Measuring directivities of musical instruments for auralization

Martin Pollow
Institute of Technical Acoustics, RWTH Aachen University, Germany
Email: mpo@akustik.rwth-aachen.de

Introduction

Auralization is an important tool in room acoustics, making it possible to subjectively judge the sound quality in rooms (e.g. a concert hall) without the necessity of being present at these rooms. Having measured the room impulse response, any recorded sound source can be made audible as if it would have been recorded in that room. However, any directivity of the auralized source is lost, as room impulse response measurements are usually done with omnidirectional sound sources and the result cannot be modified afterwards to take a specific directivity into account. Nevertheless, natural sources like musical instruments owe a substantial part of their characteristic to the directivity. It seems obvious that measuring room impulse responses with respect to specific directivities has the potential to improve the quality of auralization of musical instruments. To do so, we have to evaluate the radiation pattern of the instrument and measure the impulse response of a room with a source of that instrument-specific directivity [1].

Setup for directivity measurements

Whereas directivities of technical sound sources can be measured sequentially for many different directions, the measurement for musical instruments demands a different approach. Their radiation pattern might vary with the style and strength of playing and its exact excitation signal is not known. Therefore all directions have to be measured simultaneously, to get a correct result.

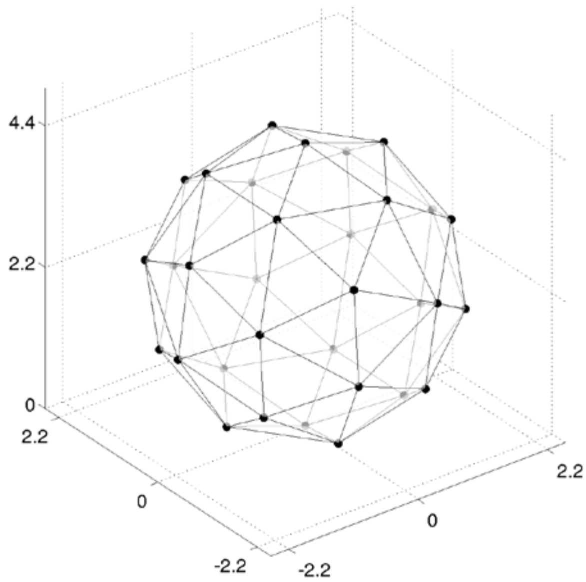


Figure 1: Geometry of the microphone array with a perfectly even distribution of 32 microphones

To accomplish this, a spherical microphone array was constructed, placing 32 microphones regularly distributed on a sphere of a diameter of 4.4 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). As recordings done with that type of microphones are too noisy to be used as input material for high quality auralization, an additional studio microphone can be used to create a suitable audio track for the convolution.

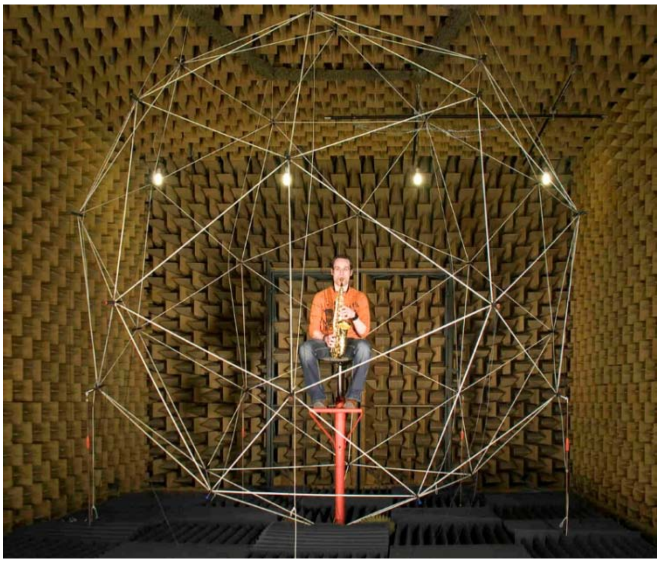


Figure 2: Microphone array during the measurement of the radiation pattern of a saxophone [2]

In Figure 2 the array is depicted as used for the measurements in the hemi-anechoic chamber in Aachen. To avoid disturbing ground reflections, the floor was covered with absorbing mats. The musicians played a piece of music, while keeping still in a fixed position in the sphere, and a 32-channel track was recorded. Using short-time Fourier transform yields a frequency spectrum for each microphone per evaluated time interval. These spatially discrete values can be interpolated to gain the directivity of the instrument as a spatially continuous function. It varies over time and is only valid for that specific track which was recorded.

Misplacement of the sound source

A problem arises if the radiated sound does not emerge in the centre of the array, as different travelling times from the actual sound source to the microphones yield phase displacement. Assuming the source wrongly in the centre, 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, the faultier the results are. This phenomenon is called spatial aliasing and is caused by an insufficient spatial resolution of the microphone distribution [3]. As every instrument has certain dimensions, it is difficult to state its acoustical center. Furthermore, some instruments may have different acoustical centre points for different tones, making an exact positioning of a modeled point source impossible.

But there is another way to minimize the artifacts caused by misplacement: working only with the magnitudes of the measurement creates only a smaller error, which is caused by the different decay for the different distances from source to microphones. This has only a small effect compared to the artifacts created by the differing phase of complex computation. As the phase is not audible by the human ear, it should not reduce the quality of the measured room impulse responses. The phases can then be chosen conveniently to create the best-possible approximation to the sound pressure magnitudes measured.

Using a static directivity

To gain a static room impulse response in respect to the instrument directivity, we have to simplify the measurement data to get a time-invariant general directivity. Of course, any dependency on the strength and style of playing is hereby neglected. Furthermore, the different spectral parts of the energy (as fundamental frequencies and higher harmonics) are assumed to have a similar radiation pattern, to allow averaging. First comparisons of different tones show that this seems to be the case. Nevertheless, further research has to be done to quantify the difference of the averaged radiation pattern over frequency to the actually measured patterns over frequency and time.

Using the magnitude-only method, averaging can be done by simply taking the root-mean-square of the pressure signals. This weights the radiation with the spectral energy and sums them up to the time-invariant result. Thus, loud parts of the played track are dominating and result in an averaged energy spectrum for each microphone.

Figure 3: Measurement loudspeakers for different frequency ranges, each with 12 independently excitable membranes

Adjustable sound source

To be able to perform the measurement of room impulse responses for several instruments with the same technical source, we have to build an electronically adjustable sound source. To do so, conventional dodecahedron loudspeakers (widely used as an omnidirectional source) were modified to be able to excite each membrane individually. In Figure 3 both mid-frequency unit (f = 150 to 1.5 kHz) and high-frequency unit are depicted. They can be mounted on top of a subwoofer unit to be centered exactly in the same position. To avoid internal coupling of the membranes each driver works on its own encapsulated air volume.

The model used for calculation of the 12-channel spherical sound source assumes all membranes as perfect spherical caps [4]. The total surface velocity of the source can then be calculated as a superposition of the aligned aperture functions with the complex valued, scalar membrane velocities (see also [5]). This information is sufficient to know the sound pressure encountered in any arbitrary distance from this source [6]. It is thus possible to compute the pressure arising at the positions of the microphone array, if we assume the spherical sound source placed in the center of the array. Writing the set of membrane velocities for the spherical loudspeaker as $\vec{v}_{mem}$, a matrix M exists that maps a set of given velocities to the resulting pressure values at the microphones [7]:

$$\vec{p}_{re} = M \cdot \vec{v}_{mem} \qquad (1)$$

In our setting we have 12 variables to match the radiation measured on 32 points. Instead of matching onto the interpolated directivity, we are able to match only on the measured values at the microphone points and use the physical properties of the radiation of the single membranes.

Aiming for a best-possible match of the reconstructed pressure values $\vec{p}_{re}$ created by the loudspeaker to the measured pressure values $\vec{p}_{mic}$ on those points, we can define the residual $\vec{r}$ as their deviation:

$$\vec{r} = \vec{p}_{re} - \vec{p}_{mic} \qquad (2)$$

To get a solution matching only the magnitudes of the sound pressure independent of their phase, the residual can be defined in alternatively as [7]

$$\vec{r}_{abs} = |\vec{p}_{re}| - |\vec{p}_{mic}| \qquad (3)$$

A weighted least-mean-squares approach that minimizes the 2-norm of the residual vector, allows gaining a solution for the complex membrane velocity vector that matches the sound pressure at the microphone points in an energetically optimal way. As we deal with a perfectly regular spatial distribution of the microphones, the solution doesn't change by omitting the weighting factors. Using the vector $\vec{r}_{abs}$ as the residual, this minimization is a non-linear problem and can be solved by using an optimization routine. Using the definition of the residual from Eq. 2, it is sufficient to calculate the pseudo-inverse of the matrix M to gain a solution.

To visualize the differing results of the two methods, the sound pressure values on the microphones are simulated for an ideal dipole, whose location is perfectly centered or vertically displaced in the array. The membrane velocities of the loudspeaker in the center of the array are then calculated to match these simulated values. Plotting the far-field directivity of the loudspeaker, the effects of source displacement depending on the computation method used can be seen exemplarily in Table 1 for a frequency of 500 Hz. The radius of the plots is the magnitude of the radiation and the color expresses the phase. Whereas the result for the centered dipole doesn't differ much for both methods, after displacement the effects of spatial aliasing can be clearly seen. For complex matching at a displacement of 25 cm in direction of the positive z-axis, the magnitude of the directivity begins to distort, whereas the phase gets a continuous change over the vertical angle. The displacement of 50 cm is sufficient to totally destroy the desired dipole characteristics. Matching only the magnitudes of the sound pressure on the microphones gives as a better-natured directivity of the loudspeaker. The shape is still recognizable as a typical dipole radiation. Only at higher displacements of the dipole a slight upward bend of the resulting lobes are noticeable. This is due to the shorter distances from the displaced dipole to the upper microphones in the array. Compared to the complex valued matching, this method is much more promising for the measurement of musical instruments, where a perfect alignment to the center is usually not possible. As with the simulated dipole, the real radiation pattern of the instruments should be found without large errors from spatial aliasing, if focusing on the magnitudes of radiation only.

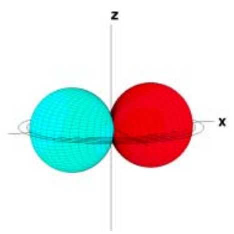
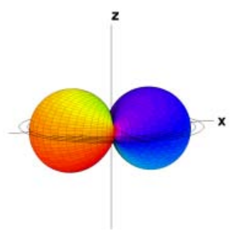
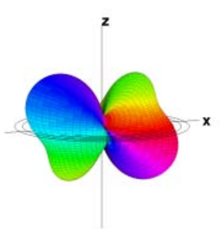
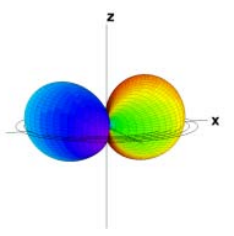
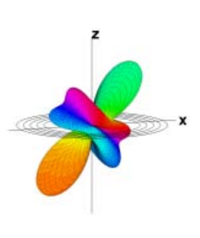
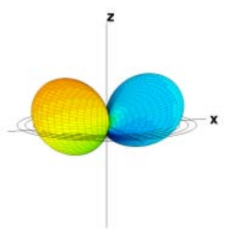
	Complex matching	**Magnitude-only matching**
$\Delta z = 0cm$		
$\Delta z = 25cm$		
$\Delta z = 50cm$		

Table 1: Effect of the displacement of a ideal dipole in the z axis and reconstruction of its directivity by the spherical loudspeaker at 500Hz

Conclusions and future work

In this paper it is shown that measurements of directivities of musical instruments are possible, even if difficulties have to be overcome. One of these difficulties is the influence of reflections in the room. As in the practical experiments a hemi-anechoic chamber was used, the influences of ground reflections were minimized by the use of porous absorbing material. To gain proper results also for very low frequencies, the use of a fully anechoic chamber is advisable.

If the exact position of the sound source is not known, a displacement from the center leads to errors due to spatial aliasing. The critical displacement distances are proportional to the frequencies, so this fact constitutes a problem mainly at higher frequencies. In practice, a perfect alignment of a musician with an instrument to the center is not possible. By neglecting the phase of the measurement results, differing distances from the source to the microphones create a smaller error, caused only by the different decays. The missing phases can then be used by an optimization routine to gain a good match in reproducing the radiation with the spherical sound source.

For improved results, new measurements of directivities of musical instruments are planned in a fully anechoic chamber. Furthermore, the directivities of the occurring tones of the recorded tracks have to be analyzed on their dependency on style, strength and played pitch. Will the actual directivity at certain times deviate too much from a generalized and averaged directivity? Statistical evaluation and listening tests will have to answer this question. Similarly, the assumption of non-audibility of phases has to be confirmed, if the sound source is used for room acoustical measurements.

References

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

[2] Reuter, S.: "Simulation of natural sound sources in immersive virtual environments", diploma thesis, Aachen, 2008

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

[4] Zotter, F.; Höldrich, R.: "Modelling radiation synthesis with spherical loudspeaker arrays ", 19th International Congress on Acoustics; Madrid, 2007

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

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

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