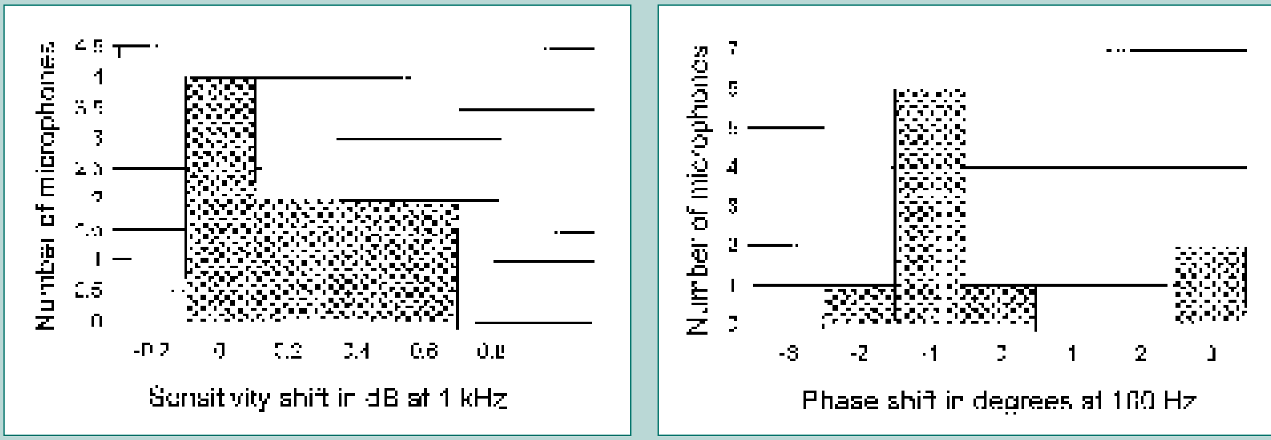

Figure 3. *Frequency of occurrence of (a) sensitivity and (b) phase drift among 10 microphones during accelerated aging tests.*

the principle of physical separation between microphone openings. This places a physical limit on how small the hearing aid can be. On the other hand, in a cosmetically driven market, it is important to design a directional hearing aid that is both functional and cosmetically acceptable. Kuk et al. showed that microphone distances as small as 5 mm would yield a good directivity index with manageable noise level.[1]

(8) Consistent performance over time

In order to achieve the SNR advantages and to meet all the wearer expectations listed above, the application of adaptive digital signal processing (DSP) techniques seems essential.

An adaptive algorithm is one that continuously updates its parametric values based on the output in order to maintain optimal performance. Accomplishing this requires inputs from at least two microphones. However, a requirement of using multiple microphones is that they be precisely matched in sensitivity and phase responses. Although most manufacturers attempt to choose the best-matched microphones to form the dual-microphone system, it is not always possible to match the microphones with perfect precision. Furthermore, drifting may occur over time in a once perfectly matched microphone pair.

In a series of accelerated aging studies (60° C with 100% humidity over 1 month) conducted at Widex's quality inspection lab, we witnessed a range of 0.6 dB drift in amplitude sensitivity and 3° drift in phase response among 10 randomly selected hearing aid microphones (Figure 3). Figure 3a shows that 4 of the 10 microphones drifted by more than 0.2 dB in amplitude response whereas Figure 3b reveals that 2 of the 10 microphones showed more than 2° in phase deviation. While this magnitude of drift (0.2 dB and 2°) is within the tolerance limit set for dual-microphone pair, it could eliminate most of the directivity in the low frequency.[1]

DSP technology has the potential to allow one to monitor the amplitude and phase characteristics of the two omnidirectional microphones and make the compensation necessary to ensure that they are perfectly matched at all times.

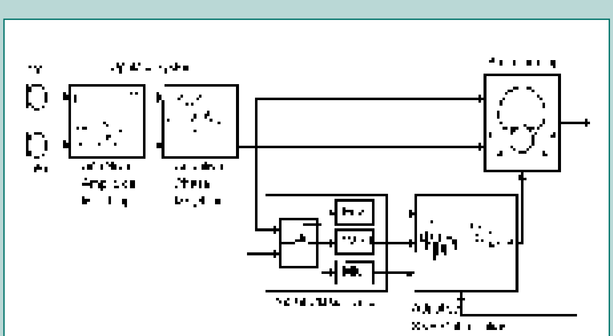

Figure 4. *Functional block diagram of the Senso Diva Locator microphone.*

FUNCTIONAL DESCRIPTION OF THE DIVA LOCATOR

The considerations discussed above prompted the design of the Diva Locator directional system. The Locator is a dual-microphone, adaptive directional system (beamformer) that uses DSP algorithms to control and manipulate the desired polar pattern. Enhanced dynamic range compression (EDRC) with a CT below 20 dB HL is used in all 15 channels of the hearing aid. This ensures audibility while maximizing the directional SNR advantage.

The system is available in BTE, ITE, and ITC hearing aids. In the BTE style, the two omnidirectional microphones are approximately 10 mm apart, while in the ITE and ITC models they are 5 mm apart. These distances yield excellent SNR improvement, acceptable cosmetics, and minimal circuit noise.[1,14] A functional block diagram of the Locator microphone system is shown in Figure 4.

The Locator has four functional components that control the input to the Diva signal processing unit: the OptiMic System, a Noise Classification System, an Adaptive SNR Optimization unit, and a Beamforming unit.

OptiMic System

The OptiMic System is a patent-pending microphone-matching system that includes two units: an Adaptive Amplitude Matching unit and an Adaptive Phase Matching unit. These units compare the outputs from the two microphones continuously at 32,000 samples per second. They effectively minimize any amplitude and phase differences between the two microphones in 2 to 30 seconds, thus ensuring that the two microphones are always perfectly matched to yield the desired polar characteristics.

The microphone-matching system also means that any pre-existing differences between the two microphones are unimportant. The implication is that any two microphones (of the appropriate type) can be used to form the dual microphone. No microphone matching is necessary at the manufacturing stage. And, if one microphone needs replacement, it can be done without replacing the other. Furthermore, because the unit is adaptive, any possible drifting over time will be corrected immediately to preserve the stated directivity. For wearers, it means that they can expect the same directivity from the hearing aid even after years of use.

Output from the two microphones enters the Beamforming unit, which sets the polar pattern of the directional microphone. In turn, the processing of this unit is controlled by results of the Noise Classification System and the Adaptive SNR Optimization unit.

The Noise Classification System

While a directional microphone may be effective in minimizing environmental sounds that originate from anywhere but the front, it is ineffective in minimizing wind noise and circuit noise. Indeed, one drawback of directional microphones is increased perception of wind noise and circuit noise.

The Noise Classification System uses the statistical properties of the different noise types to identify them and then uses the results to control the beamforming process. For the analysis, it takes inputs from both microphones. If the analysis at the Noise Classification System suggests the presence of speech and/or environmental noise (Env), such information will be used in the Adaptive SNR Optimization and the Beamforming units so that the polar pattern is set according to the azimuths of the speech and noise sources.

If the Noise Classification System analysis suggests that the primary noise source is turbulence from wind passing the hearing aid (Wind), such information will cause the Beamforming unit to set an omnidirectional polar pattern. The change in polar pattern could reduce the level of wind noise by 30 dB or more over the directional mode.[1]

Figure 5 shows the reduction of wind noise accomplished by the Locator microphone when measurements were made in a custom-made wind tunnel at the Widex Audiological Research Laboratory. The top solid line shows the level of the wind noise through the Locator when the Noise Classification System is inactive (i.e., in directional mode). The middle line shows the same output when the Noise Classification System has identified the input as wind noise. Note that a reduction in output of 20 dB to 30 dB is seen. The bottom line shows the same output when wind noise is processed by both the Noise Classification System and the noise-reduction algorithm used in the Diva hearing aid. An additional reduction of 10 dB to 12 dB in the low-frequency output is seen. This feature makes it convenient for wearers to use the hearing aid both inside and outside without needing to switch between microphone modes.

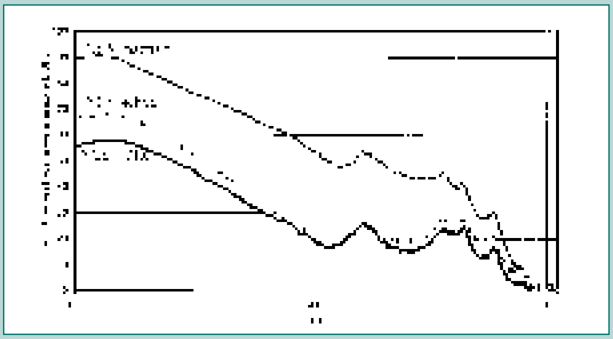

Figure 5. *Magnitude of wind noise measured at different stages of processing. The top solid line is the wind noise measured from the Diva Locator without the activation of the Noise Classification System. The hearing aid is in the directional mode. The second line is the wind noise measured with the activation of the Noise Classification System. The bottom dashed line is the final wind noise output from the hearing aid, showing the combined effects of the Noise Classification System and noise reduction.*

In a quiet environment, the primary noise that the Noise Classification System encounters would be circuit noise from the hearing aid microphone (Mic). This noise type forces the beamformer into an omnidirectional mode. In contrast to other dual-microphone designs in which only one omnidirectional microphone is used in the omnidirectional mode, the Locator uses two omnidirectional microphones in its omnidirectional mode. The advantage of adding inputs from two omnidirectional microphones is that the speech level will be increased by 6 dB even though the noise floor is increased by 3 dB. The net result is improving the signal-to-circuit noise ratio by 3 dB over the single-microphone omnidirectional mode.

This microphone mode, along with the low CT below 20 dB HL, ensures audibility of the softest speech sounds that originate from any direction. As in all Senso hearing aids, microphone noise is managed by the Microphone Noise Reduction algorithm, which uses expansion processing below the CT to minimize microphone noise.

The Adaptive SNR Optimization

To ensure that the polar pattern of the Beamforming unit changes in the intended manner (i.e., to follow the noise as it changes locations or as the nature of the noise source changes), some mechanism to monitor the output of the beamformer is necessary. This monitoring is achieved through the Adaptive SNR Optimization unit.

This unit adaptively regulates the control parameters of the beamformer so that it has the appropriate polar pattern (and least noise) at all times. By continuously monitoring the combined signal coming out of the Beamforming unit, the Adaptive SNR Optimization produces a running set of parameters that control the polar pattern, the amount of attenuation, and the speed of change of the Beamforming unit.

The Beamforming unit

The Beamforming unit changes the polar pattern of the dual microphones based on the control parameters specified by the Adaptive SNR Optimization. In contrast to other adaptive directional systems that show adaptive polar pattern change in the directional mode only (i.e., not from the omnidirectional mode), the Beamforming unit results in polar patterns that include omnidirectional, cardioid, supercardioid, hypercardioid, and bi-directional, plus any intermediate polar patterns depending on the noise locations.

To prevent a possible abrupt change in loudness and the potential loss of binaural and localization cues as the system changes its polar pattern, the patent-pending Beamforming unit maintains the same frequency-gain response for sounds coming from the front, regardless of its polar pattern. This means that the hearing aid has the same frequency-gain characteristic consistent with the fitting target in all polar patterns. It also preserves the phase characteristic when adjusting its directivity.

This preservation of low-frequency amplification and phase relationship is important for the localization of sound sources and possibly for ensuring optimal binaural advantages (e.g., masking level difference, MLD). The time required for the directional microphone to change its polar pattern was empirically determined to be up to 10 seconds in order to avoid abrupt perceptual changes.

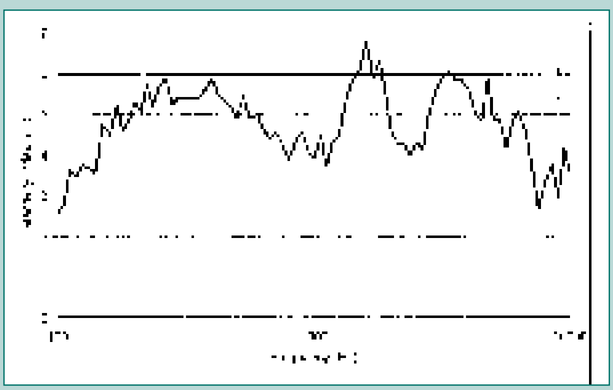

Figure 6. *Directivity index across frequencies measured with a Senso Diva ITC on KEMAR.*

MEASURED *IN-SITU* DIRECTIVITY INDEX

The adaptive Beamforming unit can be especially useful when there is only one noise source. However, in a reverberant room or where there are multiple noise sources (i.e., diffuse field), the effectiveness of the Beamforming unit may be compromised.

An alternative measure of the effectiveness of a directional microphone is its directivity index (DI). The DI reflects the SNR advantage of a directional-mic system, assuming that the signal is in the front and the noise is everywhere but the front. The Diva ITC has a DI of 5.8 dB at 400 Hz and almost 6 dB at 2500 Hz when measured on KEMAR (see Figure 6). This DI is close to the theoretical maximum of 6 dB. The high DI ensures optimal SNR improvement even in diffuse listening environments. In addition, the adaptive nature of the microphone system means that a similar DI may be expected when different people wear the Diva Locator.

To further ascertain that the directional advantage of the Locator can be realized in an ITC style hearing aid, we performed a series of parametric studies on KEMAR and live subjects in which the depth of insertion of the ITC hearing aid and the deviation of the microphones from the horizontal and sagittal planes were manipulated. Our results indicate that optimum directional properties reported in the previous section can be maintained even when the surface of the hearing aid is inserted 3 mm beyond the edge of the tragus and the angle of deviation is within ±20°. Compromises in the reported directivity may be experienced in cases that exceed these tolerance limits.

CONCLUSIONS

The use of new adaptive digital signal processing techniques allows one to take fuller advantage of dual microphones in meeting the needs of hearing aid wearers. As a result, many of the limitations of traditional directional microphones are overcome. Improvements can be summarized in the following areas:

(1) Improvement in signal-to-noise ratio seen in more situations:
- ❖ Wind noise reduction
- ❖ Circuit noise reduction by 3 dB
- ❖ Frequency-independent polar patterns
- ❖ Adaptive polar patterns for single noise source—stationary or moving
- ❖ High *in-situ* DI of 5 dB to 6 dB across frequencies for diffuse sound fields

(2) Ensuring audibility
- ❖ Low compression threshold in all 15 channels
- ❖ Automatic switching to omnidirectional mode when in quiet

(3) Maintaining consistency of performance
- ❖ Automatic, continuous microphone amplitude and phase matching

(4) Minimizing perceptual artifacts
- ❖ Smooth transition into different polar patterns
- ❖ Low-frequency compensation in all polar patterns
- ❖ Maintain loudness cues
- ❖ Maintain localization cues
- ❖ Maintain binaural cues

(5) Ease of use

- ❖ Smooth, automatic adjustment of polar pattern according to noise condition. HJ

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