Article

Microphone and Loudspeaker Array Signal Processing to Build a "Radiation Keyboard" for Authentic Samplers

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Abstract: To date electric pianos and samplers tend to concentrate on authenticity in terms of temporal and spectral aspects of sound. They barely recreate the original sound radiation characteristics, contribute to the perception of width and depth, vividness and voice separation, especially for instrumentalists, who are located in the near field. This paper describes an operational procedure to measure, store, and synthesize the complete sound of a harpsichord, including its spatial sound radiation characteristics. First, actuators excite the instrument at the intersection point of each string with the bridge with an exponential sine-sweep. Then, the radiated sound field is recorded in the near and the far field with microphone a rrays. The pressure distribution in the near field is propagated back to the soundboard of the instrument, using Minimum Energy Method. The vibration of each single string is captured with lightweight contact microphones. The soundboard is then replaced by an array of 128 loudspeakers. The loudspeaker signal is a convolution of the back-propagated sweep recording with the string recording to perform a wave field s y nthesis. A bove the s patial Nyquist frequency, the Radiation Method is applied to perform a sound field synthesis which is valid for the listening region of the instrumentalist. The result is an electric harpsichord, that approximates the sound of a real harpsichord precisely in time, frequency, and space domain. Applications for such a radiation keyboard are music performance, instrument and synthesizer building and interactive psychoacoustic research.

Keywords: microphone array; wave field synthesis; acoustic holography; sampler; synthesizer

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1. Introduction

Synthesizers tend to focus on timbral aspects of sound, which contains temporal and spectral features [1,2]. This is even true for modern synthesizers that imitate musical instruments by means of physical modeling [3,4]. Many samplers and electric pianos on the market use stereo recordings, or pseudostereo techniques [5,6] to create some perceived spaciousness in terms of *apparent source width* or *perceived source extent*, so that the sound appears more natural and vivid. However, such techniques

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do not capture the sound radiation characteristics of musical instruments, which may be essential for an authentic experience in music listening and musician-instrument-interaction.

Most sound field synthesis approaches synthesize virtual monopole sources or plane waves [7,8]. Sound field synthesis methods to reconstruct the sound radiation characteristics of musical instruments tend to use no precise, but sparse recordings of the sound field that the musical instrument creates [5], like far field recordings from circular [9,10] or spherical [11] microphone arrays with 24 to 128 microphones. In these studies, a nearfield mono recording is extrapolated from a virtual source point. But instead of a monopole point source, the measured radiation characteristic is included in the extrapolation function, yielding a so-called *complex point source* [9,12,13]. Complex point sources are a drastic simplification of the actual physics of musical instruments; but still, complex point sources able to radiate frequencies with individual amplitude and phase in each direction, which yields plausible physical and perceptual results [5,14]. This is an important quality of sound radiation characteristics, because it creates interaural incoherence in terms of interaural level- and phase differences. These are assumed affect source localization and apparent source width the most [5,15,16]. Experiments have shown that when implementing the radiation characteristics of musical instruments including interaural sound differences, a natural sound is created. This sound is perceived as much more spatial than mono sources or stereo phantom sources [17]. To date, sound field synthesis methods for musical instrument radiation characteristics do not imply interaction with the musical instrument. A recording is rendered afterwards, and not during playing the instrument. The *radiation keyboard* suggested in this paper synthesizes the precise sound radiation characteristics of a musical instrument in real-time during playing. To achieve this, the digital signal processing of established methods of microphone and loudspeaker array technology are combined. Such a radiation keyboard can be a powerful tool for interactive psychoacoustic investigations that are more ecological to musicians than passive listening tests with artificial sounds.

Motions of head and torso are essential for the instrumentalist: they are necessary to reach the lowest and highest registers, improve the instrumentalist's timing and fluency, and help him or her to hear out nuances of the played notes. It has even been demonstrated in [18] that different notes are radiated by different regions from the harpsichord sound board, which improves voice separation in polyphonic pieces. For research in the field of human-instrument-interaction and gesture studies, sociology, and psychoacoustics, a radiation keyboard is a necessary tool to investigate how the sound radiation characteristics influences motions of the instrumentalist during playing, how important radiation characteristics are for synchronization, improvisation, belongingness and group identity in ensemble music, and how radiation characteristics affects perceived timbre, naturalness and plausibility when moving the head and torso during playing. Furthermore, there may be a market for more authentic samplers, electric pianos and alike. Radiation characteristics may be an appreciated new parameter for sound designers to tune the sound of their keyboard, in addition to timbral aspects, like brightness and bandwidth, and temporal aspects, such as ADSR-curves. A radiation keyboard is also ideal for practicing, because it combines the benefits of electric pianos — namely the possibility to lower the volume without affecting the timbre of the instrument — with the benefits of a real piano — namely its vividness and width, due to variations of sound when moving the head during playing.

This paper presents the theoretic foundation of a radiation keyboard that synthesizes the sound radiation characteristics of a musical instrument for the region around the instrumentalist's head while performing. It combines methods of microphone array technology for the nearfield and far-field, sound field synthesis approaches and regularization methods for inverse problems. The described radiation keyboard may be used as a tool for research and, ultimately, as a powerful extension of samplers and synthesizers.

2. Methods

This paper suggests a chain from measuring, over storing, to synthesizing the complete sound of a harpsichord, including temporal, spectral, and spatial characteristics. The harpsichord is used as an

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exemplary instrument, because it creates basically one note for each key that is pressed. In contrast to a piano, level and timbre of this note are barely affected by the pressing velocity, so the harpsichord has comparably few degrees of freedom.

The approach starts with an impulse response measurement to record the sound that each key radiates into the nearfield. The recorded sound field is propagated back to the surface of the instrument using Minimum Energy Method. Then, the soundboard is replaced by a loudspeaker array. Each loudspeaker signal is a convolution of the back-propagated nearfied recording with the string recording to perform a wave front synthesis. Above the Nyquist frequency of the loudspeaker array, the Radiation Method is applied to perform a sound field synthesis which is valid for the region around the instrumentalist.

All single methods are not only well elaborated in theory, but have proven their value in numerous practical applications. They are discussed in this section, accompanied by some illustrations, showing their use in practice.

2.1. Impulse Response Measurement

To reconstruct the vibration of the harpsichord, a measurement setup is installed in a free-field room as illustrated in Fig. 1 for a piano soundboard. In the harpsichord, the largest contributor to the sound radiation is the soundboard. Consequently, a microphone array with square grid points is installed at a distance of 5 cm above the soundboard. The grid density is 4 cm. An acoustic vibrator excites the instrument at the intersection point of string and bridge; the *string termination point* [19].

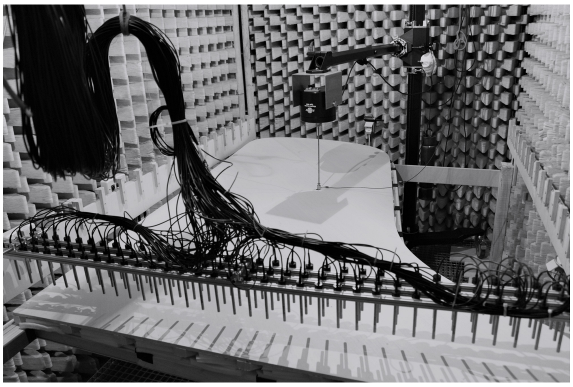

Figure 1. Measurement setup including a microphone array above a soundboard and an acoustic vibrator installed in a free-field room.

2.1.1. Driving Signal

The vibrating soundboard can be considered as a weakly nonlinear system, so a swept-sine of the form

$$s(t) = \sin(\omega(t) \times t) \tag{1}$$

is used, where the angular frequency ω increases over time, like

$$\omega(t) = \omega_0 e^{\frac{t \ln \frac{\omega_{\max}}{\omega_0}}{\tau}} . \tag{2}$$

Here, τ is the duration of the signal and lies in the order of several seconds. The deconvolution process is realized by a linear convolution of the measured output $p(t)$ with the temporal reverse of the excitation sweep signal $s(t)$, i.e.,

$$q(t) = p(t) \otimes f^{-1}(t) . \tag{3}$$

This eliminates nonlinear distortion products and gives additional information about the level of nonlinearity of the considered system.

"Sensors at the driving points measure both input force and acceleration for input mobility calculations."

The driving signal and the convolution are reproducible, i.e., repeated measurements can be carried out to sample the radiated sound field with a grid density of 4 cm.

2.1.2. Nearfield Microphone Array

All following equations are frequency-dependent, but for matters of readability, the term ω is omitted. The harpsichord soundboard is a continuous radiator of sound, but can be simplified as a discrete distribution of N radiating points $\vec{Y}$, referred to as *equivalent sources* [20]. The validity of this simplification is restricted by the Nyquist-Shannon theorem, i.e., two equivalent sources per wave length in the wood are necessary. Then, the relationship between the radiating soundboard and the microphone recordings $P_{\vec{X}}$ can be described by a linear equation system

$$P_{\vec{X}} = \mathbf{G} P_{\vec{Y}} \,, \tag{4}$$

where

$$\mathbf{G} = \frac{e^{i(kr)}}{r} \tag{5}$$

is the Free field Greens' function, describing the equivalent sources as monopole sources. Here, i is the imaginary unit, defined as $i^2 = -1$. Eq. 4 is closely related to the Rayleigh I integral which is applied in acoustical holography and sound field synthesis approaches, like wave field synthesis and ambisonics [21]. One problem in Eq. 4 is that the linear equation system is ill-posed. The radiated sound $P_{\vec{X}}$ is recorded but the source sound $P_{\vec{Y}}$, which created the recorded sound pressure distribution, is sought. When solving the linear equation system, e.g., by means of Gaussian elimination or a pseudo-inverse matrix of $\mathbf{G}$ [22], the resulting sound pressure levels tend to be huge due to small rounding errors and measurement noise. The reason for that is that the propagation matrix is ill-conditioned when microphone positions are close to one another compared to the considered wavelength. In this case the propagation matrix condition number is high. A regularization method relaxes the matrix and yields lower amplitudes. An overview about regularization methods can be found in [20,22]. For musical instruments, the Minimum Energy Method (MEM) [20,23] is superior. The MEM is an iterative approach, gradually reshaping the radiation characteristic of $\mathbf{G}$ from monopole at $\Omega = 0$ to a ray at $\Omega = \infty$ using the formulation

$$\Psi\left(\omega, \varphi, \vartheta\right) = 1 + \Omega \times \left(1 - \alpha\right) \,, \tag{6}$$

where Ψ is a reshaped complex transfer function replacing $\mathbf{G}$ in Eq. 4. The terms φ and ϑ are the azimuth and polar angle and α describes the angle between $\mathbf{Y}_n$ and $\mathbf{X}_m$ as inner product of both position vectors

$$\alpha_{m,n} = \left| \frac{\vec{Y}_n \vec{X}_m}{\left|\vec{X}_m\right| \left|\vec{Y}_n\right|} \vec{n} \right| \,. \tag{7}$$

The angle α is given by the constellation of source- and receiver positions and is 1 in normal direction $\vec{n}$ of the considered equivalent source position and 0 in the orthogonal direction. The correct value for Ω minimizes the reconstruction energy

$$E \propto \sum_{n}^{N} \left| \frac{\partial P_{\vec{Y}_n}\left(\omega\right)}{\partial n} \right|^2 = \min \,. \tag{8}$$

The energy E is proportional to the sum of the squared pressure amplitudes on the considered structure. In a first step, the linear equation system is solved for integers from $\Omega = 0$ to $\Omega = 10$ and the reconstruction energy is plotted over Ω. Around the local minimum, the linear equation system is again solved, this time in steps of 0.1. Typically, the iteration is truncated after the first decimal place. Alternatively, the parameter Ω can be tuned manually to find the best reconstruction visually; the correct solution tends to create the sharpest edges at the instrument boundaries, with pressure amplitudes near 0. This is a typical result of the truncation effect: the finite extent of the source causes an acoustic short-circuit. At the boundary, even strong elongations of the sound board create hardly any pressure fluctuations, since air flows around the sound board.

The radiation of a number of musical instruments has been measured using the described microphone array setup and MEM, like grand piano [24,25], vihuela [20,26], guitars [26,27], drums [20,23,28], flutes [23,28], and the New Irland lounuet [29]. The method is so robust that the geometry of the instruments becomes visible, as demonstrated in Fig. 2.

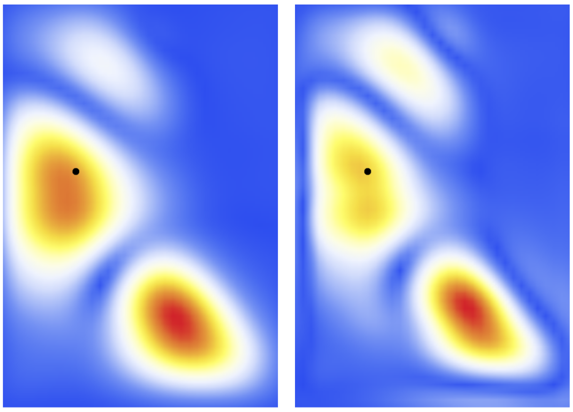

Figure 2. Recorded sound field (left) and vibration of a grand piano soundboard, reconstructed from minimum energy method (right).

As mentioned above, the harpsichord is excited at the intersection point of each string with the bridge, so the nearfield measurement and back-propagation are carried out for each note. For a harpsichord with 4 octaves, this yields 48 measurement series.

2.1.3. Far field Microphone Array

In addition to the near field recordings, the radiated sound is also recorded with a microphone array that samples the region in which the instrumentalist's head may be located during playing. We refer to this region as the *listening volume* and to the discrete sample points as *listening points*. Strictly speaking, this is still the near field for low but the far field for high frequencies. In the near field measurement one planar microphone array samples a planar region parallel to the sound board. In the far field measurement the planar microphone array samples a rectangular cuboid. These listening points are depicted as small spheres in Fig. 3. As for the near field measurements, the spatial Nyquist frequency (or spatial aliasing frequency) in this volume is 4 kHz.

2.2. Creation of Loudspeaker Driving Signals

For synthesizing the sound field, it is meaningful to divide the harpsichord signal into three frequency regions: Frequency region 1 lies below 1.5 kHz, the spatial Nyquist frequency of the loudspeaker array. Frequency region 2 ranges from 1.5 kHz to 4 kHz, the spatial Nyquist frequency of the microphone array. Frequency region 3 lies above these Nyquist frequencies. Different sound field synthesis methods are ideal for each region and should be combined as described in the following. We create individual impulse responses for each frequency region and finally combine them to one

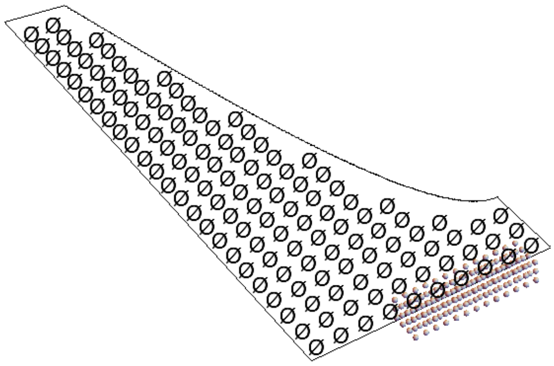

Figure 3. Setup with exemplary loudspeaker positions (⊘) and discrete listening points (small spheres) sampling a listening volume.

broadband impulse response for each loudspeaker. A convolution of the broadband impulse response with a string recording yields the loudspeaker driving signals.

2.2.1. String recording

For each note, the vertical polarity of the transverse string acceleration is recorded with a lightweight piezoelectric transducer. In contrast to a microphone recording, this gives a clean source sound for each note, which is independent of sound radiation characteristics. Alternatively, the vertical polarity can be recorded by means of a high speed camera recording as described in [25,30].

2.2.2. Frequency Region 1 (Wave Field Synthesis)

In principle, a convolution of the back-propagated impulse response of each equivalent source with the transient string acceleration can serve as a loudspeaker driving signal. This approach resembles the idea of an *acoustic curtain* [31], which is considered as one of the earliest wave field synthesis approaches [8,21,32]. In physical terms, this situation is a spatially truncated discrete Rayleigh integral which is the core of wave field synthesis [7,8,21,33]. A prerequisite is that all loudspeakers are homogeneous radiators in the half-space above the soundboard. For loudspeakers without a cabinet, low frequencies approximate dipoles fairly well in the low frequency region. While truncation creates artifacts in most wave field synthesis setups, referred to as *truncation error* [8,21,32], no artifacts are expected in the described loudspeaker grid, due to natural tapering: At the boundaries of the loudspeaker array, an acoustic short-circuit will occur, because the loudspeakers have no cabinet. However, the acoustic short-circuit also occurs in real musical instruments, as demonstrated in Fig. 2. This is because compressed air in the front flows around the sound board towards the rear, instead of propagating as a wave. The acoustic short-circuit of the outermost loudspeakers acts like a natural tapering window, which is artificially created in wave field synthesis installations to compensate the truncation error.

Loudspeaker placement

At this point we have over one thousand equivalent sources. Replacing all of them by an individual loudspeaker is not ideal, because the spacing is too dense for broadband loudspeakers, and it is challenging to synchronize thousands of channels for real-time audio processing. Audio interfaces including D/A-converters for 128 synchronized channels in audio-cd quality are commercially available, using e.g. MADI or Dante protocol. In wave field synthesis systems, regular loudspeaker distributions have been reported to deliver the best synthesis results [34]. Covering the complete soundboard of a harpsichord with a regular grid consisting of 128 grid points, yields about one loudspeaker every 12 cm. This is a typical loudspeaker density in wave field synthesis systems and yields a spatial Nyquist frequency of about 1.5 kHz for waves in air [7]. It means that every third

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equivalent source can be replaced by a loudspeaker without affecting the radiated sound field of frequencies below 1.5 kHz. If we consider wave lengths in wood, the Nyquist frequency may be even higher.

At the ideal location of the loudspeaker grid, all loudspeakers would lie at regions of high amplitude of the soundboard for all frequencies of all notes. This way, all loudspeakers would contribute efficiently to the wave front synthesis, by constructive interference of the wave fronts. This ideal location can be approximated by calculating the energy of the convolved signal at each grid point for each note, as in Eq. 8. But this time, the position of the loudspeaker grid that yields maximum energy is considered as ideal. This way we find the ideal location of 128 loudspeakers that synthesize the sound field by means of wave field synthesis below the Nyquist frequency of 1.5 kHz. An exemplary loudspeaker distribution is illustrated in Fig. 3

2.2.3. Frequency Region 2 (Radiation-Method)

The spatial Nyquist frequency of the loudspeaker array lies around 1.5 kHz. For reproduction of higher frequencies, wave front synthesis is an inadequate means.

The far field recordings from the harpsichord described in Sect. 2.1.3 provide a sample of the desired sound field in the region in which the instrumentalist is moving the head. Alternatively, one could calculate the desired sound field. This can be achieved by forward-propagation of the sound pressure distribution that has been calculated from back-propagation of the nearfield recordings via Minimum Energy Method as described in Sect. 2.1.2 and demonstrated, e.g., in [35]. Both methods should yield the same pressure distribution within the listening volume for each frequency of each played note: the *desired sound field*.

Now that the desired sound field is known and the loudspeakers are distributed over a soundboard-shaped surface, the far field recording is repeated. Only this time the swept sine is radiated by each individual loudspeaker, instead of the soundboard. The relationship between the sound field at the listening points $P_{\vec{X}}$ and the 128 loudspeaker locations $P_{\vec{Y}}$ can be described by Eq. 4. As for the nearfield microphone array measurement, this is an inverse problem which may be ill-conditioned and needs relaxation. The MEM described above is one possible solution, which has been examined already in a comparable scenario in [17,33]. In this case the sound radiation characteristics of the loudspeakers would be assumed to lie somewhere between a monopole and a ray. Other regularization methods [36], mainly leveraged in higher order ambisonics approaches, are conceivable as well. However, the radiation method [17,21,33,37] is preferred in the given constellation, as it comes closer to the physical reality of the given problem. The radiation method takes the sound radiation characteristics of the loudspeakers into account.

The complex transfer function $\mathbf{R}$ between each loudspeaker $\vec{Y}$ and each listening point $\vec{X}$ is determined by simply recording the propagated swept sine, Eq. 2, of each loudspeaker at each listening point, followed by the deconvolution, Eq. 3. The transfer function is neither the idealized monopole source radiation $\mathbf{G}$, nor the energy-optimized radiation function $\mathbf{\Psi}$. Instead, $\mathbf{R}$ is the actual transfer function as measured physically. It includes the frequency and phase response of the loudspeakers, the amplitude decay and the phase-shift from each loudspeaker to each receiver. It can thus be considered the true transfer function. It includes the sound radiation characteristics of the loudspeakers, which tend to deviate from $\mathbf{G}$ and $\mathbf{\Psi}$. Accounting for the actual transfer function from each loudspeaker to each listening point has the advantage that the rows in the linear equation system tend to deviate stronger in reality compared to idealized monopole radiators. This has been demonstrated in [17,33]. The radiation method is a robust regularization method that has been demonstrated to relax the linear equation system [9,21,33]. It leads to a) low amplitudes and b) solutions that vary only slightly, when the source-receiver constellation or the source signal is varied slightly. The solution is valid for the considered frequency region, i.e., between 1.5 and 4 kHz. For higher frequencies the solution is valid for the discrete listening points but not necessarily for the space in between. Consequently, the critical frequency does not depend on the loudspeaker distribution but on the distribution of listening points that sample the listening area. It is possible to approximate solutions for an over-determined equation system. In this case, the solution would be valid for more listening points than loudspeakers present in the radiation keyboard, so the listening volume can be enlarged or the critical frequency could be increased. This way a solution for frequencies above 4 kHz can be found. An approximate solution for an over-determined linear equation system with more listening points than loudspeakers could be chosen to be a least-square solution or a solution in which the length of the impulse response is minimized. However, another method may be preferred, as it ensures a good impulse fidelity of the loudspeaker driving signals. This method is described below.

2.2.4. Frequency Region 3 (Maximize Impulse-Fidelity)

The auditory system is insensitive to phase and interaural phase difference of frequencies above 1.2 kHz [5,38]. Consequently, the amplitude in the listening region is more important than the phase. Therefore, the phase of high frequencies should be aligned with the phase of low frequencies as derived in Sect. 2.2.2. The more frequencies are in phase, the shorter the resulting impulse response. A short impulse response maximizes the impulse fidelity. This is necessary to retain the impulsive character of a harpsichord.

2.3. Real-Time Sound Field Synthesis

We combine the three spectrally truncated impulse responses to one individual broadband impulse response for each loudspeaker and each note. These are convolved with the source signal as described in Sec. 2.2.1. This yields one sample for each loudspeaker and note. For each loudspeaker one instance of a sampler is initialized in a sequencer. The signal for the corresponding loudspeaker is loaded for each note. Technologies like VST and Direct-X are able to handle this parallelism, and several multi-channel sequencers (like Steinberg Cubase, Ableton Live and Magix Samplitude) can handle the high number of output channels. Finally, the original keyboard of the harpsichord is replaced by a MIDI-Keyboard, whose note-on command triggers the 128 samples for the corresponding note.

As the effect of key velocity on the created level and timbre is negligibly, the harpsichord is the ideal instrument to start with; only one sample per note and loudspeaker is necessary. For other instruments, like the piano, the attack velocity affects the produced level and timbre. Here, several samples per note, or one attack-velocity controlled filter has to be applied.

3. Conclusion

The theoretic foundation of a radiation keyboard has been presented. It includes the complete chain from recording the source sound and the radiated sound of a harpsichord to synthesizing its temporal, spectral and spatial sound within an extended listening volume, controlled in real-time. To achieve this, established methods of impulse-response measurements, nearfield and far-field recordings, forward- and backward-propagation, and sound field synthesis are combined. The initial outcome of such a radiation keyboard is a sampler that mimics not only the temporal and spectral aspects of the original musical instrument, but also its spatial aspects. Moreover, the radiation keyboard allows for control over the source sound and over the sound radiation characteristics, which then becomes an artistic parameter to play with as a sound designer and music composer.

4. Outlook

The paper presented the theoretic framework of our current research project. We have already built and used many of the single approaches, and started combining them.

The presented radiation keyboard will serve as a research tool to carry out interactive listening experiments that are more ecological than passive listening tests with artificial sounds in a laboratory environment. Note that the radiation keyboard is not restricted to harpsichord sounds. Solely the distribution of the loudspeakers was chosen to be most energy-efficient for harpsichord notes. In principle, any arbitrary sound file can act as source signal and be radiated like a harpsichord. And

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the other way around: Capturing the harpsichord note from the vertical string motion separated the source signal, i.e., the oscillator, from the resonator. On the one hand, this is a simplification, because the string itself is a resonator, which is coupled to the soundboard. On the other hand, the string tends to enslave the soundboard to radiate with the string's eigenfrequencies [39, ch. 2 and 7]. So separating the source sound from the sound radiation characteristics also enables us to radiate this source sound with various sound radiation characteristics. To achieve this, the impulse responses of the loudspeakers have to be altered for each note of the alternative radiation characteristic. Being in phase, for example, approximates a plane wave, which is assumed to sound much narrower than a structure with a complex vibration pattern. Amplitude- and phase randomization, on the other hand, is assumed to sound wide.

Loading different source sounds while keeping the sound radiation fixed, could reveal which temporal and spectral parameters affect the perception of source extent and naturalness in the direct sound of musical instruments. The radiation keyboard could answer the question, whether a saxophone sound with the radiation characteristics of a harpsichord sound larger than a real saxophone. To date, physical predictors of apparent source with disagree strongly, which frequency region is of major importance for the listening impression. Different predictors and the discourse are examined in [5]. Creating different sound radiation characteristics, while keeping the source sound constant, could help creating a transfer function between physical sound radiation characteristics and perceived source extent. This knowledge may become a powerful, expressive tool for sound designers and instrument builders of physical musical instruments, samplers, electric and electronic instruments. *Sound radiation characteristics*, or *source width* may become a new parameter in synthesizer design and audio mixing consoles for immersive audio.

The strength of a real-time capable radiation keyboard is the interactivity: musicians can actively play the instrument instead of carrying out passive listening tests. Interactivity creates a dynamic sound and allows for a natural interaction in an authentic musical performance scenario. This is a necessity in the field of performance, gesture, and human-machine-interaction studies and a prerequisite for ecological psychoacoustics.

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