OBJECT FUSION AND LOCALIZATION DOMINANCE IN REAL-TIME SPATIAL PROCESSING OF ACOUSTIC SOURCES USING HIGHER ORDER AMBISONICS

Giso Grimm, Volker Hohmann and Stephan Ewert

AG Medizinische Physik Carl-von-Ossietzky Universität Oldenburg D-26111 Oldenburg, Germany g.grimm@uni-oldenburg.de

ABSTRACT

The psycho-acoustic properties *fusion* and *localization dominance* were measured for a rotating target and a fixed-position distractor, which was a delayed copy of the target. Higher-order Ambisonics was used for spatial presentation of the target. The relative delay between distractor and target, the listener position within the playback system and the angular speed of the target were varied. For measurement of the localization dominance a pointer device was developed, which allows measurement of the perceived direction in real-time, synchronized to the target motion.

The aim of this study was to find the limitations of the higherorder Ambisonics setup regarding the position of listeners and sources, as well as the potential influence of source speed and continuity on these limitations. A small effect of continuity on the breakdown of localization dominance was found. The results of this study are qualitatively in line with the predictions of the precedence effect. They can directly be used to optimize an artistic concert installation where acoustic sources are processed and presented as virtual moving sources. The setup is also suited for new hearing aid evaluation methods.

1. INTRODUCTION

In many conventional concert amplification and public address systems, the precedence effect is considered and utilized to achieve an appropriate localization. However, in a concert setup with dynamic real-time spatial processing of acoustic sources and presentation through a multi-speaker presentation technique, the influence of the precedence effect is unclear.

The precedence effect is an important phenomenon in spatial acoustics. The first major studies on the precedence effect with a detailed description were made more then 60 years ago [1]. Fifty years later, Litovsky et al. [2] gave a comprehensive overview over the precedence effect and previous studies on that topic. They described the precedence effect to consist of mainly three phenomena: Fusion, localization dominance, and discrimination suppression. In a later study, Litovsky and Shinn-Cunningham [3] compared the three phenomena fusion, localization dominance and discrimination suppression in the same subjects. However, as in most other studies, they used headphones for stimulus presentation. While some other studies [1, 4, 5] also used loudspeaker presentations in anechoic rooms, and another study used virtual

acoustics presented via headphone [6], no reference was found for a systematic evaluation of precedence effect phenomena in virtual acoustics using loudspeaker presentation.

Complex spatial presentation of sounds plays an increasing role in many audio-related fields. Several techniques can be used for this purpose. Ambisonics is a technique that offers high quality spatial presentation with relatively small computational and technical complexity [7]. Using higher order Ambisonics has the advantage of good spatial precision even for off-center listening. Most studies of precedence effect and Ambisonics are related to the speaker layout [8], room acoustics [9], or room simulation [10]. A study on application of Ambisonics in concerts exist [11], but the impact of precedence effect was not discussed.

This study aims to address the gap between fundamental psycho-acoustic research on the precedence effect and the application of virtual acoustics. The study prepares the ground for further studies in this area and provides direct input for the preparation of a concert setup with real-time spatial processing of acoustic sources as well as for an evaluation setup for hearing aids.

2. METHODS

In the following sections, the apparatus, technical methods and subjective measures used in this study are described.

2.1. Apparatus

The test setup (see Fig. 1) consisted of an third-order Ambisonics system with ten loudspeakers arranged on a circle in the horizontal plane. The highest possible order for a horizontal speaker layout would be four, however, due to limitations in the decoder software only a third-order Ambisonics system was chosen. One of the ten speakers was also used to play a distractor stimulus, which was a delayed version of the target stimulus. Its virtual distance was adjusted by the relative delay between the target and distractor. The test subject was seated within the circle, on two alternative positions between the center and the distractor speaker. The circle diameter was 5 m, and the listener position was 1 and 2 m from its center, respectively. Small head movements were allowed, to improve localization. The room of the setup (see Fig. 2) was an acoustically treated medium conference room, approximately 7 m wide and 13 m long (Communication acoustics simulator, with the active acoustics switched off [12]). The reverberation time T_{60} was about 0.4 s at all frequencies.

The audio signal was processed by a personal computer running a Linux operating system optimized for low-delay audio pro-

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Figure 1: Schematic setup of test system. Ten speakers were arranged on a circle with a 5 m diameter. The test subject was seated at position 1 and 2 (1 and 2 m from the center), respectively. The moving target source was played back via the Ambisonics system, and was rotating counter-clockwise. The distractor source was played from one of the speakers and its distance was accounted for by different delays. Subjects indicated the direction of the moving source by a hand-held pointer device.



Figure 2: Photograph of the speaker setup in the measurement room.

cessing. The test stimuli were digitally sampled with a rate of 48 kHz, processed by the spatial processing software, and converted to the analog domain using an RME HDSP9652 sound card with two synchronized Behringer ADA8000 DA and AD converters. For playback, ten identical active studio monitors (KRK ROKIT RP5) were used. Simultaneously five microphone signals were recorded for tracking the position of a handheld pointer device used by the subjects. The HörTech Master Hearing Aid [13] was used as real-time software platform to create the virtual sound sources in Ambisonics format, to control the delay of the distractor, and to estimate the pointer position. The Ambisonics format was decoded to the specific loudspeaker layout with the open source application AmbDec [14], developed by Fons Adriaensen. The measurement was controlled by a MATLAB application.

2.2. Pointer device

In order to measure the localization dominance, subjects indicated the perceived dominant direction of the sound source by pointing with a hand-held pointer device. The pointer consisted of a small ultrasonic loudspeaker on the tip of a short stick, continuously playing log sine sweeps in the frequency range from 17 to 24 kHz [15]. It was verified that the subjects could not hear this sound unless they pointed the device directly towards their ears. The sound was picked up by five microphones, arranged on the edges of a pentagon. The distance between the pointer tip and the five microphones was calculated at the frame rate of the block processing in the real-time software (23.4 Hz). The delays were found by searching the maximums of the impulse responses from the loudspeaker to the microphones. The impulse responses were cut after 17 ms to reduce the influence of room reflections. This corresponds to a maximal distance between pointer and microphones of 5.8 m. If distance jumps of more than 0.2 m per frame (4.7 m/s) occurred, the values were replaced by the median of two previous and two later distances measures. The position of the pointer z_p on the complex plane was calculated for all ten possible combination of microphone pairs at the positions z_1 and z_2 , with the corresponding distances d_1 and d_2 :

$$\alpha = \frac{\arcsin(|z_2 - z_1|^2 + d_1^2 - d_2^2)}{2|z_2 - z_1|d_1}$$
$$z_n = z_1 + d_1 e^{i(\angle (z_2 - z_1) + \alpha - \pi/2)}$$

It was assumed that the pointer was on the plane spanned by the microphones. In the first step, a most likely position was calculated by taking the median x and y positions of all ten position estimates. In a second step, the average of the five positions closest to the most likely position was calculated and used as a final position. The position information was converted into a direction by taking the difference to an individually measured reference point, see also section 3.1 for details. Errors caused by this conversion from position to direction were the largest limitation in the use of the pointing device.

2.3. Stimuli

The effects of fusion and localization dominance was investigated with two different stimuli: One stimulus was white noise, band-limited to the frequency range from 180 Hz to 7 kHz with a 7th order Butterworth-filter. The noise was sinusoidally amplitude modulated with 4 Hz and a modulation index of 1. The second stimulus was a short musical sequence, played on one string instrument, and was also band-limited to the same frequency range. Although the modulation depth was lower, the temporal structure of the rhythmic part of the music stimulus. The signal power in third-octave bands is shown in Fig. 3, the spectrogram is shown in Fig. 4. Both stimuli were played at 67 dB SPL (unweighted) at the center of the circle. The overall level at the listener positions as a function of the direction is given in Fig. 5.

2.4. Parameter set

The influence of three factors on the precedence effect was assessed. The factors were (i) the relative attenuation of the two sources, (ii) the delay (or virtual distance) of the two sources, and (iii) the angular speed ω of the target movement. Accordingly



Figure 3: Signal power of test stimuli in third-octave bands. The noise stimulus was a sinusoidally amplitude modulated (4 Hz) white noise, band-limited to 180 Hz to 7 kHz. The music stimulus was a short recording of a string instrument.



Figure 4: Spectrogram of modulated noise (upper panel) and music stimulus (lower panel).

three measurement parameter sets, and an additional anchor parameter set were employed. A summary of the four parameter sets is given in Table 1.

In the anchor parameter set (S_{anchor}) either the distractor or the target was played, respectively. This parameter set was required for calibration of the pointer device and to ensure that all subjects were generally able to locate a virtually moving sound source presented with the Ambisonics system.

In the second parameter set (S_{att}) , the attenuation of the distracting source relative to the target source was varied. The virtual distance of the distracting source was 0 m, which means that the distracting source was played simultaneously (delay = 0 ms) with the target source for a target direction of 0 deg and was leading for all other target directions. This condition was tested only at the listener position one.

In the next measurement parameter set (S_{dist}) , the virtual distance of the distracting source was varied by varying the relative delay, at a fixed level ratio between the distracting and target source. This paradigm is closest to classical measurements of the precedence effect, where the delay between the lead and the lag stimulus is the main parameter. This condition was tested at listener position 2. The nominal attenuation of the distracting source was 0 dB, however due to the small distance between the physical loudspeaker and the listener, the actual level ratio between distracting and target source varied significantly with the target direction.

The fourth measurement parameter set (S_{ω}) was designed to investigate the influence of the angular speed ω of the moving target source on the precedence effect. The rotation period was varied between 5 s and 20 s. The target and distractor source had the same sound pressure level. The virtual distance of the distracting source was 1 m, which means that the distracting source was lagging (delay = 2.9 ms) for a target direction of 0 deg and was leading by the same amount for a target direction of 180 deg.

In each trial, the target source rotated two full circles counterclockwise. In all parameter sets except for the set S_{ω} , the angular speed ω was 0.628 s^{-1} (rotation period = 10 s). The data were analyzed for the second rotation only. The measurement was performed in one block for each task and each listener position. The duration of each block was between 7 and 11 minutes, with a short break between the blocks. Within a block, the parameter sets and all trials were played in randomized order. Written instructions were handed to the test subjects, with the possibility to ask questions. Before the first block, a set of sound examples was presented to the listener.

2.5. Subjective measures

Localization dominance was assessed with the help of a pointing device: The test subjects were asked to hold the pointer towards the dominant direction of the source. The target localization performance p of subjects in one trial was defined as the vector strength of the difference between nominal target direction φ and estimated direction $\hat{\varphi}$:

$$p = \left| \left\langle e^{i(\varphi - \hat{\varphi})} \right\rangle \right| \tag{1}$$

The average $\langle \cdot \rangle$ was taken over the last full rotation of the virtual sound source movement. With this definition, constant differences like systematic estimation errors and a lag between estimated and nominal target source are ignored. Values of p near one are reached if the difference between nominal and estimated target direction was constant, i.e., the target source was tracked with the pointer. Values near zero mean that the difference between target and pointer direction are time dependent, i.e., the pointer was at a fixed position throughout the trial. Intermediate values indicate that the source was partly tracked.

To assess the effect of *fusion*, the subjects were asked to push a control button whenever they heard two sources, and to release the button as soon as they heard only one source. The button state was recorded continuously together with the current target source direction. As a scalar measure of fusion, the control button state was averaged across the last full rotation of the virtual sound source. The average value ranges between 1 and 2, indicating the perception of one or two sources.

	S_{anchor}	S_{att}	S_{dist}	S_{ω}
Listener position / m	1, 2	1	2	1
Stimulus	noise, music	noise, music	noise, music	noise, music
Attanuation / dB	-60, 60	-20, -15, -10, -5, 0, 5	0	0
Distance / m	0	0	0, 1, 2, 3, 4, 5, 8, 12	1
Period / s	10	10	10	5, 6.67, 10, 20
Conditions	8	12	16	8

Table 1: The four measurement parameter sets (see text for details). The last row indicates the resulting number of conditions per parameter set.



Figure 5: Level in dB SPL of target and distracting source at listener position 1 (left panel) and 2 (right panel), as a function of target direction. The target-to-distractor level ratio (SNR) is the difference between the solid and the dashed line.

2.6. Test subjects

Ten normal hearing listeners participated in this study, three female and seven male. The average age was 32.7 years, with a standard deviation of 6.9 years. Most of the subjects had previous experience with psycho-acoustic tests. One of the subjects was the main author. The subjects not affiliated with the medical physics group received a compensation for their participation on an hourly basis.

3. RESULTS

3.1. Anchor condition and calibration

For validation of the setup calibration, the sound pressure level of the target source was measured at both listener positions as a function of direction, see Fig. 5. At listener position one, the level was maximal for a target direction of 0 deg (68.7 dB SPL), and minimal for a target direction of 150 deg (65.2 dB SPL). The SNR is here defined as the target-to-distractor level ratio. The SNR, defined as target-to-distractor level ratio, thus varied between -0.3 and -3.8 dB (depending on the direction) for a nominal distractor attenuation of 0 dB. For the second listener position, the minimal and maximal level was reached at the same directions (73.2 dB and 64.1 dB, respectively). Here, the level of the distracting source was 75.6 dB SPL, and thus the SNR varied between -2.4 dB and -11.5 dB.

The pointing device was calibrated with the data recorded in the anchor parameter set with the moving target without distracting source. The listeners could clearly follow the moving source. All subjects moved the pointer more or less on a circle. However, the center of the circle and the radius showed large inter-individual differences. Thus for each listener, the center and radius of a circle was fitted to the individual pointer trajectory. The pointer positions and the individual center was then used to calculate the pointer direction estimate in all other conditions. The difference between estimated pointer direction and nominal target direction in degrees is shown in Fig. 6, averaged across subjects. The data of the noise stimulus is shown as a solid line, and the music stimulus as a dashed line.

3.2. Attenuation of distracting source

The effect of the level ratio between target and distractor was measured by attenuating either the distractor (negative nominal attenuation values) or the target (positive nominal attenuation). In this measurement, the delay between target source and distracting source was zero. The delay at the listener position depended on the target direction, and varied between 0 ms for 0 deg target direction and 5.9 ms delay of the target for 180 deg target direction. The results are shown in Fig. 7. The data in the left panel show the localization performance p as a function of the distractor attenuation. It can be observed that within a range of 10 dB (-5 dB to -15 dB), the localization performance jumps from about zero to almost 100 percent. The right panel of Fig. 7 shows the detection rate for two sources. The data indicates that most subjects noticed two sources for at least part of the target source rotation at a distractor attenuation of -10 dB (SNR of 10 dB). At distractor attenuation values below -10 dB the target source was dominating, and only one source was noticed. At distractor attenuation values of 5 dB for the noise stimulus and -5 dB and above for the music stimulus, only the distracting source is detected. The average number of recognized sources as a function of target direction is shown in Fig. 8. For intermediate SNRs a second source was slightly more often detected if the moving source came from the left hand side.

3.3. Distractor distance

Fig. 9 shows the effect of distractor distance on localization performance and detection rate for a second source. In the left panel only little effect on the localization performance is observed. Only for the largest distance of 12 m, corresponding to a delay of 35.3 ms for 0 deg target direction and 23.5 ms for 180 deg target direction, the target source was tracked by at least a quarter of the listeners, as indicated by the quartile bars. However, as can be observed in the right panel, the fusion of target and distractor broke down for distances above 5 m, corresponding to a distractor delay of 13.7 ms at 0 deg and 2.9 ms at 180 deg. For distances of 8 and 10 m most



Figure 6: Pointer performance (difference between estimated pointer direction and nominal target direction) in the anchor setup, averaged across test subjects. In the upper panel, the data for listener position 1 is shown, in the bottom panel for position 2. With a thin solid line, the corrected target direction (listener positions were off the center of the circle) for the respective listener position plotted. The gray area denotes ± 1 standard deviation.

subjects detected two sources for the biggest part of the rotation, with the noise source. For the music stimulus, most subjects noticed two sources only at about 70% of the rotation.

3.4. Period time

The effect of period time on localization dominance and fusion was less clear than the effect of attenuation and distance. Fig. 10 shows the results in the same style as in Fig. 7 and 9. The interindividual differences are much larger than in the other parameter sets. As a trend, it can be observed that localization performance (left panel) was maximal for target rotation durations of 6.7 and 10 s ($\omega = 0.942 \, s^{-1}$ and $0.628 \, s^{-1}$, respectively), and decreasing for larger and shorter rotation durations. The localization performance was smaller for music than for noise. For music also the detection of the second source (right panel) was less often than for noise (not detected by most subjects).



Figure 7: Results of localization performance (left panel) and detection rate of second source (right panel) as function of distractor attenuation. Median values with lower and upper quartiles are shown.



Figure 8: Average number of recognized sources as function of target direction. Left panel was the noise stimulus, right panel the music. The second source was more likely to be detected if the target came from the left side, i.e., the distance between distractor and target was larger.



Figure 9: Localization performance (left panel) and detection rate of second source (right panel) as function of distractor distance. Again, median values with lower and upper quartiles are shown.

4. DISCUSSION

4.1. Pointing accuracy

The localization performance was measured with a pointer device. Its position was recorded continuously. However the interpretation of absolute pointer position as a pointing direction is unclear, and



Figure 10: Localization performance (left panel) and detection rate of second source (right panel) as function of duration for full target rotation. Median values with lower and upper quartiles are shown.

it turned out that it substantially varied across subjects. This was partly caused by different individual reference positions and rotation centers of the handheld pointer device. Additionally, subjects reported difficulties to point steadily behind the head. Some subjects swapped the hands used to hold the pointer during a trial. Despite these problems with the pointer device, on average the virtual sound source could be tracked very well. Larger errors occurred on the right hand side where the distractor speaker was placed. Reasons for this might be (a) that the physical distance to the subject was low and hindered a free pointer movement, and (b) that the anchor condition was played randomized within all other trials. The subjects might have learned that a fixed source could be expected from the right side, and tended to stick to that direction until a different direction became obvious.

For an Ambisonics plavback system, the perception of a nominal direction at off-center listener positions might be frequency dependent: At low frequencies, a planar wave field can be synthesized, and the perceived direction is equal to the nominal direction. However, at high frequencies the wave field synthesis does not work outside the sweet spot of the system, and the direction of maximum signal energy is located. For off-center listener position this results in an angular difference. In Fig. 6, the difference to the nominal direction is shown. In addition, the difference of the energy based direction at the respective listener position is plotted as a thin solid line. For the back-facing hemisphere, on average the energy based direction was estimated. For the frontal hemisphere the estimated direction is closer to the nominal direction. This might be an indication that the reproduction of a planar wave field works only for frontal direction. However, due to the large localization errors in the direction of the distractor source the data can not reliably support this indication.

The physical resolution of the pointing device could be increased by reducing the directivity of the ultrasonic loudspeaker. With the current device, the sound beam was highly directional, which led to a reduced signal-to-noise ratio at those microphones not reached by the beam and to a larger influence of room reflections. With an omnidirectional device, the shadow effect of the listener could also be reduced. A simple solution would be to place additional loudspeakers at the side of the stick.

An alternative to the acoustically tracked pointer tip would be an optical (laser) pointer tracked by cameras. The advantage of such a pointer would be that the pointer position can be interpreted as a direction, and the pointing resolution would be higher than with the acoustic pointer, because the subjects have visual feedback of the pointing direction. However, the visual feedback limits the usable range of an optical pointer to the visual plane of the subjects.

4.2. Level effects

One outcome of this study is that the effect of level ratio between target and distractor is larger than expected. A reason for this might be that most precedence effect related studies presented the stimuli either via headphones, or via distinct loudspeakers in anechoic environments. In this study, virtual acoustics has been used to create the target source. This might cause a reduced temporal precision for listener positions outside the sweet spot of the system. Additionally, room reflections might contribute to a reduced effect of the principle of the first wavefront. Furthermore, the test setup calibration was done in a way that the distractor was louder than the target in most conditions. For a better evaluation of the temporal effects, this should be considered in further measurements.

4.3. Continuity and localization dominance

The assumption that continuity of motion leads to a more robust localization even in conditions where the precedence effect would predict localization of the distractor was supported. The measured effects, however, are small, and require further investigation. Specifically, it is not clear from the presented results whether the effect of a reduced localization accuracy at high rotation speeds was caused by a decreased localization perception, or rather by limitations of pointing correctly at higher speeds. The underlying mechanisms are not yet clear.

4.4. Relevance for hearing aid evaluation

In hearing aid technology spatial processing and performance in the three-dimensional space plays an increasingly important role, partly because of the availability of binaural processing and advanced multi-microphone techniques, but also because of an increased awareness of the problems and downsides of directional processing. The pointing device developed in this study can be used for evaluating full three-dimensional localization performance and similar tasks in hearing aid research. Together with a higher-order Ambisonics playback system this opens paths to new evaluation strategies.

The results of localization performance, specifically the continuity aspects of localization, should be considered in the improvement of direction-of-arrival estimators.

5. SUMMARY AND CONCLUSIONS

In this study two aspects of the precedence effect, *localization dominance* and *fusion*, have been explored using an Ambisonics system to present a rotating sound source. In contrast to most literature data the localization dominance was measured by tracking a moving source. The results are qualitatively in line with literature data on the precedence effect. The level ratio between distractor and target source was shown to be a dominating factor: At level differences of about 10 dB or more, the louder source was always dominating the localization. This holds even for extreme target-distractor delays. The effect of fusion was larger for music than

for the modulated noise stimulus, which was expected due to less pronounced envelope fluctuations in the music. Although the effect of angular speed of the rotating target source on localization dominance was small, it might give evidence that moderate angular speeds in the range of 0.6 to 0.9 s^{-1} increase the localization performance compared to higher or lower speeds.

The conclusion for a practical application in a concert setup with real-time spatial processing of acoustic sources is to provide sufficient amplification on the electrically presented target source, to avoid localization dominance of the acoustic sources. If the target source is weaker than the distracting acoustic source, then even large distances won't improve the localization. However, if the levels are chosen properly the localization performance can be good even for listener positions extremely close to the loudspeakers. The effects of angular speed on localization performance should be considered in the design of spatial processing concepts.

The suggested test setup including the handheld pointer device have generally proven their suitability for estimation of rotating virtual sound sources. The setup can be used for evaluating full three-dimensional localization performance and similar tasks in hearing aid and psychoacoustic research.

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