A Three-Dimensional Acoustic Simulation for the Development and Evaluation of Active Noise Control Systems using the FDTD Method

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Abstract

The development of Active Noise Control Systems to reduce noise in certain two- or three-dimensional areas or spaces, respectively, requires good knowledge of the desired installation location as well as the understanding of the systems properties and behaviour. In this paper simulation software is presented which allows the user to build virtual rooms or open air test sites and to place noise sources and active systems in an arbitrary manner to reduce the noise. A FDTD (Finite Differences Time Domain) algorithm was implemented to compute sound field behaviour, including reflection, absorption, transmission and diffraction in different media. The active systems can consist of an arbitrary number of sensors (microphones) and actuators (loudspeakers) which are combined and connected via different signal paths and signal tool boxes. This allows the creation of various system setups to find the best solution and to predict the effects of noise cancellation for a given problem.

Introduction

In the field of Active Noise Control (ANC) the development of wide area and wide frequency band noise reduction systems is desired. Compared to applications within small, well controllable boundaries like headphones, an ANC application for an arbitrary room considerably depends on the spatial characteristics and the placement of sensors and actuators. Every room or site has different acoustical properties, which have influence on the quality of an ANC system.

Computer simulations of acoustic fields can help developers to understand these properties and detect certain characteristics given by a surrounding environment and improve accurate sensor- and actuator-placement. Therefore, an application was developed which combines two important parts: The acoustic simulation of rooms and the simulation of active systems for realizing arbitrary noise cancelling algorithms.

Software Features

The simulation software's graphical user interface (GUI) mainly consists of three parts. The first screen, shown in figure 1, enables the user to model three dimensional rooms, place sound sources and sinks. It is possible to move inside this artificial environment and manipulate it. Simple objects like quadrics, walls or cylinders are available to build user defined scenarios. In order to involve more complicated structures the user is able to import computer aided design (CAD) models from software like Pro/ENGINEER or any other CAD-tool using the surface tessellation language (STL) format. The place and dimensions of the solution area for the sound field simulation are also defined within this screen.

In the second part of the application the user can build an active system setup by placing different signal blocks and connecting them to each other, like demonstrated in figure 2. For example a signal generator can be connected to a loudspeaker. The speaker itself also exists within the three-dimensional environment, delivering the signal of the generator to the sound field simulation. Microphones in contrast take pressure values from the simulation at every simulation step and therefore function as signal sources for the active system. In addition, several signal blocks are available to perform basic mathematical operations or to influence signals within the active system and within the simulated sound field. To give an example, the sound recorded by a microphone can be delayed or filtered and routed to a speaker, which could be located inside another room within the three-dimensional environment than the microphone. To evaluate the system at every part of the setup, the user can place virtual scopes and connect them to any signal path to record and display the appropriate signal.

The settings for the simulation process are modifiable via the third tab. Parameters like sampling time or the speed of sound can be set. Furthermore, the user can decide whether he or she wants to simulate the active system setup alone or combined with a Finite Differences in the Time Domain (FDTD) simulation for the sound field.

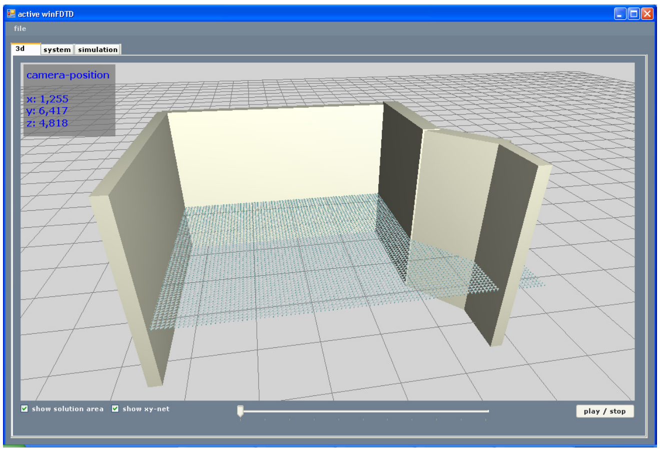

Figure 1: The three-dimensional environment editor showing a simple room configuration with a solution area in it.

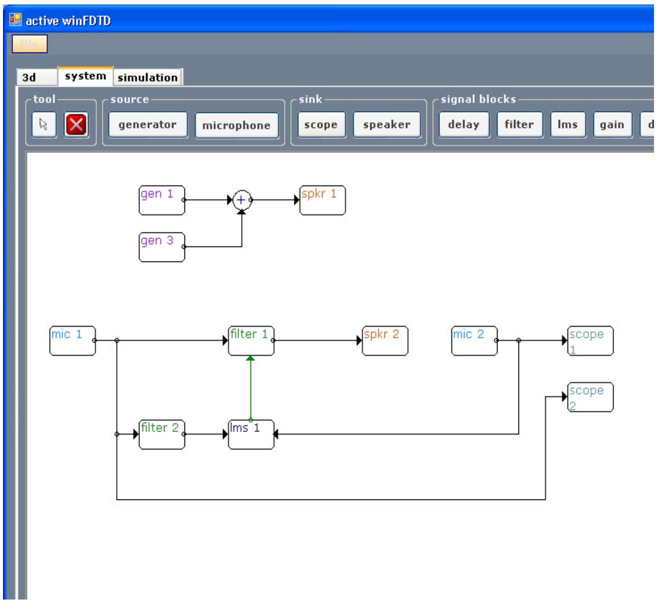

Figure 2: The active system setup editor including a FxLMS system (bottom) and two signal generators connected with a speaker to produce the primary noise.

drawing and connection of the different signal blocks. The whole simulation process is initiated and watched by the third class. The different manager classes can communicate and exchange data with each other, since objects like sound sources or sinks exist in all three sections of the software. Hence, a loudspeaker, which was removed from the three-dimensional scene, has to be removed from the system setup by the supervising class as well.

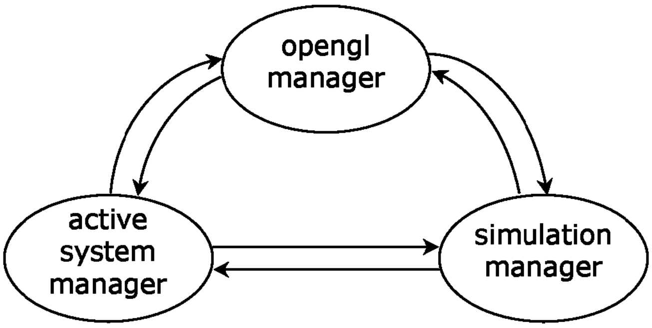

Figure 3: Demonstration of the software concept using three manager objects to administrate the objects of the different software subdivisions and to exchange data.

To increase performance and to prevent extensive memory usage during long time simulations it is possible to define start and stop values for recordings of the simulated sound field at every vertex in the solution area and for every time step.

The simulated and recorded sound pressure values are animated within the three-dimensional environment. Every pressure value is mapped to a colour- and a z-value, resulting in a moving, coloured wave, as shown in figure 4. The user can start and stop the animation or use a slider to pick a certain frame. Additionally, plots of input signals of microphones and scopes or output signals of loudspeakers can be viewed in a simple signal explorer.

Development of the Software

The idea behind the software is to create a powerful tool for the simulation of ANC systems as close to reality as possible with an easy-to-understand and easy-to-handle GUI. Due to the implementation of a three-dimensional environment editor and the wish to use a completely object oriented approach combined with the possibility of multi processor use, C# was chosen as programming language [1]. The project was built using Microsoft® Visual Studio® 2008 Professional Edition and the .Net framework. Three-dimensional graphics are rendered in OpenGL®, which was wrapped for C# by the Tao framework [2, 3]. To render the active system's symbols and connections, the Graphical Device Interface plus (GDIp) was included [4]. In addition to that, the ZedGraph library was employed to display function plots and graphs [5].

The main programming concept is based on different object manager classes for the three different parts of the software as shown in figure 3. The first class contains all objects of the three-dimensional scene and performs OpenGL® rendering. The second one manages all tasks like the

Simulation Process

The simulation process consists of two interacting parts to compute the state of the active system setup and the state of the sound field.

Active System

The active system setup holds several signal blocks, each representing a completely independent object. These blocks are connected via one or more input- and/or output-ports. When objects are linked by the user the information about the connection is saved in the objects and the connecting line, so each signal block can determine the other signal blocks it is connected to. Hence, no partial differential equations have to be solved during the simulation of the system but the signal blocks process data on their own and exchange the processed data among each other. Therefore, the system organizes the calculation itself.

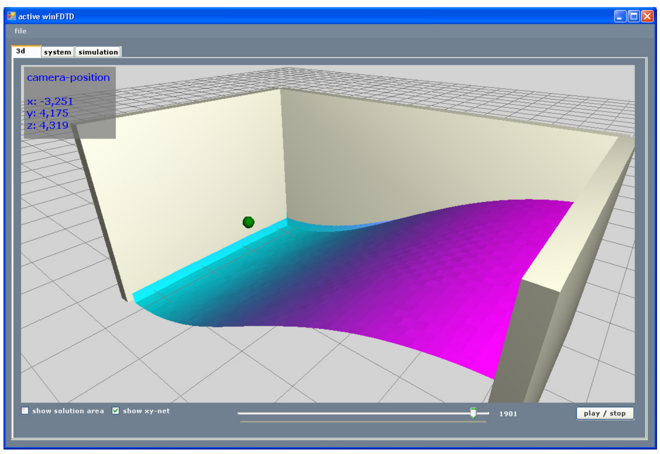

Figure 4: An animation of a sound field excited by a spherical source. Pressure values are mapped to the plane's z- and color-values.

Sound Field Simulation

A two-dimensional FDTD algorithm was implemented for the simulation of the sound field. This method is well suited for the computation of low-frequency sound propagation, which is the main application area for ANC systems [6]. Another advantage of the FDTD method is that calculations are performed in the time domain. Hence, the pressure values at a certain time step can be modified or used by the active system simulation. In addition to acoustic pressure, particle velocity can also be evaluated.

Thus, a FDTD method was chosen, which employs two coupled first order differential equations for pressure p and velocity $\vec{v}$. The first equation is the linear Euler's equation of motion, which describes the relation between the pressure gradient and the derivation of velocity with respect to time (1). It is valid for small variations of pressure [7].

$$\frac{\partial \vec{v}}{\partial t} = -\frac{1}{\rho}\vec{\nabla} p \qquad (1)$$

In this equation ρ denotes the density of the medium in kg/m³ and ∂t an infinitesimal small time step in seconds.

The second equation is the linear continuity equation and describes the relation between the velocity gradient and the derivation of pressure with respect to time coupled by the medium's density and speed of sound c in m/s (2).

$$\frac{\partial p}{\partial t} = -\rho \cdot c^2 \vec{\nabla} \vec{v} \qquad (2)$$

The differential operators in these equations are approximated via finite differences and therefore discretized in space h in meters and in time T in seconds (3), (4). The resulting grids for pressure and velocity are dislocated against each other at a half step in space and a half step in time.

$$v^{i}_{i+\frac{h}{2}}\left(t+\frac{T}{2}\right) = v^{i}_{i+\frac{h}{2}}\left(t-\frac{T}{2}\right) - \frac{T}{\rho \cdot h}\left[p_{i+h}(t) - p_i(t)\right] \qquad (3)$$

$$p_i(t+T) = p_i(t) - \frac{\rho \cdot c^2 \cdot T}{h}\left(v^{i}_{i+\frac{h}{2}}\left(t+\frac{T}{2}\right) - v^{i}_{i-\frac{h}{2}}\left(t+\frac{T}{2}\right)\right) \qquad (4)$$

Figure 4 shows a spatial grid for the two-dimensional FDTD implementation. To avoid numerical errors guarantee stability, the relation between step size in time and space for the two-dimensional FDTD simulation has to fulfil equation (5) [9].

$$cT \le 1 \Big/ \sqrt{\frac{1}{h_x^2} + \frac{1}{h_y^2}} \qquad (5)$$

The simulation software in this paper uses a grid with same step sizes in x- and y-direction. The spatial step size is related to the step size in the time domain, relative to the shortest wavelength.

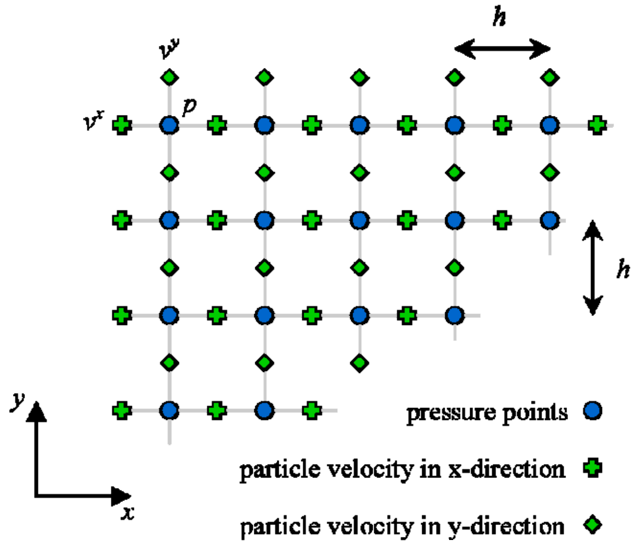

Figure 5: Depiction of a two-dimensional grid for the FDTD simulation (modified from [8]).

Perfectly Matched Layers

To avoid reflections from solution area borders, which result from the FDTD method, a perfectly matched layers (PMLs) algorithm was implemented [10]. A sound wave propagating into the outmost areas of the grid is attenuated gradually and therefore absorbed at the border with minimal reflections. The PMLs can give the user the ability to compute acoustic simulations for free-field-like environments.

Results and Future Prospects

Results

During a first test run of the simulation tool a duct-like configuration, open at one side, was build in the environment editor, including a sound source. The source emits noise and the aim is to reduce this emitted noise by an ANC system. For this purpose, a typical system setup using a filtered-x least mean squares (FxLMS) algorithm was simulated within the software application, like proposed in [11] (see also figure 2). The ANC installation consists of a reference microphone, a speaker to produce the cancelling signal and an error microphone. Typically, the chosen method consists of two steps: First, the secondary path from the cancelling speaker to the error microphone is adapted by a finite impulse response (FIR) filter using the LMS algorithm and second, the system is activated, trying to adapt on the noise in the duct.

The simulation showed a good result for the adaption of the secondary path, depicted in figure 5 as filter coefficients. The ANC application achieved little damping of the noise within the duct, which can be the outcome of several issues, for example the missing feedback compensation. Therefore, further testing has to be undertaken.

Future Prospects

By now a two dimensional FDTD simulation was implemented. In the near future the software will be extended to simulate and display three-dimensional sound fields with additional plotting tools. Furthermore, the consideration of wall impedances and different media within the simulation and the use of several solution areas are planned. The improved simulation of sound sources will be needed for reliable results of ANC system evaluation.

Within the active system setup, besides further signal blocks, a new interface will be implemented to give the user the ability to create signal blocks with custom functionality.

The simulation process will be enhanced for more flexibility, better performance and improved memory usage. Multi-threading and the use of multi-processor systems will be implemented.

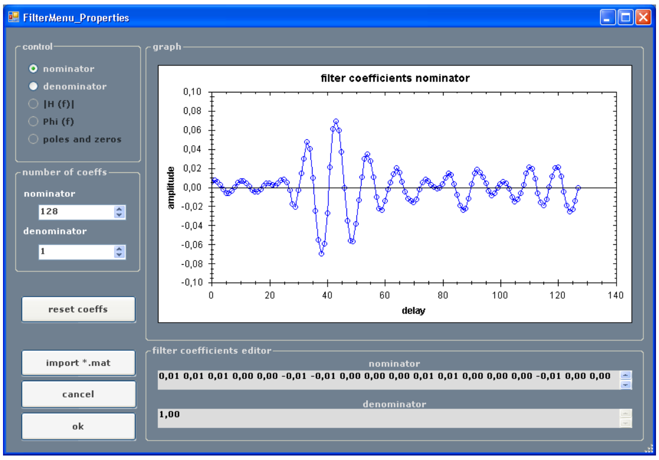

Figure 6: The properties menu of a filter object. The coefficients represent a virtually adapted secondary path of an ANC application which is to reduce noise in a duct using the FxLMS algorithm.

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