Directivity pattern measurement of a grand piano for augmented acoustic reality

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Introduction

In augmented acoustic reality the authentic or at least plausible reproduction of sound sources is key to achieve a high level of acceptance by the user, whereas the latter is easier to reach. Hence, source directivity and the measurement thereof are an important part of the augmented acoustic reality system employed in the project Augmented Practice-Room [1]. Low-latency spherical harmonics domain directivity filters were derived for most instruments based on freely available measurements [2]. However, no directivity measurements were available for the grand piano at that time. Therefore, we present a practical directivity measurement setup for the grand piano under non-anechoic conditions and a modelling approach by utilizing mimimum-phase design.

Augmented acoustic reality

Augmented reality can be defined as an enhancement of reality by the use of technology, such that the "real" and the virtual world become fused. Virtual objects such as sound sources can be presented without losing access to the real world. In a newly developed e-learning tool, the Augmented Practice-Room, virtual acoustics enhances the real instrument sound in a music practice room. The system comprises either of modified open-back headphones (to reduce the isolation of sound from the played instrument to a minimum) or loudspeakers [1]. The acoustic properties of a room can have considerable effects on the sound of an instrument and how it is perceived by a performing musician [3]. In the design of a virtual or augmented acoustics system, one has to consider whether the aim is to create an authentic or a plausible reproduction of the room. Authenticity would require the simulation to be perceptually indistinguishable from the real room, while plausibility requires the reproduction to meet the user's expectations about the simulated sound [4]. Aiming for an authentic reproduction in an real-time system is currently out of reach and not required in our case.

Simplified source directivity

Dynamic source directivity modelling was shown to improve plausibility, already at limited accuracy. In case of an auralized voice for example, plausibility was improved by including a dynamic directivity model based on 12 beam-directions and SH-interpolation [5]. In [6] a VBAP based approach was used to interpolate between 6 control points of a loudspeaker directivity pattern, causing only a small decrease of plausibility compared to a 10°-resolution reference.

For ease of rotation and incorporation into our real-time system, we favour an order-truncated spherical harmonic (SH) source-representation. Experiments have shown that in the present case of nearly coincident source and receiver, the order of a highly directive source can be truncated to 3rd-order, without any noticeable difference to a high-order reference directivity [7].

Methods

Measurement setup

Measurements of the grand piano (Fig. 1) were taken in a sound treated multi-purpose room called the IEM CUBE at the Institute of Electronic Music and Acoustics. The IEM CUBE is a 10.3 m×12 m×4.8 m large room with a reverberation time $T_{60} = 700\,\mathrm{ms}$.

Microphone Positions

The microphone array consists of 22 NTI 2230 microphones, where 20 microphones are aligned on three rings, one microphone positioned is placed at the zenith, and one at the nadir (cf. Fig. 2, Tab.1). The microphone array center was inside of the piano and the array radius was 2 m.

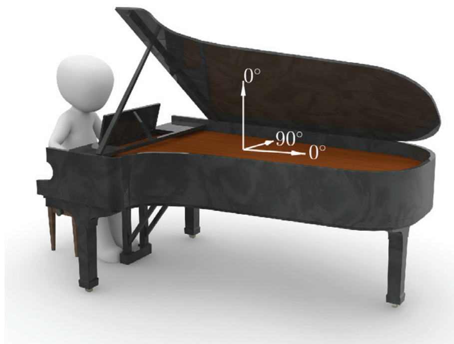

Figure 1: Orientation of the grand piano.

Table 1: Azimuth and elevation angles of the microphone positions in degree. Radius all microphones was $r = 2\,\mathrm{m}$, except for $r_{22} = 1\,\mathrm{m}$ at nadir.

m	φ_m	ϑ_m	m	φ_m	ϑ_m
1	0	0	12	210	40
2	45	0	13	270	40
3	90	0	14	330	40
4	135	0	15	0	90
5	180	0	16	30	-30
6	225	0	17	90	-30
7	270	0	18	150	-30
8	315	0	19	210	-30
9	30	40	20	270	-30
10	90	40	21	330	-30
11	150	40	22	0	-90

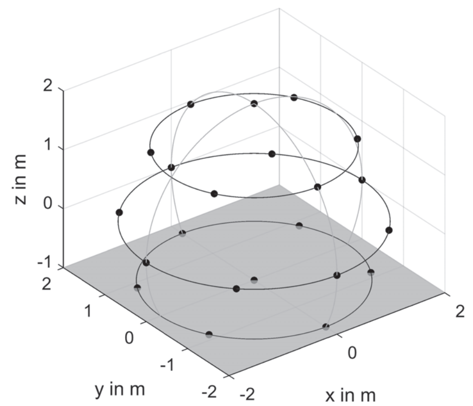

Figure 2: Microphone positions are represented by black dots. The center of the grand piano is located at the origin of the coordinate system. The orientation of the grand piano is shown in Fig. 1. The grey plane indicates the floor.

Floor microphones

The lower microphone ring and the microphone positioned at the nadir can be seen as boundary microphones if omnidirectional microphones are used. The floor as a boundary can be considered sufficiently large for the relevant frequency range of the instrument. The gain adjustment for the microphones at the lower ring is therefore -6 dB and at the nadir position by -12 dB.

Spherical harmonic transform

The spherical harmonic (SH) representation of sound pressures measured at the set of microphone directions $\boldsymbol{\Theta}_M$ can be found by solving the least squares problem

$$\min_{\boldsymbol{\psi}} ||\boldsymbol{p} - \boldsymbol{Y}_N(\boldsymbol{\Theta}_M)\boldsymbol{\psi}||_2^2. \quad (1)$$

This is can be done for example by computing the pseudo inverse of the SH matrix $\boldsymbol{Y}$, which contains the spherical harmonics up to order N, with the sound pressure $\boldsymbol{p}$ evaluated at the measurement positions $\boldsymbol{\Theta}_M$.

$$\boldsymbol{\psi} = \boldsymbol{Y}_N(\boldsymbol{\Theta}_M)^{\dagger}\boldsymbol{p} \quad (2)$$

While determining a microphone setup, the challenge is to find a layout, that is at the same time easy to set up but still leads to a well-conditioned matrix $\boldsymbol{Y}$. The present setup fulfills these requirements with a condition number $\mathrm{cond}(\boldsymbol{Y}_3) \approx 1.78$, allowing to calculate spherical harmonics up to an order of 3. The 3rd-order resolution is advantageous in regard to computational load and is sufficient for the application, as our previous studies have revealed application [7, 8].

Measurement data

In order to achieve a suitable data set over the whole frequency range of the grand piano with (i) an open and (ii) closed lid, recordings were made of single notes starting from the lowest key A0 (27.5 Hz) up to C8 (4186 Hz) at a sampling rate of 48 kHz. Power spectra of the single notes are calculated with a resolution of 11.72 Hz and then the data is merged to generate magnitude responses over the discussed frequency range. Linear interpolation is used for frequency bins between semitones to avoid errors due to a too low signal-to-noise ratio. The magnitudes for frequencies above the highest key are kept constant at the measured magnitudes for the highest played note on the grand piano.

Filter design

As we have outlined above, our analysis method and our data set is limited and due to the calculation of the power spectra all phase information is lost. Although, even if we could retrieve the phase information from the measurement data, the measured phase would most likely be erroneous because of the acoustic centering problem [2]. However, the retrieved magnitude spectra describe well how sound is radiated from the grand piano and can be used to model source directivity, if suitable filters can be derived to resemble it. This can be done for example using minimum-phase modeling. We calculate minmum-phase filters from the zero-phase magnitude spectra. The minimum-phase design allows us to generate stable directivity filters, which do not introduce additional system delay. A filter length of 512 samples is chosen to reduce the computing load necessary for the e-learning tool. In order to identify whether our approach models the underlying data correctly, we compare the minimum-phase directivity filters with the zero-phase filters in the SH domain with a reduced dynamic of 25 dB. The reduced dynamic range is ought to be perceptually sufficient according to our findings in [7].

Results

Mean absolute gain error

We investigate the deviations of the used minimum-phase approach by analyzing the mean absolute gain error for different modeling steps, whereas the dynamic has been also reduced to 25 dB. The modeling steps include encoding the microphone signals into the SH-domain, shortening of the impulse responses, and minimum-phase filter modeling. In Fig. 3 and 4 we present the mean absolute gain error in dB for the different modeling steps. Deviations in the magnitude of 3.5 dB maximum occur if the length of the impulse responses is reduced (dark grey).

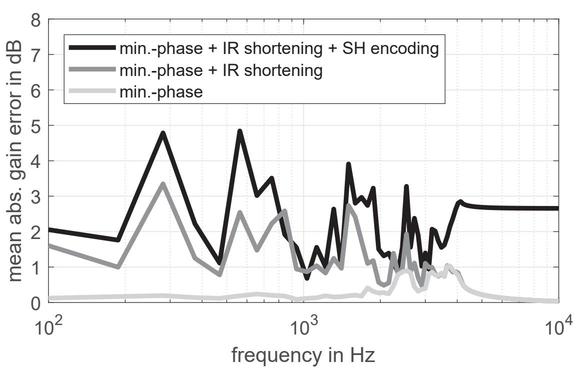

Figure 3: Mean absolute gain error between magnitudes calculated from the minimum-phase filters (includes encoding) to the measured magnitudes (solid black), to the magnitudes from the zero-phase filters (dark grey), and to the magnitudes from the shortened zero-phase filters (light grey) for the grand piano with an open lid.

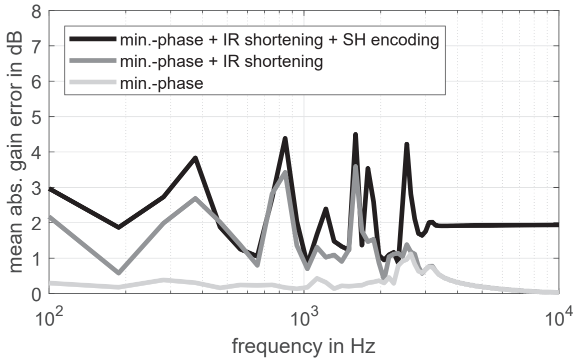

Figure 4: Mean absolute gain error between magnitudes calculated from the minimum-phase filters (includes encoding) to the measured magnitudes (solid black), to the magnitudes from the zero-phase filters (dark grey), and to the magnitudes from the shortened zero-phase filters (light grey) for the grand piano with closed lid.

If the modeling step of encoding the microphone signals into the SH-domain is considered we see even larger deviations up to 4.8 dB (black). Interestingly, almost no differences are found due to the minimum-phase modeling (light grey).

In general, there occurs no larger gain deviation than 4.8 dB over the whole modeled frequency range for the grand piano with an open or with a closed lid. The largest errors for the open lid measurement occur at rather distinct frequencies at 280 Hz, 560 Hz, 750 Hz, 1500 Hz, 1900 Hz and 2500 Hz, which exceed 3 dB. The largest errors for the closed lid measurement occur at 380 Hz, 840 Hz, 1600 Hz, 1800 Hz, and 2500 Hz.

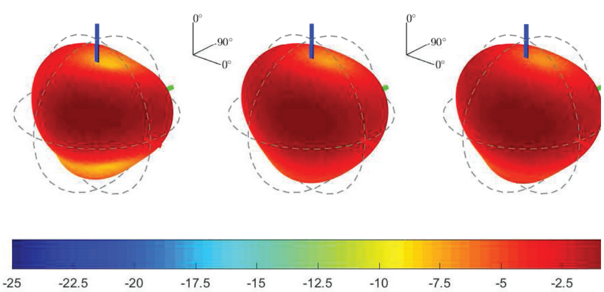

Figure 5: SH-interpolated directivity patterns for the grand piano with an open lid at 500 Hz (left to right) calculated from (i) zero-phase filters, (ii) shortened zero-phase filters, (iii) and minimum-phase filters.

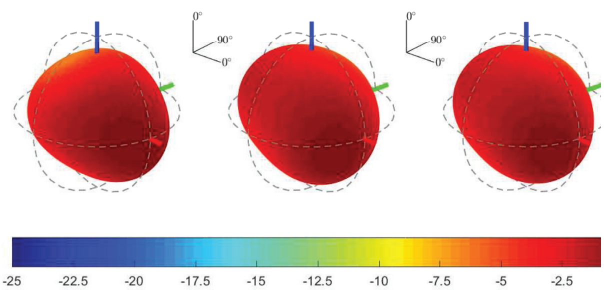

Figure 6: SH-interpolated directivity patterns for the grand piano with an open lid at 1000 Hz (left to right) calculated from (i) zero-phase filters, (ii) shortened zero-phase filters, (iii) and minimum-phase filters.

Directivity plots

In our analysis we present directivity plots (SH domain) for the grand piano from zero-phase filters, shortened zero-phase filters (512 samples filter length), and minimum-phase filters (512 samples filter length) at 500 Hz, 1 kHz, and 3 kHz for the grand piano with an open and a closed lid, respectively. The dynamic of all directivity plots is limited to 25 dB.

Open lid

At low frequencies the directivity plots for the grand piano show that sound is radiated into all directions in the horizontal plane and slightly upwards around 270° (cf. Fig. 5), which means sideways of the piano where the lid is opened. A reduction of the filter length leads to expected deviations in the directivity patterns at low frequencies, see Fig. 5. With increasing frequency, there is less upward radiation; it is slightly more focused towards the 0°-azimuth direction (Fig. 6), which is the opposite side to the fingerboard. For higher frequencies the sound is mainly radiating upwards and towards the side of the grand piano, see Fig. 7.

Closed lid

In Fig. 8 we show the directivity plots calculated from minimum-phase filters at 500 Hz, 1 kHz, and 3 kHz. At 500 Hz and 1 kHz the patterns resemble the results for the mimimum-phase variant for the grand piano with the open lid. The largest differences occur for higher frequencies (see Fig. 8, rightmost), where the sound of the grand piano mainly radiates towards the floor and the 0°-azimuth direction.

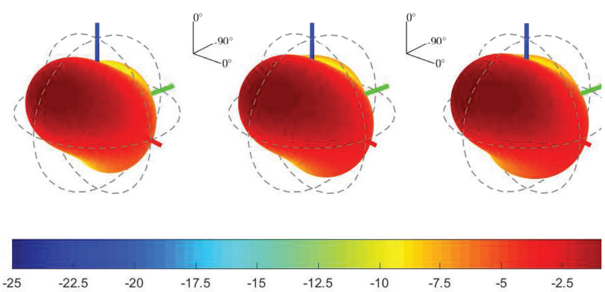

Figure 7: SH-interpolated directivity patterns for the grand piano with an open lid at 3000 Hz (left to right) calculated from (i) zero-phase filters, (ii) shortened zero-phase filters, (iii) and minimum-phase filters.

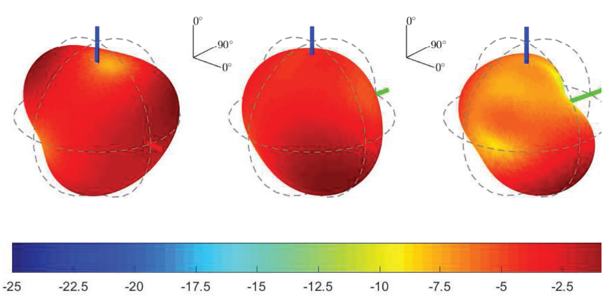

Figure 8: SH-interpolated directivity patterns for the grand piano with a closed lid at 500 Hz, 1 kHz, and 3 kHz (left to right) calculated from minimum-phase filters.

Discussion

Previous studies have investigated the directivity of instruments by including the phase. As mentioned in the method section we propose an analysis and modeling solely by utilizing the magnitude responses. The directivity plots calculated from the measurement data after merging single notes provide reasonable results over the whole frequency range. A comparison of the zero-phase filters, the shortened zero-phase filter, and the minimum-phase filters show that the largest deviations occur due to the shortening of the impulse responses in the time domain, whereas even less errors occur for modeling the shortened zero-phase filters as minimum-phase filters, see Fig. 3.

The analysis of the mean absolute gain error reveals that largest errors in the magnitude of 3 dB and above occur at distinct frequencies for the open and the closed lid. The maxima of the mean absolute gain error for the modeled directivites for the grand piano stays below 4.8 dB for the whole frequency range. The largest deviations of the magnitudes result from the encoding step and the impulse response shortening. Minimum errors are introduced by the minimum-phase modeling.

Conclusion

The aim of this paper was to present a practical measurement setup for the grand piano under non-anechoic room conditions. Furthermore, we investigated the approach of calculating minimum-phase filters to model the directivity. The study has identified that larger deviations of the average gain error occur if the resolution is reduced (i.e length of the modeled filters) and neglectable gain deviations for modeling directivity with minimum-phase filters on the basis of shortened zero-phase filters. In spite of the limitations of the current approach and the lack of high-resolution, anechoic data for comparison, preliminary findings indicate that the approximation of the source directivity by the minimum-phase filter approach on the basis of measured magnitude spectra is valid for the use of plausible reproduction of an instrument in augment acoustic reality: Users accept the simulation and can hear clear differences between the closed lid and the open lid condition.

Acknowledgments

This work is supported by the project Augmented Practice-Room (1023), which is funded by the local government of Styria via Zukunftsfonds Steiermark (future fond of Styria).

This research has received funding from the European Union's Horizon 2020 research and innovation programme under the Marie Skłodowska-Curie grant agreement No. 812719.

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