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Improving on the Speech-in-Noise Problem with wireless array technology

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In this paper, a novel FM system, the Lexis, is presented. It combines the virtues of a traditional FM system with unsurpassed directionality, brought about by an array of directional microphones and digital signal processing. With the recent advances in technology, the Lexis is able to provide directionalty with fully maintained low-frequency audibility, in a handy and versatile package. Thus, it allows adults and children to overcome the challenges they meet in their everyday noisy environments by providing consistent access to speech.

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Speech recognition in noisy environments

People with hearing impairments have difficulty hearing in noise, in reverberant conditions, and at a distance. Our daily environments are frequently filled with noise and reverberation and we are often required to listen from a distance. People need to be provided with conditions in which speech can be easily perceived and people with hearing impairments can enjoy the lifestyle they choose. If access is not achieved, people with hearing impairments experience decreased social participation, difficulties with employment, psychological effects, and compromised physical health status which lead to a decreased quality of life.

These obstacles are encountered in our everyday listening environments. Speech understanding is affected by the masking effect of background noise. As background noise increases this problem gets worse. Distance also decreases the ability to understand speech, as the level of the signal drops by 6 dB with every doubling of the distance. Distance and reverberation rarely occur in isolation in our environments. The combined effect of distance and reverberation compound the problem; as distance increases the ratio of direct sound to reverberant sound decreases. Reverberation smears the internal energy and the signal masks itself. Research has demonstrated that even small amounts of reverberation significantly decrease perception of speech with increased reverberation time (Nabelek et al., 1981).

People with hearing loss cope with the challenges of noise, distance and reverberation they face either by withdrawing from social interaction, manipulating their environment, or by maximising their residual audition through using hearing aids and assistive listening devices. Communication situations can be manipulated by using clarification strategies or by changing acoustic environments. While this coping strategy can be effective, it can also be seen as intrusive and requires the person with a hearing impairment to be assertive. It is common for many patients to be unwilling to be assertive. In this regard, the use of hearing aids and assistive listening devices, such as Frequency Modulated (FM) systems, is an appealing, less intrusive alternative.

Hearing aids and assistive listening devices

Basically, there are only two ways to improve speech understanding: optimise audibility, and improve the signal to noise ratio (SNR) for speech. Note that benefit from an improved SNR is completely dependent on audibility being achieved. The technologies that people use to obtain these improvements include hearing aids (for audibility), and hearing aids with directional microphones and assistive listening devices (for improved SNR). With respect to SNR improvement it is worth noting that noise reduction algorithms aimed at improving the SNR in hearing aids are largely unsuccessful (Schum, 2003a; Schum, 2003b). In contrast, directional microphones are effective in suppressing noise and focusing on the speaker in front. Likewise, a transmission technology like an FM system is very effective in overcoming noise, distance, and reverberation. The speech signal is picked up by the FM system while it is not contaminated by reverberation and background noise.

It has been demonstrated that directional microphones increase speech perception performance in noisy listening conditions. Thus, improvements in SNR of 4-5 dB, as compared to omni-directional microphones, have been found by e.g. Ricketts et al. (2001), Valente (2000), and Amlani (2001). Recent research (Walden et al. 2003) indicates that directional microphones are beneficial when the signal source is in front and near and the noise source is spatially separated from the signal source. Users of directional microphones need to be counselled to know in which situations to use this technology to achieve maximum benefit. The situations where there does not seem to be a preference for the use of a directional microphone are when the signal is at a distance and there is reverberation present.

Every day, adults and children find themselves in acoustic environments where the signal is at a distance and there is reverberation present. The use of personal FM systems increases speech perception in such listening conditions – improvements of 20-25 dB SNR have been demonstrated (Hawkins, 1984; Smaldino & Crandell, 2000).

The use of FM systems enhances the quality of life for individuals with hearing impairment. Users are able to access the speech signal, participate and engage in conversations, and experience less fatigue. People have reported increased quality of life with the use of FM systems (Erber & Osborn, 1994; Jerger et al., 1996) as well as many opportunities where the use of an FM system would be advantageous in gaining access to otherwise inaccessible sound (Ross & Yuzon, 1994). However, FM systems can be intrusive, bulky, and not flexible with the user's listening lifestyle. A solution to this challenge is a handheld transmitter that is flexible and easy to use combined with a small ear level receiver. These handheld transmitter solutions are effective in increasing speech understanding in people with hearing impairments (De Laat & Bonnet, PF 28).

The first generation ear level receiver technology is small in size, but does not allow for the features of the body worn receivers. The ability to change the gain at the receiver level is crucial to achieving maximum benefit and comfort from an FM system. A gain trimmer allows the FM system to be adjusted for an optimal match to each individual's hearing aid and general user situation (eg. handheld or boom microphone).

With the recent advances in digital signal processing, the use of directional microphones in an array has been investigated. Using an array to process sound has many advantages including increased SNR due to improved directionality compared to conventional directional microphones. Soede (1993) demonstrated a 7 dB improvement in SNR with a 5 microphone array over an omni-directional microphone on a Behind The Ear (BTE) hearing aid; while a 4-5 dB improvement in SNR was shown over a directional microphone on a BTE. Saunders and Kates (1997) have also demonstrated this benefit in 'real life' situations. These situations included an office room and a conference room. These rooms were both highly reverberant with direct-to-reverberant ratio for the speech stimuli of about -6 dB and -10.5 dB, respectively.

Introduction of Lexis

The idea of an FM system with a highly directional handheld transmitter is promising (Saunders & Kates, 1997), but until now products have not taken full advantage of this promise.

Oticon responded to the need of an attractive, ergonomic handheld transmitter that utilises the advanced technology that achieves high directionality without the negative side effects of directional microphones as used in hearing aids (low-frequency roll-off and a high level of internal noise) while maintaining low-frequency response. The Lexis handheld transmitter is flexible and is able to fit into a listener's lifestyle. It can be held in the hand and pointed at what the listener wants to hear, placed on a table for group gatherings, worn around the neck of a speaker in a Lavaliere style, or used in conjunction with a direct audio input cable or boom microphone. The DAI/auxiliary cable can be plugged into a television, computer, or CD player. The ear level receiver is also flexible and fits well into everyone's lifestyle because it is lightweight. It can be rotated by the hearing care professional to adjust to the user's hearing aid to obtain the best reception. Finally, the ear level receiver features a gain trimmer, which enables the FM system to be adjusted with respect to gain to provide each user maximum benefit.

The amount of directionality can be chosen with the Lexis handheld transmitter. There are two directional modes (Focus and Superfocus) realised with 4 directional microphones and digital signal processing. In addition to the array, there is one separate omni-directional microphone for the omni-directional mode. The next section will go into the technical aspects behind the directionality of Lexis.

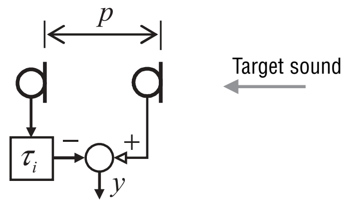
Technical description of Lexis directionality

Before the description of Lexis directionality, the baseline technology of conventional directional microphones will be briefly reviewed.

Starting point – the traditional directional microphone

Directional microphones have been used in hearing aids since the 1970'ties. Basically, a directional microphone can be either a single microphone with two acoustic ports or an electronic directional microphone composed from two omni-directional microphones, an electrical delay circuit, and a subtraction circuit. In any case, both can be described by the block diagram in Figure 1.

Figure 1. Block diagram of a standard first-order directional microphone composed from two omni-directional microphones, a time delay and a subtraction circuit



As seen in this figure, there are two parameters to look at: the port spacing p (the space between two inlets for sound) and the time delay τ_i (the delay applied to the rear signal). Depending on the choice of τ_i, one of the three well known characteristic polar patterns is obtained. Examples are cardioid (see Figure 2 (left)), hypercardioid, and dipole (figure-of-8). This polar pattern is valid at low frequencies, where the port spacing is small compared to the wave-length of sound. At higher frequencies where the wave-length becomes comparable or similar to the port spacing, directionality breaks down, as is seen in Figure 2 (right). Note that the results in Figure 2 are theoretical curves, which assumes a free acoustic field, with no head bias. In practice, when the directional microphone is situated in a hearing aid, which sits on a person's head, the results will be less smooth.

Figure 2. Left: Polar plot of a cardioid microphone (valid at low frequencies). Right: Directivity Index (DI) and AI-DI values of cardioid microphones with port spacings as indicated.

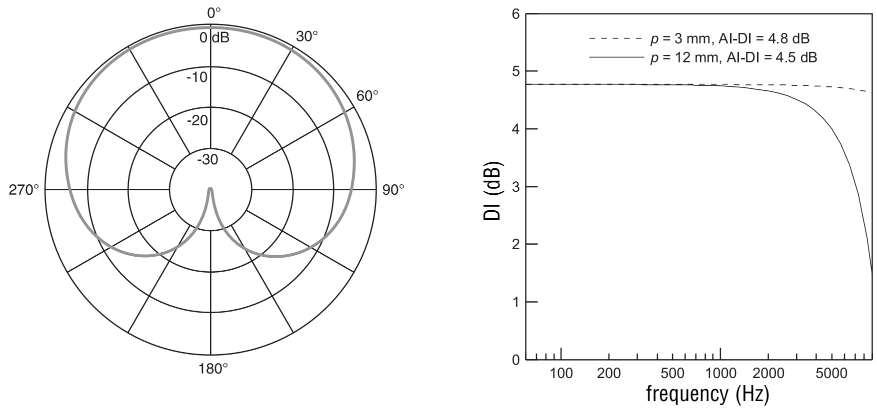
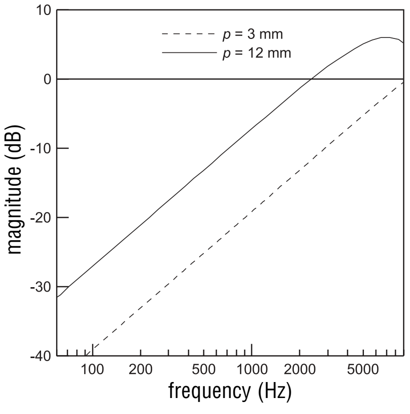


Figure 2 (right) shows the Directivity Index (DI) of cardioid microphones with port spacings of 12 mm (a value typically found in hearing aids) and 3 mm (a typical value for HiFi use). Recall that the DI is a front-to-random index, that is, the DI expresses (in dB and as a function of frequency) the directional microphones's response towards the target direction relative to its average response towards noise, which is assumed to impinge from all directions with equal probability. Note that with smaller port spacing, the directional microphone maintains its directionality higher up in frequency.

A more condensed description of the performance of the directional microphone is the Articulation Index weighted DI (AI-DI). The Articulation Index provides the relative amount of speech information that is carried in the different frequency regions. It is weighted by the importance of each frequency in speech understanding, with focus on 2000 Hz. Thus, the AI-DI is a weighted average of the DI across frequency that accounts for this known distribution of information. Example results are included in Figure 2 (right).

In the discussion of how a directional microphone is evaluated by means of the DI and the AI-DI, the actual frequency response of the microphone has been disregarded entirely. However, the very important down-side of the directional microphone is in fact its inherent low-frequency response roll-off. This is illustrated in Figure 3, which shows the frequency response (assuming sound impinging from the target direction) of the two cardioid microphones considered in Figure 2, relative to the response of one of the component omni-directional microphones, see Figure 1. The directional microphones' waning output at low frequencies is due to the fact that the output is formed as a subtraction of two sound pressures, which are picked up at adjacent points. As frequency goes down, the wavelength increases and the difference in sound pressure at the two points becomes smaller and hence the microphones output also becomes smaller – under the assumption of a constant wave amplitude. This also explains why the output of the directional microphone is less for the smaller of the two port spacings considered in Figure 3; this is because of the less advantageous ratio between port spacing and wavelength.

Figure 3. Magnitude response of cardioid microphones with port spacings as indicated, relative to an omni-directional microphone (0 dB line).



The roll-off of the low frequencies seen in Figure 3 is not a problem in itself, because it may easily be compensated for by an electrical filter with the opposite frequency response as that shown in Figure 3. However, bringing the low-frequency response of the directional microphone up to that of the omni-directional microphone also brings up the directional microphone's internal noise as well as wind noise and thus makes noise in the system louder. This is very problematic and explains why full low-frequency compensation is rarely seen in hearing aids. Note that the problem with low-frequency roll-off, compensation and amplified internal noise is smaller for the larger port spacing, because more gain is required for compensation with the smaller port spacing.

In the above, one of the fundamental compromises regarding directionality is outlined. Thus, the results in Figure 2 suggest a small port spacing so as to obtain good directionality high up in frequency, whereas the results in Figure 3 suggest a large port spacing in order to limit the problems with frequency response compensation and amplification of internal noise.

The above fundamental compromise can be dealt with in several ways by manipulating directionality, internal noise, and frequency response. The most commonly chosen option in directional hearing aids is to

- maintain full directionality, maintain internal noise level and sacrifice low-frequency response.

This is, however, a very poor solution for users of hearing aids with severe and profound hearing impairment, because such people usually are dependent on the information found in the low-frequency part of the speech spectrum, as this is the only area of aidable residual hearing (Hogan & Turner, 1998; Ching et al. 1998). An alternative option is to

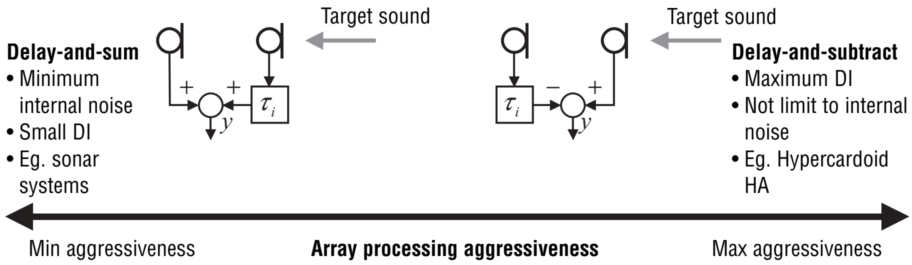
- maintain full directionality, sacrifice low internal noise level and maintain low-frequency response.

This should in principle be a better solution for the users with hearing impairment, but as already noted above it is very rarely used because wind noise and handling noise also are amplified. For completeness it should be noted that intermediate solutions with, say, 50% compensation of low-frequency response are seen in some directional hearing aids. However, there is actually another alternative to the two above strategies for dealing with the fundamental compromise. This may be described as

- sacrifice directionality, maintain (or even reduce) internal noise level and maintain low-frequency response.

In its fundamental form, this alternative is the so-called delay-and-sum beamforming illustrated to the left in Figure 4, together with the traditional directional microphone, which may be characterised as delay-and-subtract.

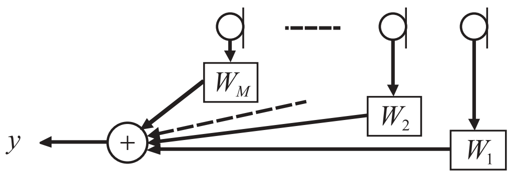
Figure 4. Block diagrams of delay-and-sum (left) and delay-and-subtract (standard first-order directional microphone, right) processing strategies, with key features listed. These strategies are shown in their capacity of being the two extremes of array processing strategies with respect to processing aggressiveness.



Introducing array processing

In reality, the two approaches to directionality that are illustrated in Figure 4 are two extremes that exist at each end of a continuum of array processing strategies. These strategies are characterised by how aggressively directionality is captured at the cost of increased internal noise. In general, array processing involves any number of microphones which are combined through flexible processing filters, as sketched in Figure 5. It is interesting to note that if the number of microphones in Figure 5 is reduced to $M = 2$ and if the processing blocks W_1 and W_2 are chosen appropriately, the two directional schemes from Figure 4 appear as special cases of the array processing shown in Figure 5.

Figure 5. Sketch of a general array structure comprising a number of microphones 1,…, *m*,…, *M*, the signals from which are passed through individual processing blocks W_m before the final summation that forms the array output *y*.



The main asset of the flexible array structure in Figure 5 is the possibility to position the directionality of the array anywhere between the two extremes from Figure 4, with exactly the right compromise between directionality, internal noise and frequency response. It is important to note that in practice, realising this compromise requires a digital implementation of the array processing. In this respect, it is noteworthy that another handheld directional device on the market employs the maximally aggressive delay-and-subtract processing, realised with analog electronics in an array with two directional microphones.

The Lexis array microphone

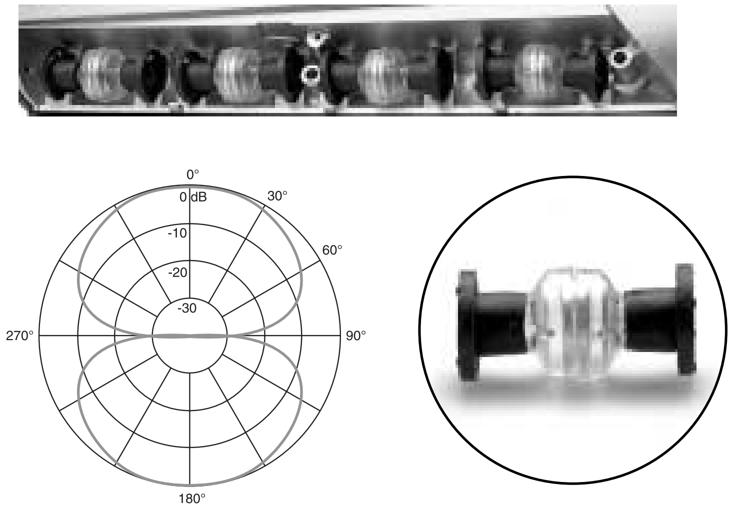
There are many components in microphone arrays that affect their directionality. These factors include microphone configuration, physical array design, as well as the actual array processing. Each factor of the Lexis microphone array has been carefully determined in a global combined optimisation procedure. This means that breaking down the performance of Lexis into a number of separate contributions is impossible, as such. However, in the following the results of the global optimisation will be discussed one by one.

Individual directional microphones

Each of the four microphones in Lexis is a traditional first-order directional microphone of the acoustical two-port type, as discussed above. However, both the spacing between the two sound ports of each microphone and the polar pattern have been determined in conjunction with the other parameters of Lexis, so as to obtain optimal performance.

The port spacing has been set to 15 mm, by adequate design of the microphone suspensions seen in Figure 6. As discussed previously, the port spacing is determined as a compromise between microphone response at low frequencies (suggesting large port spacing) and maintaining the nominal directionality throughout the frequency range of interest (suggesting a small port spacing).

Figure 6. The four directional microphones in Lexis pictured, with close-up of a single microphone and the nominal polar plot.



The optimal polar pattern for a first-order directional microphone on its own in the free field is the hypercardioid pattern. The hypercardioid has a nominal DI of 6.0 dB. However, it has turned out that in combination with array processing the optimal polar pattern is the dipole (also known as bi-directional or figure-of-8), see Figure 6. On its own, the dipole microphone has a nominal DI of 4.8 dB. In broad terms, the combined optimisation of polar pattern and array processing is able to take full advantage of the dipole microphones' outstanding directionality towards the sides, because the array processing provides directionality towards the rear. The combined effect of dipole microphones (directionality towards the sides) and array processing (directionality towards the rear) can be seen in the polar plots shown in Figure 10 below.

Physical array design

The physical design of Lexis involves a number of decisions related to the microphone array.

The most fundamental of these regards the orientation towards the target. Microphone arrays can be designed for any target direction, but the two most common configurations are the so-called 'endfire' and 'broadside' ones. In these, the microphones are arranged on a straight line that either points towards the target ('endfire') or is perpendicular ('broadside') to the target. As is seen in Figure 7, using a handheld device like Lexis to point and listen intuitively suggests the endfire configuration. This turns out to be fortunate, because it has been shown (Stadler & Rabinowitz, 1993) that a given number of microphones is more effectively exploited in an endfire configuration than in broadside.

Figure 7. Lexis in a typical handheld situation.

Another important parameter is the total length of the array. From a purely acoustical point of view, the array should be as long as possible in order to obtain the best possible directionality, particularly in the important low-frequency range. However, practical considerations of handling and cosmetics impose limitations on the total array length – in Lexis the distance from the front to the rear sound port is 75 mm or 3 inches. Note that this is at least 5 times the total 'array length' found in BTE or ITE hearing aids.

Next, the number of microphones to be distributed within the array needs consideration. Ideally, more microphones mean more directionality (Stadler & Rabinowitz, 1993; Dillon, 2001). However, the distance between the microphones needs to be reduced as more microphones are added within the same total array length, see Figure 8.

Figure 8. Illustration of the reduction in microphone distance *d* with adding more microphones, when total array length *L* is fixed.

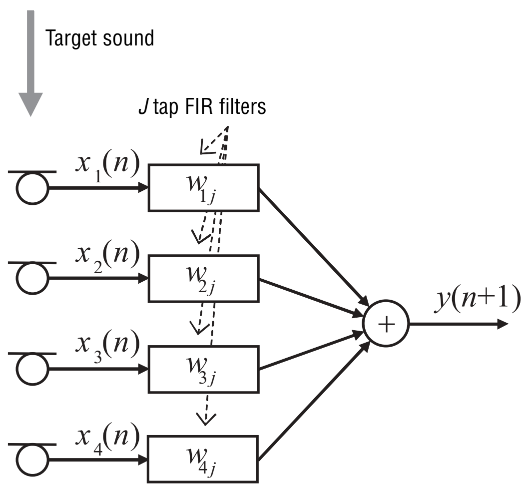
In agreement with the observations previously made with regard to the standard directional microphones, the problem with frequency response and internal noise becomes worse when the distance between each microphone is reduced. This means that there is a limit to how many microphones it is worthwhile adding to the array, when frequency response and internal noise is taken into consideration. In Lexis, with the given total array length of 75 mm, and with the given frequency range of interest (speech frequencies), the optimal choice turned out to be 4 microphones, in the sense that adding more microphones did not lead to an improved AI-DI.

Digital signal processing

Having decided on digital signal processing, there are a couple of fundamental choices that needs to be made.

First up is the choice between fixed-weight and adaptive array processing. Adaptive directionality can perform very impressively when a dominant noise source is present. Further, in a fully automatic hearing aid, adaptive directionality makes good sense because the hearing aid is supposed to react to the environment. However, with a handheld FM device like Lexis, which the user actively points to specific targets, adaptive directionality will be perceived as confusing; the user should be in complete control of Lexis. Hence, fixed-weight array processing has been chosen for Lexis. This allows for the processing to perform predictably, with good performance in a variety of conditions, and ensures that the user of Lexis is in complete control of the device. The fixed-weight array processing is realised as a bank of digital Finite Impulse Response (FIR) filters that act on each of the 4 microphone signals, as illustrated in Figure 9. The term filter-and-sum has been used to describe this processing strategy – as inputs are added after being filtered.

Figure 9. Block diagram of the digital signal processing that realises the array processing in Lexis.



Digital filters can be implemented in two fundamentally different ways; which are known as time-domain or frequency-domain implementations. Both methods are used in different digital hearing aids manufactured today. The frequency-domain implementation offers an advantage in terms of the required amount of internal computations, which translates into longer battery lifetime. However, this is obtained at the expense of increased processing delay, compared to a time-domain implementation. For Lexis, a time-domain implementation with very low processing delay (about 2.3 msec in total) is chosen. This is because Lexis will operate in series with a potentially digital hearing aid, which has its own processing delay, and the sum of them needs to be considered. The extended processing delay may disrupt lip-reading, which is a very important consideration for many of the expected users of Lexis. According to the literature (Dillon, 2001) only delays in excess of 40 msec have actually been shown to be disruptive to lip-reading (McGrath & Summerfield, 1985), but more recent anecdotal evidence indicate that the limit may be as low as 10 msec.

Finally, the digital FIR filters shown in Figure 9 must be designed. The first step is to choose an adequate number of taps for each filter (the filter taps can be thought of as free parameters through which the response of the filter is designed). For Lexis it turns out that $J = 32$ taps for each filter are sufficient to realise the full potential of the present four-microphone array. Thus, there are a total of $4 \cdot 32 = 128$ filter parameters to decide. The design of the array processing filters is – once again – a compromise between obtaining as much directionality as possible, while maintaining low-frequency response and keeping internal noise below a specified limit. However, it turns out that yet another issue needs to be taken into account. If the array processing is designed very aggressively to yield very high directionality (with a high noise limit), the price in terms of internal noise will be dear but on top of that the nominal directionality will easily be destroyed by microphone mismatch. The higher the noise limit, the more vulnerable to microphone mismatch (Stadler & Rabinowitz, 1993). This means that for a given variation in microphone data there will be a limit to how aggressively the array directionality should be designed. Thus, the expected amount of variation in the physical parameters of the microphones used for Lexis has been a determining factor for the chosen level of internal noise.

The result from all these considerations of internal noise and robustness towards microphone mismatch is what may be described as a 0 dB noise limit, which means that the Lexis array will be as noisy as one of its component microphones alone. This yields an input related noise level of 33 dB SPL(A), which is slightly above the noise floor of the FM transmission system. However, what is perhaps more important is that this noise limit makes it possible to guarantee an AI-DI of 8.5 dB in production.

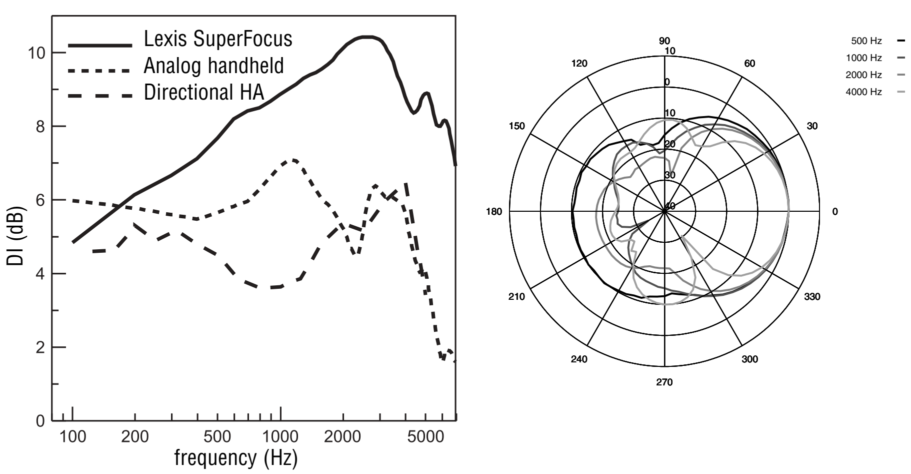
Performance evaluation

The performance of Lexis that results from the above optimisations is visualised in Figure 10, in terms of directionality, and in Figure 11, in terms of frequency response.

The DI curve to the left in Figure 10 shows a dramatic increase in directionality over a directional hearing aid, particularly in the frequency range 500 to 3000 Hz, which is crucial to speech understanding. This is also reflected in Lexis' nominal AI-DI value, which is 8.5 dB compared to the 4.7 dB of the directional hearing aid. (In Lexis' Focus mode the AI-DI is 5.9 dB.) It is interesting to compare this result with that from another handheld directional FM device on the market, which is also included in Figure 10. As discussed in paragraph 4.1.1, that device features a very aggressive processing strategy – with a nominal DI substantially better than that shown in Figure 10. However, because of the aggressiveness of the array processing, the resulting performance is very vulnurable to microphone variations, microphone positioning errors, and the presence of the device's casing. This is presumably the explanation for the sub-standard result shown in Figure 10 (which was obtained as an average across measurements taken on three devices).

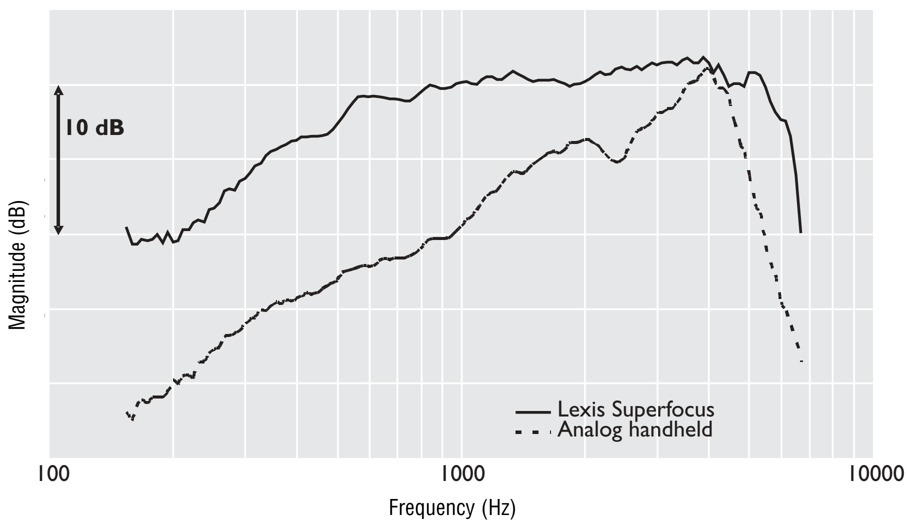
Recall the original polar plot of a single dipole microphone, shown in Figure 6. In that polar plot pronounced directional nulls are observed towards the sides. In the Lexis result, shown to the right in Figure 10, these nulls have been smeared slightly, although Lexis still shows very strong directionality towards the sides. The smearing occurs because of the inevitable deformation of the directional response due to the casing in which the Lexis array is mounted. The suppression of sound to the rear brought about by the array processing improves as frequency goes up, in agreement with the DI curve. Note, however, that in the handheld mode (which is where SuperFocus is expected to be used) the body of the user will provide even further suppression of sound from the rear.

Figure 10. Left: Directivity Index and AI-DI of a typical Lexis, another handheld device and a typical directional hearing aid. Right: corresponding polar plot of Lexis at four frequencies, as indicated.



While the directionality provided by Lexis is outstanding in itself, its usefulness is only fully realised in combination with the broad frequency response of Lexis, shown in Figure 11. As already noted above, directional hearing aids typically roll off the low frequencies, which means that even if an improved SNR has been obtained with directionality, the (low-frequency part of the) resulting signal may be inaudible to the user – particularly if a severe to profound hearing loss with narrow dynamic range is present. In Lexis, the roll-off of the low frequencies is only 6 dB/oct and it does not take effect until below 500 Hz. This means that with Lexis an audible signal with dramatically improved SNR can be presented to the ear of the user throughout the important range of speech frequencies, and where useable residual hearing is present. Once again, the corresponding result for another directional FM device has been included, see Figure 11 (left). As a consequence of the very aggressive analog processing strategy employed in that device, the issues of internal noise and frequency response have been a concern. As seen in Figure 11 this has been addressed in that device by letting low frequencies roll off from 4 kHz and downwards. This obviously has a profound impact on the audibility of the low-frequency parts of the output signal.

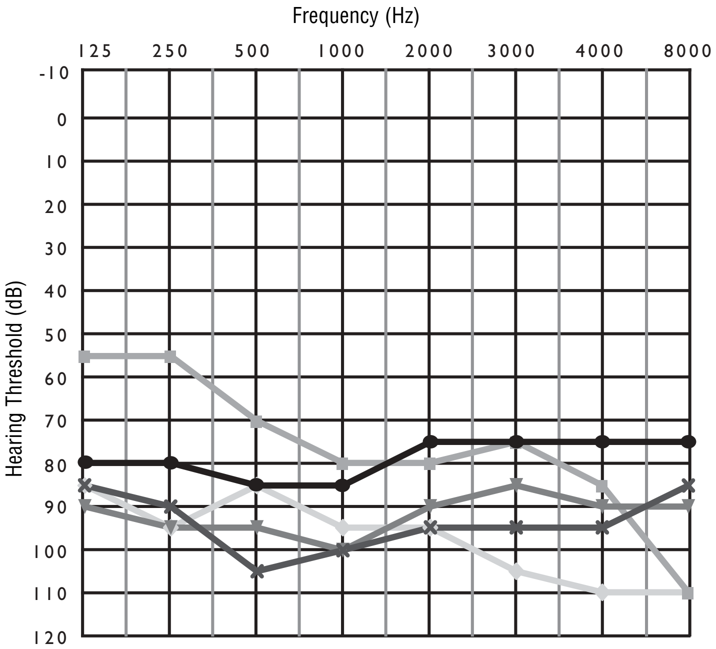
Figure 11. Frequency response of Lexis and an another handheld FM device for sound incident from the target direction. Both transmitting microphones are in their most directional mode.



Clinical evaluation of Lexis

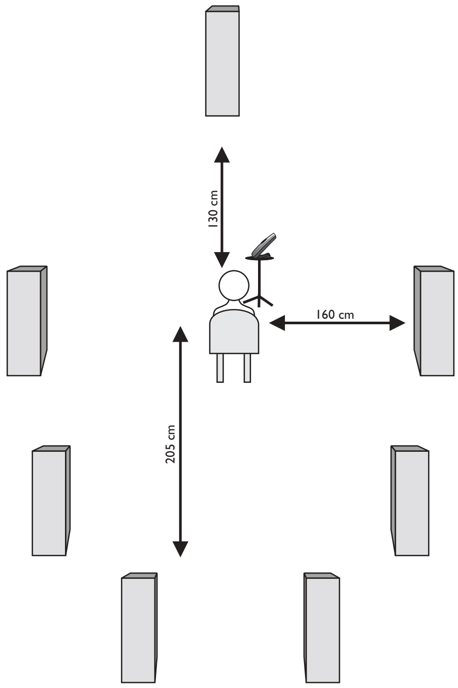
Designing a highly directional FM system based on theoretical analysis and technical measurements alone is naturally not enough. Hence, a clinical evaluation of the Lexis system has been carried out to demonstrate how the Lexis works in clinical and real world environments. Lexis was evaluated during a four week trial. Five active adults (aged 32-53 years) with severe to profound hearing impairment participated in the evaluation, see Figure 12. Following four weeks of use, the users completed speech perception testing and hearing aid versus FM system benefit questionnaires.

Figure 12. Audiograms of the test participants' better ear.



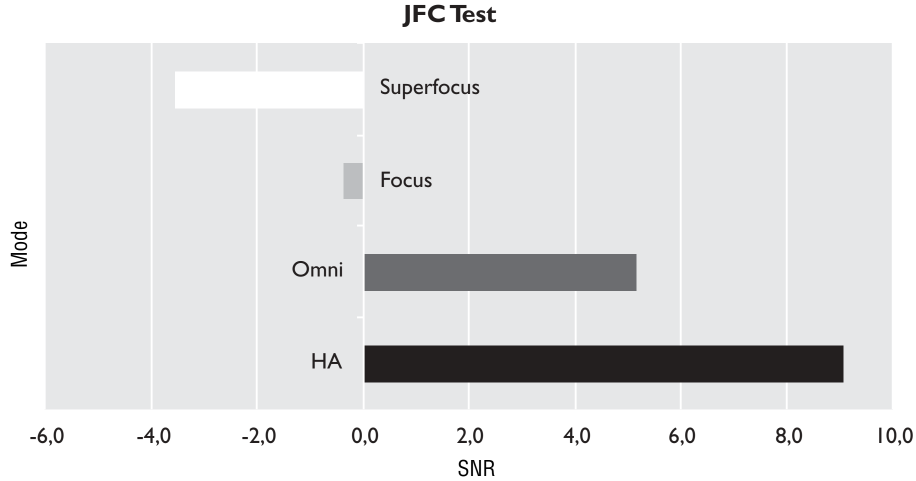
For the speech perception testing, an adaptive discourse speech perception test was used. The user was surrounded by 6 loudspeakers that emitted uncorrelated speech weighted noise. In front of the user, speech was emitted from a single loudspeaker at 65 dB SPL. The user was instructed to adjust the level of the noise until they understood 50% of the information, or so they could 'Just Follow the Conversation' (JFC test) (Hawkins and Stevens, 1950; Larsby and Arlinger, 1993).

Figure 13. Speaker set up for the JFC test. Speech weighted noise was emitted from the 6 loudspeakers surrounding the user and speech was coming from the front speaker. The user was asked to adjust the level of noise until they understood 50% of the information. Speech was fixed at 65 dB SPL.



A significant difference was revealed between the directional modes of Lexis, as compared to their own hearing aid and the omni-directional mode of Lexis. Also, there was a difference between the user's own hearing aid and the omni-directional mode of Lexis. This difference is because the Lexis microphone was closer to the speaker (as it was in handheld position), and because the user's body was blocking some of the noise in the Lexis omni-directional condition.

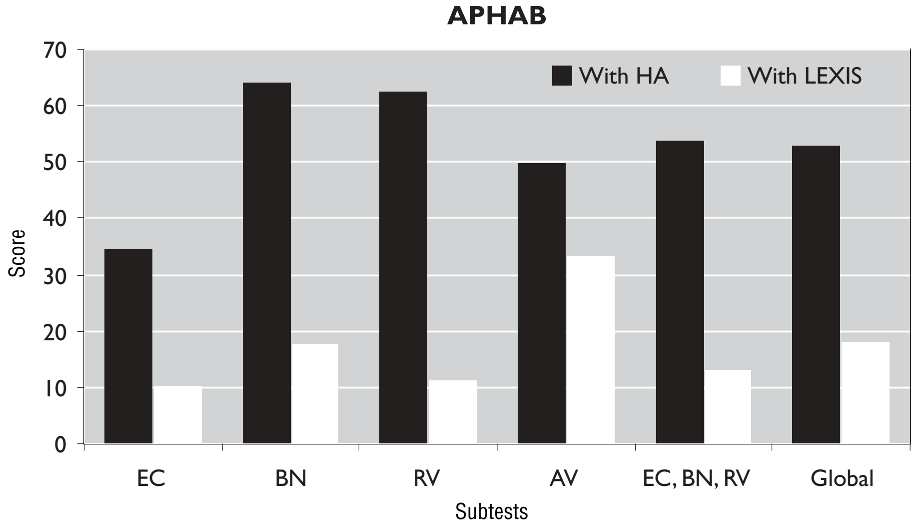
Figure 14. Results from the JFC test.



It is very encouraging that the average improvements in SNR as measured by the JFC test are 8.7 and 5.6 dB for the Superfocus and Focus modes of Lexis, relative to the omni-directional mode of Lexis. These numbers are in near perfect agreement with the technically measured AI-DI values reported above.

The Abbreviated Profile of Hearing Aid Benefit (APHAB, Cox, 2000) was adapted for this study by the users responding to hearing aid only use as compared to FM (Lexis) use.

Figure 15. Results of hearing aid versus FM from the Abbreviated Profile of Hearing Aid Benefit.



The Background Noise, Reverberation, and Ease of Communication subtests and Global test scores demonstrated a positive significant difference (p=0.1) between the hearing aid only and the use of the FM system. The Aversiveness of sounds subtest is the only one that is not statistically significant. The users indicated several situations where the Lexis system was beneficial to them, including: doctor's office, cinema, lecture, dinner party, in traffic, and at a theatre.

Paediatric applications

The first years of life are the critical period for developing children's auditory, speech, and language skills (Sharme et al, 2002). If children with hearing impairment are not fitted with appropriate amplification, they are not able to access spoken language during this crucial learning period. Children need clear, distinct, and consistent exposure to spoken language throughout their day. To achieve this optimal input, children with hearing impairment need amplification that will provide a consistently adequate signal. This can be achieved with the use of hearing aids coupled with FM systems.

In fact, the auditory systems of children with normal hearing do not fully mature until late adolescence. With a developing system, the input signal needs to be clear and accessible, as they are developing their phonemic repertoire and need to hear fine auditory differences (for example, "pa" verses "ba"). An FM system gives a clear signal and improves audibility to children with hearing impairment.

Research has demonstrated that using FM systems in the home is beneficial to children with hearing impairment and their caregivers. In a study by Moeller et al. (1996), children's rate of language development, especially grammatical complexity, matched their chronologically aged peers after using an FM system at home. The use of an FM system reduces the delay in language development that is usually seen in children with hearing impairment. Another benefit of using an FM system, has been an increased feeling of security by children with hearing impairment when their caregivers were not able to be seen.

Caregivers have also reported using an FM system to be beneficial. They preferred using the FM system during tutorial sessions at home, where background noise can be distracting; as well as for listening to the TV or a speaker (e.g. Sunday school) (Moeller et al., 1996).

Lexis can support and benefit children and their caregivers by providing constant and clear exposure of speech throughout the day. The flexibility of the transmitter can be used in the active listening environments that children encounter in their daily lives. The high directionality and preservation of the low frequencies delivers speech clearly to the child's ear, while the gain trimmer on the receiver allows for an individual fit for each child. A secure fit is made with the dedicated receiver as it interlocks with an Oticon hearing aid (SUMO). With Lexis, children will obtain constant exposure to language during the day as there is valuable language input and communication interaction taking place.

Summary

Compared to people with normal hearing, people with hearing impairment have difficulties understanding speech when there is noise and reverberation present, and when the speaker is at a distance. This is the case even when the issue of audibility has been taken care of by means of a well-fitted hearing aid. Hearing aids with directional microphones can be beneficial, but are often not enough. Furthermore, directional microphones are often of very little benefit to people with severe and profound hearing impairment. This is because the frequency response of directional hearing aids rolls off in the low frequencies that are so important for this group, as this is where they generally have residual hearing. As an alternative, an FM system that picks up the speech close to the speaker's mouth and transmits the speech signal directly to the user's ear is very effective. However, such systems are often bulky.

In this paper, a novel FM system, the Lexis, has been presented. The Lexis allows adults and children to access speech consistently throughout their day, as they encounter noisy environments. It is flexible because it accommodates each user's individual listening needs, as well as offering an ear level receiver that gives the ability to individually fit the FM system to maximise residual hearing.

Lexis may be used in small groups (table stand), and in large groups and noisy environments (handheld). The Lexis can also be used for a single speaker by using it in Lavaliere style, on a pocket with the pocket clip, or via boom microphone through the direct audio input. Television, computer, or CD player may also be connected to the direct audio input. Depending on the distribution of speakers and the amount of noise, the Lexis can be used in Omni, Focus or Superfocus mode.

While the Omni mode employs a separate omni-directional microphone, the directionality in the Focus and Superfocus modes is realised by means of 4 directional microphones and digital array processing. The digital array processing is pivotal in obtaining the optimal balance between directionality and its side effects, which are internal noise and low-frequency roll-off. Thus, in Lexis a directionality that amounts to an AI-DI of 8.5 dB has been obtained in conjunction with a low level of internal noise and a fully maintained low-frequency response, which means that the Lexis will be useful also for the severely and profoundly hearing impaired.

The performance of the directional modes in Lexis has been confirmed in a clinical study, with excellent agreement between the technical AI-DI measure and the functional improvement in speech perception. The results from a subjective evaluation were also very encouraging.

In conclusion, it is the combination of a very flexible system with respect to both fitting and ways of use, unsurpassed directionality, and a

fully maintained audio bandwidth towards low frequencies that makes Lexis what it is: The ultimate solution to the problems of understanding speech in noise!

Testimonial

One user said, "One day I was using my Lexis while at lunch with my normal hearing friends. A very large group of people entered the restaurant and sat down behind us. They were very loud. I experienced to my surprise that I could hear more or less everything my friends said, but they had problems hearing in all the noise! My friends were thrilled because they have seen the problems I used to have in similar situations."

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