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Amplification of interaural level differences improves sound localization in acoustic simulations of bimodal hearing

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Users of a cochlear implant and contralateral hearing aid are sensitive to interaural level differences (ILDs). However, when using their clinical devices, most of these subjects cannot use ILD cues for localization in the horizontal plane. This is partly due to a lack of high-frequency residual hearing in the acoustically stimulated ear. Using acoustic simulations of a cochlear implant and hearing loss, it is shown that localization performance can be improved by up to 14° rms error relative to 48° rms error for broadband noise by artificially introducing ILD cues in the low frequencies. The algorithm that was used for ILD introduction is described.
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I. INTRODUCTION

For cochlear implant (CI) users with residual hearing in the contralateral ear (bilateral bimodal stimulation), combined electric acoustic stimulation may be a feasible alternative to bilateral cochlear implantation. With clinical devices, it has been demonstrated that combining a CI with a contralateral hearing aid (HA) can improve sound source localization and speech perception performance (Ching *et al.*, 2004, 2007; Dunn *et al.*, 2005; Seeber *et al.*, 2004). While sound localization performance with a single CI improves by adding a contralateral HA, it is still poor compared to normal hearing (NH) listeners. Only a small fraction of the subjects tested in the studies cited above showed performance better than side discrimination.

Francart *et al.* (2008a, 2009) have shown that many users of a bilateral bimodal hearing system are sensitive to the main localization cues with stimuli under direct computer control that are matched in time and frequency. Their just noticeable difference (JND) in interaural level difference (ILD) was in the order of 2 dB and their JNDs in interaural time difference (ITD) were in the order of 100–350 μs. However, current signal processing strategies in clinically available CI speech processors and HAs do not allow the use of these cues by the subject. The ITD cues are weak since (1) the CI speech processor does not encode ITD information in the pulse timing and (2) the CI and HA stimulation does not occur synchronized in time due to different processing delays of the two devices and the additional traveling wave delay at the acoustic side (Francart *et al.*, 2009). The ILD cues are not available because in most subjects residual acoustic hearing is only available at the lower frequencies, and ILD cues are mainly physically present at the high frequencies (Feddersen *et al.*, 1957).

Nevertheless, if sufficiently large ILD cues are available between the low frequencies of the residual acoustic hearing and the broader spectrum of the electric stimulation via the CI, bimodal listeners may improve on localization performance. The current study assesses whether localization performance can improve when ILD cues between the low frequencies of the residual hearing and the broader spectrum of the contralateral electric stimulation are artificially introduced. This is done using acoustic simulations of bimodal hearing, presented to NH listeners. The performance improvement is measured between conditions with and without application of an algorithm that determines the ILD from the full-band signal and introduces it into the low frequencies.

II. METHODS

The steps used to simulate directional hearing using bimodal devices in normal hearing subjects are illustrated in Fig. 1. First, directional hearing was simulated using head related transfer functions (HRTFs) measured with behind the ear (BTE) devices. Second, bimodal hearing was simulated using a noise band vocoder and a low pass filter. The first step is described in Sec. II B and the second step is described in Sec. II C. The signals that were subjected to these transformations contained broadband information and are described in Sec. II A.

Two conditions were evaluated. In the first condition (noamp), stimuli were generated, simulating bimodal hearing (see Sec. II C). In the second condition (amp), a custom ILD

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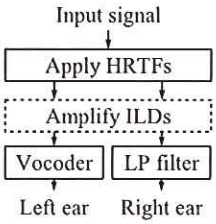


FIG. 1. Signal flow for the simulation of directional hearing for NH subjects.

introduction algorithm was applied to the signal before simulating bimodal hearing. The ILD introduction algorithm is described in Sec. II D.

Six normal hearing subjects were tested (see Sec. II E) using the procedures described in Sec. II F.

A. Stimuli

Two broadband signals were used as input for the processing described in Secs. II B and II C. The first signal (designated as noise14000) was created by filtering a 400 ms long white noise signal with an eighth order low pass Butterworth filter with a cutoff frequency of 14 000 Hz. The second, more realistic, signal (designated as telephone) was a 1000 ms long alerting signal of an old-fashioned telephone. An important feature of this signal is the prominent modulation of about 16 Hz. Its main acoustic energy is situated between 500 and 3000 Hz. Its properties are extensively described by Van den Bogaert *et al.* (2006). A cosine gate of 50 ms was applied to the start and end of both signals.

B. Simulation of directional hearing

To simulate directional hearing, the method of headphone simulation as described by Wightman and Kistler (1989) was applied. Since localization with bimodal hearing is investigated, HRTFs were measured using omnidirectional microphones (type Knowless TM-6436-D46) positioned in BTE HA shells. This BTE configuration is typically used in high power hearing aids and CI speech processors. To avoid the time consuming process of measuring HRTFs for each subject, HRTFs were measured using an artificial head of type Cortex MK2, and the same set of HRTFs was used for each subject. HRTFs were measured for each angle of incidence in an anechoic chamber using the same loudspeaker configuration as in the testing room. The input signal was processed by filtering it with the corresponding HRTFs for each angle of incidence.

C. Simulation of bimodal hearing

To simulate the amount of spectral information that would be perceived by a CI user in optimal circumstances, an eight-channel noise band vocoder (Shannon *et al.*, 1995) was used. It does not provide an exact model for loudness perception using a CI. With proper settings of the CI and HA signal processing, loudness growth between electric and acoustic stimulation is linear (Francart *et al.*, 2008a). Therefore, for the current purpose of demonstrating improvement of localization performance by ILD introduction, it suffices to have equal loudness growth with the simulated electric and acoustic stimuli.

The noise band vocoder mimics the characteristics of a typical CI speech processor by sending the input signal through an eight-channel analysis filter bank, performing envelope detection in each channel and finally adding together bands of noise with different cutoff frequencies, after modulating them with the corresponding envelopes.

Severe hearing loss was simulated in the contralateral ear by the use of a sixth order low pass Butterworth filter with a cutoff frequency of 500 Hz. The signal was calibrated at 65 dBA, such that for our NH subjects the frequencies up to 500 Hz were clearly audible, 1000 Hz was just audible, and higher frequencies were inaudible. This filter simulates the hearing loss of an average bimodal system user from our clinic, as for most of these patients the frequencies below 500 Hz and in some cases up to 1000 Hz can be sufficiently amplified by a HA.

No systematic ITDs could be found in the resulting stimuli using cross correlations on either the full-band stimuli or per critical band.

D. ILD introduction algorithm

An ILD introduction algorithm was developed that measures the ILD on the full-band signals from the microphones of both the HA and CI devices (in the current study microphones mounted in BTE HA shells). It then accordingly increases or decreases the level of the low-frequency signal to be emitted by the HA. If A_{CI} is the root mean square (rms) amplitude of the signal at the microphone of the CI device and A_{HA} the rms amplitude of the signal at the microphone of the HA, then the ILD is defined as $ILD=A_{HA}/A_{CI}$. The ILD is then introduced into the acoustical signal by multiplying it by ILD/2. The idea behind this algorithm is that with bimodal stimulation the head shadow effect is only reduced in the acoustically stimulated ear, and therefore the ILD needs to be increased by half the ILD present in the broadband signal.

Note that if the subject has more residual hearing than in the current simulations, it can be useful to amplify only the low frequencies (e.g., using a shelving filter) instead of amplifying the entire frequency range of the acoustic signal.

The effect of our simulations and the ILD introduction algorithm is illustrated in Fig. 2. The "before sim" lines show the levels of the unprocessed signals at the left and right ears for different angles. Around 0° the ILD (difference in level between both ears) varies approximately linearly with angle while at larger angles the curve flattens. The "vocoder L" and "LP filter R" lines show the levels of the signals at both ears after a simulation of bimodal hearing. The curve for the left (CI) ear remains approximately the same before and after simulation, but the curve for the acoustical signal is severely flattened because of limited ILD cues at low frequencies.

The "LP filter + amp R" line in Fig. 2 shows the same low pass filtered acoustic signal as before, but now the ILD introduction algorithm was applied before the simulation of bimodal hearing. The overall ILD after processing is now as prominent as in the before sim stimuli.

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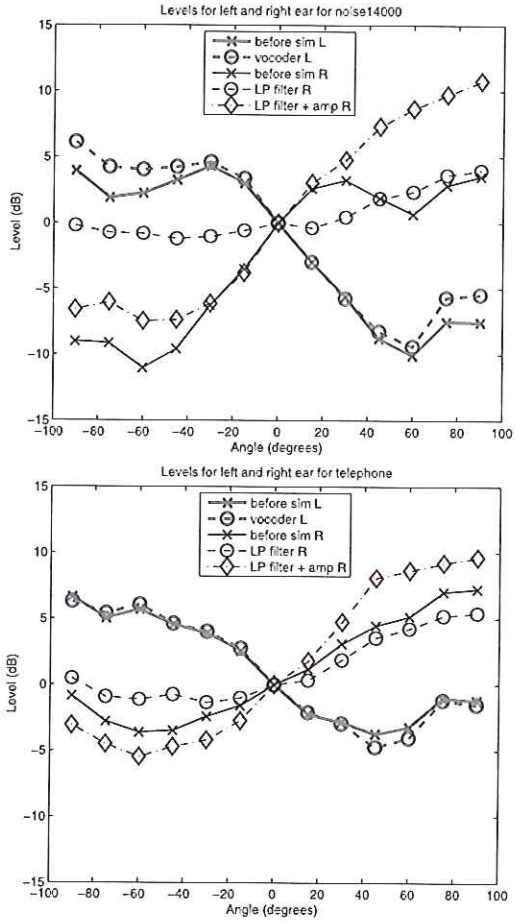


FIG. 2. Levels of the wide band signals (noise14000 and telephone) after filtering with BTE HRTFs, with and without simulation of bimodal hearing, and with and without application of the ILD introduction algorithm. The noise band vocoder simulation (CI) was applied to the left ear signal, and the low pass filtering (HA) was applied to the right ear signal. The ILD at a certain angle can be obtained by subtracting the respective levels in decibels for the left and right ears.

E. Subjects

Six normal hearing subjects aged 21–27 years participated in the current study. Their pure tone thresholds were better than 20 dB hearing level at the octave frequencies between 125 and 8000 Hz. The subjects came to the laboratory for at least two sessions of about 1.5 h, including training and excluding breaks.

F. Procedure

The subject was seated in a chair in the middle of an array of loudspeakers placed at a distance of 1 m from the subject. The chair was adjusted such that the cones of the loudspeakers were at ear height. Identical speakers were positioned at 15° intervals, yielding a total of 13 speakers and spanning 180° in front of the subject. The loudspeakers were labeled with numbers 1–13 and served only as visual cues. Stimulation was delivered using headphones of type Sennheiser HD650 connected to an RME Hammerfall DSP sound card. The experiment was controlled by the APEX 3 program developed at ExpORL (Francart *et al.*, 2008b).

Localization performance was assessed with and without introduction of ILD cues into the low pass acoustic signal using the ILD introduction algorithm described in Sec. II D. A single run consisted of three presentations of a stimulus from each angle, resulting in a total of 39 presentations per run. The chance level for the rms error in this setup is 79°. For each stimulus presentation a random angle was selected. Stimulus levels were roved trial by trial by ±6 dB to reduce monaural level cues.

As the ILD introduction algorithm introduces artificial ILD cues, subjects needed some training before being able to associate the ILD cues with the correct angles. Therefore, for each combination of stimulus and condition, some training runs were performed. A training run was the same as a normal run, but after the subject's response, feedback was shown on the screen. At least three training runs were done for each stimulus/condition before performing a normal run. After this training period, three or four normal runs were done for each subject, stimulus, and condition. Before each normal run, a training run was performed. Only the normal runs were included in the reported results. If T is a training run and N a normal run, an example test sequence for a certain stimulus/condition could be TTTNTNTN.

Calibration was done separately for each stimulus type using the stimulus from angle of 0°. The left and right outputs were calibrated separately using an artificial ear such that the level for the left and right headphone channels was 65 dBA. This resulted in a stimulus that was approximately balanced in loudness at 0°. This roughly reflects a CI and HA fitting strategy in which both devices are balanced for a sound source in front of the subject.

III. RESULTS

The results of the localization experiments are shown in Fig. 3. In the noamp condition the rms errors were 48° and 41° for, respectively, the noise14000 and telephone signals. In the amp condition the rms errors were 34° and 39°, respectively. The rms errors for each subject and each normal test run were used in a repeated measures analysis of variance (ANOVA) with main factors signal (noise14000 and telephone), condition (amp and noamp), and repetition (test run number). No main effect of or interactions with the factor repetition were found, indicating the absence of learning effects during normal test runs. Since the ANOVA showed a significant interaction between condition and signal $[F(1,67)=17.49, p<0.01]$, separate ANOVAs were performed per stimulus type with factors condition and repetition. As expected, neither of these ANOVAs showed significant main effects of or interactions with the factor repetition. For the telephone signal, no significant difference was found between the amp and noamp conditions. For the noise14000 signal, the rms error improved significantly by 14° rms error when amplifying ILDs $[F(1,32)=46.92, p<0.01]$.

The bottom panel of Fig. 3 shows how the total rms error is generated by the combination of bias and variability. In the top row, the standard deviation of the response per stimulus angle of incidence is shown. It is an indication of the variability of the subject's responses, excluding the bias.

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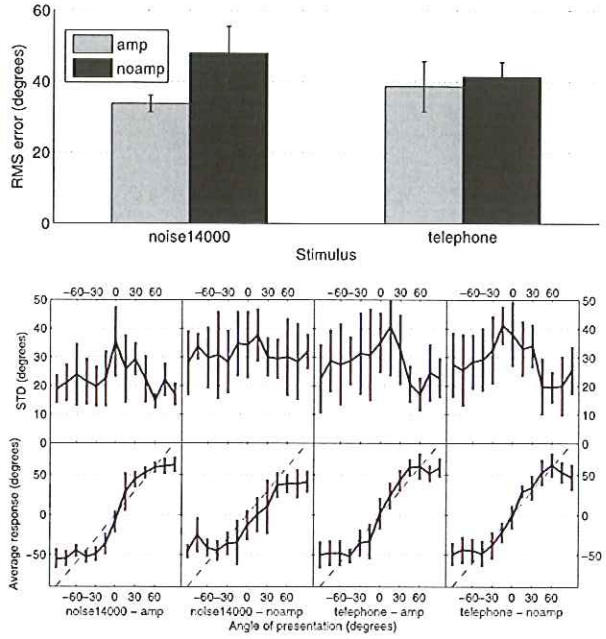


FIG. 3. Results of the localization experiments averaged per signal and condition. amp is the condition with application of the ILD introduction algorithm and noamp is the condition without ILD introduction. The error bars are between subjects' standard deviations. The top panel shows the global rms localization error. The bottom panel shows the standard deviation and the average response per presented angle.

The bottom row shows the average response. It is an indication of the local bias and the extent of laterality. The main observation from this panel is that the decrease in rms error for the noise14000 signal after ILD amplification is caused by both a decrease in bias and a decrease in variability. The bias is largest for stimuli at either side of the head.

Visual inspection of the results of the training runs shows an improvement in performance over the first few training runs of the first condition that was tested. No sufficient data were available to assess differences in learning across conditions.

The global bias was very small for each condition and stimulus (average $-0.7°$). This is likely due to the dBA calibration of the stimulus at 0° and the subjects' training. When considering change in bias separately for stimuli either presented from the left or right hand side, there is a slight, non-significant tendency of decrease with increasing run number.

IV. DISCUSSION

For both the white noise and telephone signals, the conditions without ILD introduction (noamp) yield worse performance than for NH listeners in the free field and virtual free field conditions reported by Van den Bogaert *et al.* (2006) and Francart (2008). This can be explained by the severe signal degradation caused by the simulation of bimodal hearing. The processed left and right signals are largely uncorrelated, yielding a diffuse sound image, and do not contain consistent ITDs, disabling the use of ITDs for localization. Moreover, the reduction in spectral detail by the noise band vocoder and the bandwidth restriction by the low pass filter restrict the use of ILDs, such that only limited spectral cues and very small ILD cues remain, yielding low localization performance. While rms error values cannot be compared directly, qualitatively our results are comparable to those for bimodal listeners found in the literature (Ching *et al.*, 2001; Dunn *et al.*, 2005; Seeber *et al.*, 2004), which confirms the validity of the used simulations.

The use of ILD amplification improved localization performance significantly for the noise14000 stimulus. While level roving was done over a range of 12 dB, it is still possible that monaural level cues were used to some extent. However, considering the white noise signal, in the noamp condition the maximal monaural level difference in the CI simulation (left ear) was 15 dB. In the amp condition, the maximal monaural level difference was 18 dB (see Fig. 2, right ear). Even under the assumption that the listener focuses on the ear with the largest range of level fluctuations, it seems unlikely that this 3 dB difference would have been the cause of the 14° performance improvement of the amp condition over the noamp condition.

One of the reviewers pointed out that the increased monotonicity of either monaural or binaural cues in the amp condition could have been the cause of the performance improvement. However, if the performance improvement for the noise14000 signal would have been based on monaural cue monotonicity, one would expect a larger improvement in performance after ILD amplification for stimuli from the right hand side than for the left hand side. From an analysis of error patterns per angle of presentation (not shown), it seems that this was not the case. We therefore assume that mainly binaural cues were used for localization in the current study.

The lack of a significant effect of ILD introduction for the telephone signal is probably due to the following reasons. (1) The difference in ILD between the amp and noamp conditions is smaller for the telephone signal than for the noise14000 signal (see Fig. 2). (2) Monotonicity of the ILD cues in the noamp condition is better for the telephone signal than for the noise14000 signal and does not improve with ILD introduction, while monotonicity for the noise14000 signal does improve with ILD introduction. (3) Signals with clear modulations merge better between the ears (Francart *et al.*, 2009), increasing the performance in the condition without ILD introduction [repeated measures ANOVA with factors stimulus and repetition, $F(1,36)=9.83, p<0.01$] and therefore reducing the potential improvement. However, better merging for the telephone signal is not evident in our data and therefore warrants further research.

The current simple ILD introduction algorithm suffices for showing feasibility of localizing sounds using artificial ILD cues. For use in practice, however, it should be further optimized. There are two main steps in the optimization. The first is to optimize the amplification function. The optimal ILD versus angle function could be a natural ILD versus angle function or a modified version without ambiguities and with enlarged ILDs. Enlarging ILD cues may further improve localization performance, which is necessary because ITD cues are largely unavailable, both for bilateral CI users

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and for users of a bilateral bimodal hearing system. A second step in the optimization of the algorithm is to use a signal processing system to determine the location of the most prominent sound source and then use the optimal ILD versus angle function to introduce an unambiguous and sufficiently large ILD into the signal. This would make the algorithm also useful for stimuli containing only low frequencies, which do not contain sufficient ILDs to be amplified.

V. CONCLUSIONS

We investigated the effect of a practical ILD introduction algorithm in NH subjects. When introducing ILDs determined on the broadband signal into the low-frequency signal, there was no significant effect on localization performance for a telephone alerting signal, but for broadband noise, performance improved by 14° rms error relative to 48° rms error. While the usability of a similar algorithm still needs to be tested with bimodal listeners and while aspects of combined fitting of both devices are still to be considered, the current results demonstrate that it is perceptually feasible to use ILD introduced into the low frequencies.

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