

Degradation of spatial sound by the hearing aid

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It is well known that the hearing aid distorts the spatial cues used to localize sound sources and this has severe consequences for sound localization and for listening in noise. However, it is not clear how the different components in the hearing aid contribute to the degradation of spatial sound. In this study we investigate how the spatial sound is degraded by four hearing aid components: 1) the microphone location, 2) the directionality (beamforming), 3) the compressor, 4) the real ear measurement. Head Related Transfer Functions from an artificial KEMAR head are convolved with appropriate excitation sounds and processed through the respective hearing aid algorithm. The performance metrics under investigation are: 1) interaural level difference (ILD), 2) interaural time difference (ITD), 3) monaural spectral cues. It is found that the main source for ILD degradation is the position of the microphone around the pinna which distorts the ILD by up to 30 dB. It is also found that the real ear measurement compensation severely affects the monaural spectral cues.

INTRODUCTION

It has been known for more than 100 years that the acoustic signals at the ears contain a multitude of information about the spatial nature of any of the sources in the acoustic wave field. This spatial information is encoded in interaural time differences (ITD), interaural level differences (ILD), spectral cues, and reverberation cues (Blauert, 1997). Binaural processing by the brain, when interpreting the spatially encoded information, results in several positive effects: better signal-to-noise ratio (SNR), direction of arrival estimation, depth/distance perception, and synergy between the visual and auditory systems. Therefore, better localization performance will improve sound quality as well as hearing in noise (Hawley *et al.*, 1999).

Even though the benefits of spatial sound are well known, it is not clear how the different components and algorithms of a state-of-the-art hearing aid will distort the spatially encoded information. Previous studies have mainly focused on the localization performance of hearing-impaired test subjects when wearing different types of hearing aids (e.g., Van den Bogaert *et al.* (2006)). Using real hearing aids gives realistic test results but makes it difficult to identify the true sources of any

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degradation of spatially encoded information.

In this study we investigate how four of the main components of a state-of-the-art hearing aid distort the spatially encoded information. The components are: 1) the spatial position(s) of the microphone(s) on the ear, 2) the hearing aid directionality system, 3) the hearing aid compressor system, 4) the real ear measurement (REM) procedure.

A mathematical model of the algorithm package in a state-of-the-art hearing aid will be used to investigate the degradation of spatially encoded information. The input signal to the model will be encoded with head-related transfer functions (HRTFs) based on the corresponding microphone positions measured on a artificial KEMAR (Knowles Electronics Manikin for Acoustic Research) head. The spatially encoded information will be described by ILD, ITD, and HRTF information and any degradation of these cues will be expressed relatively to the open ear response.

SETUP

The HRTFs were recorded on a KEMAR manikin located in an anechoic room in accordance with ISO 3745 and approved for measurements between 30 Hz and 10 kHz. KEMAR was rotated with a B&K 5960 turntable in steps of 2° covering a full 360° rotation and a KEF Q85S speaker was used to transmit a 5 seconds code length 13 maximum-length-sequence (MLS) signal (Golomb and Gong, 2005). The hardware used for the sound recordings and the play back was a Tucker-Davis-Technologies RX8 sound processor running at a sampling frequency of 48828 Hz. All hardware components were controlled from Matlab.

The microphones used for the recordings were placed at 45 different positions on a small female ear and a large male ear respectively. The microphone positions and the ears can be seen in Fig. 1.

The speaker and microphone transfer functions were removed from the HRTFs offline using a standard deconvolution algorithm implemented in Matlab. Furthermore, the open ear responses were recorded with a 711 coupler.

Fixed directionality (FD) beam forming using a hyper cardioid beam pattern was applied on the following microphone positions $\{[1, 11], [11, 21], [21, 24]\}$. The beam forming filters were derived from the microphone data when they were mounted on KEMAR. The directionality beam forming was implemented using finite-impulse-response (FIR) filters with a length of 101 samples (at $f_s=15625$ Hz). The compressor algorithm was implemented using a warp band delay line with compressor knee-point at 50 dB.

REM compensation was applied to ensure that the output of the hearing aid had the same amplitude spectrum as the open ear response (the coupler measurement). The REM compensation filter was implemented as a 1501-taps FIR filter which transformed the open ear impulse response at 0° angle into the corresponding head-

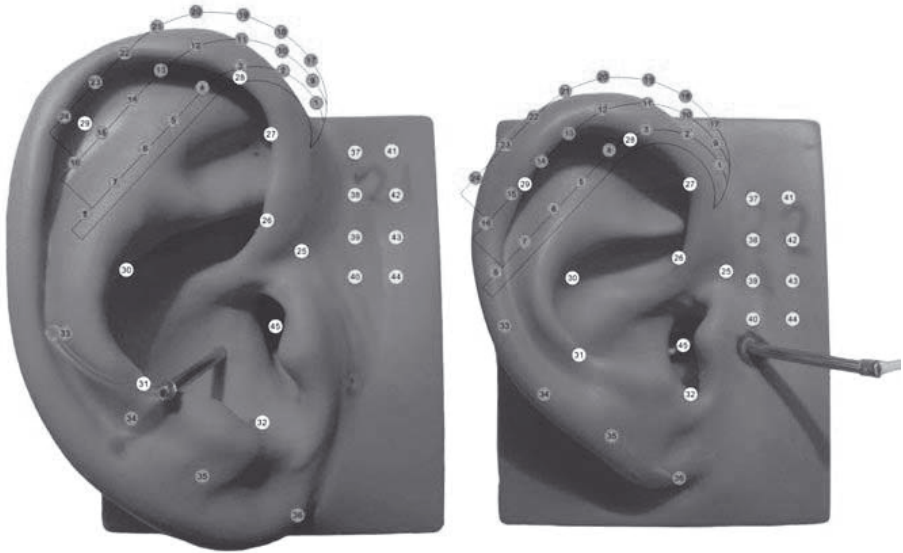


Fig. 1: The two artificial ears used on KEMAR and the 45 microphone positions. The last microphone position is the coupler microphone. All the 'blue' positions are located behind the pinna. Position 45 is located at the entrance of the ear canal.

related-impulse-response measured at 0° in a least-square sense. This filter was applied on all the HRTF data for that given microphone position.

RESULTS

The microphone positions

In Fig. 2 the broad band ILD and ITD (ITD only below 2 kHz) for all 46 microphone positions are shown. It is clear that especially ILD is significantly affected by the change in microphone position, and the corresponding ILD error (deviation between the ILD for a given microphone and the corresponding open ear ILD) is as big as 5-8 dB for angles around 90° and 270° . If the ILD of the open ear response was distorted by 5 dB at these angles, it corresponds to an error on the open ear angle estimate of more than 50° (found from visual inspection of Fig. 2). The maximum ITD for the open ear (coupler) is approximately $750 \mu\text{s}$ which is approximately $100 \mu\text{s}$ less than the maximum ITD values for certain microphone positions. If the open ear ITD is distorted by $100 \mu\text{s}$ this will result in an error of up to 50° (Found from visual inspection of Fig. 2). It should also be noted that the human auditory system is sensitive to ITD changes as small as $13 \mu\text{s}$ (Hartmann, 1999)

In Fig. 3 the frequency dependent ILD for 6 selected microphone positions are shown

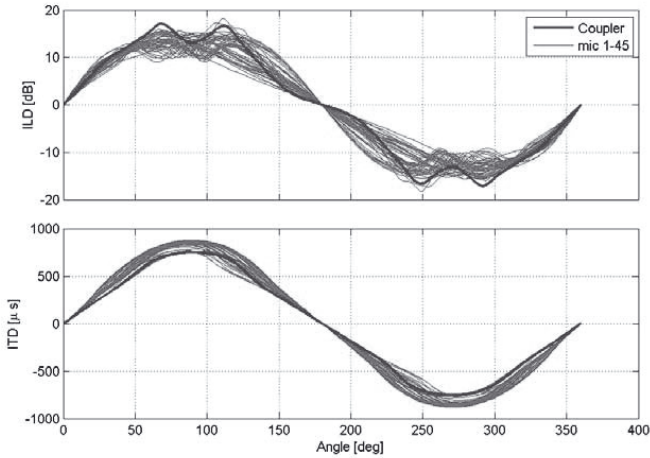


Fig. 2: Top: Broad band ILD (0-10 kHz) for the 45 different microphone positions (red curves) as well as the coupler microphone (thick blue curve). Bottom: The corresponding ITD evaluated between 0-2 kHz using a cross correlation estimator.

where all 6 microphone positions are relevant from a hearing-aid perspective. Based on Fig. 3, it is clear that the ILD error can get much larger at some frequencies than the corresponding broad band ILD error. Especially the microphone positions located behind the pinna have ILD errors of ~ 30 dB at many frequencies. Humans are sensitive to ILD changes of 0.5 dB (Hartmann, 1999) so it is reasonable to assume that an ILD error of 30 dB is noticeable. However, care should be taken here since an increase in ILD at 90° from 40 dB (which is the natural open ear ILD at 5 kHz) to 70 dB will not move the perceived angle of the sound source in space. It is more likely that the listener will experience the sound as internalized. This is often the result when the human auditory system is presented to signals processed through non-personalized HRTFs (Hartmann and Wittenberg, 1996).

Fixed directionality beam forming

Fig. 4 shows the broad band ILD and ITD (ITD only under 2 kHz) as a function of angle for the three different microphone pairs tested for fixed directionality using the large male ear on KEMAR. For comparison also the ILD/ITD for the open ear and for microphone position 1 are shown. The ITD is up to $300 \mu\text{s}$ higher for the fixed directionality signals compared to the open ear ITD at 260° . This corresponds to an ITD error of $100 \cdot 300 \mu\text{s} / 750 \mu\text{s} = 40\%$. Even though this ITD error is large it is not clear what the perceptual consequences are. The maximum error occurs at angles where the sound source is either directly to the left or to the right of the listener. An

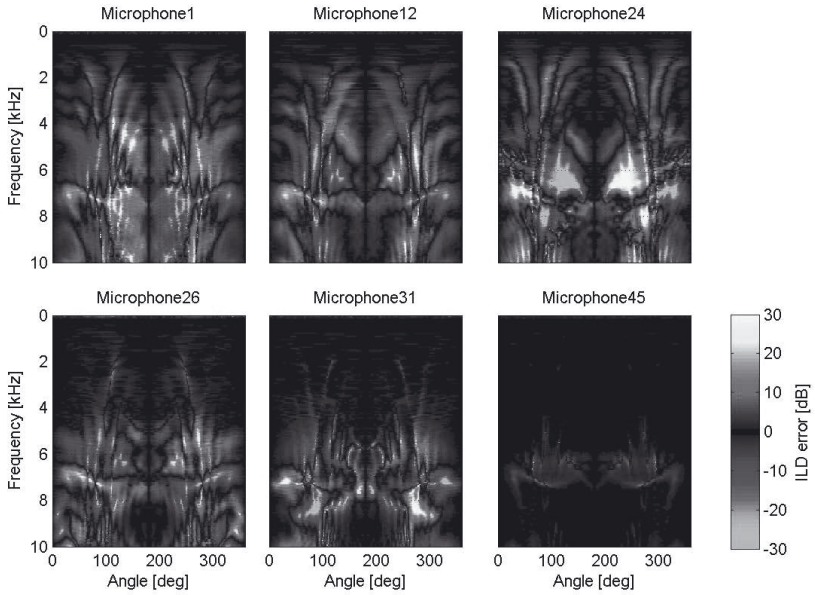


Fig. 3: The ILD error (difference between the open ear ILD and the estimated ILD) for 6 different microphone positions

ITD larger than the natural one at these angles does not move the source location in space, it rather creates a unnatural sound impression.

Fig. 4 also shows that fixed directionality distorts the ILD significantly for certain angles. At 120° , where the null in the beam pattern is positioned, the ILD becomes positive for all three tested microphone positions. This means that high-frequency (above 1.5 kHz) signals coming from the right hemisphere at an angle around 120° will be perceived as coming from the left hemisphere. The worst performance is achieved for the microphone pairs (21, 24) which are located on the ‘back’ of the pinna (pointing backwards). Here the broad band ILD error is 22 dB, which should be compared to the smallest ILD difference of 0.5 dB detectable by humans (Hartmann, 1999).

The compressor

The compressor was tested with four compressor ratios 1, 2, 3, 4, which corresponds to the slope of the input/output gain curve of the hearing aid. The result on broad band ILD and ITD for both male speech and white noise as input signals can be seen in Fig. 5. Here the microphone position 45 was used on the large male ear on KEMAR and SPL was 65 dB (for 0°). The graphs in Fig. 5 show a very clear trend where

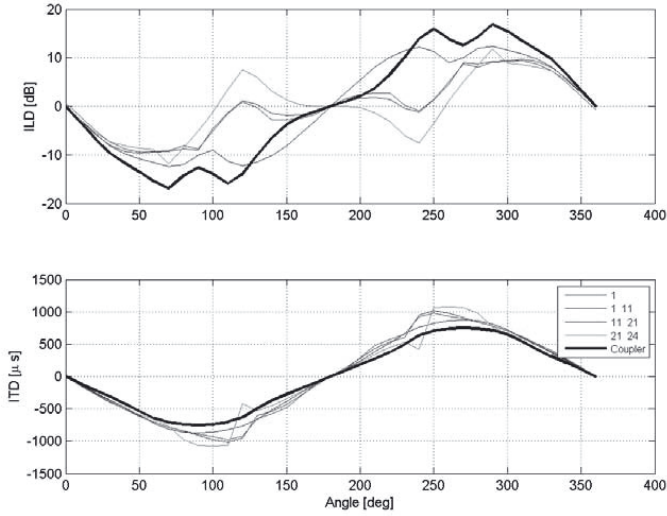


Fig. 4: The ILD and ITD for the fixed directionality beam forming. As reference also the open ear ILD/ITD are shown as well as the ILD/ITD for microphone position 1. Peaks in the ITD plot around 120° is due to the poor signal to noise ratio.

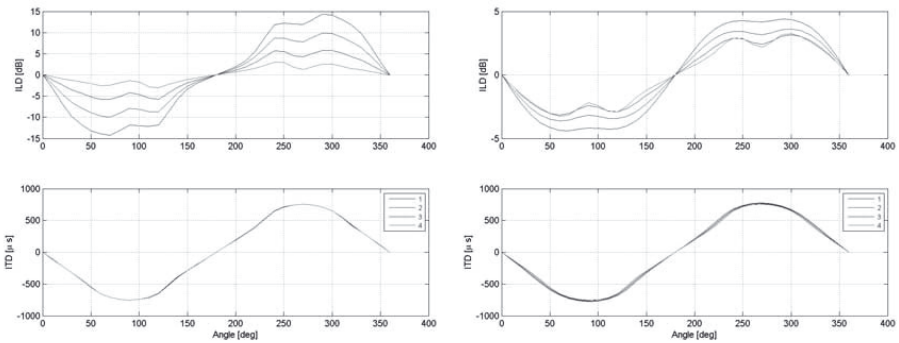


Fig. 5: Left: The broad band ILD and ITD for 4 different compressor ratios when white noise was used as input signal. The microphone at position 45 was used to record the signal on the male ear. Right: same as left but with male voice as input signal.

ILD is decreasing when the compressor ratio is increased. The compressor distortion effect on broad band ILD is up to 12 dB when the input signal is white noise and the uncompressed ILD is 14 dB. The corresponding ILD error is $12 \text{ dB}/14 \text{ dB} \cdot 100 = 85\%$. The ILD distortion is significantly less when male voice is used as input signal. Here the ILD error is less than 2 dB and the relative error on the ILD estimate is no more than $2 \text{ dB}/4 \text{ dB} \cdot 100 = 50\%$. It should also be noted that Fig. 5 shows that ITD does not change when the compression ratio is increased.

The Real-Ear-Measurement compensation

In Fig. 6 The HRTFs for 6 different microphone positions {1, 12, 24, 26, 45, Coupler} are shown where REM compensation is applied except on the Coupler HRTFs. Fig. 6 shows that REM compensation influences monaural spectral cues which are responsible for front-back localization and externalization of the sound image (Hartmann, 1999). Fig. 6 also shows that there is nearly no difference between the HRTFs for microphone 45 located at the entrance to the ear canal and the Coupler microphone. This result proves that the ear-canal transfer function is not a function of angle to the external sound source but can be regarded as a stationary FIR filter. According to the basic laws of physics this holds true as long as the diameter of the ear canal is much smaller than the wave length.

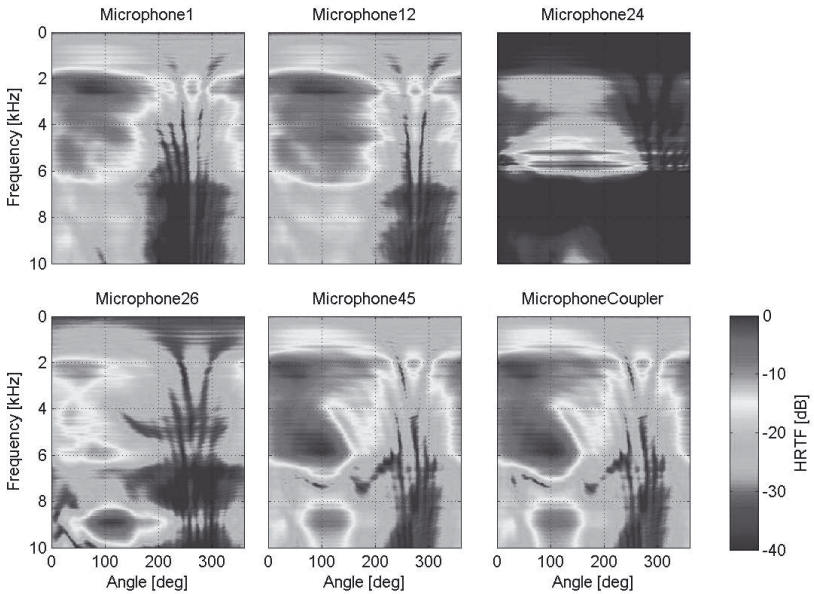


Fig. 6: The HRTFs for microphone positions {1, 12, 24, 26, 45, Coupler} when REM compensation is applied on the large male ear.

When the microphone position moves further away from the entrance to the ear canal the error introduced by applying the REM compensation grows larger. Microphone position 26 shows a significant high-frequency amplification from 90°-180° which most likely will have a huge effect on sound quality. The worst result is obtained with microphone position 24. Here the raw microphone response has a 40-dB narrow dip located around 0° and 5-6 kHz. The REM compensation amplifies this dip by 40 dB at all angles and the result is an HRTF pattern which is significantly different from the open-ear response (except at 0°).

CONCLUSION

It was found that the microphone positions had a significant effect on ILD and ITD. At the entrance to the ear canal the distortion was moderate (less than 10 dB) but behind the pinna microphones introduced ILD errors up to 30 dB at frequencies from 6-8 kHz. Also the ITD error was significant; for some microphone positions it was up to $\sim 100\mu\text{s}$. Fixed directionality introduced significant (~ 20 dB) broad band ILD distortion when sound sources were located around 100°-150°, at other angles the effect was moderate. The compressor had the effect of systematically decreasing ILD. When compression ratio 4 was tested with white noise as input signal the resulting ILD curve had a maximum of 3 dB (compared to 15 dB with no compression). REM compensation did not show any effect on either ILD or ITD but the monaural spectral cues were significantly affected. A positive effect of REM compensation on monaural spectral cues was seen on microphone positions close to the entrance of the ear canal and significant artifacts were introduced when microphones behind the pinna were used.

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