# FP6–004171 HEARCOM

# Hearing in the Communication Society

**INTEGRATED PROJECT**

**Information Society Technologies**

# D-7-3: Specification spatial-hearing test

| | |
|---|---|
| Contractual Date of Delivery: | 31 August 2006 (+45 days) |
| Actual Date of Submission: | 12 October 2006 |
| Editor: | Koen Eneman |
| Sub-Project/Work-Package: | SP3-WP7 |
| Version: | 2.0 |
| Total number of pages: | 44 |

| Dissemination Level | | |
|---|---|---|
| PU | Public | X |
| PP | Restricted to other programme participants (including the Commission Services) | |
| RE | Restricted to a group specified by the consortium (including the Commission Services) | |
| CO | Confidential, only for members of the consortium (including the Commission Services) | |

Project co-funded by the European Commission within the Sixth Framework Programme (2002-2006)

# Deliverable D-7-3

| VERSION DETAILS | |
|---|---|
| Version: | 2.0 |
| Date: | 12 October 2006 |
| Status: | Final |

| CONTRIBUTOR(S) to DELIVERABLE | |
|---|---|
| **Partner** | **Name** |
| BE-LEU | Koen Eneman |
| BE-LEU | Tim Van den Bogaert |
| BE-LEU | Ellen Boon |
| BE-LEU | Heleen Luts |
| NL-TNO | Mark Houben |
| | |

| DOCUMENT HISTORY | | | |
|---|---|---|---|
| **Version** | **Date** | **Responsible** | **Description** |
| 0.1 | 28 March 2006 | Koen Eneman | First Draft and Outline |
| 0.2 | 6 June 2006 | Tim Van den Bogaert | Manual ALP-HCOM |
| 0.3 | 19 June 2006 | Koen Eneman | Draft version 0.3 |
| 0.4 | 27 July 2006 | Koen Eneman | Draft version 0.4 |
| 1.0 | 13 September 2006 | Koen Eneman | Version ready for review |
| 2.0 | 12 October 2006 | Koen Eneman | Revised final version |

| DELIVERABLE REVIEW | | | |
|---|---|---|---|
| **Version** | **Date** | **Reviewed by** | **Conclusion*** |
| 1.0 | 14 September 2006 | Mark Lutman | Modify |
| 1.0 | 2 October 2006 | Wouter Dreschler, Joyce Vliegen | Modify |
| | | | |

∗ e.g. Accept, Develop, Modify, Rework, Update

# Table of Contents

# List of Figures

## Acknowledgement

Supported by grants from the European Union FP6, Project 004171 HEARCOM. The information in this document is provided as is and no guarantee or warranty is given that the information is fit for any particular purpose. The user thereof uses the information at its sole risk and liability.

# Pre-Amble

The present report summarizes the research activities that have been conducted in the frame of workpackage 7 – task 3 of the HearCom project.

Whereas workpackage 5 focuses on the technical evaluation, optimisation and implementation of a number of promising signal enhancement algorithms for speech quality and intelligibility improvement in hearing-aid devices, workpackage 7 aims at the definition of common evaluation procedures to validate and compare the envisaged enhancement algorithms. In a first task within workpackage 7 suitable speech materials have been defined. The results of this work have been described in deliverable D-7-1. A second task dealt with the definition of appropriate environmental conditions and procedures for speech-in-noise tests, which are summarized in deliverable D-7-2. A fourth, still ongoing task, concentrates on listening effort and runs more or less in parallel with task 3, which is presented in this report. Bringing all this expertise together will enable us to conduct profound evaluation tests with the signal enhancement approaches supplied by workpackage 5. This will be done based on listening tests with hearing-impaired subjects in task 5 of workpackage 7.

More specifically, in the original project proposal the research activities of workpackage 7 – task 3 are described as follows:

1. A sound localization test with representative everyday signals has to be defined to estimate the functional benefit of a given type of signal processing. We have to agree upon a restricted number of such conditions, which can easily be implemented both in ‘hardware’ (e.g., by loudspeakers in a test room with appropriate acoustics) and in software (spatial virtual acoustics).

2. A clinical test on spatial hearing will be developed. Various approaches will be implemented, based on sound presentation by loudspeakers or via headphones. The results obtained with groups of normal-hearing and hearing-impaired subjects will be related to their experience in sound localization in daily life. The higher the correlation, the more relevant the test.

After a first revision of the project proposal by the European Commission the project has been scaled down, which implied among other things, that workpackage 5 and 7 mainly had to concentrate on unilateral, rather than bilateral hearing-aid devices. As a consequence, the development of a sound localization test to validate the benefits of bilateral processing schemes became somewhat redundant. Accordingly, the amount of person months assigned to this workpackage 7 – task 3 was cut down and the focus of workpackage 7 – task 3 was limited to the development of a clinical test on spatial hearing, corresponding to point 2 in the above list. The results that are presented in this report describe the achievements with respect to this part of the research.

The presented results are related to other HearCom activities that have been described in deliverables D-1-2, D-1-3, D-2-1 and D-2-1b. Similar to the report at hand, these deliverables address spatial hearing and localization issues. However, they primarily aim at offering an overview and implementation of state-of-the-art localization tests for screening and diagnostic purposes. In deliverable D-7-3, on the other hand, we focus on aided conditions, i.e. tests to evaluate the spatial-hearing abilities of hearing-aid or cochlear-implant users. Given the high cost of many of the existing test setups and a lack of standardization it was our objective to develop a low-cost spatial-hearing test intended for clinical usage.

# 1 Executive Summary

In this report a test setup is proposed that is intended for the assessment of spatial localization abilities of hearing-aid and cochlear-implant users. Most of the spatial-hearing tests presented in the literature, an overview of which is given in HearCom deliverable D-2-1, make use of high-quality hardware components (loudspeakers, amplifiers, sound cards) and dedicated software packages. They are typically designed and tuned towards the needs of the research institute for which they have been developed. As a consequence, spatial-hearing test setups are rather expensive, they are often too dedicated and specific, and are not readily commercially available.

In this deliverable a test setup is proposed that uses low-cost, readily available hardware components such as surround sound cards and active PC loudspeakers. Thanks to the modularity of the approach several sound cards can be combined, offering a true multi-loudspeaker setup. Furthermore, a user-friendly software package, called ALP-HCOM, has been developed to communicate with the sound card(s). The software is capable of conducting spatial localization experiments both in free field and via headphones and provides test result reports upon completion of the test.

In order to conduct proper localization experiments dedicated sound material is needed. To this aim, sound files and Head-Related Impulse Response databases are provided. They can be loaded into the ALP-HCOM software environment to set up spatial localization experiments.

To check the validity of the approach, a representative low-cost surround sound card has been evaluated. The conducted performance measurements proved that readily available low-cost surround sound cards fulfill the technical specifications that are required for sound localization performance assessment. In order to evaluate the accompanying software environment a spatial localization setup was constructed by combining multiple sound cards and active loudspeakers following our specifications. The setup has been validated by means of listening experiments and met our expectations.

# 2 Introduction

Selective hearing is a useful mechanism for extracting desired signals in complex acoustic environments, such as a cocktail party. The ability to understand speech in these complex scenarios has been largely attributed to the binaural processing strategy used in the auditory system (Bronkhorst and Plomp, 1988, 1989). Information from both ears interacts at various subcortical structures thereby providing the listener with the information needed to reconstruct the auditory scene. This helps the listener to stay focussed on one sound source and to cancel out unwanted sound sources.

One of the binaural processes is sound localization. The main mechanisms used for sound localization are fairly well known. Localization involves binaural processing of very small differences in time (10–700 μs), intensity (0–20 dB), and spectrum between the two ears (Stevens and Newman, 1936; Blauert, 1997; Gilkey and Anderson, 1997; Hartmann, 1999; Langendijk and Bronkhorst, 2002).

Extensive psychoacoustical research has been done on localization: experiments to measure localization performance of normal-hearing (Makous and Middlebrooks, 1990; Hofman and Van Opstal, 1998; Lorenzi *et al.*, 1999b) and hearing-impaired subjects (Hausler *et al.*, 1983; Noble *et al.*, 1994; Lorenzi *et al.*, 1999a) with different stimuli and in different test conditions, experiments via headphones with isolated or conflicting cues (Wightman and Kistler, 1992; Lorenzi *et al.*, 1999b), comparing performance of a monaural hearing-aid or cochlear-implant configuration with a bilateral hearing-aid or cochlear-implant configuration (Dillon, 2001; Van Hoesel and Tyler, 2003), and many others.

Although a lot of work has been done on localization with normal-hearing and hearing-impaired subjects, little work has questioned the effect of a bilateral hearing-aid system on the binaural potential of the hearing-aid user. In the human auditory system, the acoustical input signals of both ears are linked to the binaural centers where the binaural cues are interpreted and processed. Adding independently working hearing aids, each using its own compression and signal processing scheme, introducing its own time delay (in the order of 5 to 10 ms) (Dillon *et al.*, 2003) and having independent noise reduction schemes, could have a destructive effect on the binaural cues. Correspondingly, the hearing-aid user’s localization performance and speech perception in a complex environment could also be degraded.

In the work of Hausler *et al.* (1983), the question was raised as to whether hearing aids could have an impact on sound localization performance. The authors concluded that the usage of behind-the-ear hearing aids results in a reduction of Minimum Audible Angle (MAA, see also HearCom deliverables D-1-2 and D-1-3) compared to unaided conditions. In-the-ear hearing aids, on the other hand, showed smaller reductions in localization performance. Noble and Byrne (1990) tested localization performance in the frontal horizontal and vertical planes with bilateral behind-the-ear (BTE), in-the-ear (ITE), and in-the-canal (ITC) hearing aids with omnidirectional microphone configurations. A normal-hearing group was used as a reference. Intrasubject analysis did not show significant differences between unaided and aided performance for all three groups. These analyses were done on an error measure in which both vertical and horizontal errors were included. No statistical analysis on only horizontal or on only vertical localization errors was presented in the study. However, Noble and Byrne stated that for the control group, i.e. a group of six normal-hearing subjects, horizontal localization performance dropped from nearly 100% correct unaided to 73% correct wearing BTE hearing aids. In the same study, it is mentioned that the hearing-aid users tended to show poorer aided than unaided localization performance in the frontal horizontal plane, except for the ITE hearing-aid users, when wearing their own hearing aids. The difference in horizontal localization performance was not quantified in the study. Later, Noble *et al.* (1998) and Byrne *et al.* (1998) showed that better performance could be obtained by using open earmolds instead of closed earmolds for subjects with a moderate high-frequency (and a severe low-frequency loss) or a moderate low-frequency (and a severe high-frequency) hearing loss. By using open earmolds, the subject can use the direct sound field in the region of the moderate hearing loss for localization. For subjects with a moderate high-frequency loss, improvement in the vertical plane was found. For subjects with a moderate low to midfrequency hearing loss, improvement in the horizontal plane was found and performance was restored to unaided performance. These studies suggest that bilateral BTE hearing aids do not preserve localization cues. In all three studies, a broadband pink noise target stimulus was used and no jammer sources were present.

The available processing power in hearing aids increases as technology evolves. One of the main benefits is that more complex noise reduction algorithms can be implemented, improving speech understanding performance in acoustically challenging scenarios. In recent years, good results have been obtained using adaptive filtering techniques. These techniques adapt according to changes in noise scenario or acoustic condition, but are typically designed and evaluated monaurally. An important question is whether using these techniques bilaterally can have an impact on a binaural process, in particular on localization.

In Van den Bogaert *et al.* (2006) the localization performance of bilateral BTE hearing-aid users is validated in the frontal horizontal plane. The main cues for sound localization in these tests are interaural time differences (ITD) and interaural level differences (ILD). This study questions and quantifies the effect of current signal processing techniques on localization performance. Thereto, the hearing aids were fitted with a nonadaptive omnidirectional and an adaptive directional microphone configuration. Low-frequency, high-frequency, and broadband signals were used to separate the effects on ITD and ILD processing. Jammer sources were added to the condition with the broadband signal to evaluate the impact of the noise reduction system. First, it was shown that hearing-impaired subjects localize sounds less accurately than normal-hearing subjects. However, the results showed that hearing-impaired subjects can still use binaural cues, which motivates further research on binaural hearing-aid systems. Second, it was shown that current state-of-the-art bilateral hearing aids have a negative impact on binaural cues, thereby degrading localization performance. The decrease in localization performance with hearing aids could not be fully explained by microphone placement. Third, noise reduction techniques, such as an adaptive directional microphone, can have an additional significant negative impact on localization performance. Whether a significant drop in localization performance occurred with the noise reduction technique depended on the stimulus and noise scenario used in the localization test. The fact that hearing-aid users receive distorted binaural cues leads to degraded speech perception in noisy listening situations, in which binaural cues are very important. The question that arises is whether research, testing and design of monaural hearing aids is sufficient for bilateral hearing-impaired persons. Future research towards the preservation of binaural acoustical cues in hearing aids is therefore needed.

Given the increasing number of people using bilateral hearing aids and the impact of such systems on spatial localization performance, the assessment of the spatial-hearing abilities of hearing-impaired subjects gains importance both in research labs, as well as in clinical environments. Spatial localization ability assessment is required for the optimal fitting of modern, advanced (bilateral) hearing-aid systems and furthermore provides additional information to the subject's auditory profile. Due to the high cost of many of the existing setups and a lack of standardization it was our objective to develop a low-cost spatial-hearing setup intended for clinical usage. This deliverable presents and describes the proposed system both in terms of hardware and software specifications.

# 3 Hardware specifications for a spatial-hearing test that is intended for clinical usage

Existing, dedicated spatial-hearing setup solutions make use of high-quality hardware components. The quality constraints, and more importantly, the specificity of the setup require the usage of non-mass production hardware components, such as dedicated multi-channel sound cards and amplifiers. Typically, this hardware is rather expensive. Furthermore, the required hardware components are not readily available on the market for immediate delivery. In general, however, sound localization experiments are less demanding than speech-in-noise tests as far as hardware requirements are concerned. Low-cost solutions can hence be considered as a viable alternative.

## 3.1 Low-cost multi-channel sound cards

The solution we advise makes use of consumer-electronic, hence low-cost, hardware. Instead of relying on dedicated multi-channel A/D converters, we propose to use 5.1 or 7.1 surround sound cards, which are primarily designed for home cinema applications. As a consequence, they can generally be purchased at a low price in basically any PC shop.

### 3.1.1 Surround sound cards

Surround sound cards give the customer more ambience feeling than a mere stereo configuration. A 5.1 card for instance provides 6 channels, each referred to with a specific name, derived from the home cinema context. In this way, one center and two front channels (left + right) are defined, which create 3-channel stereo. Furthermore, there are two surround channels (left + right) giving ambience feeling and providing room impression to the listener. They are complemented by a subwoofer channel to generate low frequencies. A 7.1 surround sound card on the other hand, offers 8 channels, additionally providing two so-called rear surround channels. The applicability of surround sound cards is not limited only to home cinema systems, as typically in practice, the channels can be independently controlled by the sound card driver and can basically output whatever sound stream that is needed. Furthermore, several multi-channel sound cards can be combined and be independently accessed. In this way, basically any kind of multi-loudspeaker configuration can be realized. Depending on the number of loudspeakers that is finally required in the spatial-hearing test, the type of sound card (5.1 or 7.1) and the number of cards can be easily determined.

### 3.1.2 Sound card requirements

Given the application we have in mind, the most important requirements for the sound card are an almost flat frequency response, low cross coupling between the different channels, low harmonic distortion, no channel-dependent delays and low background noise levels. Therefore, when purchasing a sound card the following items should be checked:

- flatness of the frequency response
- Total Harmonic Distortion
- cross-coupling between the different channels
- inter-channel delays
- background noise level with respect to the signal level[1]

Furthermore, make sure that you purchase sound cards whose decoder units can be turned off. Decoder units (DTS or Dolby Digital) perform operations on the signals, such as lowpass or highpass filtering etc., which are undesired for the application we have in mind. Also verify that the master volume control of the audio mixer software is fully open to maximally exploit the dynamic range of the cards.

Apparently, commercially available 5.1 or 7.1 sound cards can easily meet the specified requirements. To verify this we purchased two 7.1 Sound Blaster®Live! 24-bit PCI sound cards from Creative Labs (Figure 1). They cost €50 each.

Also stand-alone solutions exist. These external sound cards typically communicate with the PC using a USB connection. External sound cards are expected to be less sensitive to the internal noise generated by the PC. Furthermore, unlike their PCI counterparts, stand-alone cards can be used in combination with a laptop PC. They are typically more expensive, however: the Creative Labs 7.1 Sound Blaster Audigy2 NX USB sound card, the stand-alone counterpart of the Sound Blaster®Live! card, costs about €125. A nice feature of this USB sound card is that you can switch off the built-in 5.1 or 7.1 decoders very easily.

---

[1] In order to check the amount of background noise, a “blocked input” condition (e.g. by connecting a suitable resistor across the input terminals) can be recorded. In this way, it can be determined whether the recorded output from the sound card is greater than the blocked condition. The output can be measured electrically using an instrumentation quality voltmeter with adequate dynamic range, or acoustically with a sound level meter when the output of each channel is connected to a loudspeaker.

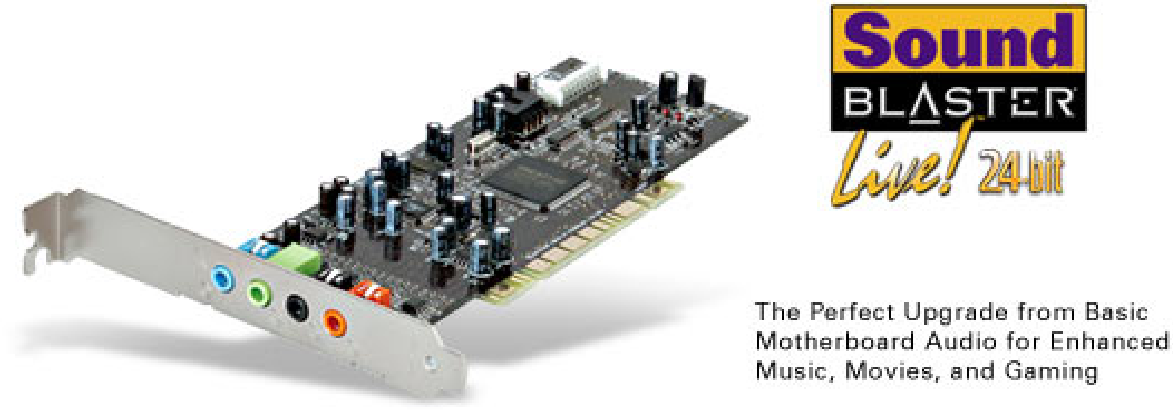


**Figure 1: Creative Labs 7.1 Sound Blaster®Live! 24-bit PCI sound card**

### 3.1.3 Performance evaluation of the 7.1 Sound Blaster®Live! 24-bit sound card

To prove the viability of our approach we measured and evaluated the Sound Blaster®Live! 24-bit sound card.

First, a Gaussian white noise signal sampled at 48000 Hz and linearly quantized with 16 bits per sample was sent to one of the 8 channels of the Sound Blaster card, zeros were sent to the other channels. Care was taken not to exceed the maximum amplitude level provided by the 16-bit data format, i.e. $2^{15}$-1. In this way clipping of the data and, hence, the introduction of severe distortion could be avoided. Then, the eight signals that came out of the Sound Blaster card were recorded with a high-quality 8-channel sound card (RME Hammerfall) using a sampling frequency of 96 kHz and 32-bit linear quantization. Based on the recorded data the frequency spectrum of the system, as well as the cross-coupling between the different channels was computed. The results for the case when noise was sent to channel 1 are shown in Figure 2 and Figure 3. It can be verified that the frequency response of the system is flat and has a bandwidth of about 24 kHz, as expected. Furthermore, the cross coupling between the channels seems sufficiently low for the intended application. Similar figures were found when the white noise signal was sent to one of the other input channels.

Figure 4 shows the Total Harmonic Distortion and the frequency spectrum that is observed when a full-scale 1 kHz sine wave signal is sent to channel 1. Similar results have been obtained for the other channels. The amount of distortion seems sufficiently low for the intended application.

The spectrum has been normalized in order to compensate for the length and the type of window that is used. The reference level of 0 dB shown in the figure corresponds to the expected value of the magnitude spectrum of a Gaussian white noise signal having unit standard deviation, i.e. a standard deviation $\sigma$ that corresponds to the maximum input level of the RME sound card. This level is a theoretical reference as a Gaussian white noise signal with unit standard deviation would exceed the maximum input level for all samples that fall outside the interval [$-\sigma$, $\sigma$], and would hence be heavily distorted. If a full-scale sinusoid is processed with the window type and window length selected in Figure 4 peak levels of about 41.6 dB are observed.

It was also verified that there is no channel-dependent delay: when identical signals are simultaneously sent to each of the 8 channels, the resulting inter-channel delays are smaller than 1 sample.

Finally, Figure 5 shows the spectrum of the background noises recorded at each channel when no signal was sent to the Sound Blaster card. Compare with Figure 2.

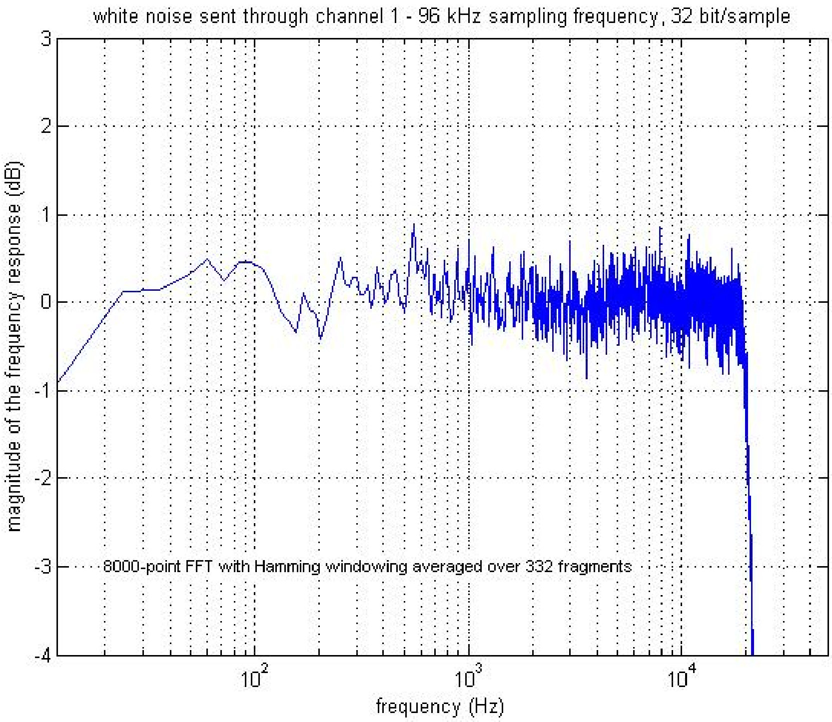


**Figure 2: Frequency spectrum of the Sound Blaster®Live! 24-bit sound card – channel 1**

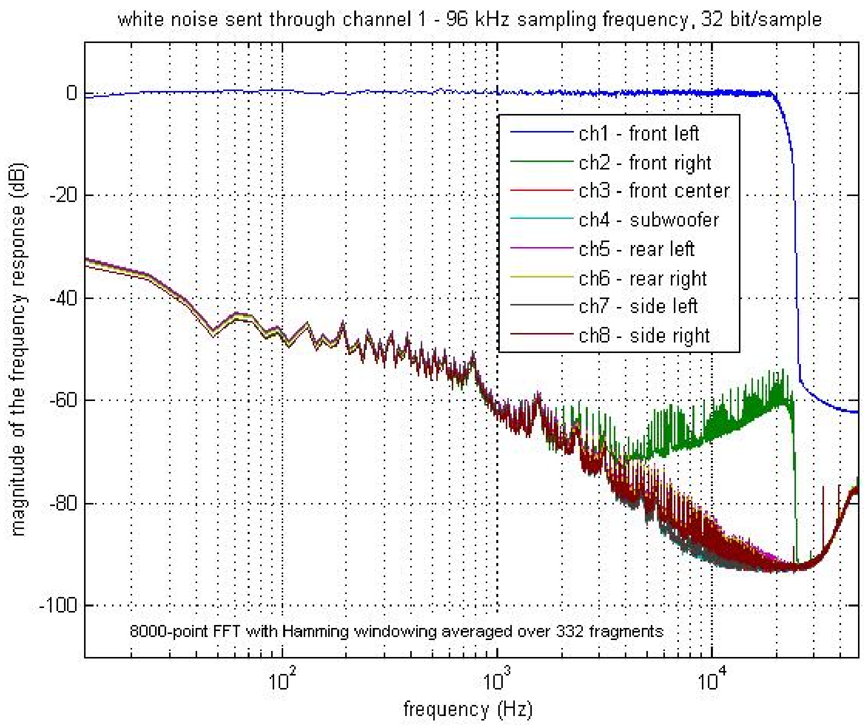


**Figure 3: Sound Blaster®Live! 24-bit sound card – cross coupling between channels when a white noise signal was sent to the first channel**

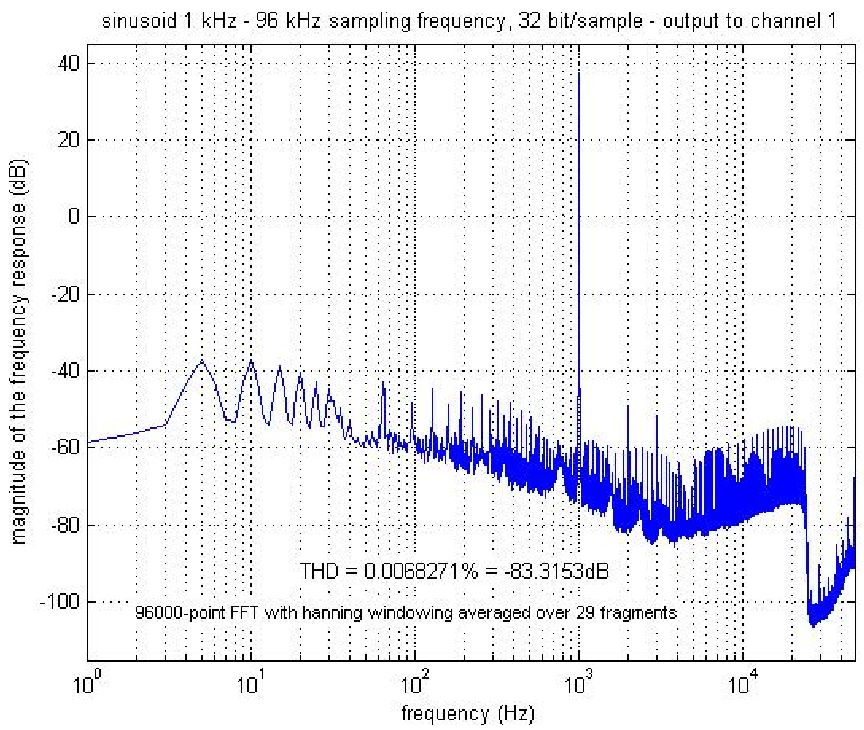


**Figure 4: Sound Blaster®Live! 24-bit sound card – Total Harmonic Distortion for channel 1**

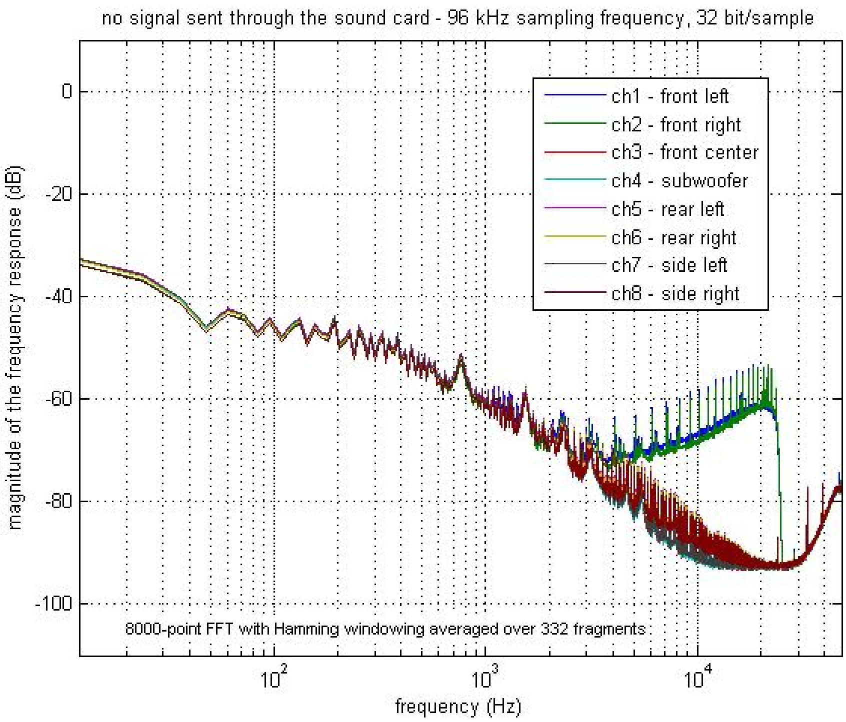


**Figure 5: Sound Blaster®Live! 24-bit sound card – background noise**

So far, the evaluation has only been performed for the 7.1 Sound Blaster®Live! 24-bit PCI sound card from Creative Labs, which we rather randomly selected and purchased from a nearby PC shop. It is of course practically impossible to validate all surround sound cards that are available on the market, a market which is anyway quite dynamic and constantly offering new releases and products. Basically, Figures Figure 2 to Figure 5 just prove that low-cost multi-channel sound cards that are readily available on the market, easily meet the technical specifications that are required for sound localization testing.

## 3.2 Amplifier and loudspeaker requirements

As far as the amplifier and the loudspeakers are concerned, a high-quality solution would require expensive (passive) speakers complemented with a high-fidelity multi-channel audio amplifier. For sound localization experiments the requirements on the frequency characteristics of the loudspeakers and the amplifier are less restrictive than in the case of speech-in-noise tests for instance. Hence, standard PC speakers can be used instead, which have an incorporated internal amplifier. In practice, it suffices to take care that the loudspeakers used for testing are of the same type and that they are appropriately calibrated. Differences in volume settings should be compensated for by applying level roving.

Furthermore, it should be verified that the frequency response does not vary too much between different speakers.

Surround sound cards often come together with dedicated surround loudspeaker sets. Be aware that these sets are primarily intended for surround sound reproduction, and are mostly not suited for localization experiments. Typically, the loudspeakers are not all of the same type, resulting in different frequency characteristics. Furthermore, as the speakers are usually passive devices, they are powered by a joined amplifier, often integrated in the subwoofer and performing undesired filtering operations on the signals.

# 4 HearCom Advanced Localization Procedure (ALP-HCOM)

The clinical spatial-hearing test setup we propose in this deliverable not only makes use of low-cost hardware (surround sound card(s) + standard PC loudspeakers), as explained in section 3, but also comes with a dedicated software environment, called ALP-HCOM.

## 4.1 Introduction

The HearCom Advanced Localization Procedure (ALP-HCOM) has been developed in the frame of Workpackage 7 – task 3 of the European HearCom project. This software program provides a user-friendly interface that allows the user to conduct localization experiments via headphones or in free-field conditions. The software can be downloaded from the HearCom portal:

http://fit-bscw.fit.fraunhofer.de/bscw/bscw.cgi/36960320

## 4.2 Installation

The program was developed under Windows XP using the DirectSound (part of Microsoft DirectX) libraries. If DirectSound has not yet been installed on your PC, download the DirectSound libraries from

http://msdn.microsoft.com/downloads/

and run dx90bsdk.exe. This will unpack DirectX to a folder you specify. DirectX 9.0 can be installed by clicking the install.exe file in the folder you selected. It is possible that you have to reboot your PC after the installation.

Once the DirectX library has been installed you can copy the ALP-HCOM directory to your hard disk and run ALPHCOM.exe.

## 4.3 Using the program

Defining a full localization test procedure consists of three main parts:

- Defining the hardware to be used during the test
- Defining the test stimuli
- Defining the test conditions: how should the stimuli be presented (random order / level roved / number of repetitions / …)?

All of these actions can be set in the startup screen window (Figure 6).

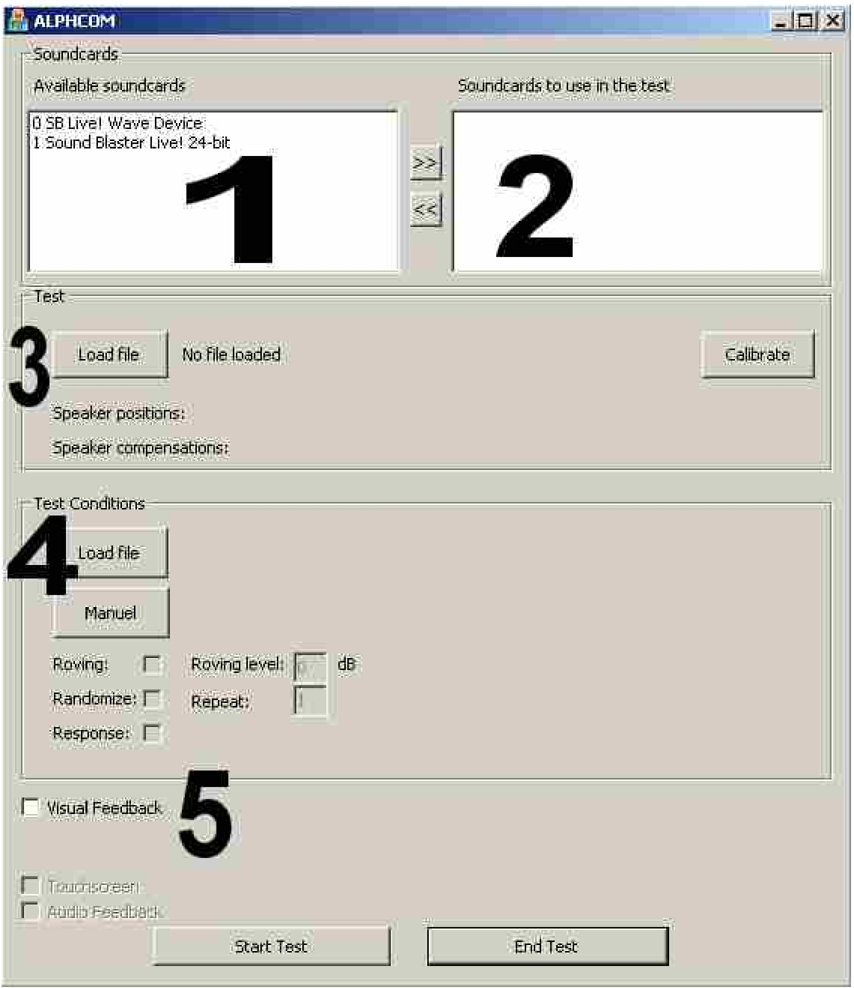


**Figure 6: ALP-HCOM startup screen window**

### 4.3.1 Defining the hardware (nrs. 1,2 in Figure 6)

The program is able to communicate with PCI and USB sound cards using the DirectSound standard. It was tested with 7.1 Sound Blaster®Live! 24-bit PCI sound cards from Creative Labs (see section 3.1.3 for an evaluation of this type of sound card) and 7.1 Sound Blaster Audigy2 NX USB sound cards, also from Creative Labs.

In the left column at the top of the startup screen all sound cards are shown that are available in your PC (“Available soundcards”, nr. 1 in Figure 6). To define the sound cards to be used during the experiment, select a desired sound card and add it to the right column (“Soundcards to use in the test”). Selected sound cards can be removed from the right list (nr. 2 in Figure 6) using the arrows pointing to the left.

The software has been validated using two 7.1 sound cards, giving access to 16 sound channels in total. When doing localization tests, make sure that all sound effects have been switched off. Often these effects are part of the software that comes with your sound card. Also set the volume of the audio devices you wish to use to its maximum level. Otherwise, you won't use the full dynamic range of the sound cards.

Remark: Problems with Windows have occurred when multiple sound cards are accessed at the same time. When one e.g. tries to communicate with two identical PCI Sound Blaster cards, some PCs will crash and give a blue screen. If, on the other hand, two identical USB Sound Blaster cards are used, Windows does not crash, but still will fail to control the sound cards (this also depends on the drivers you are using). Both problems can be explained by the fact that Windows is not able to handle two identical devices (also depending on the drivers). The reason for this is that cheap identical devices carry the same serial number. This serial number is used by the driver and by Windows to make a distinction between the audio devices. Hence, possible malfunctioning of the setup in the case of multiple sound cards is not caused by the ALP-HCOM software, but is rather due to implementation errors by Microsoft and the sound card manufacturers. Therefore, if your application requires multiple sound cards we recommend using non-identical cards (which doesn't seem logical, but anyway is the best guarantee of a stable system). Also take into account that Creative Labs will not give support if multiple identical sound cards are used at the same time.

### 4.3.2 Defining the stimuli (nr. 3 in Figure 6)

The program can be run in two modes:

- Free-field experiment
- Headphone experiment (any 2-channel sound card supporting DirectSound can be used for this purpose)

When doing free-field experiments, typically, sound files will be randomly presented through one of the loudspeakers that are physically positioned in the test room. After each stimulus the subject indicates through which loudspeaker the sound was perceived. The test can be conducted under supervision of a test leader or can be autonomously performed by the subject using a touch screen (see Figure 7).

When doing headphone experiments stereo stimuli are presented to the subject. The subject then has to guess the location of the virtual stimulus and input the answer to the computer (see Figure 7). The stimuli are typically generated by convolving a mono sound file with a pair of Head Related Impulse Responses (HRIRs), also called Head Related Transfer Functions (HRTFs), which in fact are impulse responses from a certain 3D-position in space to both the left and the right ear of an average human listener. In this way, the direction-dependent filtering of sound waves around an average human head can be taken into account and the listener can be given the impression that the sound is coming from a specific direction in space.

The following parameters should be entered (in random order) in a file with extension .htf (HearCom test file). Use notepad or another plain text editor to generate the file and save it as a *filename*.htf

#### 4.3.2.1 Free-field experiment

First, an example is given of a valid test file for a free-field experiment. The different options will be explained after the example.

```
[speaker positions]= 90, 75, 60, 45, 30, 15, 0, -15, -30,-45,-60,-75,-90,

[speaker compensation]= -1, -2, -3, -4, -5, -6, -7, -8, -9, -10, -11, -12,
-13,

[headphones]=no

[sounds directory]=C:\ALP\localisation\sounds\

[overall gain]=0

[stimuli]

1,    13th500Hz_200ms_20dB.wav,
*

2,    13th500Hz_200ms_20dB.wav,
*

3,    13th500Hz_200ms_20dB.wav,
*

4,    13th500Hz_200ms_20dB.wav,
*

5,    13th500Hz_200ms_20dB.wav,
*

6,    13th500Hz_200ms_20dB.wav,
*

7,    13th500Hz_200ms_20dB.wav,
*

8,    13th500Hz_200ms_20dB.wav,
*

9,    13th500Hz_200ms_20dB.wav,
*

10,   13th500Hz_200ms_20dB.wav,
*

11,   13th500Hz_200ms_20dB.wav,
*

12,   13th500Hz_200ms_20dB.wav,
*

13,   13th500Hz_200ms_20dB.wav,

[end stimuli]
```

<u>The different options:</u>

`[speaker positions]`

The angles (°) indicating the position of the loudspeakers relative to the subject. In the case of a free-field experiment the angles correspond to the real position of the loudspeakers. Buttons will appear on the screen corresponding to the specified locations, so that the subject can indicate from which direction the sound was perceived (see Figure 7). The angle values have to be separated by a comma. The first speaker position will be linked to speaker number one, the second position to speaker number two, etc.

`[headphones]`

Indicates whether or not your experiment is a headphone experiment (‘yes’ in case of a headphone experiment, ‘no’ in case of a free-field test).

`[speaker compensation]`

The amount of correction gain (in dB) that should be applied to each single loudspeaker (these gains should be negative to avoid clipping of the sound file). There should be as many correction gains as there are loudspeaker positions (each speaker has a correction gain). The first correction factor will be used when playing sounds through loudspeaker number one, the second one when playing through loudspeaker number two, etc. The gains are applied in software. Hence, be aware that too much attenuation will increase the amount of quantization distortion.

`[sound directory]`

The absolute path indicating where the sounds are located on your hard disk drive. Always end the path with a ‘\’ (e.g. c:\sounds\).

`[overall gain]`

The overall gain (in dB) applied to the test signals. This gain is especially useful when doing the same test at a different sound pressure level (dB SPL). The gain is applied in software, hence it should be negative to avoid clipping of the data. Again, since the gain is applied in software, too much attenuation will increase the amount of quantization distortion. The overall gain should be set in accordance with the signal level of the stimulus files and the volume settings selected during calibration.

`[stimuli]`

The stimuli (mono 16-bit files) that will be used in the free-field experiment, together with the corresponding loudspeaker where the sound should be played on. A number of representative stimuli files are provided together the software, as explained in section 5.1. The loudspeakers are referred to as integer numbers, corresponding to the entries of [speaker positions], indicating where the loudspeaker is positioned relative to the test subject. Each stimulus is separated from another stimulus by an asterisk * (do not put an * after the last stimulus because then the program will generate an error message). The loudspeaker identifier is separated from the stimulus file name by a comma. In the example only one stimulus will be used during testing (*c:\ALP\localisation\sounds\13th500Hz_200ms_20dB.wav*). It will be played on loudspeakers 1 to 13 (at angles 90° to -90° w.r.t. the subject, separated by 15°).

#### 4.3.2.2 Headphone experiment

Next, an example is given of a valid test file for a headphone experiment.

```
[speaker positions]=90, 75, 60, 45, 30, 15, 0, -15, -30, -45, -60, -75,-90,
[headphones]=yes
[headphone calibration left]=-5
[headphone calibration right]=-10
[sounds directory]=C:\ALP\localisation\sounds\
[overall gain]=-2
[stimuli]
2,    sinusstereo2.wav,
*
3,    sinusstereo3.wav,
*
4,    sinusstereo4.wav,
*
5,    sinusstereo5.wav,
*
6,    sinusstereo6.wav,
*
7,    sinusstereo7.wav,
*
8,    sinusstereo8.wav,
[end stimuli]
```

<u>The different options:</u>

[speaker positions]

The angles (°) indicating the virtual position of the loudspeakers relative to the subject. Buttons will appear on the screen corresponding to the specified locations, so that the subject can indicate from which direction the sound was perceived (see Figure 7). The angle values have to be separated by a comma. The first speaker position will be linked to speaker number one, the second position to speaker number two, etc.

`[headphones]`: see section 4.3.2.1

`[headphone calibration left]`
`[headphone calibration right]`
The amount of correction gain (in dB) applied to the left and right headphone speaker. These correction gains will be applied in software to all stimuli played through the headphones. These gains can be used to calibrate the left and right speaker of the headphones. The gain values should be negative. In this way clipping of the sound file can be avoided. However, one should be careful not to apply too much attenuation since the gain is applied in software. It therefore reduces the dynamic range and hence increases the amount of quantization distortion.

`[sound directory]`: see section 4.3.2.1

`[overall gain]`: see section 4.3.2.1

`[stimuli]`

Here the stimuli (stereo 16-bit files) are defined that will be used in the headphone experiment together with the stimulus position. A number of stereo stimuli files are provided together the software, as explained in section 5.1. The stimulus position is a number that refers to the entries of `[speaker positions]`, indicating the virtual speaker where the stimulus is coming from (being the correct answer). The stimuli are separated from each other by an asterisk `*`. The stimulus position is separated from the stimulus file name by a comma.

In the example 7 different stimuli will be used (*C:\ALP\localisation\sounds \sinusstereo2.wav* till *C:\ALP\localisation\sounds\sinusstereo8.wav*). The files could for instance be generated by convolving a stimulus with a set of 7 Head Related Impulse Response pairs. Head Related Impulse Response (HRIR) or Transfer Function (HRTF) databanks provide impulse responses from a set of 3D-positions in space to both the left and the right ear. A number of HRIR databanks are provided, as explained in section 5.3.

The correct answer when playing stimulus sinusstereo2 is speaker location 2 (given in the left column), which is 75° in this case (see `[speaker positions]`). In the given example the user will see a screen with 13 different speaker locations (from 90° to -90° in steps of 15°) to choose from (see Figure 7). Normally, one uses the same amount of stimuli as there are answering options. In this way, there is one sound file for each speaker location. This is however not mandatory, as shown by the example.

### 4.3.3 Defining the test conditions (nr .4 in Figure 6)

After specifying the test stimuli still some test parameters have to be defined. The program needs to know for instance whether level roving should be applied, whether the stimuli should be presented in a randomized way, and how many times each stimulus has to be repeated. By switching off the user response option all the stimuli will be presented sequentially without waiting for an answer of a test subject. This is convenient during testing, evaluating or calibrating the setup.

The test parameters can be loaded from a file or can be entered manually (press “manuel”, choose your options and then press ok). To load the parameters from a file, store the settings in a text file with extension .set and press “load file” to read the file.

An example of such a pre-defined file:

```
[randomize]=yes

[repeat]=3

[response]=yes

[roving]=yes

[roving level - in dB]=5
```

The roving level is entered in dB. In this case level roving will be applied from 0 to -5dB (negative to avoid clipping of the wav file).

### 4.3.4 Other options

- As an extra option (Nr. 5 in Figure 6) the user can activate the visual feedback functionality at the bottom of the screen. Sometimes the test leader wants to know the correct answer corresponding to a given stimulus. Thereto, a visual feedback option has been added. When switching off this option, the test can be performed by the subject in an unsupervised way using a touch screen (no information is given to the test subject).

- In a future release a child-friendly version of the software will be made available if necessary (already tested by partner BE-LEU). This gives the user the possibility to show pictures (.jpg) on the screen instead of buttons with numbers.

- In a future version audio feedback will be made available if necessary (needed in the full child-friendly version developed by partner BE-LEU)

### 4.3.5 Starting and completing the test

When starting the test the definitions of the stimuli are combined with the given test conditions and the localization experiment is compiled. The user will see the screen shown in Figure 7.

The user has to press the start button to initiate the test (nr. 2 in Figure 7). The test leader can always stop the test by pressing “end of test” (nr. 3 in Figure 7). A progress bar is shown just above the “end of the test”-button. When the test subject presses the “start” button the first trial will be initiated: the play-back of the first stimulus will be started and once the stimulus has been played, the buttons on the different speaker locations will highlight. Now the subject or the test leader can enter the response by pressing one of the buttons. When a button is pressed the next trial will be presented. When the option “show visual feedback” is selected (nr. 5 in Figure 6) the buttons will be all highlighted except for the correct one, thereby showing the correct answer to the test leader. If the user chooses to disable the “response option” (section 4.3.3), all trials will be presented sequentially without waiting for a response.

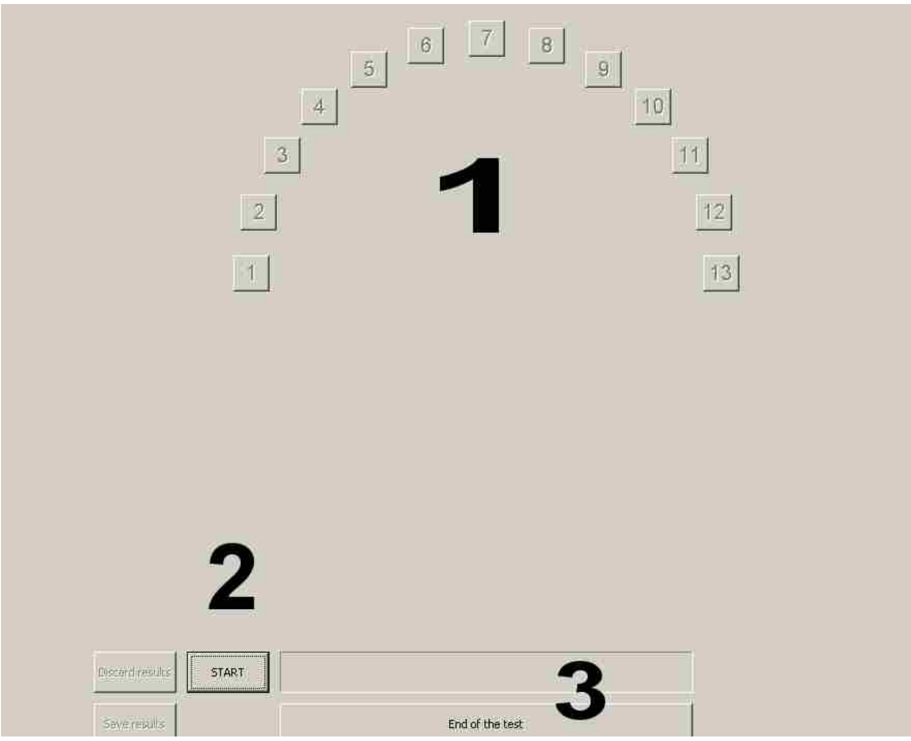


**Figure 7: User screen during the actual localization experiment. An experiment with 13 discrete locations was defined, with loudspeakers placed at -90° to +90° in steps of 15°.**

### 4.3.6 Test results and performance measures

Upon completion of the test, the user might want to save the test results to a file. The results will then be stored in two files with extension .drr and .drrs, the .drrs file being the shorter version of the .drr file. In the .drr file all options and test definitions are included. At the end of the file, the stimuli-response pairs and the error measures can be found. This information represents the full test. The confusion matrix enables the user to recalculate the error measures or —if desired— to introduce other error measures. It is recommended to use one file per test subject. If a storage file is selected that already exists, the results will be appended at the end of the file (an example is given below). If the user prefers not to save the data the cancel button should be pressed when the saving-data dialog is shown. This action needs to be confirmed by pressing the “discard results” button.

Example of a .drr and .drrs file, which contain the test results of 2 tests:

**.drr-file (long version of the test results)**

```
________________________________________________________________________________
------------------------------ TEST --------------------------------
test description:test 2 randomized / roved
date: 12/3/2006
starting hour: 16:6
ending hour:   16:6

Filename:C:\13 Speakers Bow full.htf
[speaker positions]          =90,75,60,45,30,15, 0, -15, -30 ,-45,-60,-75,-90,
[speaker compensation]       = -1, -2, -3, -4, -5, -6, -7,   -8,   -9,   -10,  -11,  -12,  -13,
[headphones]=no
[sounds directory]=C:\ALP\localisation\sounds\
[overall gain]=-10
[stimuli]
1,      13th500Hz_200ms_20dB.wav,
*
2,      13th500Hz_200ms_20dB.wav,
*
3,      13th500Hz_200ms_20dB.wav,
*
4,      13th500Hz_200ms_20dB.wav,
*
5,      13th500Hz_200ms_20dB.wav,
*
6,      13th500Hz_200ms_20dB.wav,
*
7,      13th500Hz_200ms_20dB.wav,
*
```

8, 13th500Hz_200ms_20dB.wav,
*
9, 13th500Hz_200ms_20dB.wav,
*
10, 13th500Hz_200ms_20dB.wav,
*
11, 13th500Hz_200ms_20dB.wav,
*
12, 13th500Hz_200ms_20dB.wav,
*
13, 13th500Hz_200ms_20dB.wav,
[end stimuli]

Randomize: true
Repeat:1
Response: true
Roving: true
Roving level:4
TestResults:

[confusion matrix]
Stimulus location
1 1 1 1 1 1 1 1 1 1 1 1 1
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0Response
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0
0 0 0 0 0 0 0 0 0 0 0 0 0

[chronological stimulus and response list with the details of the played stimulus (3rd column is roving level)]

| | | |
|---|---|---|
| 7 | 1 | -2.000000 |
| 6 | 1 | 0.000000 |
| 4 | 1 | 0.000000 |
| 2 | 1 | -1.000000 |
| 8 | 1 | -4.000000 |
| 12 | 1 | -2.000000 |
| 5 | 1 | -3.000000 |
| 13 | 1 | -2.000000 |
| 1 | 1 | -3.000000 |

| | | |
|---|---|---|
| 9 | 1 | -4.000000 |
| 11 | 1 | -4.000000 |
| 3 | 1 | -4.000000 |
| 10 | 1 | -3.000000 |

[measurement values]
RMS error: 106.0660
Mean absolute error: 90.0000
Mean error: -90.0000- means shifted to the left
Percentage correct: 7.6923
------------------------------ END TEST ----------------------------

---

------------------------------ TEST --------------------------------
test description:test 3 no roving
date: 12/3/2006
starting hour: 16:6
ending hour: 16:7

Filename:C:\13 Speakers Bow full.htf
[speaker positions] =90,75,60,45,30,15, 0, -15, -30 ,-45,-60,-75,-90,
[speaker compensation] = -1, -2, -3, -4, -5, -6, -7, -8, -9, -10, -11, -12, -13,
[headphones]=no
[sounds directory]=C:\ALP\localisation\sounds\
[overall gain]=-10
[stimuli]
1, 13th500Hz_200ms_20dB.wav,
*
2, 13th500Hz_200ms_20dB.wav,
*
3, 13th500Hz_200ms_20dB.wav,
*
4, 13th500Hz_200ms_20dB.wav,
*
5, 13th500Hz_200ms_20dB.wav,
*
6, 13th500Hz_200ms_20dB.wav,
*
7, 13th500Hz_200ms_20dB.wav,
*
8, 13th500Hz_200ms_20dB.wav,
*
9, 13th500Hz_200ms_20dB.wav,
*

```
10,     13th500Hz_200ms_20dB.wav,
*
11,     13th500Hz_200ms_20dB.wav,
*
12,     13th500Hz_200ms_20dB.wav,
*
13,     13th500Hz_200ms_20dB.wav,
[end stimuli]

Randomize: true
Repeat:1
Response: true
Roving: false
Roving level:4
 TestResults:

[confusion matrix]
        Stimulus location
 1 1 1 0 1 0 1 1 1 0 1 0 1
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0Response
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 0 0 0 0 0 0 0 0 0 0
 0 0 0 1 0 1 0 0 0 1 0 1 0

[chronological stimulus and response list with the details of the played stimulus (3rd column is roving level)]
        7       1       0.000000
        9       1       0.000000
        3       1       0.000000
        13      1       0.000000
        2       1       0.000000
        8       1       0.000000
        5       1       0.000000
        1       1       0.000000
        11      1       0.000000
        10      13      0.000000
        12      13      0.000000
        4       13      0.000000
        6       13      0.000000
```

[measurement values]

RMS error: 97.9207

Mean absolute error: 80.7692

Mean error: -34.6154- means shifted to the left

Percentage correct: 7.6923

------------------------------ END TEST ----------------------------

**.drrs file (short file) for the same two tests:**

test description:test 2 randomized / roved
date: 12/3/2006
starting hour: 16:6
ending hour:   16:6

RMS error: 106.0660
Mean absolute error: 90.0000
Mean error: -90.0000- means shifted to the left
Percentage correct: 7.6923

test description:test 3 no roving
date: 12/3/2006
starting hour: 16:6
ending hour:   16:7

RMS error: 97.9207
Mean absolute error: 80.7692
Mean error: -34.6154- means shifted to the left
Percentage correct: 7.6923

### 4.3.7 Calibration

The program also gives the opportunity to calibrate the loudspeakers or the headphones. When pushing the calibration button (right-hand side of Figure 6) a screen will appear which makes it possible to play a certain .wav file through the different speakers (choose a mono calibration file and press the button of the speaker you wish to calibrate) or through the headphones (choose a stereo file and click the button) for calibration purposes. The stimuli are played with the overall gain and compensation (headphone left and right compensation for headphone calibration and speaker compensation for free-field testing) that is specified (roving is not applied during calibration). The user can then measure the intensity output of the headphones or the loudspeakers and recalculate the compensation values to calibrate the setup.

## 4.4 Information

This program was developed by partner BE-LEU in the frame of Workpackage 7 of the HearCom project. More information can be obtained from tim.vandenbogaert@med.kuleuven.be

# 5 Sound material and Head-Related Impulse Responses databases

Dedicated sound material is needed to conduct proper spatial localization experiments. More specifically, to set up a free-field or a headphone experiment within the ALP-HCOM software environment, appropriate sound files and Head-Related Impulse Response databases are required.

## 5.1 Stimuli

In a free-field localization experiment stimuli are successively sent to one of the loudspeakers in the test room. The choice of an appropriate stimulus file is thus of great importance. Typically used are low-frequency bandlimited noises, high-frequency bandlimited noises and broadband signals such as telephone ringing sounds (Van den Bogaert *et al.* 2006). A number of representative stimuli, provided by HearCom partner BE-LEU (ExpORL, Dept. Neurosciences, Katholieke Universiteit Leuven, Belgium), can be found on

http://fit-bscw.fit.fraunhofer.de/bscw/bscw.cgi/36960320

## 5.2 Head-Related Impulse Responses

Free-field experiments make considerable demands in terms of hardware. Headphone experiments are then often preferred, as they are much easier to perform. In order to set up a headphone experiment stereo headphone signals have to be generated from a mono stimulus file to give the listener the impression that the sound is coming from a specific virtual direction. This can be done through the use of Head-Related Impulse Responses (HRIRs), which describe the direction-dependent filtering of sound waves around an average human head.

## 5.3 Databases

In the frame of this Workpackage several HRIR databases are provided. They can be found on the HearCom portal:

http://fit-bscw.fit.fraunhofer.de/bscw/bscw.cgi/36960320

The databases can for instance be used to generate the stimuli that are required by the ALP-HCOM software during a headphone experiment.

### 5.3.1 BE-LEU databases

HearCom partner BE-LEU (ExpORL, Dept. Neurosciences, Katholieke Universiteit Leuven, Belgium) provides four sets of HRIRs. Three data sets were recorded in a sound-treated room with size 6 m x 3 m x 3.5 m (length x width x height). The reverberation time T60 of this room is 540ms and hence corresponds more or less to a living room. Additionally, a set of HRIRs is provided that is recorded in a room with low reverberation: the room dimensions are 5 m x 3.5 m x 3 m (length x width x height) and the reverberation time T60 is 150ms. In all cases, the signals were recorded with a Cortex MK II artificial head and sampled at 96 kHz. For the reverberated room three sets (see below) of 13 head-related impulse response pairs were derived, corresponding to 13 azimuth angles in the horizontal plane: -90°, -75°, -60°, -45°, -30°, -15°, 0°, 15°, 30°, 45°, 60°, 75° and 90°. For the low-reverberant room 18 head-related impulse response pairs are provided for the following azimuth angles in the horizontal plane: 0°, 15°, 30°, 45°, 60°, 75°, 90°, 120°, 150°, 180°, -150°, -120°, -90°, -75°, -60°, -45°, -30° and -15°. Angle 0° corresponds to the direction right in front of the manikin. Positive azimuth angles are to the right side of the manikin.

The following HRIR databases are available:

1. for the reverberant room (T60=540ms)
    a. head-related impulse responses generated based on recordings with the microphones inside the ears of the manikin
    b. head-related impulse responses based on recordings with the front microphone of two GNResound Canta 7 behind-the-ear hearing aids put on the manikin’s ears
    c. head-related impulse responses based on recordings with the rear microphone of two GNResound Canta 7 behind-the-ear hearing aids put on the manikin’s ears
2. for the low-reverberant room (T60=150ms)
    a. head-related impulse responses generated based on recordings with the microphones inside the ears of the manikin

The HRIR data are available in the original sampling format (96 kHz) and were also resampled to 44.1 kHz. They were written to wave files, quantized with 32 bits.

### 5.3.2 NL-TNO database

HearCom partner NL-TNO (TNO Human Factors, Soesterberg, The Netherlands) provides a HRIR database that was recorded in an anechoic room. The signals were recorded with an ARTIFICIAL HEAD HMS II.3 and were sampled at 50 kHz. In total 72 head-related impulse response pairs were generated, corresponding to 72 azimuth angles in the horizontal plane between -175° and 180° in steps of 5°. Angle 0° corresponds to the direction right in front of the manikin. Positive azimuth angles are to the right side of the manikin. The HRIR data are available in the original sampling format (50 kHz) and were also resampled to 44.1 kHz. They have been written to wave files, and quantized with 32 bits.

# 6 Listening tests

In order to validate the proposed setup, the software and the sound material, listening experiments have been conducted with a number of normal-hearing subjects.

## 6.1 Listening experiment specifications

### 6.1.1 Subjects

The study has been performed with five normal-hearing subjects between 20 and 33 years old, with an average age of 25. Their hearing thresholds are equal to or better than 20dBHL (hearing level) for all octave frequencies from 125 Hz to 8000 Hz.

### 6.1.2 Setup

Tests were carried out in a reverberant room with dimensions 5 m x 3.5 m x 3 m (length x width x height) and a reverberation time T60 of 0.15 s, as determined for a speech weighted noise spectrum. The test subjects were placed inside a semicircular array of 13 FOSTEX 6301 BX single-cone loudspeakers. The radius of the array was 95 cm. The speakers were located in the frontal horizontal plane at equidistant angles, ranging from -90° to +90° in steps of 15° and were labeled from 1 to 13. The FOSTEX speakers are of high quality, showing an almost flat frequency response, deviating only about 1 to 2 dB between different speakers. Both a free-field and a headphone localization experiment were conducted. For the free-field tests, signals were successively played through one of the 13 loudspeakers using two sound cards: a 7.1 Sound Blaster®Live! 24-bit PCI sound card (see Figure 1), combined with a 7.1 Sound Blaster Audigy2 NX USB sound card, both from Creative Labs. The tests were conducted using the ALP-HCOM software described in section 4.

### 6.1.3 Stimuli

Three different stimuli were presented during the test. A 200ms 1/3 octave low-frequency noise band ($f_c$ = 500 Hz), a 200 ms 1/3 octave high-frequency noise band ($f_c$ = 3150 Hz) and a 1s broadband telephone ringing signal. All target signals are cosine windowed with a rise and fall time of 50 ms. Stimuli were presented at 65 dB SPL. The test simuli that were used in the test can be found on

http://fit-bscw.fit.fraunhofer.de/bscw/bscw.cgi/36960320

For the headphone experiments the stimuli were filtered with pairs of Head-Related Impulse Responses (HRIR set 2a as described in section 5.3.1) that were recorded in the same room as where the listening tests were performed. The HRIR pairs can also be found on the same website as the stimuli.

### 6.1.4 Test protocol

Subjects were seated inside the array of 13 speakers. All the tests used three repetitions per speaker resulting in 39 presented trials per test. The stimuli were presented randomly and were roved with a roving level of 5dB. The subjects were instructed to keep their head fixed and pointed to 0° during stimulus playback. The task was to identify the speaker through which the target sound was heard. The stimuli were presented in two test conditions, with headphones (HD650) and in free field. The tests were done twice for test-retest purposes. One test took about 5 minutes, for the total test about one hour was required. Subjects could take a break whenever needed.

## 6.2 Results and analysis

A three-way analysis of variance with repeated measures was carried out on the rms data with the factors stimulus (telephone signal, low-frequency noise band and high-frequency noise band), test condition (free-field and with headphones) and test-retest. A significance level of 0.05 was used and a Bonferroni correction was applied to adjust for multiple comparisons. Average rms errors +/- standard deviations are given.

There is a main effect of stimulus (lower bound[2] p=0.019) and test condition (lower bound p=0.031). The factor test-retest is not significant (lower bound p=0.177). Moreover, none of the interactions are significant (always p>0.4). The subjects performed best with the broadband telephone signal (mean rms error of 6.6° +/- 2.7°), then the low-frequency noise band (mean rms error of 9.6° +/- 2.4°) and performance was worst with the high-frequency noise band (mean rms error of 17.5° +/- 6.3°). These results agree well with the study of Van den Bogaert et al (2006), which was done in a room with T60=0.5s (see table 1) and with a high-frequency noise band with a center frequency of 5000 Hz rather than 3150 Hz, as used in this study. The mean rms error is smaller for the free-field test condition (9.8° +/- 5.4°) than with headphones (12.6° +/- 6.7°), but the mean difference between both test conditions is only 2.8°. The mean difference in rms error between the test and retest condition is 0.5°.

[2] Lower bound is the most conservative test of within-subjects effects.

| | Current study | | Van den Bogaert et al, 2006 |
|---|---|---|---|
| | Free-field T60=0.15s | Headphones | Free-field T60=0.5s |
| Telephone signal | 5.6 +/- 2.0 | 7.6 +/- 3.0 | 6.8 +/- 3.1 |
| Low-frequency noise band | 8.5 +/- 1.4 | 10.7 +/- 2.7 | 13.5 +/- 2.1 |
| High-frequency noise band | 15.4 +/- 5.6 | 19.5 +/- 6.5 | 21.3 +/- 3.5 |

Table 1: Comparison of the current data with data obtained by Van den Bogaert et al. (2006) for normal-hearing subjects. Average rms errors +/- the standard deviations are given for three test stimuli. All data are in units of degrees (°). It should be mentioned that the high-frequency noise band has a center frequency of 3150 Hz in the current study and 5000 Hz in the study of Van den Bogaert et al. (2006).

# 7 Dissemination and Exploitation

## 7.1 Dissemination

This deliverable is a public report that will be made available through several internet sites.

The ALP-HCOM software platform described in section 4 and the hardware specifications of section 3 have been especially designed to offer a flexible, user-friendly and low-cost spatial-hearing test solution to clinical hearing centers and hearing research institutes. If combined with the appropriate test material and stimuli they provide all necessary tools to conduct proper spatial localization experiments.

The ALP-HCOM software and a representative set of test stimuli and Head-Related Impulse Response databases can be downloaded from the following website:

http://fit-bscw.fit.fraunhofer.de/bscw/bscw.cgi/36960320

## 7.2 Ethical issues

The presented software has been primarily designed to set up a spatial-hearing test with low-cost hardware. It provides a user interface to conduct spatial-hearing experiments in free field or via headphones.

Exposure to high-intensity sound fields can severely harm the ear and might cause temporary or permanent hearing loss. Be aware that the ALP-HCOM software does not provide automatic protection to prevent excessive sound level exposure. However, there are several features incorporated in the software that enable the user to control the loudness of the presented stimuli. It is hence the responsibility of the experiment supervisor to make sure that the sound levels cannot exceed the tolerated limits. Excessive sound levels can be avoided by appropriately calibrating the setup. This can be achieved by repositioning the loudspeaker volume knobs during calibration and by off-line scaling of the sound material. The software furthermore provides the `[overall gain]` parameter that can be lowered if too high sound levels occur. Additionally, the `[headphone calibration left]` and `[headphone calibration right]` parameters in the case of a headphone experiment, and the `[speaker compensation]` parameter in the case of a free-field experiment can be appropriately set to control the acoustic sound output.

# 8 Conclusions

Most of the spatial-hearing tests presented in the literature make use of high-quality hardware components (loudspeakers, amplifiers, sound cards) and dedicated software packages. They are typically designed and tuned towards the needs of the research institute for which they have been developed. As a consequence, spatial-hearing test setups are often expensive, too dedicated and specific, and are not readily commercially available.

These days, spatial-hearing and localization tests gain importance not only in research labs, but also in clinical environments. The assessment of the spatial-hearing abilities of hearing-impaired subjects is more and more often required for the optimal fitting of modern, advanced (bilateral) hearing-aid systems. Furthermore, knowledge of the spatial-hearing abilities of a subject provides additional information to the auditory profile. Given the high cost of many of the existing solutions and a lack of standardization it was our objective to develop a low-cost spatial-hearing setup that is intended for clinical usage. This deliverable presents and describes the proposed system both in terms of hardware and software.

Our solution makes use of consumer-electronics, hence low-cost hardware. Instead of relying on dedicated multi-channel A/D converters, we propose to use 5.1 or 7.1 surround sound cards, which are primarily designed for home cinema applications. As a consequence, they can generally be purchased at a low price in basically any PC shop. Furthermore, the approach is modular as several of these sound cards can be combined offering a true multi-channel setup. As far as the amplifier and the transducers are concerned, commercially available active PC loudspeakers can be used, which have an incorporated amplifier.

For the purpose of the intended application a new, dedicated software platform, called ALP-HCOM, has been developed. This user-friendly software package automatically recognizes the available sound cards in the PC and sets up the required communication with the hardware to play a desired sound file on one of the sound card channels. The software is furthermore capable of conducting a standard spatial localization experiment providing several user settings. Both a free-field experiment can be set up, which requires multiple loudspeakers installed in a room, as well as a headphone experiment, which relies on spatial virtual acoustics. Upon completion of the test a test result report is produced and some performance measures are computed.

Appropriate sound material is needed to conduct proper localization experiments. To this aim, sound files and Head-Related Impulse Response databases are made available, which can be used to set up free-field or headphone localization experiments in the ALP-HCOM software environment.

To check the validity of the approach, a representative low-cost 7.1 surround sound card has been purchased and evaluated. The conducted performance measurements proved that readily available low-cost surround sound cards easily meet the technical specifications required for sound localization performance assessment. In order to evaluate the accompanying software environment and measuring procedure a spatial localization setup was constructed by combining several surround sound cards and active loudspeakers following the specifications presented in this report. The setup has been validated by listening experiments and met our expectations.

# 9 Literature


- Blauert, J. (1997). *Spatial hearing, the psychophysics of human sound localization*. The MIT Press, Cambridge, MA.
- Bronkhorst, A. W., and Plomp, R. (1998). “The effect of head-induced interaural time and level differences on speech intelligibility in noise,” J. Acoust. Soc. Am. 83(4), 1508–1516.
- Bronkhorst, A. W., and Plomp, R. (1989). “Binaural speech intelligibility in noise for hearing impaired listeners,” J. Acoust. Soc. Am. 86(4), 1374– 1383.
- Byrne, D., Sinclair, S., and Noble, W. (1998). “Open earmold fittings for improving aided auditory localization for sensorineural hearing losses with good high-frequency hearing,” Ear Hear. 19(1), 62–71.
- Carlile, S., Leong, P., and Hyams, S. (1997). “The nature and distribution of errors in sound localization by human listeners,” Hear. Res. 114, 179–196.
- Dillon, H. (2001). *Hearing aids*, Boomerang Press, Sydney.
- Dillon, H., Keidser, G., and Silberstein, H. (2003). “Sound quality comparisons of advanced hearing aids,” Hear. J. 56(4), 1–6.
- Kuhn, G. F. (2003). “Model of the interaural time differences in the azimuthal plane,” J. Acoust. Soc. Am. 62(1), 157–167.
- Gilkey, R. H. and Anderson, T. R. (1997). *Binaural and spatial hearing in real and virtual environments*. Lawrence Erlbaum associates, Mahwah, NJ.
- Hartmann, W. M. (1999). “How we localize sound,” Phys. Today 11, 24– 29.
- Hausler, R., Colburn, S., and Marr, E. (1983). “Sound localization in subjects with impaired hearing,” Acta Oto-Laryngol. 400, 1–62.
- HearCom deliverable D-1-2 (2005), “Report on the proposed set of communication performance test”
- HearCom deliverable D-1-3 (2006), “Protocol for implementation of communication tests in different languages”
- HearCom deliverable D-2-1 (2005), “Implementation of a preliminary test set for auditory impairments”
- HearCom deliverable D-2-1b (2006), “Demo version of the preliminary test set for auditory impairments”
- HearCom deliverable D-7-1 (2005), “Speech recognition tests for different languages”
- HearCom deliverable D-7-2 (2006), “Specification on ‘standard speech test’ and ‘standard environmental conditions’ to be used for evaluation purposes within Europe”
- Hofman, P. M., and Van Opstal, J. (1998). “Spectro-temporal factors in two-dimensional human sound localization,” J. Acoust. Soc. Am. 103(5), 2634–2648.
- Langendijk, E. H. A., and Bronkhorst, A. W. (2002). “Contribution of spectral cues to human sound localization,” J. Acoust. Soc. Am. 112(4), 1583–1596.

- Lorenzi, C. S., Gatehouse, S., and Lever, C. (1999a). “Sound localization in noise in hearing impaired listeners,” J. Acoust. Soc. Am. 105(6), 3454– 3463.
- Lorenzi, C. S., Gatehouse, S., and Lever, C. (1999b). “Sound localization in noise in normal hearing listeners,” J. Acoust. Soc. Am. 105(3), 1810– 1820.
- Makous, J. C., and Middlebrooks, J. C. (1990). “Two-dimensional sound localization by human listeners,” J. Acoust. Soc. Am. 87(5), 2188–2200.
- Moore, B. C. J. (1997a). *An introduction to the psychology of hearing*, Chap. 8: Spatial hearing and related phenomena, Academic, New York, 297–345.
- Moore, B. C. J. (1997b). *An introduction to the psychology of hearing*, Chap. 7: Space perception, Academic, New York, 223–267.
- Noble, W., Sinclair, S., and Byrne, D. (1998). “Improvements in aided sound localization with open earmolds: Observations in people with high-frequency hearing loss,” J. Am. Acad. Audiol 9(1), 25–34.
- Noble, W., Byrne, D., and Lepage, B. (1994). “Effects on sound localization of configuration and type of hearing impairment,” J. Acoust. Soc. Am. 95(2), 992–1005.
- Noble, W., and Byrne, D. (1990). “A comparison of different binaural hearing aid systems for sound localization in the horizontal and vertical planes,” Br. J. Audiol. 24, 335–346.
- Stevens, S. S., and Newman, E. B. (1936). “The localization of actual sources of sound,” Am. J. Psychol. 48, 297–306.
- Van den Bogaert, T., Klasen, T. J., Moonen, M., Van Deun, L., Wouters, J. (2006). “Horizontal localization with bilateral hearing aids: Without is better than with,” J. Acoust. Soc. Am. 119 (1), 515-526
- Van Hoesel, R. J. M., and Tyler, R. S. (2003). “Speech perception, localization, and lateralization with bilateral cochlear implants,” J. Acoust. Soc. Am. 113(3), 1617–1630.
- Van Hoesel, R., Ramsden, R., and O’Driscoll, M. (2002). “Sound direction identification, interaural time delay discrimination, and speech intelligibility advantages in noise for a bilateral cochlear implant user,” Ear Hear. 23(2), 137–149.
- Wightman, F. L., and Kistler, D. J. (1992). “The dominant role of low-frequency interaural time differences in sound localization,” J. Acoust. Soc. Am. 91(3), 1648–1661.