

The test shows that one of the participants seems to experience ‘normal’ differently from the others or using the scale differently. Disregarding test person 13, a few of the sound stimuli are judged generally as normal (2, horse hooves and 10, hand washing), while all other stimuli have one or sometimes two test persons evaluating them as non-natural. Looking at the test persons three of them are judging all natural or close to normal, while the majority have one or two stimuli that they don’t experience as normal. A low judgment on the normal-scale could of course be due to a bad recording quality or poor choice of sound stimuli, but the non-systematical distribution of judgments low on the scale indicates that this is not the real problem. Basically this test underlines the problem of establishing a common ground of reference. Although the term “normal” often shows up in the fitting situation the reference for this parameter seems to have large individual variations and therefore cannot be used as a shortcut in evaluating sound quality or hearing aid performance.

DISCUSSION

The knowledge of sensory practice along with the experiments described here has indicated that obtaining a vocabulary for sound impressions might be a good idea in hearing aid fitting. It is, however, a time consuming and quite difficult task if it is to be done properly with the tools we know today. The most challenging problem might be to establish the right scale for the evaluation. To use a judgment of normality as a fast obtained reference, is not possible, since this parameter has shown to have considerable individual variances.

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Spatial cue reproduction in modern receiver-in-the-ear hearing instruments

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This study investigates the ability to preserve spatial cues in receiver-in-the-ear (RIE) instruments for six different hearing aid manufacturers. In this particular study, the instruments were fitted bilaterally assuming a symmetric hearing loss profile. In cases where the manufacturer recommended a specific programming option to maximize spatial awareness, this option was chosen. Otherwise, the default mode was applied. S2 and N4 audiograms were used to mimic hearing-loss and testing was performed in an anechoic chamber on a KEMAR head. In order to mimic the peripheral filtering of the auditory system the left and right signals were filtered using a gammatone-filterbank. ILD's were estimated at the output of each band across angles from 0-360 degrees and compared to the corresponding values of the open-ear-response. ITDs were determined by low-pass filtering the left and right input signals and using a cross-correlation technique in order to find their respective time shift. Distortions of ILDs were as large as 10 – 15 dB for certain manufacturers whereas ITD distortions lay between 20-100 μ s.

INTRODUCTION

A key element in hearing and interpreting the acoustic wave field is binaural processing in the brain (Hartmann, 1999). The two signals at the ears contain a multitude of information about the spatial nature of any of the sources in the acoustic wave field. The spatial information is encoded in Interaural-Time-Differences (ITD), Interaural-Level-Difference (ILD), spectral cues and reverberation cues. Binaural processing by the brain, when interpreting the spatially encoded information, results in several positive effects; better speech-perception; direction of arrival (DOA) estimation; depth/distance perception and synergy between the visual and auditory systems (Bronckhorst *et al.*, 1988; Bronckhorst *et al.*, 1989; Hawley *et al.*, 2004). Furthermore, even if DOA is an important aspect of spatial perception, and the most commonly investigated property of spatial hearing, preserving DOA estimation does not automatically give a natural sound impression. A sound field might contain all spatial cues needed for DOA estimation, but still will sound artificial or "inside the head". The field is said to be internalized rather than being externalized (Hartmann *et al.*, 1996). Hearing aid solutions affect the audio signal adaptively and constantly interfere with the integrity of the sound. The end users have been reported to have poorer ability to localize sounds and determine

DOA in the aided situation compared to the unaided (Van den Bogaert *et al.*, 2006). Investigating in depth how hearing aids affect the spatial perception is indeed a very complicated problem. The aim of this paper is to focus on how well the binaural cues; ILD and ITD are reproduced in state-of-the-art receiver-in-the ear (RIE) instruments measured over a wide range of manufacturers.

EXPERIMENTAL SETUP

Procedure

The experiments were carried out in the anechoic chamber in the GN Research lab in Ballerup, Denmark. A KEMAR manikin with standard male ears was used to mimic the test subject. The ear canal was simulated using a 711 coupler. The manikin was placed on a turntable to be able to control the angle of incidence relative to the source which was a KEF Q8S loudspeaker. The speaker was hanging from the ceiling, elevated approximately 2 m from the wire floor of the anechoic chamber. Sound recordings were made directly from the coupler, to simulate the open ear response of KEMAR followed by a set of recordings with the different hearing aids.

Stimuli

As stimuli, a 30 sec long female ISTS sentence was used. The sound pressure level at KEMAR was 70 dB SPL. The duration of the signal was chosen in a way that any adaptive algorithm being active in the hearing-aids would have enough time to converge to a steady state solution. For analysis, the last 10 s of the recorded ISTS sentences were used.

Hearing-aid fitting and device settings

Products from six manufacturers (GN ReSound, Oticon, Phonak, Siemens, Widex and Starkey) were included in the investigation. Each product was tested as a bilateral pair and each was programmed according to the manufacturer's recommended method for the spatial feature described.

Two standard audiograms taken from IEC 60118-15 (Electoacoustics – Hearing aids) were used. See Table 1. The first represented a moderate, steeply sloping high frequency loss (S2) and the second a moderate to severe gradually sloping loss (N4).

Audiogram	250 Hz	500 Hz	1 kHz	2 kHz	3 kHz	4 kHz	6 kHz
S2	20	20	25	55	75	95	95
N4	55	55	55	65	70	75	80

Table 1: Audiograms applied in the study

In all cases the S2 audiogram was tested using both an open dome and an occluded ear canal to facilitate the analysis of the sound processed by the hearing aid without

interference of direct sound. For the N4 audiogram an occluding power type of dome was used in all cases. Audiometric data are included in the table below:

The following list shows the names of the processing schemes in the devices that were activated during measurement. Note that the term "processing scheme" is applied here in a broad sense also including e.g. microphone configurations.

Oticon Agil Pro mini RITE™	Siemens Pure 700™	Phonak Audeo IX™
Spatial Sound 2	True Ear	Real Ear Sound
Spatial Noise Management	Sound Smoothing	Soundflow
Auto Tri mode	Speech and Noise Management	Voice Zoom
Binaural Broadband	Feedback Manager	Sound Recover
Noise Management	Wind Noise Blocker	Echo Block
My Voice	Sound Brilliance	Wind Block
		Digital Feedback Control

Widex Clear 440™	Starkey S series IQ9™	GN ReSound Alera™
Digital Pinna	InVision Directionality	Natural Directionality II
Spatial Sound Tracer	Acoustic Scene Analyzer	Directional Mix set to "low"
Speech Enhancer	Adaptive directional mode in "medium" setting	Digital Feedback Control DFS ultra™
Impulse Noise Features		Noise Tracker II set to "Per Environment"
Digital Feedback Control		
Wind Noise Blocker		
InterEar features		

Table 2: Processing schemes of the hearing-aids applied during measurements

DATA PROCESSING AND SPATIAL CUE CALCULATION

To estimate the interaural-time-difference (ITD) and the interaural-level-difference (ILD) the following approximations and assumptions were made:

- ITD is an effect that is dominant below 1500 Hz. Above this frequency; the timing difference is ambiguous and cannot be used.
- The distortion in ITD when using a hearing aid cannot exceed 4 ms.
- ILD is a broadband effect and is equally important for all frequencies.
- The 711 coupler is only valid as ear canal simulator up to approximately 8000 Hz.

ITD estimation

Extracting the ITD is achieved by locating the maximum of the cross-correlation function between the signals in the left and the right ear respectively. The DC component of the measured signals is removed by applying a linear phase FIR of order 101. The second step is to remove the spectral content above 1500 Hz. Above this frequency, the ITD will be ambiguous and can no longer be used by the brain to decode direction of the sound source (Hartmann *et al.*, 1999). The spectral content was removed by low pass filtering the signals using a linear phase FIR filter of order 501. The output of this filtering is (note, that here we drop the dependence on the type of measurement, as the processing is the same for all experiments)

$$x_{R,L}(\theta, n) = y_{R,L}(\theta, n) \otimes h_{DC}(n) \otimes h_{LP}(n), \quad (\text{Eq. 1})$$

where $h_{DC}(n)$ is the DC removal filter, \otimes denotes digital convolution, $h_{LP}(n)$ is the low pass filter and $y_{R,L}(\theta, n)$ are the signals recorded in the right and left coupler, respectively. The lag is found where the cross-correlation of the left and right response has its maximum value

$$n_\theta = \arg \max_n \sum_{m=M_1}^{M_1+M-1} x_R(\theta, m) x_L(\theta, m+n) \quad n = -N, -N+1, \dots, N \quad (\text{Eq. 2})$$

where M_1 is the sampleindex corresponding to recorded signal length of 20s and M the sampleindex corresponding to a recorded signal length of 30 s. The true ITD will normally generate a sub sample delay and therefore, the above equation will not be accurate enough. Therefore, the samples around the found peak are extracted and a second order polynomial $p(n) = an^2 + bn + c$ is fitted to these samples. The coefficients are found by

$$\{a_\theta, b_\theta, c_\theta\} = \arg \min_{a,b,c} \sum_{n=n_\theta-L}^{n_\theta+L} [r_{xR,xL}(\theta, n) - p(n)]^2 \quad (\text{Eq. 3})$$

and consequently the ITD for the given angle is given by

$$ITD(\theta) = -\frac{b_\theta}{2a_\theta f_s} \quad (\text{Eq. 4})$$

where f_s is the sampling frequency of the system. In the experiments, the sampling frequency was 24414 Hz, the number of samples N calculated in the cross correlation function was 100 which correspond to approximately 4 ms. The number of samples used to interpolate the cross-correlation peak was $L = 5$.

ILD estimation

As above the DC component is removed using the same DC removal filter. The signal is then filtered again by a gammatone-filterbank from 200 to 8000 Hz with an equivalent-rectangular-bandwidth (ERB) (Glasberg *et al.*, 1990) spacing. The reason for only analysing the ILDs up to 8 kHz is because the 711 coupler is only valid as an ear canal simulator below this frequency. The processed signal is given by

$$z_{R,L}(\theta, f, n) = y_{R,L}(\theta, n) \otimes h_{DC}(n) \otimes g(f, n) \quad (\text{Eq. 5})$$

where $g(f, n)$ is a gamma-tone filter with center-frequency f . The ILD is estimated at the output of the filterbank by

$$ILD(\theta, f) = 10 \log_{10} \left(\frac{\sum_{n=M_1}^{M_1+M-1} z_R^2(\theta, f, n)}{\sum_{n=M_1}^{M_1+M-1} z_L^2(\theta, f, n)} \right) \quad (\text{Eq. 6})$$

The ILD estimation is thus identical to those from e.g. Raspaud *et al.* (2010).

MEASUREMENT RESULTS

The measurements were conducted at different times and the experimental setup was used for other experiments in the mean time. Thus, the absolute position of the KEMAR manikin is an uncertainty. Two assumptions are therefore made to make the results for the different hearing aids comparable to the coupler measurements:

1. The ITD should be symmetric around the zero degree direction. An offset in the absolute position of KEMAR will influence the ITD. Before presentation, the mean of the ITD estimates for each of the measurements is subtracted.
2. The ILD in logarithmic scale should be symmetric and the volume can be turned up/down in the different hearing aids independently of each other. Therefore, before presentation the results are scaled by a factor that forces the ILD estimates to be as close to 0 dB for angles 0, 180, 360 degrees as possible.

Results for ITD

The reference measurement was always the open coupler response. The mean (grey bar) as well maximum ITD error relative to the open ear KEMAR measurement are given in the figures 1-3. The six different manufacturers are displayed on the abscissa. The first measurement concerned the S2 standard audiogram with open

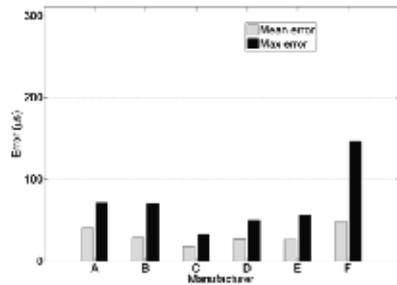


Figure 1: Mean and maximum ITD error for a S2 audiogram with an open dome.

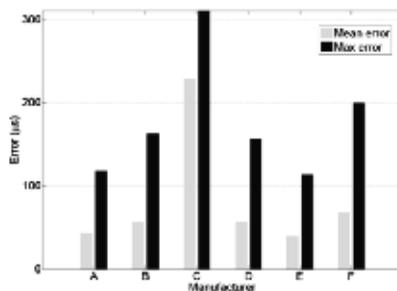


Figure 2: Mean and maximum ITD error for a S2 audiogram with an occluding closed dome.

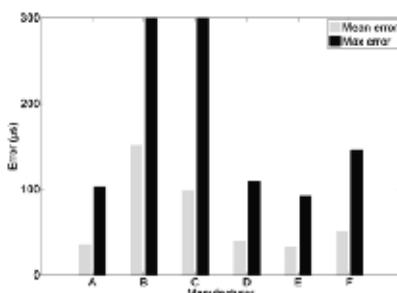


Figure 3: Mean and maximum ITD error for a N4 audiogram with an occluding power dome.

dome. This means that both the direct sound and the sound produced by the hearing instrument will be present in the recording. The estimated mean and maximum ITD errors are given in figure 1. The mean errors are relatively similar across the manufacturers. Manufacturer C performs best in terms of mean error with a relative small variation indicated by the maximum error. The second experiment applied the S2 audiogram but with a simulated occluded ear canal. Here, no dome was used, instead bluetac was wrapped around the receiver and the receiver was placed in the ear canal. The estimated ITDs can be seen in figure 2. The error in the reproduced ITD is in this case is much larger for all manufacturers compared to the corresponding open dome measurement. Note that for manufacturer C the large mean as well as maximum error was due to a large discrepancy at only one angle

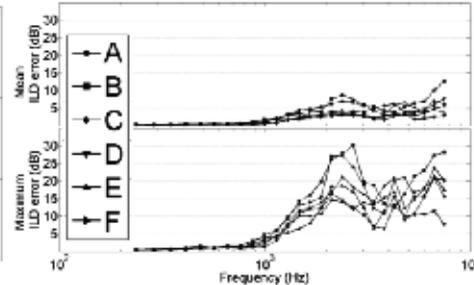


Figure 4: Mean and maximum ILD error for a S2 audiogram with an open dome.

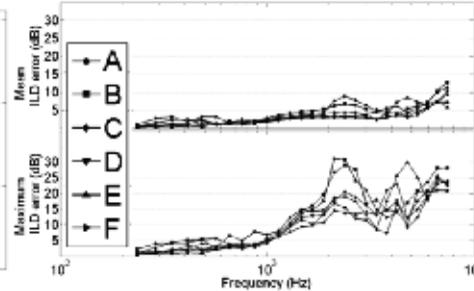


Figure 5: Mean and maximum ILD error for a S2 audiogram with an occluding closed dome.

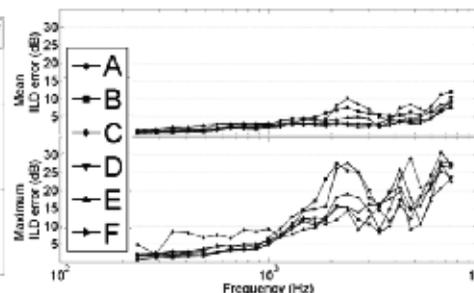


Figure 6: Mean and maximum ILD error for a N4 audiogram with an occluding power dome.

(170°). The last experiment was performed with an N4 audiogram with a closed dome and power receiver (results in figure 3)). Most prominent are the large maximum errors for manufacturer B and C. They were caused by large discrepancies at 130°, 220° and 250°.

Results for ILD

The ILD estimates are given in figures 4-6. When analyzing the results it does not seem as if the estimated ILDs for the open dome with S2 audiogram (figure 4) are improved compared to the closed dome with the S2 audiogram (figure 5) as well as with the occluded ear with the N4 audiogram even though the coupler does record a lot of the direct sound as well. Here, the results also show a large discrepancy between the reference and all manufacturers with small deviances at certain frequencies.

SUMMARY AND CONCLUSIONS

In this paper, the preservation of spatial cues for different hearing aid manufacturers has been investigated. In particular, the ITD and ILD of the reproduced sound from the hearing aids were analyzed and compared to an open ear coupler measurement on a dummy head. All hearing aids were receiver-in-the-ear devices with the housing behind the ear. Two different standard audiograms were tested; one with a mild to moderate hearing loss (S2) suitable for a open dome, and one with a more severe loss (N4) suitable for a power receiver and a closed dome. The milder loss was investigated both with an open dome but also with the ear canal blocked off so that only the sound from the hearing instrument was present at the coupler. Previous work has shown that human subjects are sensitive to a change in ITD of as little as 13 μ s, and for ILD the corresponding value is approximately 0.5 dB (Hartmann, 1999). It was shown that for the closed dome (N4) and the occluded ear canal (S2), the distortion of the ITD was more than the described perceptual threshold. The ITD distortion was smaller for the open dome measurement with S2 audiogram which presumably was due to the high amount of direct sound present in the frequencies below 1.5 kHz. The ILD was also analyzed for the three experimental conditions. It can be concluded all manufacturers displayed large ILD distortions for all measurements. The distortions tended to be smaller for the lower frequencies. The deviations from the reference ILD was for some devices and some angles as large as 5-10 dB. For the N4 audiogram, two manufacturers were having a smaller error than the other but the discrepancy is still above the detectable threshold and would very likely have a perceptual effect. In conclusion, there is still a lot of work to be done on the topic of spatial hearing for receiver in the ear hearing instruments.

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Hearing-Aid Compression: Effects of Channel Bandwidth on Perceived Sound Quality

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Several researchers have investigated the effect of hearing-aid compression (the compression speed and the compression ratio) on speech perception and the sound quality of hearing aids. Some of these experiments have revealed positive effects of fast compression. However, the majority of the experiments have been conducted on simple hearing-aid platforms with only one to four compression channels. Today, high-end hearing aids have significantly more frequency channels. The question is therefore whether the results found with wide channel bandwidths can be extended to narrower channel bandwidths.

To investigate this, 10 normal-hearing subjects were asked to rate perceived sound quality of 111 pre-processed sound recordings differing on the four parameters of compression ratio, compression speed, signal to noise ratio and channel bandwidth. The results of the study showed that increased channel bandwidth is a very important parameter in relation to improving sound quality when compression ratio and compression speed are increased. Therefore, extending positive results of fast compression with wide frequency-channel bandwidths to hearing aids with narrower frequency-channel bandwidths should be done with caution.

INTRODUCTION

Several researchers have investigated the effect of hearing-aid compression (compression speed and compression ratio) on speech perception and the subjectively perceived sound quality of hearing aids (e.g. Gatehouse *et al.*, 2006; Hansen, 2002; Neuman *et al.*, 1998). Even though the results have been inconsistent, the general picture seems to be that slow compression is preferred on subjective sound quality scales. In 2006 Gatehouse *et al.* evaluated the benefits of fast and slow-acting compression, for listening comfort and speech intelligibility. Their study concurred with the general picture, showing that slow-acting compression outperforms fast-acting compression for listening comfort, while the converse is true for speech intelligibility. Besides their own study, the article also includes a literature review of the results of fast and slow acting compression. Examining this review more closely reveals that the majority of experiments done within this area – including their own - use platforms with only one to four compression channels. This is incommensurable with today's high-end hearing aids where significantly more channels are used and it might therefore be problematic to extend the results to