ISRA 2019 AMSTERDAM
PROCEEDINGS of the
International Symposium on Room Acoustics
15 to 17 September 2019 in Amsterdam, Netherlands
NAG NEDERLANDS AKOESTISCH GENOOTSCHAP

EVAA: A platform for Experimental Virtual Archaeological Acoustics to study the influence of performance space

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ABSTRACT

Research in historical musical acoustics has for several decades focused significantly on instrument fabrication. Such research has been able to highlight the acoustical impact of material and construction choices. Musicological studies have concentrated in parallel on understanding historical notation, playing styles, and even changes in musician posture over the centuries. In studying player and listener conditions in these historical studies, little attention has been given to the acoustical conditions of the performance, aside from extreme cases such as cathedral acoustic conditions. Extending the methodologies of experimental archaeology, recent advances in computational accuracy of acoustic virtual reality simulations offer the possibility to create ecologically valid reconstructions of historic sites. We present the development of an interactive immersive real-time simulator allowing musicians to perform "live" within virtual reconstructions of historic venues, comprising real-time adaptation of source directives with performer movements, rendered in Higher Order Ambisonics. Observations of the impact of acoustic variations on player performance, and the comparisons of the resulting performance between historically suitable venues and modern performance spaces from an audience perspective will complete the feedback loop between performer and listener necessary for a full understanding of the historical musical context.

Keywords: Archaeoacoustics, Archaeological Acoustics Auralization, Virtual Reality

1 INTRODUCTION

The research and practice of "historically informed performance" (HIP) has advanced significantly in recent decades. This discipline can be summarized as a regard to performance which aims to be faithful to the approach, manner, and style of the musical era in which a work was originally conceived. Two aspects are typically considered in such efforts: performance practice of musicians and use of period instruments. However, little consideration has been given to the influence of performance space on the performance, instrument, or composition of the time. Using real-time virtual acoustic simulations, the EVAA project (Experimental Virtual Archaeological Acoustics)[1] aims to include the performance venue's acoustics as a third component of study for HIP. Placing musicians with original or historically accurate facsimiles of instruments in various virtual performance spaces, we examine the impact of the room's acoustics on performance. Such a study also informs investigations on the evolution of instruments themselves, with respect to the evolution of performance venues. Inclusion of the role of composition completes the circle, as it can be seen as both a driving force of change and a response to the changing physics of the instruments and rooms used for music performance.

This paper presents the context and overview of the EVAA project, launched in early 2019, as well as a presentation of first results : a real-time simulation platform developed to facilitate the subsequent series of studies.

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[1]EVAA project website: http://evaa.lam.jussieu.fr

2 CONTEXT

Research in historical musical acoustics has for several decades focused significantly on instrument fabrication [2, 14, 4], highlighting the acoustical impact of material and construction choices. Musicological studies have, in parallel, concentrated on understanding notation, playing styles, and even changes in musician posture over the centuries. In studying player and listener conditions in these historical studies, acoustical conditions of the performance have been largely ignored [1], aside from extreme cases such as cathedrals [17, 18].

Extending the methodologies of experimental archaeology [3, 20], recent advances in computational accuracy of acoustic virtual reality simulations [17] offer the possibility to create ecologically valid reconstructions [10] of historic sites. Generally limited to off-line rendering, simulating the audience perspective, we employ an interactive immersive real-time simulator allowing musicians to perform "live" within virtual reconstructions of historic venues. The simulator significantly advances upon previous realistic off-line renderings of historic sites. Analyzing the impact of acoustic variations on player performance, and comparisons of performances between historically suitable venues and modern performance spaces from an audience perspective, completes the feedback loop between performer and listener, for a full understanding of the historical musical context [1].

3 PROJECT OVERVIEW

Since its inception, musical acoustics has focused predominantly on the production of sound for musical purposes, with that focus being on musical instruments. Musicology extends this domain into the humanities, highlighting the cultural and societal aspects of music history as well as composition. Room acoustics concentrates on the interaction of sound sources and rooms, with a strong focus on performance spaces due to their importance for listening. EVAA uses innovative virtual reality techniques to bring together researchers from these various disciplines to work on a question at the core of their mutual interaction: the influence of the performance room's acoustics on the development of instruments, the performance of musicians, and the details of musical composition.

Based on ongoing research at the *Musée de la musique* concerning wind instruments in the context of facsimile fabrication (manufacturing techniques, materials used, acoustic behavior, playing techniques, etc.) EVAA will first focus on the reconstruction of venues of historical significance for small ensembles in the Baroque and Classical music periods, beginning with Parisian venues and the surrounding region, including the *Salle de concert du Conservatoire* and the *Château de Versailles*. Additional partners include the *Pôle Supérieur d'enseignement artistique Paris Boulogne-Billancourt* with activities on these periods and instruments and IReMus with musicological expertise on the historical approach of music and acoustics and HIP issues.

3.1 Historic Instruments

The *Musée de la musique* has in its collection over 7000 musical instruments. Its main mission is to tell the story of "Art music" (*Musique savante*) through its collection. For conservation reasons, less than 5% of these are played regularly. No wind instruments are played due to hygrothermic gradients from musician's blowing. The current solution for listening to these instruments is via facsimiles. Facsimiles are constructed from detailed information on historic instruments: acoustical measurement, historical & organological research, and material characterization. While attention is given to the music played, the appropriateness of the musical period for the instrument, all are played in the same Amphitheatre: harpsichords, oboes, flute, guitars, etc.

Current research concerns technical evolutions in the contexts of the evolution of music, playing techniques, and venues [14, 8]. In showing the history of the musical instrument construction process, a challenge is to study these changes according to venue, to propose to audiences and musicians a unique historical sound experience. EVAA will focus first on instruments such as Hotteterre flutes (17^{th} c.) and Delusse oboes (18^{th} c., one facsimile exists), from when such instruments entered into the music salon in Versailles. In 1684, French musical institutions reached their peak with the new association of violins and oboes at the Royal Academy. The Musée has at least four oboes dated end of the 17^{th} (oboe E.108, 1680). Similar oboes were played in the Chambre

Royale and the Royal Academy Orchestra. Two famous constructors (early 18th c.), have oboes in the collection. Acoustical characterization of the selected instruments with consider input impedance measurements for historical fingerings, providing acoustic information without playing. The objective is to investigate any evolution of the acoustics following evolutions in composition.

3.2 Room Acoustics

Composers of the Baroque and Classical eras, may have been aware of the effects of room acoustics on performances, adapting compositions to venues [12]. However, such observations have not been supported by objective or subjective evaluations of the rooms' acoustics. We will soon conduct room acoustical measurements, documenting these architectures and improving understanding in specific evolutions in composition during this period, providing objective acoustical analyses.

These measurements enable calibration of geometrical acoustic models, an essential step to perceptually realistic simulations. [17] has shown that calibrated simulated auralizations can be subjectively comparable to measured ones. To date, this calibration is validated for vocal sources, and needs to be extended to musical content.

The surrounding visual environment may be presented over non-disturbing VR devices, providing a multimodal environment. It has been shown that visual renderings affect auditory perception (distance, envelopment, and plausibility) [19]. The EVAA platform will facilitate further evaluations assessing effects of visual "level of detail", *e.g.* lighting conditions or images resolution on both musician and audience.

3.3 Musician at the center

The professionalism of the instrumentalists is essential. If the musicians do not have sufficient knowledge of the instrument and its playing characteristics, results will be subject to errors affecting the produced sound: *e.g.* accuracy, power, and timbre.

Characterizations of playing conditions include mechanical measurements of control parameters, like air force and movements [7]. Players may not feel exactly like in a real situation, behaving differently than in a real venue, but relative differences between two simulated venues would lead to similar differences in both real and the virtual conditions. This is evaluated through questionnaires and audio recordings in virtual and real venues.

Experiments can investigate links between the evolution of instruments, venues, and repertoire. How does performance on an instrument change with venue? How do players' evaluations of instruments change with venue? Are some venues more appropriate for some instruments? How do evaluations relate to acoustic measurements?

Musicians will play and evaluate facsimiles from different periods in simulated venues popular during these periods, with oboists and flutists from partners and external baroque ensembles, using protocols ensuring ecological validity [10, 9, 16]. Recordings will be analyzed to extract parameters that could differ: *e.g.* tempo, level, spectra, and ornamentation. Quantitative performance metrics will be compared to perceptual evaluations and the acoustic measurements. Recordings will be evaluated by musicians, music teachers, and researchers to investigate differences for audiences, helping to interpret signal analyses.

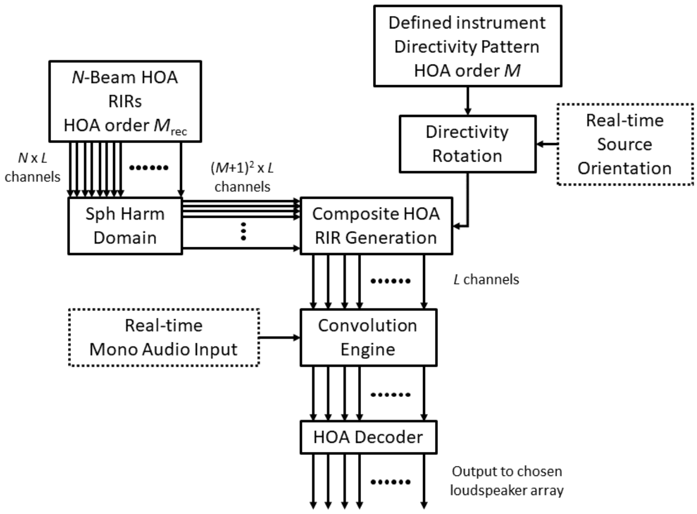
4 RENDERING ENGINE ARCHITECTURE

4.1 OVERVIEW OF RENDERING ENGINE

The aim of the rendering engine is to allow a musician or performer to be placed in different acoustical spaces in order to investigate how the room acoustics influence their performances, including the effects of instrument directivity and dynamic movement. To achieve this, performances will be captured using a wireless microphone placed in close proximity to their instrument. The directivity of the musician's instrument, incorporated via measurements or simulations, will be able to dynamically follow the motions of the performer within the virtual room in real-time. A conceptual overview is shown in Fig. 1(a).

The rendering engine detailed here provides the dynamic simulation of a directional source using a series of

(a) Conceptual visualization of a performer placed virtually in the *Château de Versailles* with the virtual room acoustics reproduced over a loudspeaker array in an acoustically controlled environment.



(b) Schematic diagram of the rendering engine for real-time dynamic source directivity.

Figure 1. Conceptual and schematic overview of the rendering engine.

pre-calculated room impulse responses (RIRs)[2] An overview of the processing architecture is shown in Fig. 1(b). The engine takes a mono sound source and processes it to a Higher Order Ambisonic (HOA) [6] audio stream used for playback. The virtual source orientation is driven by the orientation of the musician, captured using a motion tracking system. The Rendering Engine is currently implemented in C++ as a VST3 for use in the MAX real-time environment.

An HOA receiver is situated in the simulated room for RIR computation. In the current case of interest, a performer in a virtual room, this receiver position is placed at the same position as the sound source. Additional receivers can be placed elsewhere in the room, allowing for listeners other than the performer to listen to the performance, or to allow the performer to replay their performance while listening from the audience.

4.2 ROOM SIMULATION

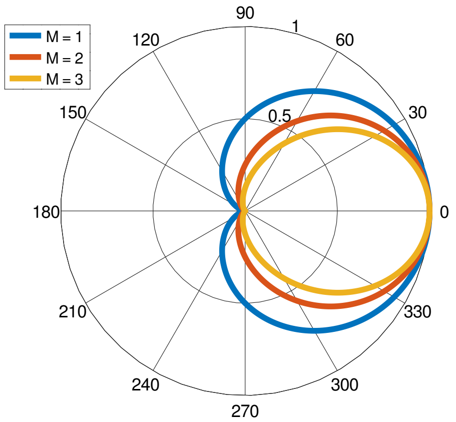
It is assumed at this point that a suitable geometrical acoustics model has been created, allowing for an accurate acoustic simulation of the room. In order to generate different source directivity patterns at run-time, an overlapping beam approach [19] is used where RIRs are calculated for a set of directional beam-sources of differing orientations. Each beam-source simulation produces an HOA-RIR of order M_{rec}, resulting in $N \times (M_{\text{rec}}+1)^2$ impulse responses for each source/receiver position combination, where N is the number of beams.

The number, directivity, and orientation of the beams depends on the reproduced spatial accuracy of the final source directivity - more detailed directivity patterns will require more, and narrower, beams be used. The spatial accuracy is a function of the spherical harmonic order M used for the representation. In [19] the beam patterns were defined using strategic control points and spline interpolation. In the current improved version, beam patterns are reformulated as M-th order cardioid patterns;

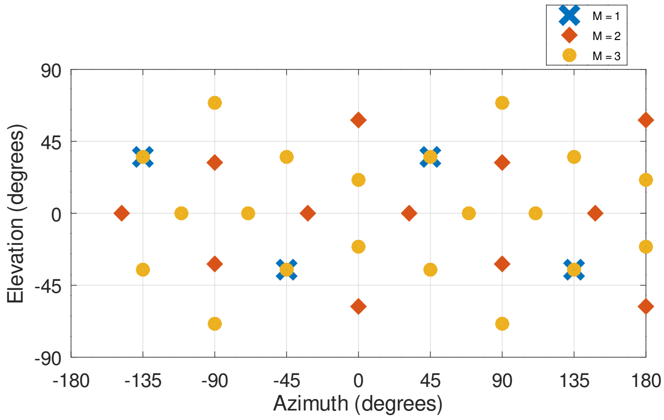
$$g(\theta,\phi) = (0.5+0.5\cos(\theta)\cos(\phi))^M, \tag{1}$$

where θ and ϕ are the azimuth and elevation directions of radiation (see Fig. 2(a) for $M \leq 3$).

[2]RIRs for this study were computed using CATT-Acoustic [5] but the methodology is not limited to any given software.



(a) Horizontal cross-section of the beam directivity patterns for order M = 1, 2, 3.



(b) The directions of the beams for orders M = 1, 2, 3.

Figure 2. The beam widths and directions used to enable maximum source directivities of M = 1, 2, and 3.

A first-order spherical harmonic approximation of source directivity requires four sources in a tetrahedral arrangement with beams of cardioid directivity. For higher order directivity patterns, the beam directivity is that of a higher order cardioid. Second-order approximation requires 12 sources arranged in an icosahedron and second-order cardioid directivity. Third-order requires 20 sources arranged in a dodecahedron and third-order cardioid directivity. Figure 2(b) shows the orientations of the set of beams for $M \leq 3$.[3] Each order's set of beams sum to produce an omnidirectional directivity pattern, ensuring that the desired directivity can be varied from that of a single beam (or any arbitrary M-th order pattern) to an omnidirectional source.

The beam directivities can be converted to spherical harmonics via:

$$\mathbf{b}_M = \frac{1}{N}\mathbf{Y}_M \mathbf{g}_n, \tag{2}$$

where $\mathbf{Y}_M = [\mathbf{y}_M(\theta_1, \phi_1), \ldots, \mathbf{y}_M(\theta_N, \phi_N)]$ is the $(M+1)^2 \times N$ matrix that encodes a function over the N beam directions to spherical harmonics of order M, $\mathbf{y}_M(\theta_n, \phi_n)$ is the vector of spherical harmonic coefficients in the n-th beam direction, and $\mathbf{g}_n = [g_n(\theta_1, \phi_1), \ldots, g_n(\theta_N, \phi_N)]^{\mathrm{T}}$ is the vector containing the n-th beam's gain in the directions of all N beams.

4.3 SPHERICAL HARMONIC SOURCE DIRECTIVITY

If the frequency-dependent source directivity to be simulated has been measured on a regular spherical grid then it can be converted to an M-th order spherical harmonic representation $\mathbf{x}_M(\omega)$. Assuming the source directivity is measured in the same directions as the required number of beams, conversion to spherical harmonics is realized via;

$$\mathbf{x}_M(\omega) = \frac{1}{N}\mathbf{Y}_M \mathbf{g}'_N(\omega), \tag{3}$$

where $\mathbf{g}'_N(\omega)$ is the vector of measured source gains in each of the measurement directions. If the measurement grid of the source is not completely regular, regularization can be used to ensure accurate conversion to the spherical harmonic domain.

[3] Orders $M > 3$ can be accommodated using t-design layouts [11], although the current implementation of the engine is limited to third-order.

4.4 IMPULSE RESPONSE PRE-PROCESSING

To improve CPU usage, the different beam RIRs are processed prior to convolution to provide a single HOA-RIR for the given instrument directivity and real-time orientation data, minimizing the number of real-time convolutions. The N-beam HOA-RIRs are first loaded into the engine and all $N \times (M_{\mathrm{rec}}+1)^2$ IRs are processed for use in the convolution engine. Due to IR lengths, an overlap-add uniform-partitioned frequency domain convolution is used [22], requiring IRs to be segmented prior to transforming to the frequency domain via FFT.

The composite source is represented by N beams, each with an HOA receiver. As $N \geq (M+1)^2$, the source representation is converted from a beam-basis to a spherical harmonic basis. The result a source represented in spherical harmonics, with each harmonic having an HOA receiver. A composite HOA-RIR is generated as a weighted sum across source-spherical harmonics (detailed in Sec. 4.5). With the potential RIR lengths and the dynamic update requirements, this approach ensures a minimum number of channels during real-time processing.

Conversion from an N-beam source representation to spherical harmonics is done on a per-segment basis. The $N \times L$ matrix transfer functions $\mathbf{H}_i(\omega)$ of i-th segment are converted to a spherical harmonic representation via;

$$\hat{\mathbf{H}}_i(\omega) = \mathbf{B}_M^{\dagger}\mathbf{H}_i(\omega) \tag{4}$$

where $\hat{\mathbf{H}}_i(\omega)$ is the $(M+1)^2 \times L$ matrix of the spherical harmonic representation of the source for each of the L playback channels for the i-th segment, $\mathbf{B}_M = [\mathbf{b}_M(\theta_1,\phi_1),\ldots,\mathbf{b}_M(\theta_N,\phi_N)]^{\mathrm{T}}$ is the matrix of beam spherical harmonics from equation (2), and † indicates the Moore-Penrose pseudo-inverse.

$\hat{\mathbf{H}}_i(\omega)$ is stored for use during real-time processing. All subsequent processing is performed in the frequency domain, avoiding costly FFTs of lengthy new impulse responses for each change of source orientation.

It should be noted that geometrical acoustic simulations can include stochastic elements due to methods of scattering/diffusion implementation. This means that the summation of the results of multiple simulations would exhibit amplitude summing in the direct and early portions of the RIR and energetic summation in the later portions. The impact of this remains to be examined.

4.5 INSTRUMENT SOURCE ROOM IMPULSE RESPONSE GENERATION

The engine allows for the rendering of a sound source with a pre-defined directivity pattern $\mathbf{x}_M(\omega)$ and variable orientation. The advantage of representing the source directivity in the spherical harmonic domain is that it allows for simple, efficient rotation via the application of a rotation matrix, ensuring that the source can have its orientation changed freely, following the motion of the musician playing in the virtual room. Assuming the directivity pattern of the source is time-invariant and only the orientation changes, the rotated spherical harmonic source directivity is obtained by;

$$\hat{\mathbf{x}}_M(\omega,t) = \mathbf{R}(\alpha,\gamma,\beta,t)\mathbf{x}_M(\omega), \tag{5}$$

where $\mathbf{R}(\alpha,\gamma,\beta,t)$ is the $(M+1)^2 \times (M+1)^2$ time-varying rotation matrix as a function of the rotation angles yaw α, pitch β and roll γ. The rotations are applied in the order yaw, pitch, then roll.

The spherical harmonic representation of the current source directivity is then used to calculate the final transfer function for each of the I segments. This is achieved by multiplying $\hat{\mathbf{H}}_i(\omega)$ (from equation (4)) by the desired source directivity pattern $\hat{\mathbf{x}}_M(\omega,t)$.

$$\bar{\mathbf{h}}_i(\omega,t) = \hat{\mathbf{H}}_i^T(\omega)\hat{\mathbf{x}}_M(\omega,t). \tag{6}$$

The transfer functions $\bar{\mathbf{h}}_i(\omega,t)$ for each of the segments are then loaded in to the partitioned convolution engine.

4.6 SOURCE RENDERING

The captured mono input is convolved with resulting set of transfer functions $\bar{\mathbf{h}}_i(\omega,t)$ to generate L HOA output channels. In order to continuously update the impulse responses, a separate thread is run that monitors the orientation of the source. If a change occurs then it generates a new composite transfer function $\bar{\mathbf{h}}_i(\omega,t)$ corresponding to the latest orientation. Once the new HOA-RIR has been generated (see Sec. 4.5), it is loaded in to the convolution engine. This process ensures that audio dropouts do not occur while the audio thread waits for a new HOA-RIR as the generation time can be longer than one I/O buffer, especially for long RIRs and higher playback orders.

A change in source directivity or orientation necessitates the generation of a new HOA-RIR. This, in turn, requires interpolation between the old and new impulse responses used in the convolution engine to avoid audible artefacts. This is achieved by cross-fading between two convolution engines: one using the old HOA-RIR and another using the new. This smooths any discontinuities and removes artefacts from the output stream at the CPU cost of requiring a second convolution engine.

4.7 PLAYBACK

The output of the convolution engine is an HOA audio stream. Higher order signals provide greater spatial and timbral fidelity, at the cost of increased CPU load. The HOA stream can be decoded to an array of loudspeakers, allowing the musician to be placed acoustically in the virtual room. Playback on loudspeakers requires an appropriate decoder to ensure low reproduction error. As an alternative to decoding to loudspeakers, the HOA stream can be decoded to binaural audio for playback over headphones using a virtual loudspeaker approach [15] or more advanced methods [25, 21]. If binaural decoding is used, the orientation of the musician's head must be tracked in addition to that of the instrument, with the sound field appropriately rotated to compensate, to ensure a realistic rendering. HOA allows for efficient rotation of the entire sound field via the application of a rotation matrix to the audio signals to generate a new audio stream, similar to the rotation of the source directivity in Eq. (5).

4.8 POTENTIAL IMPROVEMENTS AND EXTENSIONS

The current engine uses time-domain cross-fades when a new HOA-RIR is loaded to the convolution process, requiring two convolution engines to run concurrently. A more efficient method is available that applies the cross-fade in the frequency domain [23, 24] and could improve the efficiency of the current engine.

It is also possible to interpolate between different directivity patterns in the spherical harmonic domain. This would, for example, allow for the variation of directivity patterns based on the source characteristics. For example, the directivity pattern of a singer could be changed to adapt to different vowel sounds, which have been shown to produce different directivities [13].

ACKNOWLEDGEMENTS

This work was funded in part by a starter grant from the Collegium Musicae Institute, Sorbonne Université. Additional funding has been provided by a grant from the *Fondation des Sciences du Patrimoine* and a grant from the Ile-de-France Region – DIM "*Matériaux anciens et patrimoniaux*".

REFERENCES

[1] B. B. Boren. Computational acoustic musicology. *Digital Scholarship in the Humanities*, 12 2018.

[2] H. Boutin, S. Le Conte, S. Vaiedelich, B. Fabre, and J.-L. Le Carrou. Acoustic dissipation in wooden pipes of different species used in wind instrument making: An experimental study. *J Acous Soc Am*, 141(4):2840–2848, 2017.

[3] J. M. Coles. *Experimental Archaeology*. Academic Press, London ; New York, 1979.

[4] S. L. Conte. Beyond the boundaries: A multidisciplinary approach of the musical instrument. *J Cultural Heritage*, 27(SI: Wooden Musical Instruments):S1, 2017.

[5] B. Dalenbäck. *CATT-Acoustic v9 powered by TUCT user manuals*. Computer Aided Theatre Technique, Gothenburg (Sweden):, 2009.

[6] J. Daniel. *Représentation de champs acoustiques, application à la transmission et à la reproduction de scènes sonores complexes dans un contexte multimédia*. PhD thesis, University of Paris 6, 2000.

[7] A. Ernoult, B. Fabre, C. Balosso-Bardin, I. Franciosi, and P. De La Cuadra. Music in a bag? Controlling the bag of Majorcan and Galician bagpipes. *J Acous Soc Am*, 142(3):1454–1465, 2017.

[8] P. Eveno and S. L. Conte. An acoustical measurement used for the understanding of historical wind instruments. *J Cultural Heritage*, 20(SI: Cultural HELP 2014):615–621, 2016.

[9] C. Fritz and D. Dubois. Perceptual Evaluation of Musical Instruments: State of the Art and Methodology. *Acta Acust united Ac*, 101(2):369–381, 2015.

[10] C. Guastavino, B. F. Katz, J. Polack, D. Levitin, and D. Dubois. Ecological validity of soundscape reproduction. *Acta Acust united Ac*, 91:333–341, 2005.

[11] R. H. Hardin and N. J. Sloane. Mclaren's improved snub cube and other new spherical designs in three dimensions. *Discrete & Comp Geometry*, 15(4):429–441, 1996.

[12] C. R. Herr and G. W. Siebein. An acoustical history of theaters and concert halls: An investigation of parallel development in music, performance spaces, and the orchestra. In C. Barton, editor, *Association of Collegiate Schools of Architecture 86, Constructing Identity*, 1998.

[13] B. F. Katz, F. Prezat, and C. D'Alessandro. Human voice phoneme directivity pattern measurements. In *4th Joint Meeting: ASA and ASJ*, Honolulu, Hawaï, 2006.

[14] J.-L. Le Carrou, S. Le Conte, and J. Dugot. 18th and 19th French harp classification using vibration analysis. *J Cultural Heritage*, 27(Supplement):S112–S119, 2017.

[15] M. Noisternig, A. Sontacchi, T. Musil, and R. Höldrich. A 3D Ambisonic Based Binaural Sound Reproduction System. In *AES Conf 24*, pages 1–5, 2003.

[16] A. Paté, J.-L. Le Carrou, B. Navarret, D. Dubois, and B. Fabre. Influence of the Electric Guitar's Fingerboard Wood on Guitarists' Perception. *Acta Acust united Ac*, 101(2):347–359, Mar. 2015.

[17] B. N. Postma and B. F. Katz. Creation and calibration method of virtual acoustic models for historic auralizations. *Virtual Reality*, 19(SI: Spatial Sound):161–180, 2015.

[18] B. N. Postma and B. F. Katz. Perceptive and objective evaluation of calibrated room acoustic simulation auralizations. *J Acoust Soc Am*, 140(6):4326–4337, Dec. 2016.

[19] B. N. J. Postma, H. Demontis, and B. F. Katz. Subjective Evaluation of Dynamic Voice Directivity for Auralizations. *Acta Acust united Ac*, 103(2):181–184, 2017.

[20] P. J. Reynolds. *Experiment and design: archaeological studies in honour of John Coles*, chapter The nature of experiment in archaeology, pages 156–162. Oxbow Books, Oxford, 1999.

[21] C. Schörkhuber and R. Höldrich. Linearly and Quadratically Constrained Least-Squares Decoder for Signal-Dependent Binaural Rendering of Ambisonic Signals. In *AES Conf Immersive and Interactive Audio*, pages 1–9, York, UK, 2019.

[22] A. Torger and A. Farina. Real-time Partitioned Convolution for Ambiophonics Surround Sound. In *IEEE Workshop Appl of Sig Proc to Audio*, pages 195–198, New York, USA, 2001.

[23] E. Vickers. Frequency-domain implementation of time-varying FIR filters. In *AES Conv 133*, pages 8784:1–9, San Francisco, CA, USA, 2012.

[24] F. Wefers and M. Vorlander. Efficient time-varying FIR filtering using crossfading implemented in the DFT domain. In *Forum Acusticum*, pages 1–6, Kraków, Poland, 2014.

[25] M. Zaunschirm, C. Schörkhuber, and R. Höldrich. Binaural rendering of Ambisonic signals by head-related impulse response time alignment and a diffuseness constraint. *J Acous Soc Am*, 143(6):3616–3627, 2018.