AMBISONICS SYMPOSIUM 2009
June 25-27, Graz

MEASURING DIRECTIVITIES OF NATURAL SOUND SOURCES WITH A SPHERICAL MICROPHONE ARRAY

Martin Pollow, Gottfried Behler, Bruno Masiero
Institute of Technical Acoustics, RWTH Aachen University, Germany
Email: {mpo,gkb,bma}@akustik.rwth-aachen.de

Abstract: Ambisonics recording technique is based on a small microphone array, whose microphones are placed as close as possible to each other. This array will record with a certain degree of spatial information the sound field propagating in its direction. The arriving sound field can be depicted as a type of incoming spherical-like wave and this type of recording technique is ideal if the spatial characteristics of the sound field from the point of view of the listener is of interest. Now, from an opposite point of view, the point of view of a sound source; it is the spatial attributes of the sources outgoing spherical-like wave that should be recorded, leading to the need of a larger microphone array that completely encompasses the sound source. Such an array, built with 32 microphones regularly distributed in a spherical surface, was used to measure the directivity of musical instruments. To accomplish this task, musicians were invited to play their instruments inside the microphone array. The synchronously recorded signal from the 32 microphones is then processed to deliver the instruments directivity. For perfect results the sound source should be placed in the center of the array. However, due to the physical size of the instruments and practical limitations, this is an impossible task to fulfill, yielding distortions on the results of the measurement depending on the type of processing done.

Key words: directivity, musical instruments, microphone array, spherical, room impulse response, measurement

1 INTRODUCTION

Most usually natural sound sources such as musical instruments are auralized as spherical point sources, obviously neglecting their directivity patterns. To improve the authenticity of these auralizations, the room-impulse-response should be obtained with a sound source that radiates sound with the same directivity pattern as the natural sound source at issue (cf. [1]).

The first step to achieve this goal is, of course, to measure the directivity pattern of the natural sound source as accurate as possible. As these sources cannot reproduce the excitation signals usually applied in acoustical measurements (like random noise or sine sweeps), the measurement has to be done with the sounds emitted by the source. However, these natural sounds don't cover the entire frequency range equally, are irreproducible and might vary depending on the pitch, the loudness and the style of playing. This leads to the requirement of simultaneous measurement with a microphone array and some post-processing of the recorded signals. Furthermore, the exact location of the source can usually not be stated. The hereby occurring displacement from the ideally centered source can enhance aliasing errors caused by an insufficient sampling grid on the sphere, especially at higher frequencies.

To turn theory into application, a lightweight spherical array with 32 regularly distributed microphones was constructed. Placed in a full-anechoic chamber, it allows to record the musical instrument from different directions without any disturbing reflections. Several musical instruments, both those used in a comtemporary symphonic orchestra, as well as various historical instruments, were recorded in this way.

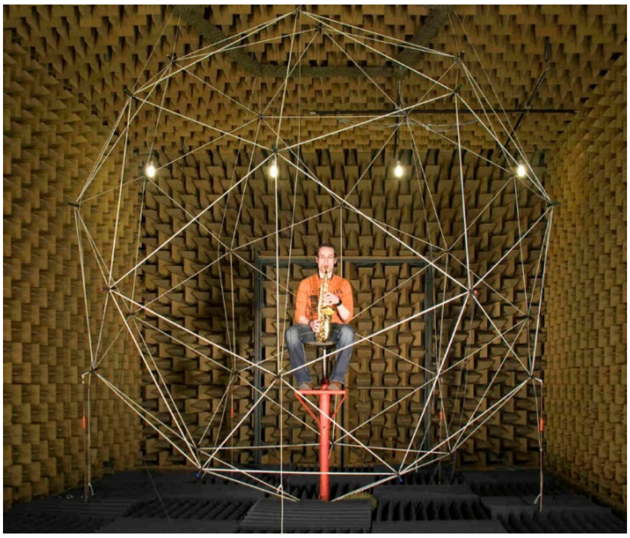

Figure 1: Spherical microphone array

2 SETUP OF THE EQUIPMENT

2.1. Spherical microphone array

The constructed spherical microphone array places 32 microphones regularly distributed on a sphere with a diameter of approximately 4.2 m surrounding the musician (Figure 1). To stay in an acceptable budget range, reasonably priced electret microphones with a flat frequency response were used (Sennheiser KE4-211-2, Figure 2). As recordings done with that type of microphones are too noisy to be used as input material for high quality auralization, additional studio microphones can be used to create additional high quality audio tracks from the recording.

Figure 2: One microphone at each joint of the spherical grid

2.2. Setup of the microphones

All of the 32 microphones need to be supplied with a phantom source which was supplied by four 8-channel pre-amplifiers (RME OctaMic). Three of these devices were connected optically with a RME Digiface, while the fourth pre-amplifier was also optically connected to a RME Multiface. Both Digiface and Multiface were linked to a PC running Nuendo as recording tool. Aditionally, two digital microphones (Neumann Solution-D) were placed inside of the array to record a stereo track in studio quality. The complete recording setup is shown in Figure 3.

2.3. Measurement in full anechoic chamber

When working with technical sound sources, it is relatively easy to filter out unwanted reflections present in a measured impulse response. But in the case of natural sources such unwanted reflections cannot be so easily filtered out. The Institute of Technical Acoustics in Aachen owns a considerably large hemi-anechoic chamber, as can be seen in Figure 1. A first round of directivity measurements was performed in this chamber, with the highly reflective floor surface covered with absorbing mats. It was though later verified that the reduction in reflection levels provided by the absorbing mats was not high enough for this measurements, most notably at frequencies below 500 Hz. The whole setup was then assembled for a second round of measurements at the Department for Audio Communication of the TU Berlin, which owns a large full-anechoic chamber, eliminating then the influence of any occurring reflection on the measurements. Musicians were again invited to play on an elevated chair inside the microphone sphere. They were positioned in a way that the instrument's expected acoustical center is close to the center of the spherical array. Of course, depending on the size of the instrument and the practical possibilities for reajustment of the elevated chair the alignment to the center can be approximated better or worse.

The musicians were asked to play single tones of a chromatic scale in two different dynamic ranges while avoiding movements as much as possible. The result is a time-continous recording with 32 channels for the spatial information.

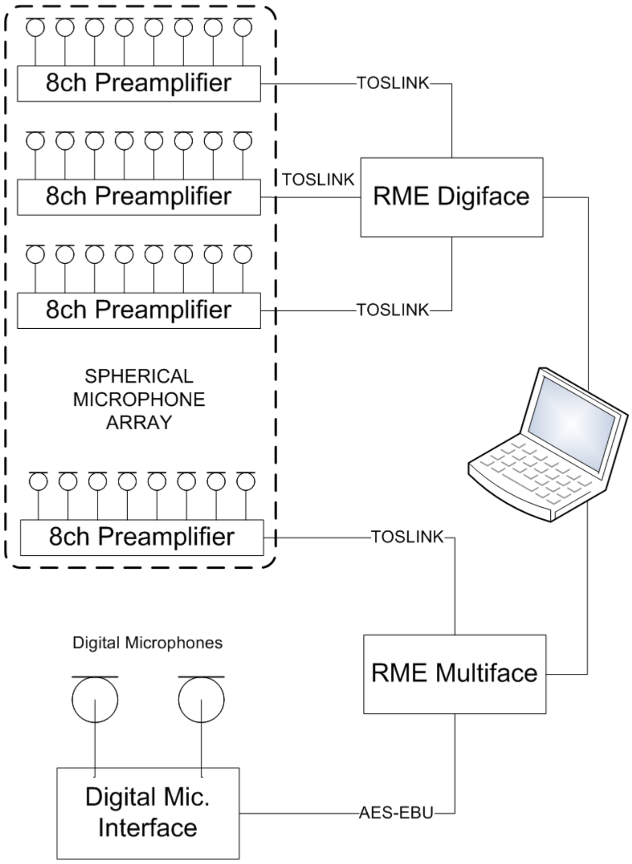

Figure 3: Connecting the microphones to the recording device

Figure 4: Frontal view of trumpete player inside the array

Figure 5: Side view of trumpete player inside the array

3 PROCESSING OF THE DATA

There are different possibilities to process the measured data in order to obtain the directivity pattern of the instrument under investigation. The spectrum of each tone played by the musician can be calculated for each microphone belonging the array. This set of spectral-spatial information can then be interpolated on a continous sphere using the spherical harmonics decomposition [2, 3]. The set of spherical harmonic coefficients used should be conveniently limited.

A problem arises if the radiated sound does not emerge from the center of the array, as different travelling times from the actual sound source to the microphones yield phase displacement. Assuming the source wrongly in the center, the complex valued radiation pattern of the instrument seems to get much more complex than it would be if correctly centered. The higher the displacement and the higher the frequency, the faultier the results are. This phenomenon is called spatial aliasing and is caused by an insufficient spatial resolution of the microphone distribution [4]. As every instrument has certain dimensions, it is difficult to state its acoustical center. Furthermore, some instruments may have different acoustical center points for different tones, making an exact positioning of a modeled point source impossible.

However, there is another way to minimize the artifacts caused by misplacement: Leaving all phase information aside and using only the magnitudes we only have to deal with the $1/r$-spread of a spherical wave if the source is displaced. Compared to the complex valued computation, this enhances the stability of the measured directivity pattern if displacements are occurring. The practical impact of this simplification, however, still has to be evaluated by listening tests.

In Figure 6 to Figure 9 the radiation of a trumpet playing a A4 (443 Hz, playing in the direction of the positive x-Axis) is depicted in different ways. Figure 6 shows the spatial distribution of the energy of the half tone A4 in a logarithmic scale; it assumes the sound source to be centered and takes the average radiated energy as the reference for 0 dB. The colored dots on the balloon plot express the measured value of the sound pressure level on the microphones. According to the radius information of the balloon plot, the radius of the location of the dots is chosen to be the value on that microphone. A dot shaped

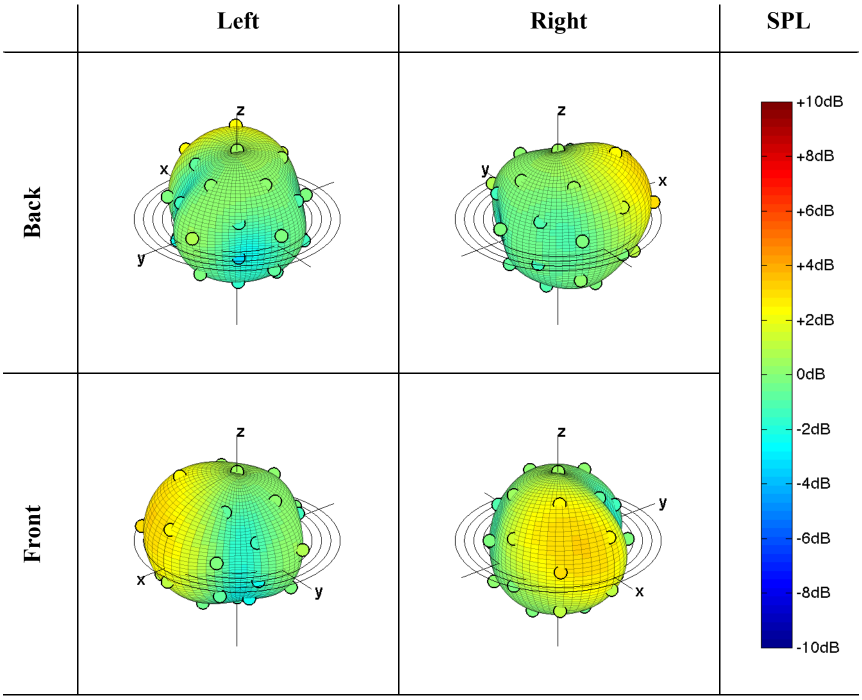

Figure 6: Radiated sound pressure level of the fundamental frequency for a trumpet playing A4 (443 Hz, source assumed to be in the center of the array)

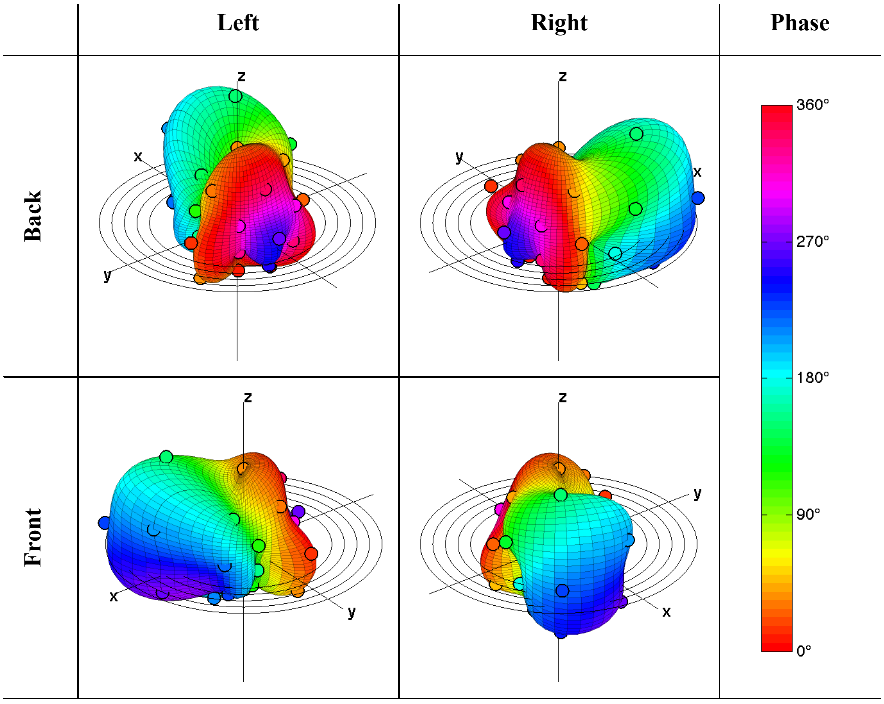

Figure 7: Radiated complex sound pressure of the fundamental frequency for a trumpet playing A4 (443 Hz, source assumed to be in the center of the array)

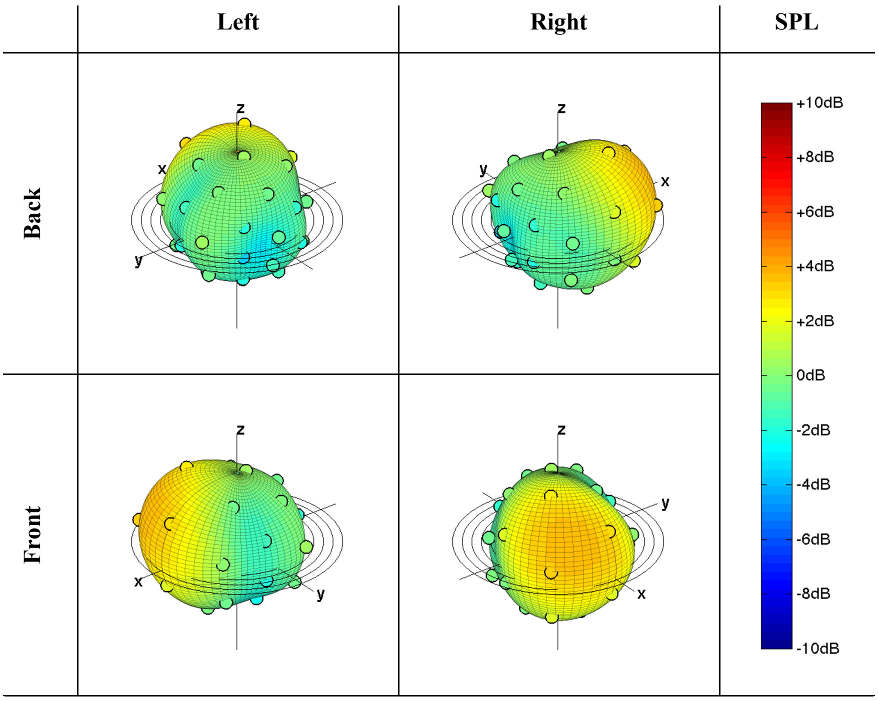

Figure 8: Radiated sound pressure level of the fundamental frequency for a trumpet playing A4 (443 Hz, source displaced by 30 cm in positive x-axis)

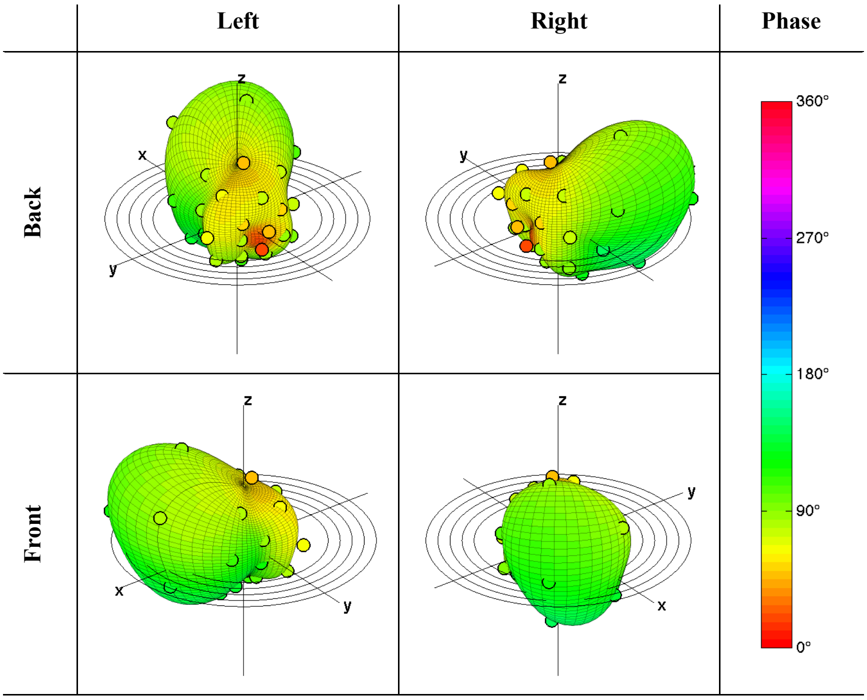

Figure 9: Radiated complex sound pressure of the fundamental frequency for a trumpet playing A4 (443 Hz, source displaced by 30 cm in positive x-axis)

like a half circle intersected by the balloon plot thus indicates a perfect match. It can be seen that in general the interpolation algorithm matches the continous directivity plot very well if sound pressure levels are considered.

In Figure 7 the radiation of the fundametal frequency of that tone is plotted linearly as a complex valued function, with its phase expressed by the used color. Again, the dots describe the values on the microphones. It can be seen that there is a phase shift on the interpolated spherical function, changing the phase when moving in direction of the x-axis. The radius of the colored dots is chosen according to the magnitude of the radiated pressure, which does not always align with the interpolated balloon plot.

Taking a look at Figure 5 we can see the musician placed in the array before the measurement started. As the center of the array is roughly behind the middle of the diagonal white stick (up left to down right), approximately at the center of the players head. The center of the sound source can be assumed to be at the trumpets bell, localized around 30 cm in front of the trumpete players head.

This displacement was compensated by virtually moving the assumed source 30 cm further to the front (in positive direction of the x-axis). The results for both the energetic level plot and the complex magnitude plot are depicted in Figure 8 and Figure 9. The first of these plots is, as expected, quite insensitive to this displacement, whereas the complex radiation pattern looses its continous phase displacement and looks in general much smoother. Note, that having applied such an displacement to the source, the geometry of the array changes. Microphones placed in front of the trumpet now seem to be spaced further apart, whereas the density of microphones on the other side becomes higher. In the plots this is most obvious by taking a look at the northpole of the balloon plot and the location of the microphone there.

4 CONCLUSIONS

In this paper the measurement setup for directivity measurements of natural sound sources was described. In total there were 41 instruments recorded, both modern and historical. Here, only one instrument was chosen, with the purpose of exemplify some processing possibilities with this set of data. The magnitude-only computation allows a higher robustness due to displacement of the real sound source from the ideal geometric center point in the array, with the drawback of lost phase information. The result of complex computation, however, shows the properties of the displaced source. Suitable post-processing can be applied to re-align the acoustical center, yielding a smoother radiation pattern without a continous change of phase on the spherical data. Applying this displacement to the plots of the sound pressure level does not reveal a large difference, though.

Practical applications to use the measured directivity data could be either improved acoustical simulations or a technical sound source with emulated musical instruments radiation pattern (cf. [5]). The latter is useful for measuring room impulse resonses in respect to the musical instrument. This allows to improve the accuracy of the auralization of musical instruments.

5 ACKNOWLEDGEMENTS

Thanks to Neumann Microphones for the courtesy of providing the utilized digital microphones free of charge. The authors also would like to thank Prof. Dr. Stefan Weinzierl and the Department for Audio Communication (TU Berlin) for the use of their measurement room and the pleasant collaboration.

REFERENCES

[1] M. Vorländer, "Auralization – Fundamentals of Acoustics, Modelling, Simulation, Algorithms and Acoustic Virtual Reality", Springer, Berlin, 2007

[2] J. R. Driscoll, J. M. Healy, "Computing Fourier transforms and convolutions on the 2-sphere", Advances in Applied Mathematics, 1994

[3] E. G. Williams, "Fourier Acoustics: Sound Radiation and Nearfield Acoustical Holography", Elsevier, London, 1999

[4] B. Rafaely, "Spatial Alisasing in Spherical Microphone Arrays", IEEE Transactions on Signal Processing, Vol.55, No.3, 2007

[5] M. Pollow, G. Behler, "Variable Directivity for Platonic Sound Sources Based on Spherical Harmonics Optimization", to be published in Acta Acustica united with Acustica