**Figure 1b:** *Photo of loudspeaker system used for research on sound field synthesis. Pictured is a 64-channel rectangular array at Signal Theory and Digital Signal Processing Group, University of Rostock*

# Sound Field Synthesis for Audio Presentation

*The use of loudspeaker arrays for audio presentation offers possibilities that go beyond conventional methods like Stereophony.*


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In this article, we describe the use of loudspeaker arrays for sound field synthesis with a focus on the presentation of audio content to human listeners. Arrays of sensors and actuators have played an important role in various applications as powerful technologies that create or capture wave fields for many decades (van Trees, 2002). In acoustics, the mathematical and system theoretical foundations of sensor and transducer arrays are closely related due to the reciprocity principle of the wave equation (Morse and Feshbach, 1981). The latter states that sources and measurement points in a sound field can be interchanged. Beamforming techniques for microphone arrays are deployed on a large scale in commercial applications (van Veen and Buckley, 1988). Similarly, arrays of elementary sources are standard in radio transmission (van Trees, 2002), underwater acoustics (Lynch et al., 1985), and ultrasonic applications (Pajek and Hynynen, 2012). When the elements of such an array are driven with signals that differ only with respect to their timing then one speaks of a phased array (Pajek and Hynynen, 2012; Smith et al., 2013). Phased arrays have become extremely popular due to their simplicity.

We define sound field synthesis as the problem of driving a given ensemble of elementary sound sources such that the superposition of their emitted individual sound fields constitutes a common sound field with given desired properties over an extended area. As discussed below, phased arrays in their simplest form are not suitable for this application and dedicated methods are required.

The way electroacoustic transducer arrays are driven depends essentially on what or who receives the synthesized field. Many applications of, for example, phased arrays aim at the maximization of energy that occurs at a specific location or that is radiated in a specific direction while aspects like spectral balance and time-domain properties of the resulting field are only secondary (Pajek and Hynynen, 2012; Smith et al., 2013). The human auditory system processes and perceives sound very differently from systems that process microphone signals (Blauert, 1997; Fastl and Zwicker, 2007). Human perception can be very sensitive towards details in the signals that microphone-based systems might not extract and vice versa. Among other things, high fidelity audio presentation requires systems with a large bandwidth (approximately 30 Hz – 16,000 Hz, which corresponds to approximately 9 octaves) and time domain properties that preserve the transients (e.g. in a speech

or music signal). Obviously, the extensive effort of deploying an array of loudspeakers for audio representation only seems reasonable if highest fidelity can be achieved given that Stereophony (stereos: Greek firm, solid; fone: Greek sound, tone, voice) and its relatives achieve excellent results in many situations with just a handful of loudspeakers (Toole, 2008).

At first glance, we might aim at perfect perception by synthesizing an exact physical copy of a given (natural) target sound field. Creating such a system obviously requires a large number of loudspeakers. Though, auditory perception is governed by much more than just the acoustic signals that arrive at the ears; the accompanying visual impression and the expectations of the listener can play a major role (Warren, 2008). As an example, a cathedral will not sound the same when its interior sound field is recreated in a domestic living room simply because the user is aware in what venue they are (Werner et al., 2013). We will therefore have to expect certain compromises when creating a virtual reality system. But we still keep the idea of recreating a natural sound field as a goal due to the lack of more holistic concepts.

The most obvious perception that we want to recreate is appropriate spatial auditory localization of the sound sources a given scene is composed of. The second most important auditory attribute to recreate is the perceived timbre, which is much harder to grasp and control. On the technical side only the frequency response of a system can be specified. As Toole (2008) puts it: "Frequency response is the single most important aspect of any audio device. If it is wrong, nothing else matters." Actually his use of the term "frequency response" encompasses also perceptual aspects of timbre, like distinction of sounds (Pratt and Doak, 1976) or identity and nature of sound sources (Letowski, 1989).

## Why Sound Field Synthesis?

The undoubtedly most wide-spread spatial audio presentation method is Stereophony where typically pairs of loudspeakers are driven with signals that differ only with respect to their amplitudes and their relative timing. Obviously, sound field synthesis follows a strategy that is very different from that of Stereophony. So why not build on top of the latter as it has been very successful?

Remarkably, methods like Stereophony can evoke a very natural perception although the physical sound fields that they create can differ fundamentally from the "natural" equivalent. Extensive psychoacoustical investigations revealed that all spatial audio presentation methods that employ a low number of loudspeakers, say, between 2 and 5, trigger a psychoacoustical mechanism termed summing localization (Warncke, 1941), which had later been extended to the association theory (Theile, 1980). These two concepts refer to the circumstance that the auditory system subconsciously detects the elementary coherent sound sources – i.e., the loudspeakers – and the resulting auditory event is formed as a sum (or average) of the elementary sources. In simple words, if we are facing two loudspeakers that emit identical signals then we may hear one sound source in between the two active loudspeakers (which we interpret as a sum or the average of the two actual sources, i.e., the loudspeakers). This single perceived auditory event is referred to as phantom source (Theile, 1980; Blauert, 1997).

Whether and where we perceive a phantom source depends heavily on the location of the loudspeakers relative to the listener and on the time and level differences between the (coherent) loudspeaker signals arriving at the listener's ears. All these parameters depend heavily on the listener's location. Thus if it is possible to evoke a given desired perception in one listening location (a.k.a. sweet spot) then it is in general not possible to achieve the same or a different but still plausible perception in another location. Note that large conventional audio presentation systems like the one described by Long (2008) primarily address the delivery of the information embedded in the source signals rather than creating a spatial scene and are therefore no alternatives.

At the current state of knowledge it is not possible to achieve a large sweet spot using conventional methods because all translations of the listener position generally result in changes in the relative timing and amplitudes of the loudspeaker signals. Interestingly, large venues like cinemas still employ Stereophony-based approaches relatively successfully. This is partly because the visual impression from viewing the motion picture often governs the spatial auditory one (Holman, 2010). Closing the eyes during a movie screening and listening to the spatial composition of the scene often reveals the spatial distortions that occur when not sitting in the center of the room. The focus lies on effects rather than accurate localization of individual sounds. Additionally, movie sound tracks are created such that they carefully avoid the limitations of the employed loudspeaker systems in the well-defined and standardized acoustic environment of a cinema.

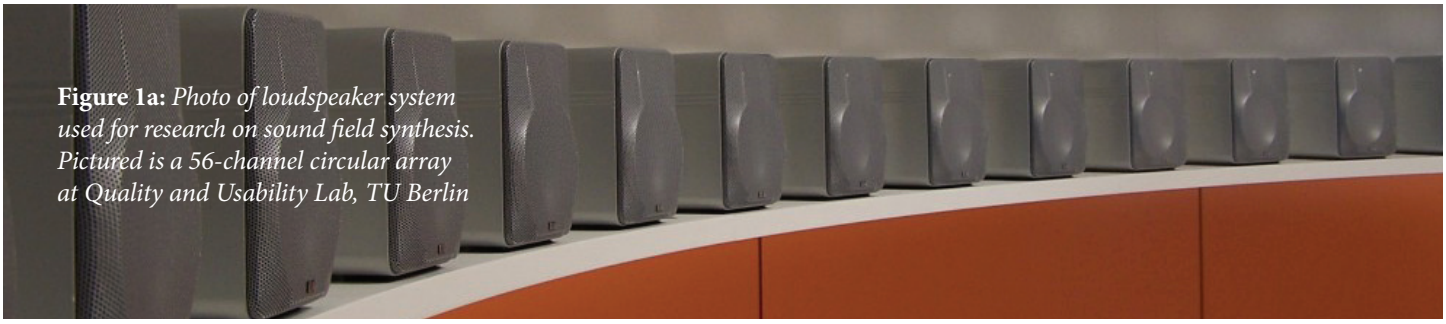


**Figure 1a:** *Photo of loudspeaker system used for research on sound field synthesis. Pictured is a 56-channel circular array at Quality and Usability Lab, TU Berlin*

In conclusion, satisfying an extended listening area with predictable and plausible perception requires approaches different than those based on Stereophony. Sound field synthesis tries to physically recreate natural sound fields so that human hearing mechanisms are addressed.

## A Brief History

The cornerstone of modern sound field synthesis theory was laid by Jessel (1973), whose work is based on some of the most fundamental integral equations in the physics of wave fields such as the Rayleigh Integrals or the Kirchhoff-Helmholtz Integral. Having been ahead of his time, Jessel did not have the means of creating a practical implementation of his work. Concurrent with Jessel, Gerzon (1973) worked with momentum on an approach that he termed Ambisonics (ambo: Greek both together; sonare: Lat. to sound). Gerzon's work used a much simpler and more intuitive theory compared to Jessel's, but Gerzon was soon able to present analog implementations based on a small number of microphones and loudspeakers.

The next big push of sound field synthesis started in the late 1980s with the work of Berkhout (1988) and coworkers (Berkhout et al., 1993) who created an approach that they termed Wave Field Synthesis. Having a background in seismology, Berkhout did not seem to have been aware of Jessel's work but he followed very similar concepts. His ideas were pursued over more than two decades and the team was able to present a ground breaking realtime implementation in 1992 featuring as many as 160 loudspeakers and dedicated digital signal processing hardware (de Vries, 2009).

The comprehensive availability of personal computing and suitable audio hardware led to the latest practical and theoretical push of sound field synthesis from the mid 2000s on resulting in more than 200 commercial and research systems worldwide. The largest one comprises more than 832 independent channels on a quasi-rectangular contour with a circumference of 86 m and fills an entire lecture hall at the University of Technology Berlin, Germany, with sound (de Vries, 2009). Refer to Figure 1a and Figure 1b for photographs of selected systems.

Especially the advancements during the last couple of years led to a mature theoretical and practical understanding of sound field synthesis and the next logical chapter is actively worked on in the audio community: The psychoacoustical study and perceptual evaluation of synthetic sound fields (Spors et al., 2013).

## Theory

Several ways of deriving an analytic solution for the loudspeaker driving signals in sound field synthesis have been presented in the literature (Berkhout, 1993; Poletti, 2005; Spors et al. 2008; Fazi et al., 2008; Zotter et al., 2009). All these solutions start with the assumption of a continuous distribution of elementary sound sources (a.k.a. secondary sources) that encloses the listening area on a boundary surface. Starting with a continuous distribution has the advantage that concepts can be developed for which a perfect solution exists. Other (imperfect) solutions can then be treated as a degenerated problem based on the perfect ones.

An obvious imperfection of practical systems is the circumstance that a continuous distribution of secondary sources is impossible to implement. We always have to use a finite number of discrete sources. Due to technical constraints it is often desired to reduce the two-dimensional boundary surface to a one-dimensional enclosing contour, preferably in a horizontal plane leveled with the listeners' ears.

The imperfections of real-world systems lead to artifacts in the generated sound field which can be described analytically or can be measured for an implemented system. However, they are not always perceptible by human listeners and can thus be tolerated. For convenience, we postpone the discussion of these imperfections and their perception and start the discussion with the ideal case.

Assuming a simply connected enclosing surface $\partial\Omega$ of secondary sources that encloses our target volume $\Omega$, we can formulate the *synthesis equation* in the temporal-frequency domain as

$$S(\boldsymbol{x}, \omega) = \oint_{\partial\Omega} D(\boldsymbol{x}_0, \omega) G(\boldsymbol{x} - \boldsymbol{x}_0, \omega) \mathrm{dA}(\mathbf{x}_0)$$

$D(x_0, \omega)$ represents the driving signal of the secondary source located at point $x_0 \in \partial\Omega$ and $G(x-x_0, \omega)$ represents the spatio-temporal transfer function of that secondary source. We use the letter $G$ because this function can be interpreted as a Green's function. The product $D(x_0, \omega)\, G\,(x-x_0, \omega)$ describes the sound field that is evoked by the considered secondary source. Integration over the entire surface $\partial\Omega$ yields the synthesized sound field $S(x, \omega)$ by summation of all contributions from the elementary sound sources. Usually, one is not interested in what sound field is created when the secondary source contour is driven in a specific way. One would rather want to know how to drive the system that a specific desired sound field arises, i.e., we want to dictate $S(x, \omega)$ and solve (1) for $D(x_0, \omega)$ . It can indeed be shown that a perfect solution exists when the boundary $\partial\Omega$ encloses the target volume and is simply connected (Poletti, 2005; Zotter et al. 2009; Ahrens, 2012; Zotter et al. 2013).

There are two different fundamental approaches for this task: 1) an implicit solution, i.e., we analyze the situation from a physical point of view and exploit our knowledge on the relation between the sound field on the boundary $\partial\Omega$ and the sound field inside the target volume $\Omega$ to derive $D(x_0, \omega)$, and 2) we manipulate (1) mathematically so that we are able to solve it explicitly for $D(x_0, \omega)$. Both approaches are outlined in the following two subsections.

## Implicit Solution

There are several ways of deriving an implicit solution to equation (1) leading to identical results, all of which start from well-known integral representations of sound fields (Berkhout, 1993; Spors et al. 2008). Here, we chose to derive the implicit solution via the Rayleigh I Integral. This derivation appears hands-on at first sight but rigorous treatments exist that prove the appropriateness of the applied approximations (Zotter and Spors, 2013).

The Rayleigh I Integral describes the sound field $P(x,\omega)$ in a target half-space $\Omega$ that is bounded by a planar surface $\partial\Omega$ and is given by (Williams, 1999).

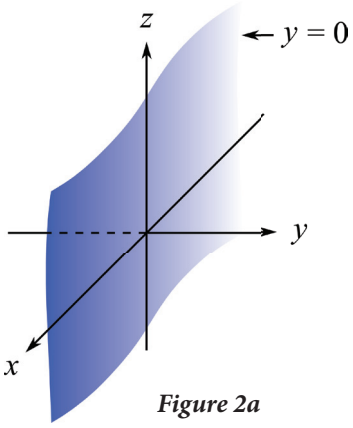



*Figure 2a*

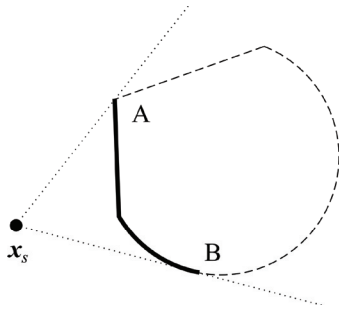



*Figure 2b*

**Schematics illustrating the theory of sound field synthesis.**

**Figure 2a:** *Planar distribution of secondary sources. The distribution is continuous and of infinite extent.*

**Figure 2b:** *Illustration of the secondary source selection that has to be performed when the physical optics approximation is applied. The virtual monopole source is located at $x_s$. The thick solid line represents the active part of the contour; the dashed part represents the inactive part.*

The geometry is depicted in Figure 2(a). In words, the integral in (2) states that we can perfectly recreate a sound field $S(x,\omega)$ that is source-free in the target half-space $\Omega$ if we drive a continuous planar distribution of monopole secondary sources with a signal that is proportional to the directional gradient $\frac{\partial}{\partial n}$ of $S(x,\omega)$ evaluated along the secondary source distribution.

So we actually have a solution for our problem assuming that we are able to implement a continuous distribution of monopole sound sources. This latter assumption is actually fulfilled sufficiently well by small conventional loudspeakers with closed cabinets (Verheijen, 1993). The inconvenience related to the above solution is that the secondary source distribution has to be planar and of infinite extent. Ideally, we want to enclose the target area with a secondary source distribution in order to be able to immerse the listener.

$$P(\boldsymbol{x},\omega) = \iint_{-\infty}^{\infty} \underbrace{-2\,\frac{\partial}{\partial \mathbf{n}}\, S(\boldsymbol{x},\omega)|_{\boldsymbol{x}=\boldsymbol{x}_0}}_{=D(\boldsymbol{x}_0,\omega)} \cdot G(\boldsymbol{x}-\boldsymbol{x}_0,\omega)\mathrm{d}x\mathrm{d}y; \quad P(\boldsymbol{x},\omega) = S(\boldsymbol{x},\omega)\ \forall \boldsymbol{x} \in \Omega$$

If we are willing to accept a far-field/high-frequency solution we can apply the physical optics approximation (or Kirchhoff approximation) (Colton and Kress, 1992). The latter is based on the assumption that a curved surface may be considered locally planar for sufficiently short wavelengths. We can then locally apply the Rayleigh-based solution. Only those secondary sources must be active that are virtually illuminated by the desired sound field as illustrated schematically in Figure 2(b).

Conveniently, the secondary source contour does not need to be smooth. Even corners are possible with only moderate additional inaccuracy (Verheijen, 1997; Ahrens, 2012). When the boundary of the illuminated area is not smooth (like case A in Figure 2(b)) then tapering has to be applied, i.e., a windowing of the amplitude of the driving function towards the end-points to smear the truncation artifacts (Verheijen, 1997).

An essential aspect is of course that the physical optics approximation holds when the dimensions of the secondary source distribution are much larger than that of the considered wavelength. This prerequisite is not always fulfilled in practice at low frequencies where the wavelength can reach several meters.

This approximated solution is much more flexible than the one based directly on the Rayleigh integral but it still requires two-dimensional surfaces of secondary sources. Implementing a surface of secondary sources is a massive effort (Reusser et al., 2013). Recall that we have to approximate a continuous distribution. A densely-spaced placement of the loudspeakers results in channel numbers that are nearly impossible to handle even for moderate sizes of the target space.

In many situations it has been shown to be sufficient to present only the horizontal information with high resolution. All other signals can be delivered by simpler conventional presentation methods or can even be fully discarded. So we seek for a solution that is capable of handling one-dimensional secondary source distributions like rectangles and circles. This solution can be obtained from our previous one by applying another approximation referred to as stationary phase approximation and that reduces the integration of the vertical dimension in Eq. (2) to a single point in the horizontal plane (Berkhout et al., 1993; Vogel, 1993). The result is then termed a 2.5-dimensional solution because it is neither 2D nor 3D, but in between.

The major limitation of the 2.5D solution is the fact that the amplitude decay of the synthesized sound field is typically faster than desired, which turned out to be inconvenience with large systems. However, we still have extensive control over the curvature of the wave fronts in the horizontal plane. Refer to Figure 3 for an illustration.

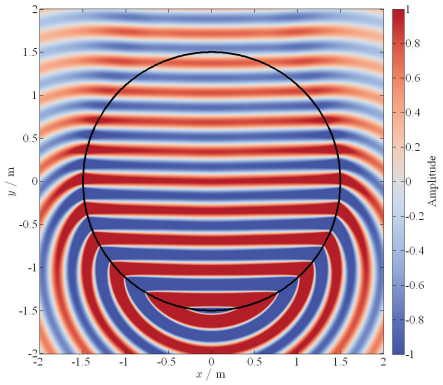



**Figure 3:** *2.5-dimensional synthesis of a monochromatic plane wave of 1000 Hz by a continuous circular distribution of monopole sources. The synthesized plane wave travels into positive -direction. The unintended amplitude decay along the plane wave's travel path is evident. (an animation of Figure 3 is available at: https://acousticstoday.org/sounds/#.U4ihSJRdUto*

A very convenient property of the solution is the fact that it can be implemented extremely efficiently: In order to drive a virtual sound source with a specific signal like a speech or music signal, one single static common filter has to be applied to the input signal: The latter is then delayed and weighted individually for each speaker (Verheijen, 1997). This implementation may be regarded as an advanced phased array.

The 2.5D solution described in the previous paragraph corresponds to the Wave Field Synthesis approach mentioned in the Introduction and proposed in (Berkhout et al., 1993). The vast majority of the existing realtime implementations nowadays use it and can handle hundreds of virtual sources using standard personal computers as processors, for example (Geier et al., 2008).

## Explicit Solution

As an alternative to the implicit solution described in the previous section, it is also possible to solve the synthesis equation (1) explicitly for the unknown function $D(x_0, \omega)$. Taking a second look at (1) reveals that the integral actually constitutes a convolution of the driving function $D(x_0, \omega)$with the

radiation function $G\,(x–x_0, \omega)$ of the secondary sources. For simple contours $\partial\Omega$ like spheres, circles, planes, and lines a convolution theorem can be found that allows for representing (1) in a suitably chosen transformed domain with spatial frequency $\upsilon$ as

$$\check{S}(\nu, \omega) = \check{D}(\nu, \omega)\check{G}(\nu, \omega)$$

Eq. (3) can be solved directly for the driving function $\check{D}(\upsilon,\omega)$ by rearranging the terms. 2.5D cases are handled by referencing the synthesized sound field $\check{S}(\upsilon,\omega)$ to a *reference contour or point* (Ahrens, 2012).

The explicit and implicit solutions are almost equivalent for simple 3D scenarios. For some secondary source geometries – for example spheres – only the explicit solution is exact (Schultz and Spors, 2014). For 2.5D scenarios, the explicit solution is exact on the reference contour or location, where the implicit solution is only an approximation. The latter aspect is not significant in practical scenarios but has been very helpful in analyzing the fundamental properties of synthetic sound fields. Most explicit solutions cannot be implemented as efficiently as Wave Field Synthesis. They rather require designing and applying an individual filter for each combination of virtual sound source and loudspeaker. Nevertheless, realtime performance is still possible (Daniel, 2003; Spors et al., 2011).

The particularly popular explicit solution for spherical and circular secondary source distributions constitutes a modern formulation of Gerzon's *Ambisonics* approach mentioned in the Introduction. The domains into which the synthesis equation is transformed by the according convolution theorems are the spherical harmonics and the circular harmonics coefficients domains, respectively.

## Spatial Discretization

As mentioned previously, practical implementations will employ a finite number of discrete loudspeakers, which constitutes a substantial departure from the theoretical requirements for the solutions outlined above. The consequences of this spatial discretization for the synthesized sound field have been studied extensively in the literature (Start, 1997; Spors and Rabenstein, 2006; Ahrens, 2012). In summary, the synthesized sound field is exact or at least well approximated up to a certain frequency termed *spatial aliasing frequency.*

Above this frequency, two fundamental cases can be distinguished:

1) Additional wave fronts arise, which are termed spatial aliasing. Wave Field Synthesis belongs to this class of approaches. Refer to Figure 4(a) and (b) for an example.

2) A region of high accuracy is still apparent at the center of the secondary source distribution but whose size diminishes with increasing frequency. Outside this region artifacts occur that are different than those in case 1). Modern formulations of Ambisonics belong to this class. Refer to Figure 4(c).

Intermediate cases can also be created. It is not clear at this stage which approach is perceptually preferable in a given scenario so that we leave this question undiscussed. We want to emphasize here that discretization artifacts are not a downside of a given driving method. They rather represent practical restrictions of the loudspeaker arrangement under consideration. The driving method only has influence on how and at what locations artifacts occur.

Typically, the loudspeaker spacing is chosen such that the aliasing frequency lies between 1500 Hz and 2000 Hz. Then the desired sound field is synthesized correctly in that frequency range where the powerful localization mechanisms based on the interaural time difference are active (Blauert, 1997). The resulting loudspeaker spacings of 9-15 cm have been shown to be a good compromise between accuracy and practicability. Recall also the systems shown in Figure 1.

## Perception of Synthetic Sound Fields

Extensive knowledge on the perception of natural sound fields has been gathered during the last century, for example (Blauert, 1997; Bronkhorst, 1999; Beranek, 2008; Toole, 2008), and simple situations are well understood. If we were able to build systems that are able to create a perfect copy of a given natural target sound field, ignoring other modalities, then we were able to predict the perception based on this existing knowledge. Looking at Figure 4 suggests that the task is not that easy because whenever we intend to create one single wave front we effectively create an entire set of wave fronts carrying closely related signals.

Fortunately, it is not necessary to create a perfect physical copy of the target sound field when human listeners are addressed. Instead a sound field is sufficient that sounds exactly like the target field (*authentic* reproduction) or which evokes a perception that is indiscernible from an implicit or

*Figure 4a*

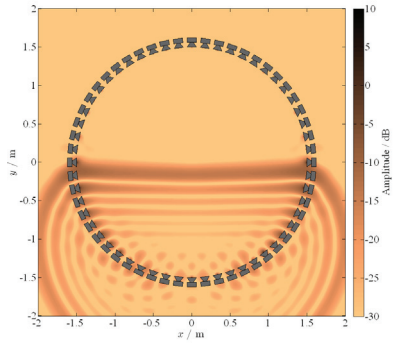



*Figure 4b*

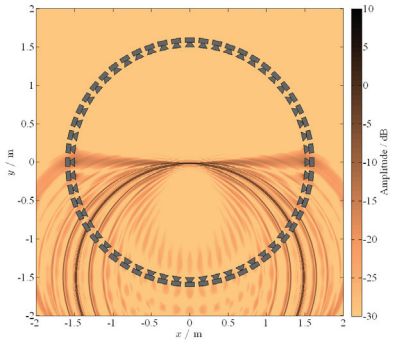
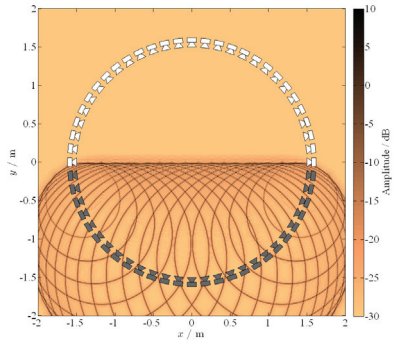



*Figure 4c*




**Time-domain simulations of a circular distribution of 56 monopole loudspeakers synthesizing a plane wave that propagates into positive -direction. (animations of Figure 4(a)-(c) are available at: https://acousticstoday.org/sounds/#.U4ihSJRdUto**

**Figure 4a:** *Bandwidth from 0 Hz to 2000 Hz, explicit solution (all loudspeakers are active); The non-zero components behind the straight wave front are due to the bandwidth limitation and are not artifacts of the driving function.*

**Figure 4b:** *Full bandwidth, case 2), explicit solution (all loudspeakers are active).*

**Figure 4c:** *Full bandwidth implicit solution, case 1); gray loudspeaker symbols represent active loudspeakers, white symbols represent inactive loudspeakers.*

explicit internal reference (*plausible* presentation) (Blauert and Jekosch, 2003). We discuss here in how far it has been proven or refuted that this goal has been achieved.

Sets of coherent wave fronts occur also in rooms where the sound emitted by a given source reaches the listener on a direct path followed by reflections off the room boundaries. After the floor reflection, the wave fronts impinging on the receiver follow the direct sound with a delay of several milliseconds or more. This is because the path of a reflection is usually at least a few meters longer than that of the direct sound and sound travels roughly one meter every 3 ms. However, the wave fronts that we are dealing with in sound field synthesis may have differences in the arrival times in the order of a fraction of a millisecond. This suggests that other hearing mechanisms than in the perception of natural reverberation might be triggered.

As indicated in the Introduction, there is a multitude of perceptual attributes that can be essential when perceptually assessing a spatial audio system. The scope of this article limits our discussion to the two most important attributes: localization and timbral fidelity.

When investigating human auditory perception it is important to distinguish the sound event that describes an event in the physical world and the *auditory event* that represents the perceived entity (Blauert, 1997). Note that a sound event does not always translate directly into an auditory event. Recall that in Stereophony we have two sound events (the loudspeakers) that are perceived as one auditory event (the phantom source). We want to achieve a similar situation in sound field synthesis as well. We would like the individual wave fronts of the loudspeakers to fuse into one auditory event. It has been shown in various places in the literature that this is indeed the case in most situations. Furthermore, it has been shown that auditory localization is accurate and reliable, for example (Vogel, 1993; de Bruijn, 2004; Wierstorf et al., 2012).

The auditory localization properties of a spatial audio system are fairly straightforward to investigate. User studies can be performed in which the listener reports the perceived location via a suitable pointing method. The perceived timbre on the other hand is composed of more abstract perceptual dimensions and can neither be measured directly nor can it be represented by a numerical value. A number of studies have been presented in the literature but the topic is still under active research so that no ultimate conclusion can be drawn. We summarize two representative sample studies in the following.

One way of assessing perceived coloration is making the subjects compare a given stimulus to a reference and making them rate the difference on a given scale (for example *no difference – extremely different*). The reference is typically a single loudspeaker at the position of the virtual source. Assuring equal conditions for all subjects – especially identical listening positions – is difficult with a real loudspeaker array. Most experiments therefore employ headphone simulations of a given loudspeaker array whose head-related impulse responses had been measured (a.k.a. *binaural simulation*) (Wittek, 2007). So did the studies mentioned below.

De Bruijn (2004) investigated the variation of timbre in Wave Field Synthesis over the listening area without assessing the actual absolute coloration. The motivation for skipping the latter was the assumption that it should be possible to compensate the system for absolute systematic coloration. This assumption is only partly true as coloration is not exclusively determined by the frequency response of a system but can also occur due to the presence of more than one coherent wave front (Theile, 1980). No methods for compensation in the latter situation are known. De Bruijn found that the variation of timbre is negligible for a loudspeaker spacing of 0.125 m but perceivable for larger spacings.

Wittek (2007) measured the variation of timbre of Wave Field Synthesis and Stereophony for different positions of the virtual source. He also included a single loudspeaker as stimulus. This gives indications on the absolute coloration of the tested methods as the coloration introduced by the loudspeakers themselves is ignored. His findings are that the coloration of Stereophony in the sweet spot and the coloration of WFS for loudspeaker spacings of 0.03 m are not stronger than the coloration produced by a single loudspeaker. Coloration is similarly strong for all larger loudspeaker spacings (tested up to 0.5 m) for the listening position that he investigated.

Above cited results give a first indication of what we can expect from sound field synthesis when it is used for audio presentation. These results are partly encouraging and partly discouraging. A fundamental problem is that it is not clear how the human auditory system integrates the various occurring coherent wave fronts into one auditory event. It is therefore not clear how we should shape the unavoidable spatial aliasing artifacts such that their perceptual impact is minimal. More fundamental psychoacoustical work is need.

Meanwhile another important aspect is under investigation: Artificial reverberation is an extremely essential component of high fidelity spatial audio signals (Izhaki, 2007). "Dry" virtual scenes lack spaciousness and plausibility (Shinn-Cunningham, 2001). It has been proposed in Ahrens (2014) to design artificial reverberation in sound field synthesis such that the additional wave fronts that occur due to spatial aliasing make up a plausible reflection pattern to thereby "hide" the artifacts in the reverberation (or actually make the artifacts part of the reverberation).

Examples for other topics under investigation are the rendering of spatially extended virtual sources (Nowak et al., 2013) as well as the combination of stereophonic techniques with sound field synthesis (Theile et al., 2003; Wittek, 2007).

## Extensions and Applications

Sound field synthesis can be performed both with virtual sound scenes, i.e., with sound scenes that are composed of individual sound sources that have an input signal, position, radiation properties, etc. that are described in metadata. Or, sound scenes can be recorded using appropriate microphone arrays such as spherical and circular ones. For convenience, we show two examples in Figure 5 of special virtual sound sources that can be used in the former case:

- *Focused* virtual sound sources: A synthesized sound field can be designed such that it converges in one part of the listening area towards a focus point and diverges behind that focus point (Verheijen, 1997; Ahrens and Spors, 2009). Refer to Figure 5(a). When a listener is located in the diverging part of the sound field they perceive a virtual sound source "in front of the loudspeakers."
- *Moving* virtual sound sources (Ahrens and Spors, 2008; Frank, 2008): As evident from Figure 5(b), it is possible to synthesize the sound field of a moving sound source so that the Doppler Effect is properly recreated, not only the frequency shift as it is the case with conventional methods.

This history of sound field synthesis as well as the overview presented in this article have been guided by a traditional application: the presentation of audio content to human listeners. This is also the application that the vast majority of the commercial systems mentioned in the Introduction focus on. However, there are also emerging applications that go beyond entertainment and infotainment. A few of these are mentioned here briefly:

- While visual rendering in the planning stage is a state-of-the-art feature of architectural software, the corresponding audio rendering of the expected noise exposure of new industrial or traffic infrastructure is still in its infancy. Here the purpose is not to please the listener with sound, but to create a virtual acoustic environment that conveys the correct level of annoyance for assessment by human listeners (Vorländer, 2010; Ruotolo et al., 2013).

*Figure 5a*

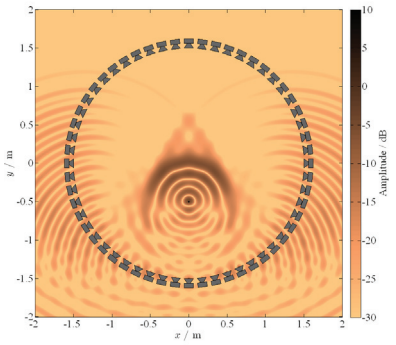
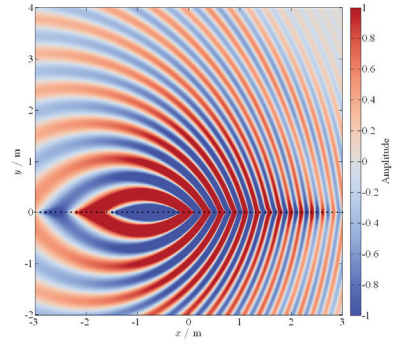



*Figure 5b*




**Simulations of synthetic sound field originating form virtual sound source with complex properties**
*(Animations of Figure 5a and 5b are available at:* https://acousticstoday.org/sounds/#.U4ihSJRdUto

**Figure 5a:** *Time-domain simulation of a focused monopole source. The black mark represents the location of the focus point at (x, y) = (0, -0.5) m. The focused source radiates in direction of the positive y–axis.*

**Figure 5b:** *Monochromatic simulation of a moving monopole sound source of 1000 Hz moving parallel to the x–axis at a velocity of 240 m/s. The marks represent the positions of the loudspeakers.*

- Testing of mobile speech communication equipment has to include also the performance in adverse acoustical environments. Rather than conducting extended outdoor test drives, the spatial and spectral structure of street noise can also be reproduced in the laboratory with suitable sound field synthesis techniques.
- Noise rendering still tries to produce some kind of reality, but there are also attempts to create sound fields that have no counterpart in the real world. An example is the creation of zones of silence for a part of the listeners while exposing others nearby to an intended acoustic content (Wu and Abhayapala, 2011; Helwani et al., 2014). This approach can also be used to deliver different kinds of auditory events to users in different locations of the listening space, for example the different seats of a car. The challenge is to provide individualized sound events with minimal crosstalk.
- As robots of various kinds are introduced to replace or extend human functions also the acoustic perception of robots is investigated. Of course the hearing systems of robots are purely technical and their abilities are by far inferior to human perception. Further developments of robot audition require reproducing sound fields with well-defined physical properties, since psychoacoustics in the traditional sense does no longer apply (Tourbabin and Rafaely, 2013). Similarly, also the research on hearing aids requires the ability to synthesize complex sound fields under laboratory conditions (Vorländer, 2010).

## Biosketches

***Jens Ahrens*** received a Diploma in Electrical Engineering/Sound Engineering with distinction (equivalent to Master of Science) from Graz University of Technology and University of Music and Dramatic Arts Graz, Austria, and the Doctoral Degree with distinction (Dr.-Ing.) from University of Technology Berlin, Germany. From 2006 to 2011 he was member of the Audio Technology Group at Deutsche Telekom Laboratories / TU Berlin where he worked on the topic of sound field synthesis. From 2011 to 2013 he was a Postdoctoral Researcher at Microsoft Research in Redmond, Washington, USA. Currently he is a Senior Researcher at the Quality and Usability Lab at University of Technology Berlin, Germany.

***Rudolf Rabenstein*** studied Electrical Engineering at the University of Erlangen-Nuremberg, Germany, and at the University of Colorado at Boulder, USA. He received the degrees "Diplom-Ingenieur" and "Doktor-Ingenieur" in electrical engineering and the degree "Habilitation" in signal processing, all from the University of Erlangen-Nuremberg, Germany in 1981, 1991, and 1996, respectively. He worked with the Physics Department of the University of Siegen, Germany, and

now as a Professor with the Telecommunications Laboratory at the University of Erlangen-Nuremberg. His research interests are in the fields of multidimensional systems theory and multimedia signal processing. Currently he is a senior area editor of the IEEE Signal Processing Letters and an associate editor of the Springer Journal Multidimensional Systems and Signal Processing.

***Sascha Spors*** received the Dipl.-Ing. degree in electrical engineering and the Dr.-Ing. degree with distinction from the University of Erlangen-Nuremberg, Erlangen, Germany, in 2000 and 2006, respectively. Currently, he heads the virtual acoustics and signal processing group as a full Professor at the Institute of Telecommunications Engineering, Universität Rostock, Rostock, Germany. From 2005 to 2012, he was heading the audio technology group as a Senior Research Scientist at the Telekom Innovation Laboratories, Technische Universität Berlin, Berlin, Germany. From 2001 to 2005, he was a member of the research staff at the Chair of Multimedia Communications and Signal Processing, University of Erlangen-Nürnberg. He holds several patents, and has authored or coauthored several book chapters and more than 150 papers in journals and conference proceedings. His current areas of interest include sound field analysis and reproduction using multichannel techniques, the perception of synthetic sound fields, and efficient multichannel algorithms for adaptive digital filtering.

Prof. Spors is a member of the Audio Engineering Society (AES), the German Acoustical Society (DEGA) and the IEEE. He was awarded the Lothar Cremer prize of the German Acoustical Society in 2011. Prof. Spors is Cochair of the AES Technical Committee on Spatial Audio and an Associate Technical Editor of the Journal of the Audio Engineering Society and the IEEE Signal Processing Letters.

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