Musical Instrument Recording for Building a Directivity Database

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Introduction

An important characteristic of the sound of musical instruments is their tonal color. It describes the spectral intensity of their sound radiation which is different for any angle of radiation. Sound reinforcement with a common technical source of different radiation pattern is thus always lacking in these specific directional tonal characteristics. This is an issue for high precision room acoustical measurements, as the directivity of the musical instruments is not included in the result of the measurement and cannot be accounted for in post-processing. To improve quality in auralization an electronically adjustable device can be used as sound source, allowing to approximate the desired directivities. This paper describes how to obtain high quality directional data of musical instruments.

State of the Art

The first approach to collect data for a directivity database goes back to the 1970s with Jürgen Meyer publishing the measured radiation of various instruments in averaged octave bands [1]. More recently also Otondo and Rindel [2], as well as Pätynen et al. [3] measured the directivity with a set of microphones distributed around the musician, using 13 and 22 microphones, respectively. The Institute of Electronic Music and Acoustic in Graz, Austria, is currently also making studies in that field of research using a 64-channel spherical microphone used to decompose sound fields in a set of orthonormal base functions [4].

Data Acquisition

As naturally played tones are not reproducible the common way to measure the directivity of musical instruments is by using a microphone array to simultaneously record the radiated sound. A regular distribution of the microphones around the musician is advantageous if interested in all possible directions. For that reason a large spherical 32-channel microphone array (diameter approx. 4.20 m) was constructed at the Institute of Technical Acoustics at RWTH Aachen University. The arrangement was chosen to be a truncated icosahedron (soccer ball shape) with one microphone at the center of each surface.

In a joint-project of TU Berlin (Audio Communication Group) and RWTH Aachen University (Institute of Technical Acoustics), a large number of musicians were invited to be recorded with the spherical array in the full anechoic chamber at TU Berlin (anechoic from 63 Hz, Volume 1070 m^3). These recordings were then evaluated on their directivity and on their spectral sound power [5] [6]. In total 41 of these instruments were measured, both modern and historical ones, covering the full range of symphonic orchestral instruments.

Measurement in Anechoic Chamber

The musicians were placed inside the spherical microphone array, located to have the assumed acoustical main source as close as possible to the geometric center of the sphere. This could be done with the help of a moveable and height adjustable chair. However, for some large instruments (pedal harp, contrabass and timpani) practical reasons did not allow alignment to the center. The musicians were instructed to play clean single tones of a chromatic scale in two dynamic ranges, pianissimo and fortissimo. They were asked to avoid the use of tremolo and vibrato to achieve stable results in terms of intensity and frequency. If desired, artificial reverberation could be provided via headphones in case the musicians felt uncomfortable with the dry environment. The recording has been performed fully calibrated. Due to the different loudness of the instruments the amplification had to be adjusted a few times to avoid clipping. The used recording levels were stored with the measured data to allow exact reconstruction of the sound pressure at the microphone positions.

Hardware Evaluation

A free-field calibration measurement of the microphone units shows significant ripples in the sensitivity above 2kHz (comb-filter effect). This is caused by the joints connecting the sticks to the microphone housings. The influence of these joints with the mounted sticks was simulated by the use of the Boundary-Element-Method.

The two types of housing (microphones with five and six joints) have a very similar pattern that differs only in

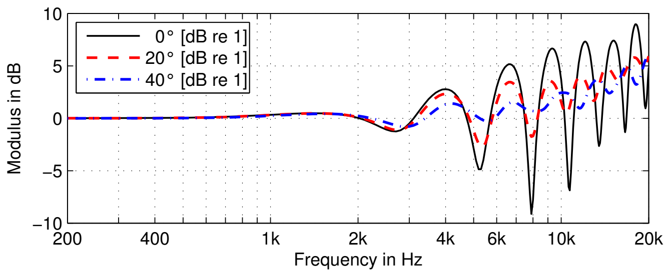

Figure 1: Influence of the housing on the microphone sensitivity, depending on the angle of sound incidence (BEM-Simulation, five joint microphone)

the sub-decibel range, whereas the angle of incidence influences the frequency response more strongly (Fig. 1). This angle dependent influence on the microphone sensitivities can be either averaged in sensible directions or used for the realignment algorithm described later.

To combine the appropriate housing sensitivities with the sensitivities of the individual capsules, a calibration measurement of the microphones mounted in an rigid wall was performed. The result is a frequency dependent microphone sensitivity that accounts for the sensitivity of the individual electret capsule as well as for the influence of its housing.

Post-Processing

The most simple approach for deriving a general radiation pattern as a function of frequency is root mean square averaging the measured spectra over all played tones. This leads to a magnitude only directivity information for all frequencies that contain sufficient energy. Usually the significant information is disrupted by frequencies of no information, as each tone is a collection of the fundamental frequency and its harmonic overtones. By using an envelope algorithm this missing spectral information can be interpolated to gain continuous spectral information starting from the lowest fundamental frequency. The advantage of this approach is the smaller influence of a possible misplaced sound source. As phase shifts are not significant, the misplacement results only in level differences (due to the decay of spherical wave propagation).

Contrarily, the complex computation accounts for phase shifts at the microphone positions and thus reflects the physical reality of the traveling wave including its instantaneous phase. By decomposition into orthonormal base functions on the sphere (spherical harmonics) it is possible to derive a spatially continuous solution for the measured radiation pattern. Phase shifts can thus be interpreted as a time delay that allows to conclude the origin of the radiated sound. In [7] it is shown how the compensation of misalignment changes the results of the measured radiation.

Subsequent Re-Alignment of the Source

As complex computation is highly influenced by source misplacement a correct alignment is vital. Misaligned sources cause a time delay (or frequency shift) on the microphones further away from the origin of radiation. This information can be used to calculate the location of the displaced source by a non-linear optimization algorithm. In general, the calculated source location is sensible if the spherical radiation has a minimum phase change over the solid angle. Ideal solutions for such spherical minimum-phase functions have not been presented until now.

Employing a non-linear optimization algorithm to maximize the similarity of interpolated complex and energetic evaluation, shows stable solutions for frequencies up to 1.5 kHz. Hereby any discrete frequencies (all fundamentals and all overtones) are considered as independent information. The solution space for the optimizer was chosen to allow all possible source locations within the spherical array. Applying more sophisticated algorithms, searching for a solution in a sensibly confined space and using information of spectrally related partial tones, are expected to stabilize the results for higher frequency as well.

The result of the realignment algorithms can be used both for the complex approach as well as for the energetic calculation. The result of the latter is enhanced by accounting for the distance dependent decay, whereas for complex computation realignment is a necessity for precise results.

Applications

The database of musical instrument directivities can be used for room acoustical simulations and for measurements of room impulse responses with an electronically adjustable sound source [8]. As the directional dependent tonal character of the instruments is preserved, a more realistic auralization can be achieved [2]. Using complex valued measurement results, the phase information can be used to approximate the point of origin of the emerging sound wave. This spatial information allows a further enhancement of both simulation and room impulse measurement.

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