Local Sound Field Synthesis by Virtual Secondary Sources

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ABSTRACT

Sound field synthesis techniques like Wave Field Synthesis and Higher-Order Ambisonics aim at the physical synthesis of a desired sound field over an extended listening area. However, for practical setups the accuracy up to which the desired sound field can be synthesized over an extended area is limited. For certain applications it is desirable to limit the spatial extent of the listening area in order to increase the accuracy within this limited region for a given loudspeaker arrangement. Local sound field synthesis aims at a higher accuracy within a local listening area. An approach to local sound field synthesis is presented that is based on the concept of using virtual loudspeakers that are placed more densely around the local listening area than the existing loudspeakers. The approach is illustrated using Wave Field Synthesis as an example.

1. INTRODUCTION

Many of the currently proposed high-quality spatial sound reproduction techniques aim at the physical synthesis of a desired sound field over an extended listening area. Two well known examples are Wave Field Synthesis (WFS) [1] and Higher-Order Ambisonics (HOA) [2]. Both utilize a high number (tens to hundreds) of loudspeakers in order to control the acoustic pressure within the entire area surrounded by the loudspeakers. However, for practical setups the accuracy up to which the desired sound field can be synthesized over an extended area is limited. WFS suffers from spatial sampling artifacts over the entire listening area above a certain frequency (spatial aliasing frequency) [3]. The area of accurate synthesis in HOA, which is approximately a circular/spherical region around the center of the loudspeaker arrangement, gets smaller with increasing frequency. For typical setups the limitations in terms of physical accuracy are quite severe for even moderate frequencies compared to the audio bandwidth. It is known that the spatial sampling artifacts of WFS might result in coloration of the virtual source [4].

For certain applications it is desirable to limit the spatial extent of the listening area in order to increase the accuracy within this limited region for a given number of loudspeakers. We will term these approaches as *local sound field synthesis*. Potential applications include e. g. sound reproduction in a typical living room where the listener(s) reside on the couch, or high-quality reproduction for one listener in the context of subjective listening experiments.

Different techniques for local reproduction have been published in the past. For instance, in [5] a technique for HOA has been developed for the translation of the area with high accuracy synthesis to arbitrary positions, or in [6] a technique is proposed for WFS which is based on adapting the number of active loudspeakers to the frequency. Other related approaches are based on optimizing the synthesized sound field at certain selected points within the listening area [7, 8, 9]. Depending on the technique and the position of these control points this might also result in a local synthesis with higher accuracy.

This paper presents a versatile concept for local sound field synthesis that can be applied to various reproduction techniques aiming at physical synthesis of sound fields. Its application to WFS is illustrated in detail.

2. LOCAL SOUND FIELD SYNTHESIS BY VIRTUAL SECONDARY SOURCES

2.1. Basic Principle

The achievable accuracy of WFS and HOA is in general linked to the spatial density of the loudspeakers (number of loudspeakers per length/area). The higher the spatial density is, hence the more loudspeakers are used for a certain region, the less prominent the resulting spatial

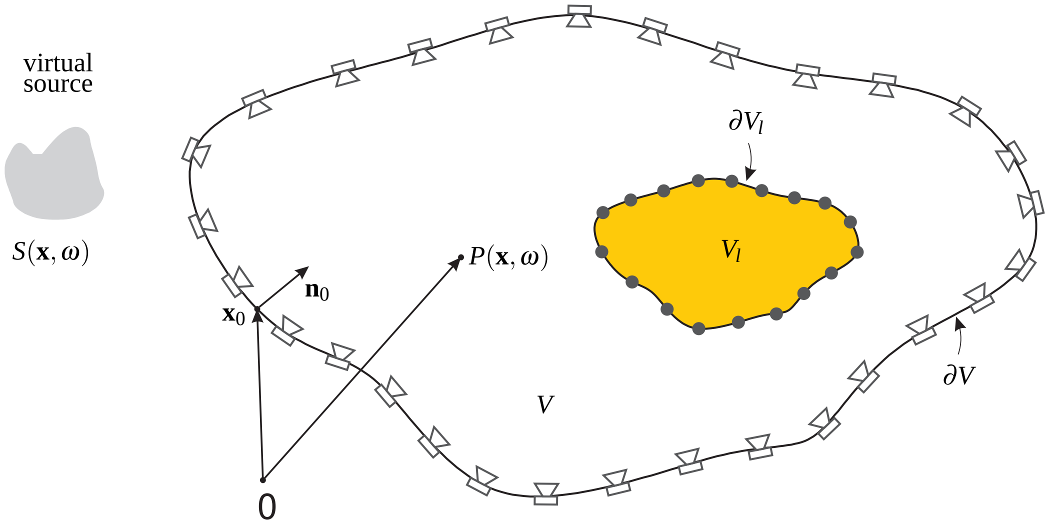


Fig. 1: Principle of local sound reproduction by virtual secondary sources. The loudspeakers are indicated by the loudspeaker symbols placed on the contour ∂V, while the virtual loudspeakers are indicated by the bullets • placed on the contour ∂V_l surrounding the local listening area V_l.

sampling artifacts will be. The same holds also for other techniques. However, the number of loudspeakers is restricted, amongst others, by their size and other practical aspects.

Techniques that aim at physical synthesis of sound fields offer an interesting possibility in this context. They allow for the synthesis of so called *focused sources* [10, 11, 12, 13]. Focused sources aim at creating the impression of a source placed in front of the loudspeakers. This is typically achieved by emitting a sound field which converges towards a focus point and diverges after. The field after the focus point resembles the field of an acoustic point source placed at the focus point. The basic concept of the proposed technique is to utilize focused sources as virtual secondary sources. Creating a set of focused sources within a given loudspeaker arrangement with a spatially more dense distribution than the secondary sources results in a higher accuracy within the area covered by the focused sources (local listening area). These virtual secondary sources have to be driven like real secondary sources placed at their positions. Figure 1 illustrates the principle. The secondary sources are indicated by the loudspeaker symbols on the contour ∂V, while the virtual secondary sources created by the focused sources are indicated by the bullets • on the contour ∂V_l.

2.2. Realization

The basic principle of the proposed approach to local sound field synthesis, as outlined in the previous subsection, is to use focused sources as virtual secondary sources. These should be distributed spatially more densely than the existing secondary sources. The realization of the proposed approach can be split into two steps: (1) the synthesis of focused sources and (2) driving of the focused sources according to the desired sound field $S(\mathbf{x},\omega)$ within V_l. Both steps have to be combined in order to compute the driving signals for the secondary sources. Figure 2 illustrates this by a block diagram which considers the model-based rendering of a virtual source. The driving function of the virtual source and the focused sources are computed according to their parameters (virtual source model, geometry of setup) using WFS, HOA, the spectral division method (SDM) [13] or any other suitable sound field synthesis technique. One could even use different techniques for the acoustic focusing and the synthesis of the virtual source. Focused sources show a limited area in which the wavefronts have the correct traveling direction due to causality. This has to be taken into consideration for the synthesis of the focused sources by using techniques which allow to control the main propagation direction of a focused source. The local listening area can be moved (e. g. to track a listener) by updating the driving signals.

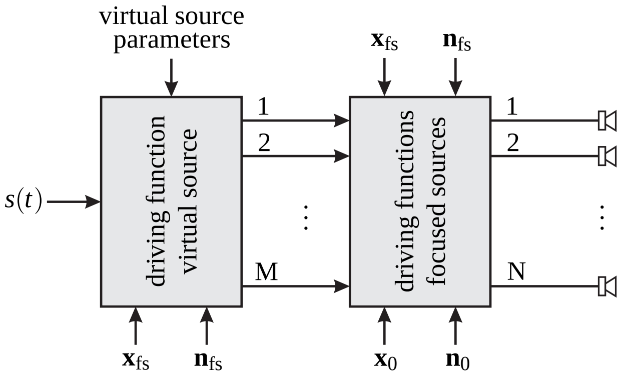


Fig. 2: Block diagram of local sound reproduction using model-based rendering.

The virtual source signal $s(t)$ is convolved with the driving function of the virtual source and the driving functions of the focused sources in order to derive the loudspeaker driving signals. Both driving functions may also be combined to a single set of driving functions.

The remainder of this paper illustrates the proposed concept by using WFS for the calculation of the driving signals for the focused sources and the virtual source. WFS allows for a very efficient implementation of the proposed technique, as will be shown.

3. WAVE FIELD SYNTHESIS

The following sections provide a short review of WFS as needed in the context of this paper. Refer to [10, 14] for a detailed treatment of the theory. Note that traditional WFS aims at synthesis within the entire region V surrounded by the secondary sources on ∂V.

3.1. Basic Principle

The concept of WFS [1] has initially been developed for linear distributions of secondary sources and has later on been extended to arbitrarily shaped convex distributions [15]. WFS is based on the Kirchhoff-Helmholtz integral, which provides the solution of the homogeneous wave equation for a bounded region V with respect to inhomogeneous boundary conditions imposed on ∂V [16]. A loudspeaker distribution surrounding the listener can be regarded as an inhomogeneous boundary condition.

For sound field synthesis it is desired to reproduce the sound field $S(\mathbf{x},\omega)$ of a virtual source inside the listening area V (see Fig. 1). The Kirchhoff-Helmholtz integral states that this can be achieved if a distribution of secondary monopole (single layer potential) and dipole (double layer potential) sources on the boundary ∂V of the listening area V is driven by the directional gradient and the pressure of the sound field of the virtual source $S(\mathbf{x},\omega)$, respectively. In this case, the sound field $P(\mathbf{x},\omega)$ inside the listening area V coincides with the sound field $S(\mathbf{x},\omega)$ of the desired virtual source.

WFS is based on a number of reasonable approximations in order to realize this principle by a distribution of secondary monopole sources. Typically a 2.5-dimensional reproduction scenario is considered where point sources are used as secondary sources for the reproduction in a plane. It is well known from WFS and HOA, that 2.5-dimensional reproduction techniques suffer from artifacts [17, 18]. Most prominent are amplitude and (slight) spectral errors in this context.

The synthesized sound field for 2.5-dimensional reproduction is given by

$$P(\mathbf{x},\omega) = \oint_{\partial V} D(\mathbf{x_0},\omega)\, \frac{1}{4\pi} \frac{e^{-j\frac{\omega}{c}|\mathbf{x}-\mathbf{x}_0|}}{|\mathbf{x}-\mathbf{x}_0|} dL_0 \;, \qquad (1)$$

where $\mathbf{x_0} \in \partial V$ denotes a position on the boundary ∂V, $D(\mathbf{x_0},\omega)$ the driving signal for the secondary sources and dL_0 a suitably chosen line element for integration. For WFS, the driving signal is given as

$$D(\mathbf{x}_0,\omega) = 2a(\mathbf{x}_0)\, c_{2.5\mathrm{D}}(\mathbf{x}_0,\omega)\, \frac{\partial}{\partial \mathbf{n}_0} S(\mathbf{x}_0,\omega) \;, \qquad (2)$$

where $a(\mathbf{x_0})$ denotes a window function, $c_{2.5\mathrm{D}}(\mathbf{x}_0,\omega)$ a correction for 2.5-dimensional reproduction and $\frac{\partial}{\partial \mathbf{n}_0}$ the directional gradient evaluated at position $\mathbf{x}_0$. The directional gradient is defined as the scalar product of the gradient and the normal vector $\mathbf{n}_0$.

The window function $a(\mathbf{x}_0)$ takes care that only those secondary sources are active where the local propagation direction of the virtual source at the position $\mathbf{x_0}$ has a positive component in direction of the normal vector $\mathbf{n}_0$ of the secondary source. It was proposed in [19] to formulate this condition analytically on basis of the acoustic intensity vector. The correction $c_{2.5\mathrm{D}}(\mathbf{x}_0,\omega)$ depends in general on the desired virtual sound field and the geometry of the secondary source distribution.

3.2. Virtual Plane Waves and Point Sources

Analytic models are typically used in WFS as models for the virtual sources. This is referred to as model-based rendering. Plane waves and point sources are used frequently. The driving functions for these two types are briefly reviewed in the following.

The driving function for a plane wave with traveling direction θ_{pw} is given by [14]

$$D_{\text{pw}}(\mathbf{x}_0,\omega) = 2a_{\text{pw}}(\mathbf{x}_0)\sqrt{j\frac{\omega}{c}}\hat{S}_{\text{pw}}(\omega)\times \\ c_{2.5\text{D}}(\mathbf{x}_0)\,\mathbf{n}_{\text{pw}}^T\mathbf{n}_0\; e^{-j\frac{\omega}{c}\mathbf{n}_{\text{pw}}^T\mathbf{x}_0}\;, \quad (3)$$

where $\mathbf{n}_{\text{pw}} = [\cos\theta_{\text{pw}}\ \ \sin\theta_{\text{pw}}]^T$ denotes the normal vector of the plane wave, $\hat{S}_{\text{pw}}(\omega)$ the spectrum of the plane wave and $c_{2.5\text{D}}(\mathbf{x}_0)$ a geometry dependent amplitude correction. The amplitude correction is approximately constant for a linear secondary source distribution and a reference line parallel to the secondary source distribution [10]. The window function $a_{\text{pw}}(\mathbf{x}_0)$ is given by

$$a_{\text{pw}}(\mathbf{x}_0) = \begin{cases} 1 & \text{, if } \langle \mathbf{n}_{\text{pw}}, \mathbf{n}_0\rangle > 0, \\ 0 & \text{, otherwise.}\end{cases} \quad (4)$$

The traditional driving function for a virtual point source placed at position $\mathbf{x}_{\text{ps}} \notin V$ is given by [10, 14]

$$D_{\text{ps}}(\mathbf{x}_0,\omega) = \sqrt{\frac{2}{\pi}}a_{\text{ps}}(\mathbf{x}_0)\sqrt{j\frac{\omega}{c}}\hat{S}_{\text{ps}}(\omega)\times \\ c_{2.5\text{D}}(\mathbf{x}_0)\frac{(\mathbf{x}_0-\mathbf{x}_{\text{ps}})^T\mathbf{n}_0}{|\mathbf{x}_0-\mathbf{x}_{\text{ps}}|}\frac{e^{-j\frac{\omega}{c}|\mathbf{x}_0-\mathbf{x}_{\text{ps}}|}}{\sqrt{|\mathbf{x}_0-\mathbf{x}_{\text{ps}}|}}\;, \quad (5)$$

where the window function can be expressed by

$$a_{\text{ps}}(\mathbf{x}_0) = \begin{cases} 1 & \text{, if } \langle \mathbf{x}_0-\mathbf{x}_{\text{ps}}, \mathbf{n}_0\rangle > 0, \\ 0 & \text{, otherwise.}\end{cases} \quad (6)$$

The driving functions for a virtual plane wave (3) and point source (5) can be realized computationally very efficient in WFS. This can be concluded after performing an inverse Fourier transformation of the driving signals. The resulting time-domain driving signal can be expressed by

$$d(\mathbf{x}_0,t) = s(t)*h(t)*w(\mathbf{x}_0)\,\delta(t-\tau_0(\mathbf{x}_0))\;, \quad (7)$$

where $*$ denotes convolution, $\delta(\cdot)$ the Dirac delta function and $s(t)$ the signal of the virtual source. The impulse response $h(t)$ denotes the inverse Fourier transformation

$$h(t) = \mathscr{F}^{-1}\left\{\sqrt{j\frac{\omega}{c}}\right\}\;. \quad (8)$$

All frequency independent weights of the driving functions for a virtual plane wave or point source are collected in $w(\mathbf{x}_0)$. The shift in the Dirac delta function is given as $\tau_{\text{pw},0}(\mathbf{x}_0) = \mathbf{n}_{\text{pw}}^T\mathbf{x}_0/c$ for a plane wave and $\tau_{\text{ps},0}(\mathbf{x}_0) = |\mathbf{x}_0-\mathbf{x}_{\text{ps}}|/c$ for a point source, respectively. Hence, the driving function for WFS can be computed by

1. filtering the signal of the virtual source $s(t)$ with the filter $h(t)$, and
2. weighting/delaying this pre-filtered signal.

This scheme is efficient with respect to computational complexity since the weighting/delay operations can be implemented by delay lines and consequently only one convolution per virtual source is required.

A spatially continuous distribution of secondary monopole sources was assumed up to now. In practice, this distribution is realized by a limited number of spatially discrete loudspeakers. This constitutes a spatial sampling process. The consequences of this sampling on the synthesized sound field have been investigated in detail by various authors. For instance in [3, 10, 20]. Spatial sampling artifacts become prominent in the synthesized sound field above a geometry and virtual source type specific frequency (also known as spatial aliasing frequency). The artifacts are typically spread over the entire listening area. The spatial aliasing frequency for typical setups is around $1-2$ kHz, which is rather low compared to the audio bandwidth of 20 kHz. Fortunately the human auditory system does not seem to be too sensible to spatial sampling artifacts. Sampling artifacts may be audible as coloration of the virtual source [4].

3.3. Focused Sources

For sound reproduction, the goal is to create the illusion of an acoustic source that is situated in front of the loudspeaker array. Hence, only contributions emerging from the desired focused source should be reproduced at the listener position in order to not confuse the auditory impression by other contributions. Since the secondary sources emit a sound field that travels towards the listener, one can only expect that the desired sound field of a focused source is correct if the focus point is located in between the active secondary sources and the listener. In the context of WFS, this is a well known limitation of focused sources [10].

The driving function for a focused source can be derived by reversing the time in the driving function of a virtual point source (5) together with a sensible selection of the active secondary sources and a pre-delay to ensure causality in practice. The listening area of a focused source in which the wave fronts travel in the correct direction is given, in the optimal case, as a half space which is limited by a line through the focus point. The selection of secondary sources has direct influence on the orientation of this line and consequently the main propagation direction $\mathbf{n}_{\text{fs}}$ of the focused source, as has been shown in [19]. This fact is quite important in the context of this paper. The driving function for a focused source at position $\mathbf{x}_{\text{fs}} \in V$ is given by [10, 19]

$$D_{\text{fs}}(\mathbf{x}_0, \omega) = \sqrt{\frac{2}{\pi}} a_{\text{fs}}(\mathbf{x}_0) \sqrt{-j\frac{\omega}{c}} \hat{S}_{\text{fs}}(\omega) \times c_{2.5\text{D}}(\mathbf{x}_0) \frac{(\mathbf{x}_0 - \mathbf{x}_{\text{fs}})^T \mathbf{n}_0}{|\mathbf{x}_0 - \mathbf{x}_{\text{fs}}|} \frac{e^{j\frac{\omega}{c}|\mathbf{x}_0 - \mathbf{x}_{\text{fs}}|}}{\sqrt{|\mathbf{x}_0 - \mathbf{x}_{\text{fs}}|}}, \quad (9)$$

where the pre-delay required for causality of the driving signal has been discarded for simplicity. The secondary source selection criteria is given by

$$a_{\text{fs}}(\mathbf{x}_0) = \begin{cases} 1 & \text{, if } \langle \mathbf{x}_{\text{fs}} - \mathbf{x}_0, \mathbf{n}_{\text{fs}} \rangle > 0, \\ 0 & \text{, otherwise.} \end{cases} \quad (10)$$

Focused sources have interesting properties which make them a good candidate as virtual secondary sources. The properties of focused sources have been investigated e. g. in [11]. We will briefly summarize the two relevant properties in the following.

(i) Focused sources exhibit an amplitude decay over distance to the focus point which is quite close to that of a point source placed at the focus point. The typical amplitude deviations of WFS for the synthesis of non-focused sources are not very prominent for focused sources.

(ii) The appearance of spatial sampling artifacts for focused sources depends on the distance to the focus point. In comparison to non-focused virtual sources, focused sources allow the accurate synthesis for much higher frequencies in the vicinity of the focus point.

This is illustrated with an example. Figure 3 shows the synthesized sound field for a focused source using WFS. For the simulated situation and non-focused sources sampling artifacts would be present throughout the entire listening area for frequencies above $f_{\text{al}} \approx 1$ kHz [3]. However, it is obvious from Fig. 3 that almost no spatial sampling artifacts are present in a circular region with a radius of about 0.5 m around the focus point for $f_{\text{fs}} = 4$ kHz.

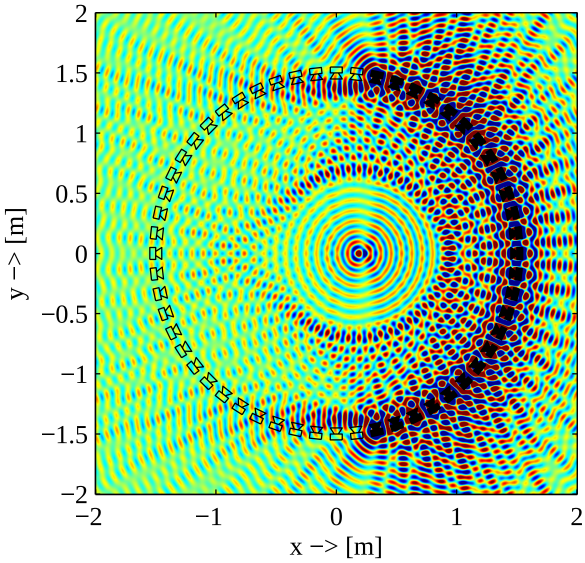


Fig. 3: Reproduction of a focused source with WFS using a circular loudspeaker array ($N = 56$, $R = 1.50$ m, $f_{\text{fs}} = 4$ kHz, $\mathbf{x}_{\text{fs}} = [0.2\ 0]^T$ m, $\mathbf{n}_{\text{fs}} = [-1\ 0]^T$). The active loudspeakers are filled.

Similar properties have been derived for the synthesis of focused sound sources using HOA [12] and the SDM [13].

4. LOCAL SOUND FIELD SYNTHESIS BY WFS

The following section illustrates the application of the proposed concept of local sound field synthesis to WFS.

4.1. Driving Signal

The local synthesis of a plane wave using a spatially discrete secondary source distribution will be considered exemplarily. The geometry shown in Fig. 1 underlies the following considerations. The driving signal for this situation is given by combining the driving signal for a plane wave (3) and the driving signals for the focused sources (9) (see Fig. 2). A total number of M virtual secondary sources and N secondary sources is assumed. The focused sources should have a flat frequency response $\hat{S}_{\text{fs}} = 1$ as they are used as virtual secondary sources. For ease of illustration the pre-delay required for causality has been discarded and it is further assumed that no correction factors $c_{2.5\text{D}}$ are used.

The driving signal of the n-th secondary source is then given by

$$D(\mathbf{x}_{0,n},\omega) = \frac{2}{\pi} a_{\text{fs}}(\mathbf{x}_{0,n}) \frac{\omega}{c} \hat{S}_{\text{pw}}(\omega) \sum_{m=1}^{M} a_{\text{pw}}(\mathbf{x}_{\text{fs},m}) \times \frac{(\mathbf{n}_{\text{pw}}^T \mathbf{n}_{\text{fs},m}) \cdot (\mathbf{x}_{0,n} - \mathbf{x}_{\text{fs},m})^T \mathbf{n}_{0,n}}{\left|\mathbf{x}_{0,n} - \mathbf{x}_{\text{fs},m}\right|^{3/2}} e^{-j\frac{\omega}{c}(\mathbf{n}_{\text{pw}}^T \mathbf{x}_{\text{fs},m} - \left|\mathbf{x}_{0,n} - \mathbf{x}_{\text{fs},m}\right|)} , \quad (11)$$

where $\mathbf{x}_{0,n} \in \partial V$ denotes the position of the n-th secondary source and $\mathbf{x}_{\text{fs},m} \in \partial V_l$ the position of the m-th focused source (virtual secondary source) with main propagation direction $\mathbf{n}_{\text{fs},m}$.

4.2. Efficient Realization

As for the synthesis of plane waves or point sources with WFS one can also derive an efficient realization for local sound synthesis using WFS. Inverse Fourier transformation of (11) yields a structure similar to (7)

$$d(\mathbf{x}_{0,n},t) = s(t) * h(t) * \sum_{m=1}^{M} w_{m,n}\, \delta(t - \tau_{m,n}) \, . \quad (12)$$

However, for the local approach multiple weighted/delayed versions of the pre-filtered virtual source signal are superpositioned. The pre-filter is given by

$$h(t) = \mathscr{F}^{-1}\left\{\frac{\omega}{c}\right\} . \quad (13)$$

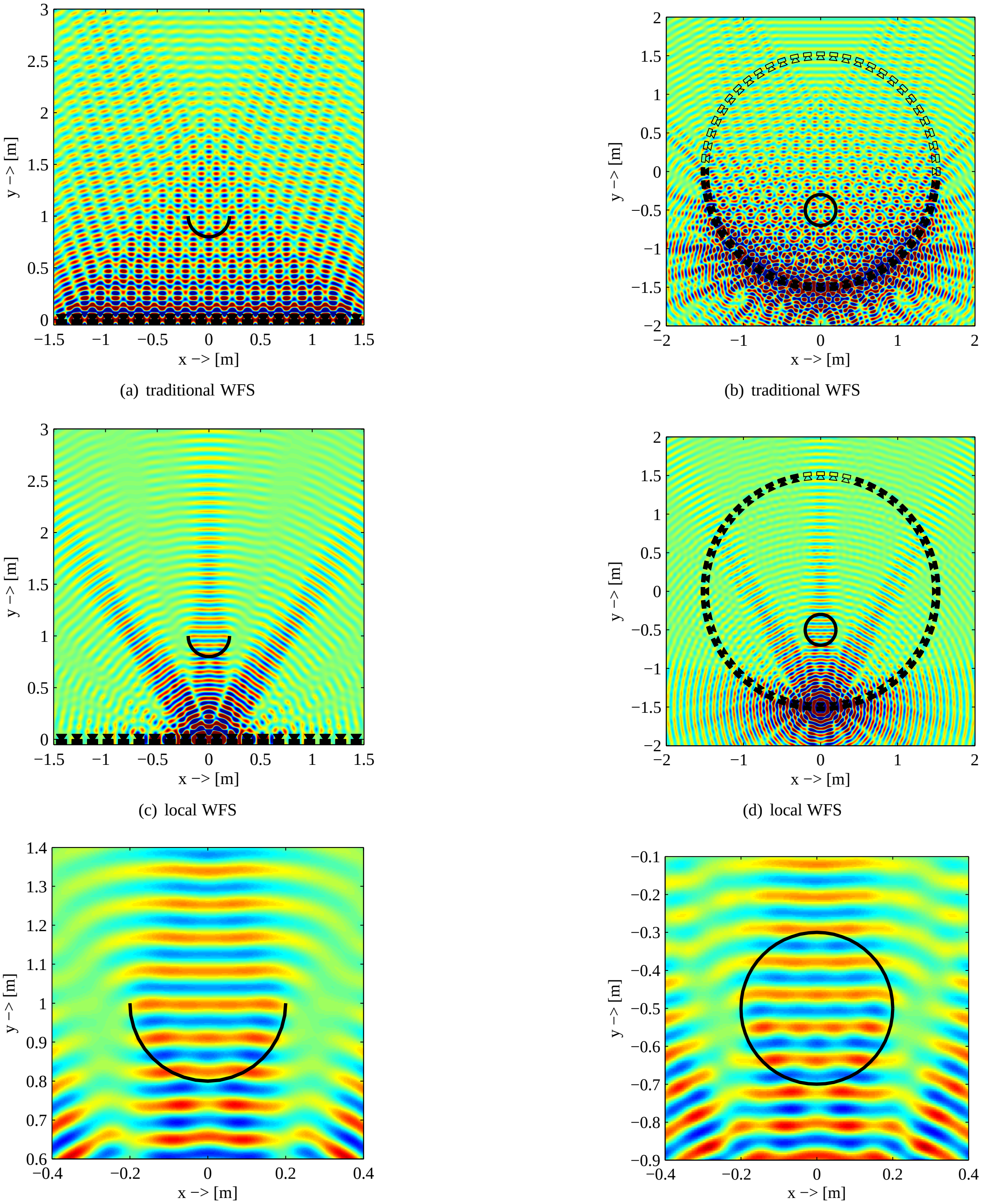
Note, the pre-equalization filter is different from traditional WFS (8). While in traditional WFS the pre-equalization can be realized by a 3dB per Octave high-pass filter, Eq. (13) states than a 6dB per Octave high-pass filter is required for local sound field synthesis.
Above considerations allow again for an efficient implementation requiring only one convolution per virtual source.

4.3. Practical Aspects

A number of practical aspects have to be considered. These will be discussed in the following.
The proposed approach requires a sensible placement of the virtual secondary sources. The zones of correct synthesis of the focused sources in combination with the geometry of secondary source distribution has to be considered. As first approximation, the resulting local listening area is given by the overlapping areas of the individual listening areas of the focused sources. As for traditional WFS, the placement of the virtual focused sources has also impact on the virtual sources that can be synthesized. For instance, if no secondary sources are located behind the listener also no virtual secondary source can be synthesized there and consequently no virtual sources can be auralized from there.
The pre-equalization filter for traditional WFS (8) should only be applied below the spatial aliasing frequency [21]. Preliminary results indicate that the same holds also for the proposed approach. However, the 6dB per Octave high-pass filter should be flattened to a 3 dB per Octave high-pass for frequencies above the spatial aliasing frequency as first approximation. This is due to the fact that spatial aliasing introduces energy and consequently results in a high-pass character. A more detailed analysis of the pre-equalization in the context of the proposed approach will be performed in the future.
The proposed approach allows to freely choose and move the local listening area, e. g. in order to track a moving listener. Figure 2 illustrates the required parameters for the (re-)calculation of the respective driving functions using (12). The efficient solution should allow for such tracking in real-time. However, care has to be taken in order to avoid artifacts due to the time variance of the resulting impulse responses. For listeners moving with reasonably low speed, overlapping of two consecutive impulse responses should be sufficient.
It has been shown in [11] that the sampling artifacts of focused sources may become audible as pre-echos for broadband virtual sources. These pre-echos get more prominent with increasing size of the secondary source distribution [22]. Without countermeasures these pre-echos might also be present in the proposed approach when using large secondary source distributions. However, it is reasonable to assume that the proposed approach will likely be applied to smaller setups where these problems are not present. Further investigations will be performed in the future.

5. RESULTS

In order to illustrate the proposed approach numerical simulations of the synthesized sound fields are presented. The driving function (11) together with a numerical simulation of the secondary monopole sources has been implemented for this purpose. Two different secondary source distributions have been simulated in oder to show the flexibility of the proposed approach: a linear and a circular one. We aim at the accurate reproduction of a monochromatic plane wave with incidence angle $\theta_{\text{pw}} = 90^{\text{o}}$ (upward traveling in the figures) and



(a) traditional WFS

(b) traditional WFS

(c) local WFS

(d) local WFS

(e) local WFS (zoom)

(f) local WFS (zoom)

Fig. 4: Synthesized sound fields for traditional WFS and the proposed local WFS for a linear ($N = 20$, $\Delta x = 0.15$ m, $N_{\mathrm{fs}} = 20$, $R_{\mathrm{fs}} = 0.20$ m) and circular distribution ($N = 56$, $R = 1.50$ m, $N_{\mathrm{fs}} = 56$, $R_{\mathrm{fs}} = 0.20$ m) of secondary sources when reproducing a monochromatic plane wave ($\theta_{\mathrm{pw}} = 90^{\mathrm{o}}$, $f_{\mathrm{pw}} = 4$ kHz). The active loudspeakers are filled.

frequency $f_{\mathrm{pw}} = 4$ kHz within a local listening area with 0.40 m diameter. The size of the local listening area has been chosen such that it covers the size of a human listener including some head movements.

Figure 4(a) shows the sound field synthesized by a linear distribution of secondary sources using traditional WFS. Severe spatial sampling artifacts are visible throughout the entire listening area. The spatial aliasing frequency for the simulated setup is $f_{\mathrm{al}} \approx 1100$ Hz [3]. Figures 4(c) and 4(e) show the resulting sound field when using the proposed approach. The virtual secondary source distribution is indicated by the black semicircle with center $[0\ 1]^T$ m. The desired plane wave is synthesized quite accurately within the listening area. Some artifacts are visible which have several potential reasons. Focused sources exhibit truncation artifacts at the limits of their listening area [19] and WFS constitutes a far-field approximation of the underlying theory [21]. Both reasons are in accordance with Fig. 4(e) since the artifacts are most prominent close to the virtual secondary sources. Although the perceptual relevance of the deviations from the desired plane wave is not clear at the current stage improved driving functions for WFS, as e. g. introduced in [21], could be used to improve the synthesized sound field. Note that the proposed approach increases the spatial aliasing frequency in the simulated situation by roughly an factor of four, at the cost of a smaller listening area. The perceptual impact of this higher spatial aliasing frequency is unclear. However, results from listening experiments conducted with similar aliasing frequencies indicate a clearly perceivable improvement in terms of coloration [4].

Figures 4(b), 4(d) and 4(f) show the synthesized sound fields for a circular distribution of secondary sources using traditional WFS and the proposed approach. A circular distribution of virtual secondary sources has been used (indicated by the black circle). Note, that Fig. 3 shows the field of one individual focused source for the same geometry.

The results for the circular secondary source distribution are quite similar to the linear one. This shows that the proposed approach can be applied straightforwardly to arbitrary geometries. This is a strength of WFS compared to approaches like HOA or the SDM which are restricted to a particular geometry.

6. CONCLUSIONS

This paper presents an approach to local sound field synthesis. The basic concept is to use focused sources as virtual secondary sources. By arranging these spatially more densely around a desired local listening area, the synthesis of a virtual source can be realized with higher accuracy there. The basic principle is applicable to all sound field synthesis techniques that allow for the physical synthesis of focused sources. For instance, WFS, HOA and the SDM can be used straightforwardly. Even combinations of these approaches are possible, e. g. using WFS to calculate the driving signals for focused sources synthesized by HOA.

Approaches to local sound reproduction have a number of applications. They are especially useful if one aims at high-quality synthesis for one or a limited number of listener(s) within a limited listening area by loudspeakers placed at some distance to this area. For instance, in a typical living room situation or car interior, in a recording studio environment or for listening experiments in a lab.

A proof of concept was provided by applying the proposed approach to WFS. The driving function for the local reproduction of a plane wave was derived and numerical simulations of the synthesized sound fields were performed. The derived results show that an accurate reproduction is possible within a limited listening area at frequencies above the spatial aliasing frequency. Furthermore, WFS allows for a very efficient implementation of the proposed technique on arbitrary geometries.

The proposed approach seems to have a number of interesting similarities to other published approaches. The synthesized sound fields of the local HOA approach published in [5] look quite similar to the one shown in Fig. 4(d). In can be further observed that the energy is inherently concentrated to a few secondary sources for the high-frequencies. The approach published in [6] seems to model this in an explicit fashion. These and other similarities to existing approaches will be investigated in the future. Listening experiments are in preparation in order to investigate on the perceptual properties of the proposed approach.

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