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Simulation and Visualization of Room Compensation for Wave Field Synthesis with the Functional Transformation Method

Stefan Petrausch[1], Sascha Spors[1], and Rudolf Rabenstein[1]

[1] *Multimedia Communications and Signal Processing, University Erlangen-Nuremberg, Germany.*

Correspondence should be addressed to Stefan Petrausch (stepe@LNT.de)

ABSTRACT

Active room compensation based on wave field synthesis (WFS) has been recently introduced. So far, the verification of the compensation algorithms is only possible through elaborate acoustical measurements. Therefore, a new simulation method is applied that is based on the functional transformation method (FTM). Compared with other simulation techniques, the FTM provides several advantages that facilitate the correct simulation of the complete wave field particularly in the interesting frequency ranges for WFS. The entire procedure, starting from the virtual "measurements" of the acoustical properties of the simulated room, via the correct excitation for the simulated wave field, towards the resulting animations is presented in this paper.

1. INTRODUCTION

Wave field synthesis (WFS) has proven to successfully recreate virtual sound fields for a large number of listeners . A number of different installations in various environments has been created by the young and growing WFS community. However, as the application range of WFS expands, new and challenging problems become visible. One of them is the compensation of unwanted room acoustics effects in the reproduction room. These effects were not present in early WFS installations, because countermeasures against reverberation in the reproduction rooms were taken. Typically passive damping has been applied, which is quite easy in a laboratory environment, but may be not possible or unfeasible in office rooms, in cars, or for mobile WFS installations.

For these possible applications of WFS active room compensation methods have been developed. The high number of loudspeakers, which are required anyway for sound reproduction, are also used for the cancellation of unwanted room reflections. The compensation algorithm is driven by measurements of the room acoustics with a microphone array. Clever use of the WFS theory allows to control the sound within the listening area by microphone array measurements along a closed curve.

While this method is attractive from a theoretical and algorithmic view point, the question remains how to optimize room compensation in different scenarios and how to verify its performance. Again elaborate acoustical measurements would be required to characterize the wave field within the auralized area. And again a high number of microphone recordings were necessary, not any more for the operation of the room compensation algorithm, but for its optimization and verification. As an alternative, accurate simulations of room acoustics are proposed to facilitate the development of room compensation algorithms. This simulation methods and examples of its application are presented in this contribution.

Section 2 gives a short introduction into the simulation algorithm. WFS is briefly reviewed in Section 3. The method for active room compensation is described in Section 4. Finally, Section 5 shows how to apply the simulation and visualization of room acoustics to investigate the performance of active room compensation. The simulation results demonstrate the need of adaptive compensation algorithms.

2. SIMULATION OF WAVE PROPAGATION USING THE FTM

In this section the simulation algorithm based on the functional transformation method (FTM) is described. In doing so, the general procedure, the specific application to the wave-equation, and several benefits of the FTM compared to other simulation techniques are given in the sequel.

2.1. General Procedure

The FTM is used so far as an efficient method for digital sound synthesis via physical modeling (see [1] for instance). In doing so, physical models in terms of partial differential equations (PDEs) are defined together with the appropriate initial (IC) and boundary conditions (BC). The following procedure, as depicted in figure 1, is based on several integral transformations that orthogonalize any given linear PDE and lead to a multidimensional (MD) transfer function model (TFM).

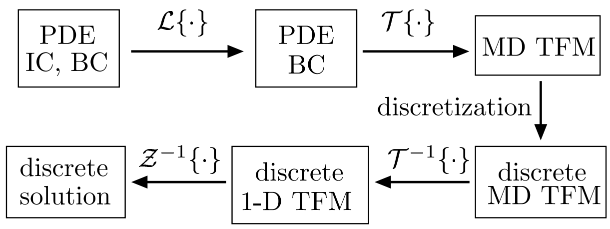

Fig. 1: *General procedure of the FTM solving initial-boundary-value problems defined in form of PDEs, initial conditions (IC), and boundary conditions (BC). Further abbreviations are explained in the remainder of this section.*

The first integral transformation is the well known Laplace transformation (indicated by $\mathcal{L}\{\cdot\}$). It replaces the temporal derivatives in the PDE by the temporal frequency variable s and leads to a boundary value problem with boundary conditions (BC) in the Laplace domain. The following Sturm-Liouville transformation (indicated by $\mathcal{T}\{\cdot\}$) performs the same for the spatial derivatives. However, there is no general form for this transformation, it has to be designed problem-specific with the help of an eigenvalue problem (see [1] for details). Nevertheless, it yields an algebraic equation without derivatives, which can be reordered to achieve the desired TFM. This TFM can be discretized and transformed back to the space domain (inverse Sturm-Liouville transformation) and to the time domain (inverse $\mathcal{Z}$-transformation). The result is a discrete solution in form of a weighted sum of the systems' eigenfunctions as described in the next section.

2.2. Application to the Wave-Equation

As a general approach, it is a straight forward procedure to apply the FTM for the simulation of wave propagation. The complete approach, together with an efficient method for the evaluation of the complete wave-field is described in detail in [2]. Here a brief overview is given.

2.2.1. The Model

The first step as it can be seen in figure 1 is the definition of a suitable model. As well known from acoustics (see e.g. [3]) wave propagation is described by the so called wave-equation, which relates pressure $p = p(\vec{x}, t)$ and particle velocity $\vec{v} = \vec{v}(\vec{x}, t)$ and can be given in the following vector form

$$\Bigg[\underbrace{\begin{pmatrix} 1 & 0 \\ 0 & 1 \end{pmatrix}}_{=\mathbf{I}_0}\nabla + \underbrace{\begin{pmatrix} 0 & -1 \\ -\frac{1}{c^2} & 0 \end{pmatrix}}_{=\mathbf{C}}\frac{\partial}{\partial t}\Bigg]\underbrace{\begin{pmatrix} p \\ -\varrho_0\vec{v} \end{pmatrix}}_{=\mathbf{y}(\vec{x},t)} = \underbrace{\begin{pmatrix} 0 \\ -f_\mathrm{e}(\vec{x}, t) \end{pmatrix}}_{=\mathbf{f}_\mathrm{e}(\vec{x},t)}, \quad (1)$$

whereas c denotes the speed of sound and $f_\mathrm{e}(\vec{x}, t)$ denotes an arbitrary excitation function.

Both initial and boundary conditions are assumed to be homogeneous. In detail the ICs are defined by $\mathbf{y}(\vec{x}, 0) = \mathbf{0}\ \forall \vec{x} \in V$ and the BCs are defined by

$$\begin{pmatrix} 0 & 0 \\ 0 & \vec{n}_\mathrm{b} \end{pmatrix}\mathbf{y}(\vec{x}_\mathrm{b}, t) = \begin{pmatrix} 0 \\ 0 \end{pmatrix} \quad \forall \vec{x}_\mathrm{b} \in \partial V\ . \quad (2)$$

The missing part is the specification of the definition region V (bounded by ∂V). As the proposed WFS-system controls only a plane in the three-dimensional space (see section 3), it is sufficient to simulate two-dimensional wave propagation. The room geometry is kept as simple as possible, too. The FTM can handle any geometry, however in order to apply the high efficient algorithms from [2] a simple geometry with separable boundary conditions is required. In result a rectangular two-dimensional region V with length l_1 in x_1-dimension and l_2 in x_2-dimension is chosen as a room model.

2.2.2. Solution in the Frequency Domain

Application of the Laplace-transformation on (1) removes the temporal derivative and yields the following PDE in space $\vec{x}$ and temporal frequency s

$$[\mathbf{I}_0\nabla + \mathbf{C}s]\,\mathbf{Y}(\vec{x}, s) = \mathbf{F}_\mathrm{e}(\vec{x}, s)\ . \quad (3)$$

The Sturm-Liouville transformation (SLT) performs similar for the spatial derivative and yields an algebraic equation, which can be reordered to achieve the solution of the problem in both, temporal and spatial frequency domain

$$\bar{Y}(\beta_\mu, s) = \frac{1}{s + \beta_\mu}\bar{F}_\mathrm{e}(\beta_\mu, s)\ . \quad (4)$$

The structure of this transfer function model (4) is always the same, differences between models become manifest in different transformation kernels $\mathbf{K}(\vec{x}, \beta_\mu)$ for the SLT and in different values of the spatial frequency variable β_μ (see [2] for details). This spatial frequency variable β_μ is restricted to a discrete set of values in order to fulfill the homogenous boundary conditions of the eigenvalue-problem, which determines both transformation kernels $\mathbf{K}(\vec{x}, \beta_\mu)$ and eigenvalues β_μ.

Discretization and inverse $\mathcal{Z}$-transformation of (4) yields a discrete convolution, which can be realized by simple first order recursive systems. Furthermore, β_μ is restricted to values below the Nyquist frequency, what results in a finite set of N_a eigenvalues β_μ. The inverse Sturm-Liouville transformation is then performed by a summation over these N_a harmonics, weighted by the transformation kernel $\mathbf{K}(\vec{x}, \mu)$ and a weighting constant N_μ (see [2] for details). The complete discrete system is depicted in figure 2. Superscript $^\mathrm{d}$ denotes time-discrete values, T is the temporal sampling interval, and k is the discrete time step. The recursive systems are excited by the time-discrete version of the Sturm-Liouville transformed excitation function $\bar{f}_\mathrm{e}(\mu, k)$.

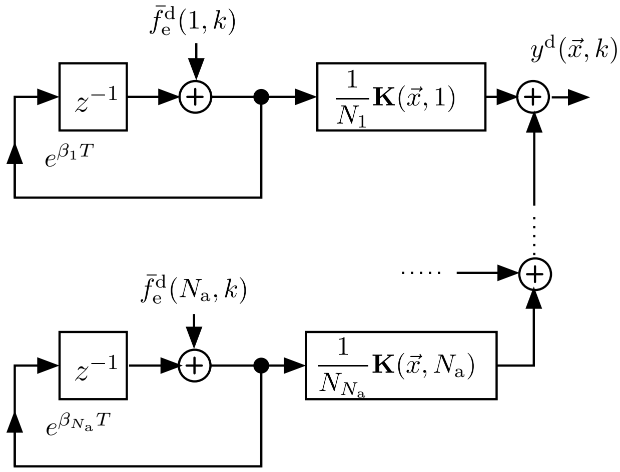

Fig. 2: *Basic structure of the FTM simulations derived from vector PDEs with several complex first-order resonators in parallel.*

2.3. Properties and Additional Features

The discrete structure from figure 2 with several additional features (which cannot be described here in detail) is implemented in the program "*Wave2D*". A full reference is provided in [2], here only a short list its features is given:

1. for rectangular models, the inverse SLT is performed by a discrete Cosine transformation type IV (see [2] for details), what yields a highly efficient evaluation of the complete wave-field,
2. arbitrary boundary conditions (with some restrictions in the free-field scenario) are implemented via boundary excitations (see [4] for details),
3. more elaborate differential equations are possible, in particular frequency independent and frequency dependent damping are implemented,
4. point-sources as well as spatially distributed sources are implemented,
5. extended graphical user interface, with load and save functionality and the possibility to export wave-files, videos, and Matlab-scripts.

Some reasons for the preferability of the proposed approach for wave-field simulation compared to other methods (e.g. mirror image method or finite difference schemes) result directly from the the application of the FTM:

1. arbitrary source and receiver positions are possible, there is no spatial grid,
2. a direct link from the physical parameters to the discrete system is given, e.g. temperature changes can be easily simulated,
3. the simulation is free of numerical dispersion, there is no favored direction in space as in e.g. finite difference schemes or wave-digital meshes,
4. especially low frequencies are simulated accurately compared with e.g. the mirror image method [5].

The FTM simulations are in good match with theoretical results (found in [6] for instance) and serve as a reasonable alternative for the simulation of wave fields synthesis, which is described in the next section.

3. WAVE FIELD SYNTHESIS

Wave field synthesis (WFS) aims at reproducing the sound of complex acoustic scenes as natural as possible. In contrast to other multi-channel approaches, it is based on fundamental acoustic principles. WFS allows a physically correct reproduction of wave fields. In doing so, WFS overcomes the limitations of the traditional multichannel systems like the "sweet-spot". This section gives a brief overview of the theory.

The theory of WFS has been initially developed at the Technical University of Delft [7] and has been developed further by a vital WFS research community over the past two decades. The intuitive foundation of WFS is given by Huygens' principle [3]. Huygens stated that any point of a propagating wave front at any time-instant conforms to the envelope of spherical waves emanating from every point on the wavefront at the prior instant. This principle can be utilized to synthesize acoustic wavefronts of arbitrary shape. Besides this more illustrative description, WFS has also a physical basis. The mathematical foundation of Huygens' principle is given by the Kirchhoff-Helmholtz integral, which can be derived from the wave equation and the Green's integral theorem [3]. The Kirchhoff-Helmholtz integral states that at any listening point within a source-free volume V the sound pressure can be calculated if both the sound pressure and its gradient are known on the surface S enclosing the volume. Straightforward application of this principle for sound reproduction would require to use a continuous distribution of secondary monopole and dipole sources enclosing the listening space V. However, two essential simplifications are necessary to arrive at a realizable reproduction system: Degeneration of the volume V to a 2D plane and spatial discretization. If these steps are performed in a sensible way, a WFS system can be realized by mounting closed loudspeakers in a linear fashion (linear loudspeaker arrays) surrounding the listening area leveled with the listeners ears. Figure 3 shows an example for a circular WFS system.

The simplifications of the Kirchhoff-Helmholtz dis-

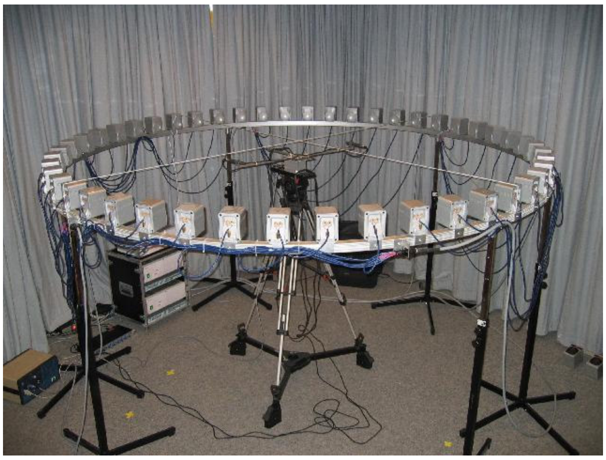

Fig. 3: Circular WFS system with 48 loudspeakers. The listening area has a diameter of $D = 3.00$ m

cussed above may produce various artifacts in the reproduced wave field [8, 9]. However, only artifacts caused by the spatial sampling will be reviewed briefly in the following. The discretization of the underlying physical and mathematical relations may result in spatial aliasing. For reproduction purposes this effect does not play a dominant role since the human auditory system doesn't seem to be too sensible for spatial aliasing. A loudspeaker distance of $\Delta x = 10 \ldots 30$ cm has proven to be suitable in practice for reproduction only purposes.
However, spatial aliasing limits the frequency up to which a proper control is gained over the wave field. Since active room compensation is built upon destructive interference, its application is limited by spatial aliasing. Thus, active listening room compensation can only be applied below the spatial aliasing frequency of the WFS system used. The spatial aliasing frequency for linear loudspeaker arrays is given in [10, 11]. For arbitrary shaped loudspeaker arrays no explicit sampling theorem can be found in the literature. In the following it will be assumed that the spatial aliasing condition is reasonable fulfilled.

4. ACTIVE LISTENING ROOM COMPENSATION FOR WFS

The WFS system used for reproduction of a recorded or virtual acoustic scene is located typically in a room. This room will be called the listening room. The theory of WFS assumes an anechoic listening room, a condition which is rarely met by typical rooms. However, it is often unfeasible due to design and cost considerations to treat the listening room by passive damping in order to fulfill this requirement. The Kirchhoff-Helmholtz integral and its simplifications state that the wave field inside a finite space V is fully determined by its pressure and/or pressure gradient on the boundary ∂V surrounding the space V. Thus, if the listening room reflections are canceled at the boundary of the listening area, the reproduced wave field within the active compensated listening area will be free of undesired reflections. A first approach would be to record the sound field in the listening area with microphones and to infer compensation signals from the microphone recordings. Besides the high number of loudspeakers used for WFS, an adequate analysis of the listening room influence also requires a quite high number of reference microphones. Since the acoustic properties of the listening room may change over time, the compensation signals have to be computed using adaptive algorithms. As a result of the high number of analysis and synthesis signals, an adaptation of the compensation signals for WFS with traditional algorithms will become a computationally very demanding task, resp. is not possible at all.

Recently a novel approach to active listening room compensation using WFS, wave field analysis (WFA) and wave-domain adaptive filtering (WDAF) has been proposed [12, 13]. Instead of using the loudspeaker and microphone signals directly to calculate room compensation filters a spatial transform of the concerned wave fields into a different representation is performed. This transformation decouples the room influence and thus lowers the computational complexity of the adaption of the room compensation filters significantly. The next section will introduce suitable representations of acoustic wave fields for this purpose. The following section will shortly introduce the improved room compensation algorithm.

4.1. Wave Field Representations and Analysis

The following discussion will be limited to wave field representations and analysis for two-dimensional wave fields as this is suitable for WFS. A wave field can be decomposed into the eigensolutions of the wave equation. These eigensolutions are dependent on the particular coordinate system used. Common choices for coordinate systems in two-dimensional space are the Cartesian and polar coordinate system.

The representations of a wave field that are connected to Cartesian and polar coordinates decompose a wave field into plane waves and circular harmonics, respectively [14]. The decomposition of an acoustic field into plane waves is given as follows [15]

$$P(\alpha, r, \omega) = \frac{1}{(2\pi)^2} \left|\frac{\omega}{c}\right| \int_0^{2\pi} \bar{P}(\theta, \omega)\, e^{-j\frac{\omega}{c} r \cos(\theta-\alpha)}\, d\theta \tag{5}$$

where α, r denote the polar representation of the Cartesian coordinates and $\bar{P}(\theta, \omega)$ denote the plane wave expansion coefficients. The latter can be interpreted as the spectrum of a plane wave with incidence angle θ.

The wave field $P(\alpha, r, \omega)$ can be alternatively decomposed into circular harmonics. This decomposition is given as follows [14]

$$\begin{aligned} P(\alpha, r, \omega) = \sum_{\nu=-\infty}^{\infty} \breve{P}^{(1)}(\nu, \omega)\, H_\nu^{(1)}(\frac{\omega}{c} r)\, e^{j\nu\alpha} + \\ \sum_{\nu=-\infty}^{\infty} \breve{P}^{(2)}(\nu, \omega)\, H_\nu^{(2)}(\frac{\omega}{c} r)\, e^{j\nu\alpha} \end{aligned} \tag{6}$$

where $\breve{P}^{(1),(2)}$ denote the expansion coefficients in terms of circular harmonics. It can be shown that $\breve{P}^{(1)}$ belongs to an incoming and $\breve{P}^{(2)}$ to an outgoing wave [14]. The decomposition into incoming and outgoing waves can be used to distinguish between sources inside and outside the measured area. While sources outside result in an incoming part which is equal to the outgoing part, sources inside the array are only present in the outgoing part. The expansion coefficients in terms of circular harmonics and plane waves are related to each other by a Fourier series [15].

Up to now, access to the entire two-dimensional wave field $P(\alpha, r, \omega)$ is required to compute the expansion coefficients. However, the Kirchhoff-Helmholtz integral states that only measurements at the boundary of the region of interest have to be taken to characterize the wave field within that region. In a practical implementation these measurements can only be taken at discrete points on the boundary. This may result in spatial aliasing if the sampling is not performed properly. Using these principles an efficient algorithm for the decomposition into circular harmonics for a circular microphone array has been developed by [15]. This algorithm will be used in the following since circular microphone arrays have many desirable properties.

4.2. Wave Domain Adaptive Filtering based Active Room Compensation Algorithm

In the following we will shortly review the room compensation system for WFS that was proposed in [12, 13]. It is based upon the concept of wave domain adaptive filtering (WDAF). The basic idea of this concept is to approximately orthogonalize the multiple-input/multiple-output (MIMO) listening room response through spatio-temporal transformations. The optimal choice of the transformed signal representation will depend on the geometry of the problem. It has been shown that circular harmonics provide a suitable basis for circular microphone arrays and typical listening rooms [12, 13]. The following description will specialize the generic concept of WDAF to a circular microphone array used for the analysis of the reproduced wave field.

Figure 4 shows a block diagram of the proposed room compensation system. The basic idea is to approximately orthogonalize the listening room response $\mathbf{R}$ through the transformations $\mathcal{C}^{-1}$ and $\mathcal{C}$. As a consequence, the matrix of compensation filters $\breve{\mathbf{C}}$ can be decomposed into a set of compensation filters, each acting on only one spatial signal component. The adaption of these compensation filters is then performed independently for each spatially transformed component. The number of compensation filters that have to be adapted is lowered significantly compared to the traditional approaches. Thus the complexity of the filter adaption is highly reduced. Please note, that nearly all one channel filtered-X adaptation algorithms can be utilized to adapt the compensation filters in the transformed domain.

In the following we will specify the transformations $\mathcal{T}$, $\mathcal{C}^{-1}$ and $\mathcal{C}$. The transformation $\mathcal{T}$ transforms the virtual source q to be auralized into its circular harmonics representation. Suitable spatial source models, like point source or plane wave propagation, allow a closed-form solution of this transformation.

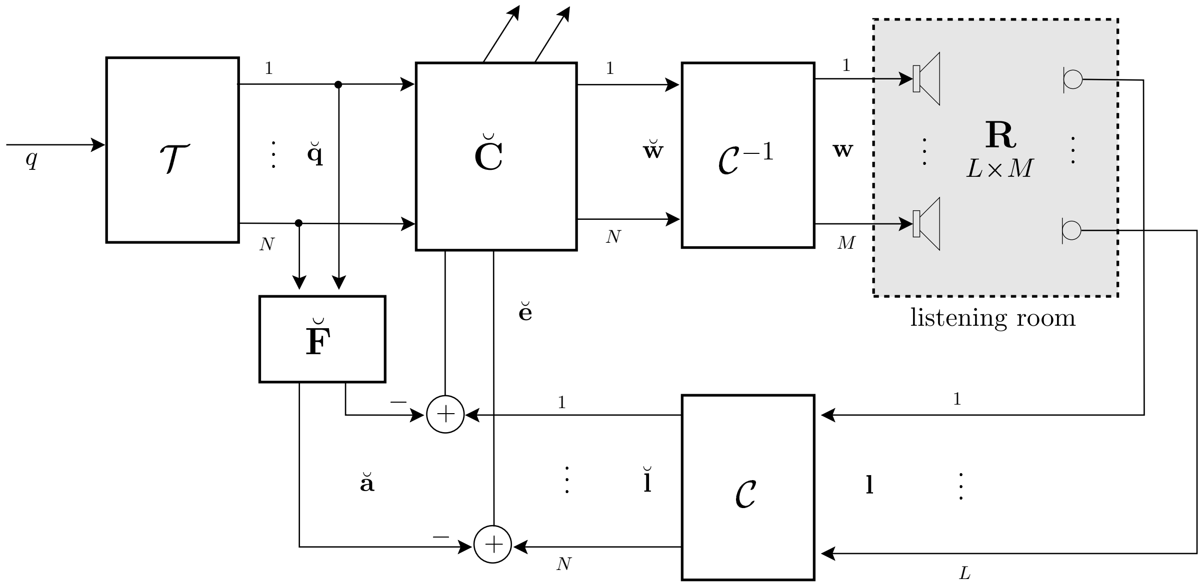

Fig. 4: Block diagram of a room compensation system based on wave domain adaptive filtering.

However, it is also possible to prescribe complex wave fields as desired wave field. The transformed signals are pre-filtered by the room compensation filters $\breve{\mathbf{C}}$. Transformation $\mathcal{C}^{-1}$ computes suitable loudspeaker signals from the pre-filtered transformed signal components. The circular harmonics decomposition, as given by equation (6), can be used for this purpose. Block $\mathcal{C}$ transforms the microphone array signals $\mathbf{l}$ into their circular harmonics representation $\breve{\mathbf{l}}$. A suitable transformation for circular microphone arrays was introduced in [15]. By using only the incoming part $\breve{P}^{(1)}$ of the recorded wave field sources inside the array are omitted for room compensation purposes.
The entire algorithm was implemented as shown in Figure 4. The compensation filters where calculated adaptively using the frequency domain adaptive filtering algorithm described in [16]. The filter adaption is thus performed in the spatial and temporal frequency domain.

5. SIMULATIONS

For the simulation of the room compensation system, virtual loudspeakers (sources) and microphones (receivers) are placed within the simulated virtual room. Then impulse responses from each source to each receiver are recorded from simulations. These impulse responses are fed into the room compensation algorithm which calculates appropriate equalization filters. These filters are used to calculate the loudspeaker driving signals for the simulation. The wave field resulting from room compensation is visualized by the simulation program and can furthermore be evaluated at certain listening points. This procedure yields e.g. animations of the wave propagation within the listening room and can be used to evaluate different aspects of the algorithms.

5.1. Simulation Setup

The simulated setup is based on the measurement setup in [12]. The setup presented there consisted of a circular loudspeaker array with a radius of $R_{\mathrm{LS}} = 1.50$ m and 48 loudspeakers. A microphone array with a radius of $R_{\mathrm{Mic}} = 0.75$ m was placed in the center of the loudspeaker array in order to analyze the auralized wave field. The circular array was realized by sequential measurement of 48 discrete positions on the circle. Figure 3 shows the loudspeaker and microphone array setup used in our demonstration room with size $5.8 \times 5.9 \times 3.1$ m (w × l × h). The wave propagation in this room is simulated using the algorithm presented in Section 2. The real room had to be simplified to a rectangular plane (2D

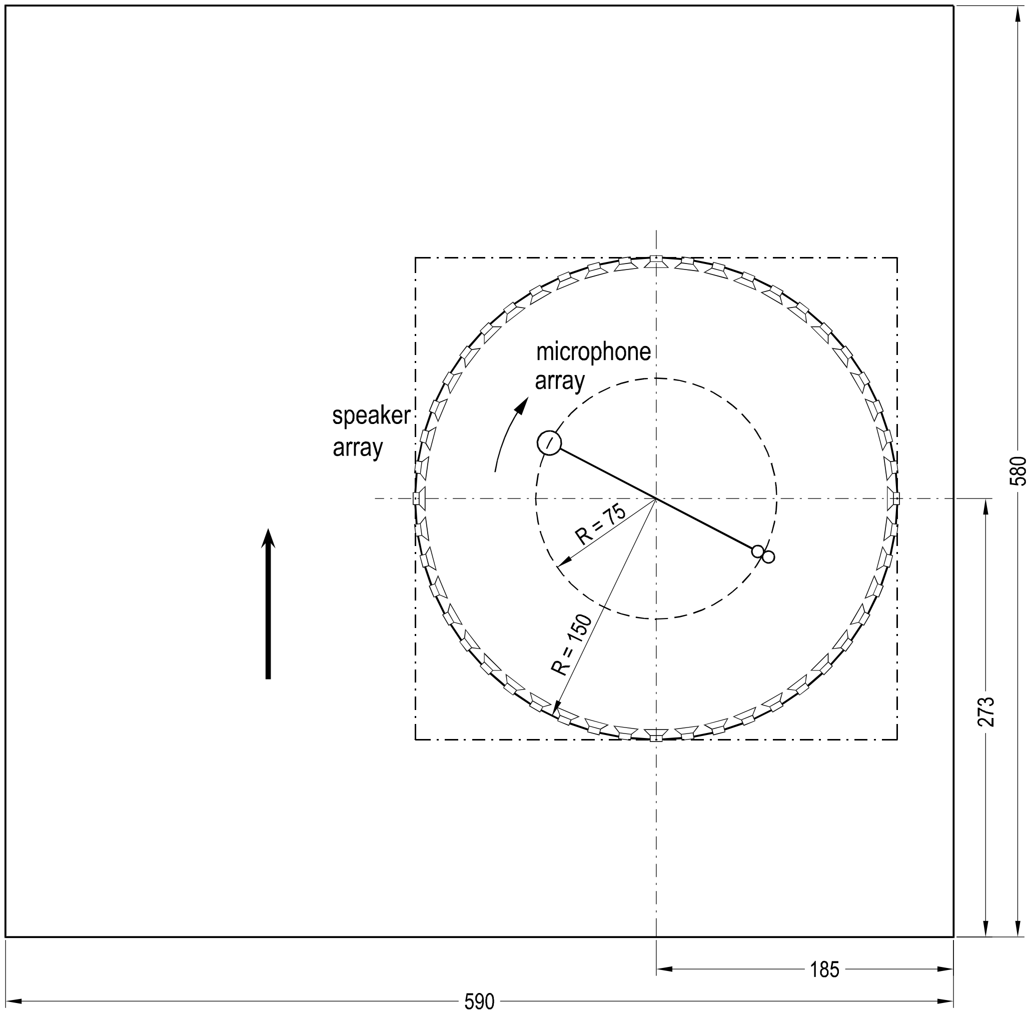

Fig. 5: Setup used for the simulations (distances in cm). The arrow indicates the traveling direction of the plane wave simulated in Section 5.2.

simulation) for this purpose. Figure 5 illustrates the geometry of the simulated setup. The position and size of the loudspeaker and microphone array in the room matches the measurement setup in [12]. This allows to compare the results obtained by measurements in the real room with the ones obtained by the proposed simulation method. A plane wave reflection factor $R_{\mathrm{pw}} = 0.8$ was chosen for all walls in the simulation, what is a good match to the real room.
The impulse responses from each loudspeaker to each microphone position were simulated, resulting in the room transfer matrix $\mathbf{R}$. The measurements were downsampled according to the spatial aliasing frequency of the loudspeaker array. An upper frequency of $f_{\mathrm{al}} = 650$ Hz was chosen for room compensation. The simulated impulse responses were then fed into the room compensation algorithm. A total of 48 circular harmonics for the wave domain representation were used. The resulting room compensation filters were used to create the input signals for the simulation in order to visualize the results.

5.2. Results

The following section will show some of exemplary results derived by the proposed simulation and room compensation methods. The following figures are screenshots of several videos generated with the program "*Wave2D*" (see section 2). All videos can be downloaded under [17].
We used a band-limited plane wave traveling upwards (as indicated in Figure 5) as desired wave field for the following results. The derived results are generic, since arbitrary wave fields can be decomposed into their plane wave contributions using (5). Four scenarios have been simulated:

1. WFS reproduction without room compensation
2. WFS reproduction with room compensation
3. variation in the speed of sound without adapting the compensation filters
4. change in the acoustic environment without adapting the compensation filters

The first two scenarios illustrate the performance of the proposed room compensation system, the third and fourth scenario illustrate the need for adaptive room compensation algorithms by applying the filters derived from the second scenario to these slightly changed environments.
The left row of Figure 6 shows the resulting wave field for the simulated reverberant room for different time-instants without applying room compensation. The upper wave field for $t = 0$ ms shows the desired plane wave in the center of the array, the middle wave field for $t = 6.8$ ms shortly before the upper wall and the lower one for $t = 18.1$ ms when the reflected plane wave enters the listening area again traveling in the opposite direction. The disturbances caused be the reflections at the walls of the listening room can be seen clearly. The right row of Figure 6 shows the results when applying the WDAF based room compensation algorithm. The wave fields after convergence of the compensation filters are shown. It can be seen clearly that the room compensation algorithm is capable of actively compensating the listening room reflections within the listening area. Only the desired band-limited plane wave is present inside the circular array.

In order to illustrate the need for adaptive room compensation algorithms, the acoustic environment has been slightly changed for the third and fourth simulated scenario. The third scenario simulated a variation in the speed of sound resulting from a 5 K increase of the temperature in the room. The fourth scenario simulated the influence of a small object (e.g. a person) entering the room. The adaptation of the room compensation filters has been halted simulating the application of a non-adaptive room compensation scheme.
Figure 7 shows the results when increasing the temperature in the room. The left row show the original results from the right row of Figure 6 for reference. The right row illustrates the impact of the temperature change. It can be seen clearly at $t = 18.1$ ms that the non-adaptive room compensation filters are not capable of suppressing the reflection of the desired plane wave caused by the upper wall of the listening room. As the speed of sound c is a little bit higher, the compensation of the reflections is to late in this changed scenario.
Figure 8 show the results when a small object, e.g. a person, is entering the room. Again the results when applying room compensation to the original scene are shown for reference in the left row. The right row shows the results for the slightly changed

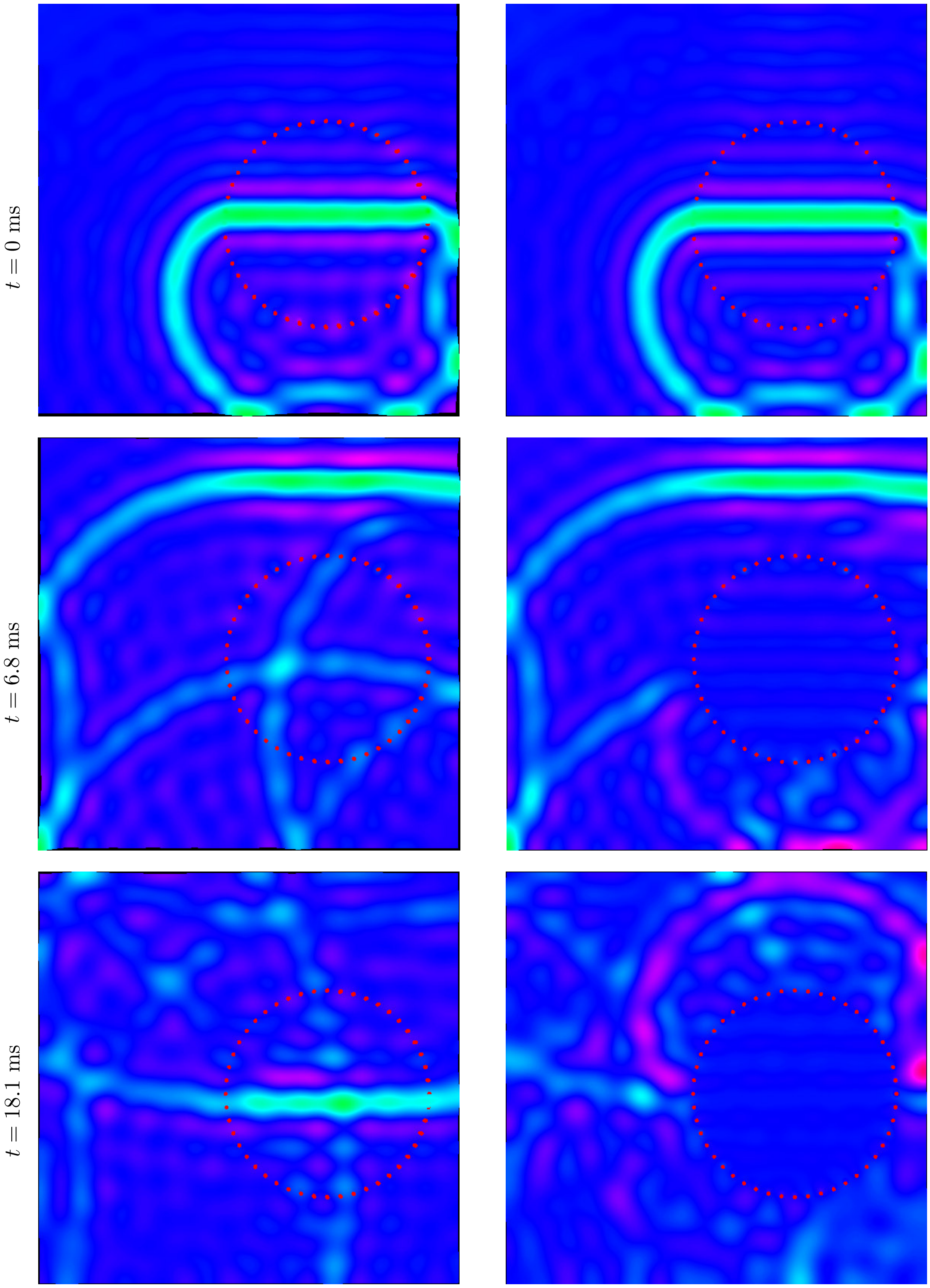

Fig. 6: The left row shows the wave field without room compensation for different time-instants. The right row show the results when applying room compensation for converged room compensation filters.

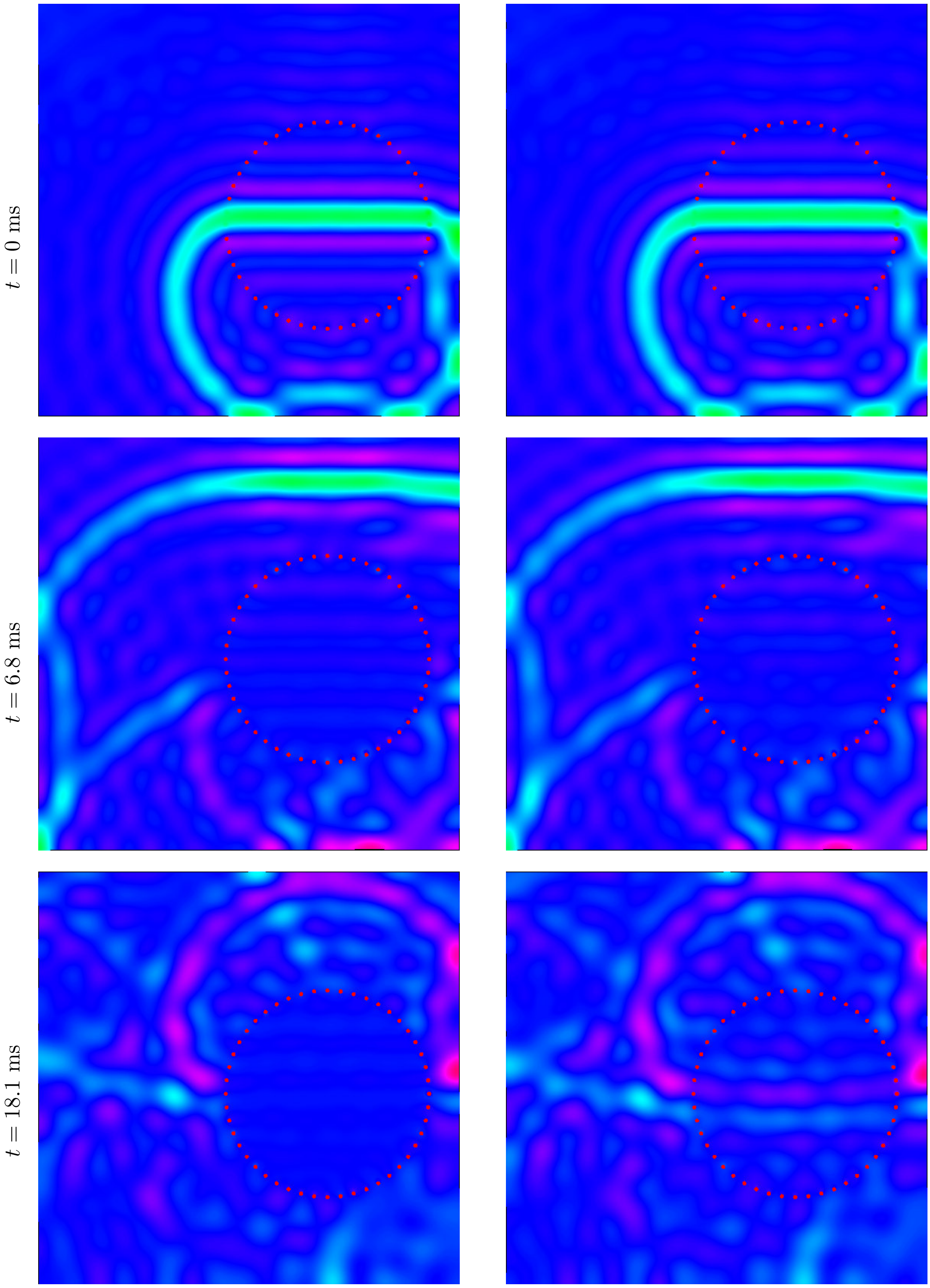

Fig. 7: Results for converged room compensation filters. The left row shows the wave field for the original temperature, the right row for $T = T_0 + 5$ K.

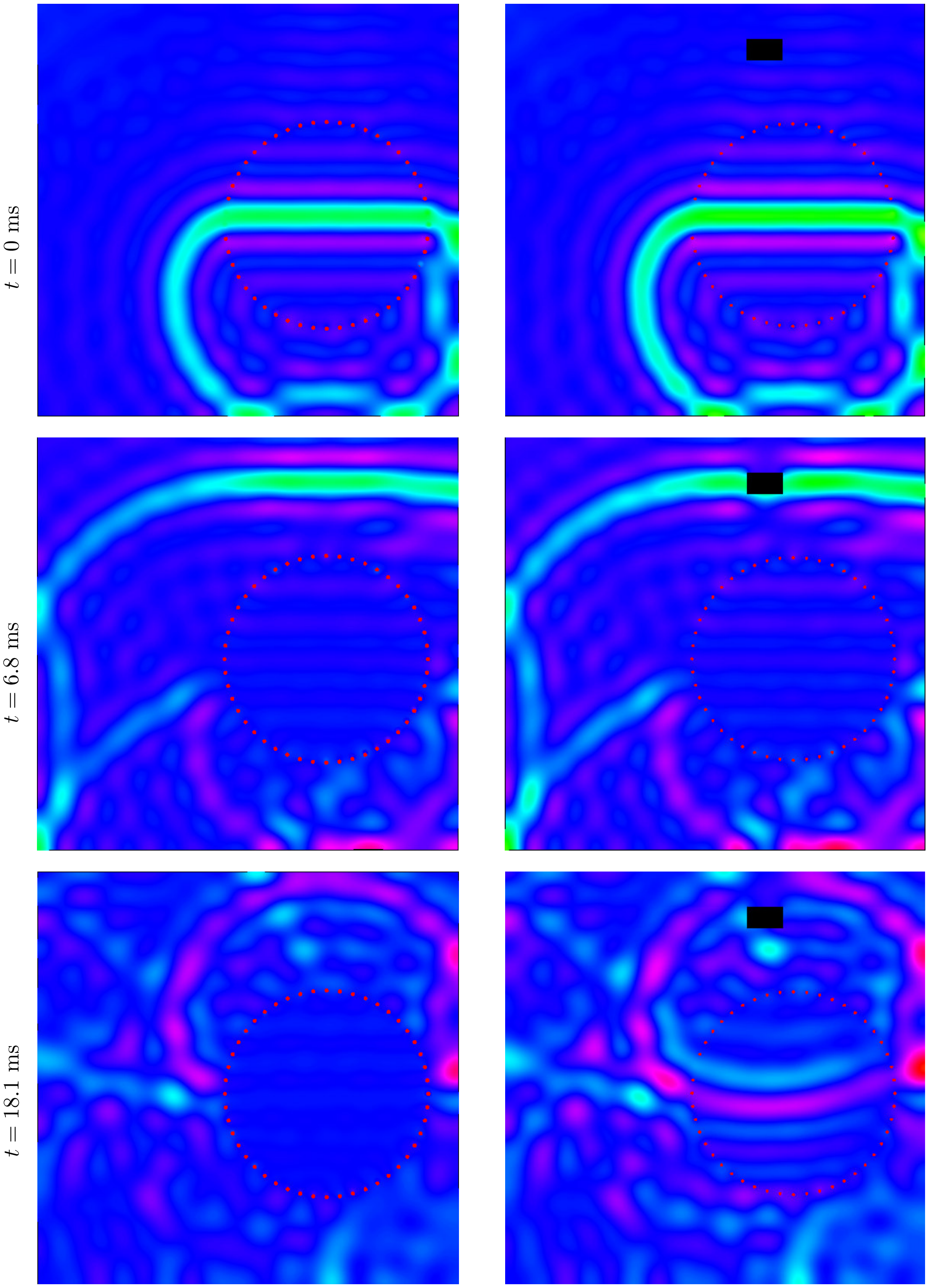

Fig. 8: Results for converged room compensation filters. The left row shows the wave field for the original scenario, the right row when a small object is present outside the array.

scene. The degradation in room compensation performance compared to the adaptive case can been seen clearly. The non-adaptive algorithm is not capable of suppressing the listening room reflections in this changed scenario. Especially the last two scenarios show the need for an adaptive room compensation algorithm as proposed in Section 4.

6. CONCLUSIONS

The presented approach demonstrates a feasible behavior for the simulation of massive multi-channel reproduction systems, i.e. wave field synthesis. Several features like a dispersion-free simulation, arbitrary source and receiver positioning, direct access to the physical parameters of the system, and a good behavior especially for low frequencies, enable even the simulation of complex algorithms. In particular it was used to visualize the effect of the listening room on the reproduced wave field in non-ideal listening rooms on one hand and to evaluate room compensation algorithms for a wide variety of scenarios one the other hand. As several effects like boundary conditions, temperature variations, and damping are included inherently in the FTM based simulation, a further development of room compensation algorithms is facilitated by the proposed procedure.

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