

SIMULATION OF ACOUSTICAL PROPERTIES OF TECHNICAL SYSTEMS USING A NETWORK-BASED SOUND-SERVER

Antje Siegel¹, Stephan Husung¹, Christian Weber¹, Albert Albers², David Landes², Matthias Behrendt²

¹Engineering Design Group, Technische Universität Ilmenau,

²IPEK – Institute of Product Engineering at Karlsruhe Institute of Technology (KIT),

ABSTRACT

Today, the development of new industrial products is often a time-consuming and cost-intensive process. Therefore, the demand for novel methods, which help to make product development more efficient, is increasing. In this regard virtual engineering can open up new possibilities. For instance, the simulation of product properties and the interactive presentation in a Virtual Reality (VR) environment allow verifying and discussing product requirements at an early stage of the development process.

The focus of this paper is on acoustical reproduction of virtual sound sources via a real-time capable, network-based sound-server. A main task of the sound-server is to reproduce the state-dependent properties of different sound sources. For this purpose the sound propagation chain of each sound source has to be reproduced. The transmission of both structure-borne noise and airborne noise has to be considered. The developed methods are explained in the paper using an automotive example.

The aim of the example application is to simulate a car passing test (according to DIN ISO 362) in a VR environment. For this purpose the auditory impression along a virtual test track is recreated. Different sound sources which cause the sound of the car (for example the wheels, the engine and the exhaust system) are described by digital filters. The filters are based on measurements carried out beforehand on an acoustical roller test bench.

1. INTRODUCTION

Acoustical properties of technical systems are becoming increasingly important particularly in the field of product development. For example, many industrial products have to satisfy noise standards and manufacturers make huge efforts to design pleasantly sounding products. In order to take both visual and acoustical information into account, a special audio-visual VR-system and audio-visual models describing the kinetic product behaviour as well as the sound generation and propagation are necessary. For current investigations a spatial, interactive, object-oriented auralisation system is used. The audio playback can be realized by a Dolby 5.1 or a wave-field-synthesis [1] framework.

One main part of the auralisation system is a so called sound server. The sound server is connected to the VR-software, which handles the scene graph and user interaction as well as the processing of the visualisation data. A main task of the sound server is to reproduce the state dependent properties of different sound sources. For this purpose the sound propagation chain of each sound source has to be modelled. The aim is to create digital filters which can describe the acoustical characteristics of different sound sources. For each sound source

several convolutions have to be executed in real time. In order to obtain an interference-free signal after the convolution, the overlap-add method is used. For the interactive VR-session a continuous filter transition is necessary. This is realised by a linear cross-fade of the filters. The developed methods are explained in the paper using an automotive example.

2. AURALISATION

To create a realistic impression of auditory events in virtual environments it is necessary to use a sound reproduction system that can reproduce the spatial properties of sound sources. Two common systems which succeed this demand are Dolby 5.1 and wave-field-synthesis (WFS) systems. The advantage of WFS over 5.1 is that the spatial audio impression is perceptible in a relatively large listener area within the loudspeaker arrangement and not only at a sweet spot. Whereas 5.1 can be realized with only six loudspeakers, which is an easy solution compared with the amount of loudspeakers that are necessary for WFS. The investigations presented in this paper were carried out with a flexible audio-visual stereoscopic projection system (FASP) [2]. The respective virtual model can be built with a commercial VR-software (see Fig. 1) and is represented in a scene graph. This scene graph contains all necessary geometrical and visual scene information and all data referring to the sound sources. Visual content is sent by the VR-software to a stereo projector system and is displayed on three moveable screens. Audio content can be auralised via Dolby 5.1 or a WFS system. Both audio systems need parameters describing the sound sources. These parameters are computed in the sound server using the scene graph data. The communication of the sound server with the VR-software and the audio system is realized by messages using the OSC (Open Sound Control) protocol [3]. Additionally the sound server generates an audio stream using audio data stored in a data base and information provided by a real-time simulation tool. The audio stream is sent to the audio playback system. [4]

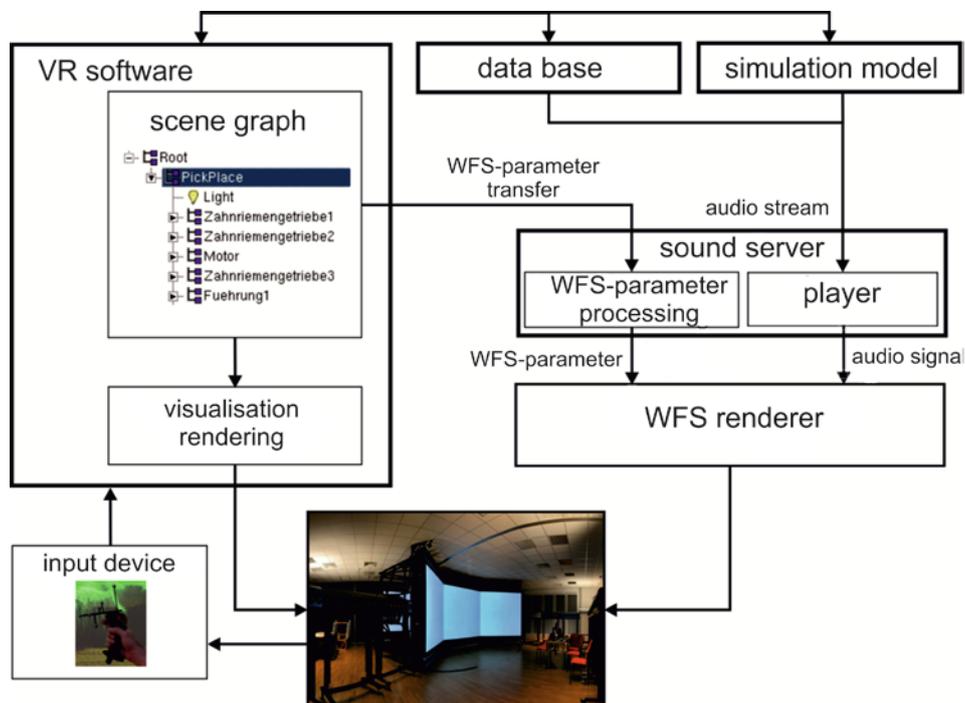


FIGURE 1: Architecture of flexible audio-visual stereoscopic projection system at Technische Universität Ilmenau

3. SOUND SERVER

The main structure of the sound server is shown in Fig. 2. For the auralisation of virtual sound sources the directional characteristics of the sound sources should be taken into account. Therefore the sound server provides the possibility of real time signal processing using digital filtering.

3.1 Convolution

Digital filters can be stored as wave files. For every sound source several digital filters can be stored. The digital filters describe the sound propagation from the source to specific positions around the sound source. The filters are loaded in the sound server when the virtual sound source is initialised. Therefore the sound server receives an OSC message from the VR-software. Dependent on the position of the user relative to the virtual sound source the right filter is chosen and a real time convolution is performed by the sound server. The convolution of the audio signal and the filter is realized block wise by the overlap-add method [4].

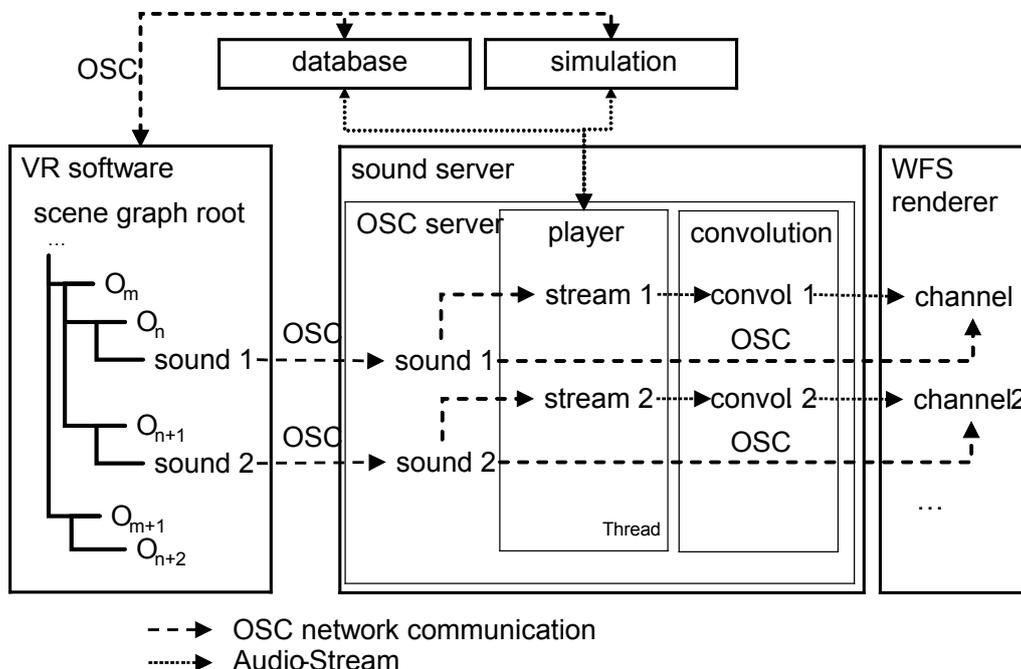


FIGURE 2: Main structure of the sound server

3.2 Filter cross-fade

In order to rebuild dynamic processes in a sufficient way filters describing several different states of the dynamic process can be used. To get a continuous signal without perceptible filter changes a linear filter-cross fade is done. This cross-fade is computed before the audio signal is convolved and sent to the audio representation system.

4. APPLICATION

A current important application of the sound server is the simulation of car exterior sound. Especially the vehicle noise level plays a central role in the development of a car. In this regard many standards and statutory regulations have to be satisfied by automobile manufacturers. These standards require time and cost consuming outdoor measurements on test tracks. In order to reduce expenses and to simplify these measurements automobile manufacturers use to carry out the measurements indoors on roller test benches. The

additional transfer of the investigations to a virtual environment is aspired to decrease the metrological effort even more. In order to reproduce the acoustical and visual properties of a vehicle in Virtual Reality a virtual model of the car has to be generated. The different sound sources of the car should be represented in the virtual model as separate elements. This allows investigating the influence of every single sound source on the overall noise of the vehicle.

4.1 Test scenario

The DIN ISO 362 standard is one of the standards which regulate the measurement of noise emitted by road vehicles. Several test scenarios are defined in this standard. The investigations presented in this section are based on a vehicle pass-by with constant speed which is one of the described test scenarios of the DIN ISO 362. The aim is to simulate a car which passes with a constant speed of 50 km/h.

4.2 Transfer functions and measurements

The virtual model of a passing car in the application consists of seven single sound sources. These single sources represent especially the engine, the intake, the four tyres and the exhaust system. For the parameter identification on the one hand the actual source signals are measured. On the other hand transfer functions which describe the transmission behaviour from the sources to different positions along the car have to be determined. For this purpose measurements were carried out at the IPEK – Institute of Product Engineering at Karlsruhe Institute of Technology (KIT) [5]. A vehicle was placed on a roller test bench and near field measurements were made to get the source signals. In a second step white noise was generated on several positions parallel to the longitudinal axis of the car (see Fig. 3). This white noise was emitted by a dodecahedral loudspeaker to approximate the directional characteristics of a point source. The transfer functions from the single sound sources to the position of the dodecahedral loudspeaker were measured reciprocally with microphones which were placed in a distance of 100 mm to the sound sources. Based on the transfer functions FIR (finite impulse response) filters can be generated.

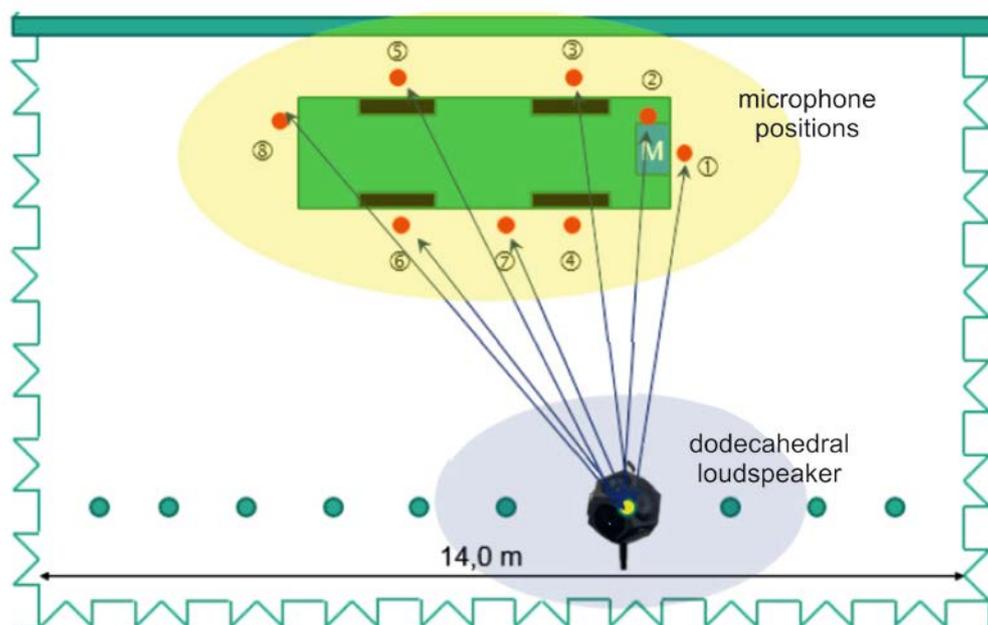


FIGURE 3: Measurements with dodecahedral loudspeaker

4.3 Audio playback

The measured source signals and FIR filters are stored as WAVE files. In order to ensure that the filters are assigned correctly to the respective sound sources the file names of the sound sources are analysed by the sound server. The file names of the filters have to end with the same characters as the respective source file names. Consequently the sound server can search for the right filter files and can load them when a new source is initialised. The sound server chooses the relevant filters as described in section 3.1 and provides the necessary data for the audio reproduction system.

5. EVALUATION

The concept presented in section 4 will be evaluated in several tests. Therefore preliminary investigations were made. The sound server was prepared for the necessary signal processing (see section 3) and first digital filters were created. The digital filters are based on a simplified virtual model which considers non-separated sound sources. This virtual model takes only into account the overall vehicle sound. It was assumed that the car is a point source and that the acoustical centre of that point source is at the position of the engine [6]. A vehicle was fixated on a roller test bench and the car noise was measured at a distance of 7.5 m from the vehicle with a linear microphone array (see Fig. 4). Since microphone 6 was nearest to the acoustical centre it was supposed for reasons of simplification that the signal at microphone 6 is equal to the vehicle sound which is emitted by the assumed point source. Therefore the signal at microphone 6 served as reference signal. By using the other microphone signals transfer functions from the reference microphone to the positions of the other microphones were computed. These transfer functions were used to build digital filters [4].

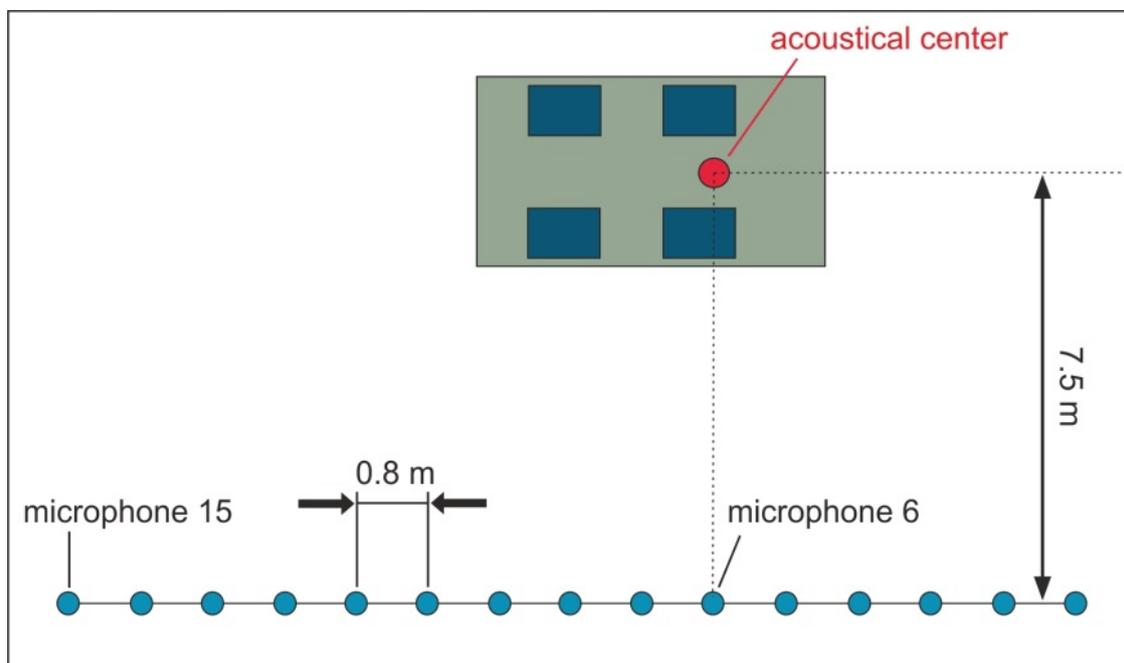


FIGURE 4: Measurement setup

The quality of the filters is checked by performing tests with volunteers. First results can be seen in Fig. 5. Since only a few test persons participated in the study the results can only be interpreted as tendencies. The test persons listened to sound examples of the filtered signals and compared them with original signals which were recorded with the linear array. The

participants had to assess if a difference between the filtered and the original signal was perceptible. For each sound example the probands could give one of the following answers:

- no differences are audible (1)
- slight differences are audible (2)
- differences are audible (3)
- considerable differences are audible (4)
- no similarities are audible (5)

The ordinate of Fig. 5 consists of the numbers 1 to 5. The numbers refer to the above listed rating which the subjects could give. One is equal to “no differences are audible” and 5 is equal to “no similarities are audible”. The numbers 1 to 15 on the abscissa correspond to the respective microphone positions from the above described measurement setup. The data point (13 | 2), for example, shows that when participants compared the original signal of microphone 13 with the respective filtered signal slight differences were audible on average. The sound example that is represented on the abscissa with the number 16 served as hidden anchor. This sound example consisted of two equal signals. For every data point also the standard deviation is displayed. As can be seen in the diagram signals of microphones which are nearer the reference microphone can be described better with the filters than the microphone signals which are further away from the reference.

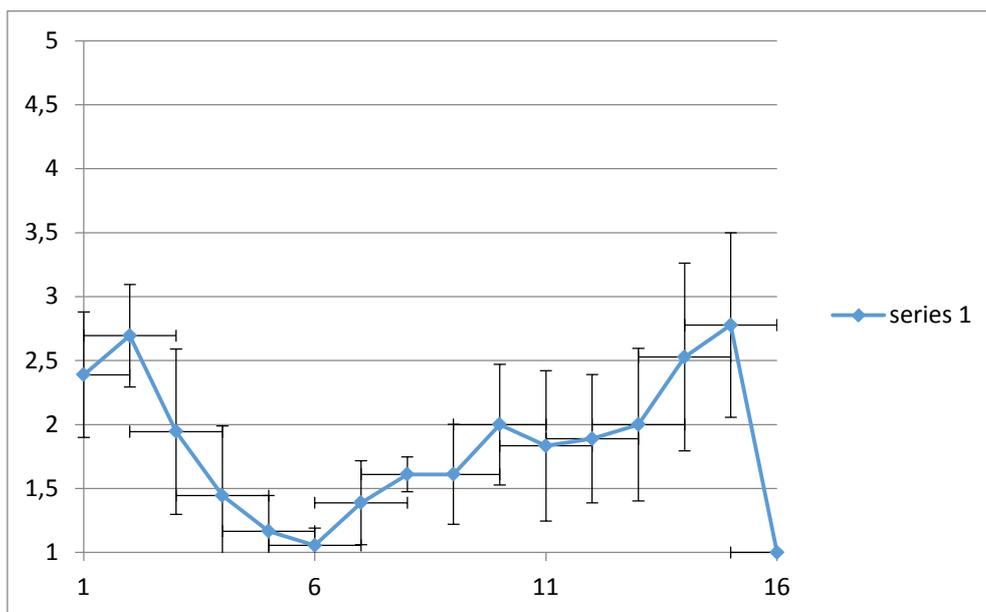


FIGURE 5: Evaluation results for different microphone positions

6. CONCLUSION

In this paper the architecture of a flexible audio-visual stereoscopic projection system is described. The focus is on a sound server which provides the possibility for real time signal processing. This is the basis for representing directional characteristics of sound sources in a VR environment. The functionality of the sound server is illustrated by an application example. The application example refers to the exterior sound emission of a vehicle. In order to auralise the car sound in VR a virtual audio-visual model is presented. The model consists of separated sound sources and digital filters which describe the acoustical properties of these sound sources. For the evaluation of the model preliminary investigations were made.

Therefor tests with volunteers were made and first results of the tests are shown. Furthermore additional measurements will be carried out to modify and improve the virtual model. The Quality of the digital filters will be evaluated and verified in further tests.

ACKNOWLEDGMENT

The authors would like to thank the members of the Zeidler-Forschungs-Stiftung for their support.

REFERENCES

- [1] SLADECZEK, Christoph ; REUSSNER, Thomas ; RATH, Michael ; PREIDL, Kar ; SCHECK, Hermann ; BRIX, Sandra: Audio Network Based Massive Multichannel Loudspeaker System for Flexible Use in Spatial Audio Research. In: Journal of the Audio Engineering Society 2013, Nr. 61, S. 235–245
- [2] WEBER, Christian ; HÖHNE, Günter ; HUSUNG, Stephan: Modelling of Acoustical Properties for Technical Systems in VR. In: DAUNYS, M. (Hrsg.): 14th International Conference Mechanika, 2009, S. 456–460
- [3] WRIGHT, Matthew ; FREED, Adrian: Open Sound Control: A New Protocol for Communicating with Sound Synthesizers. In: International Computer Music Conference. Thessaloniki, Hellas, 1997
- [4] HUSUNG, Stephan ; SIEGEL, Antje ; WEBER, Christian: ACOUSTICAL INVESTIGATIONS IN VIRTUAL ENVIRONMENTS FOR A CAR PASSING APPLICATION. In: ASME 2014 International Design Engineering Technical Conferences & Computers and Information in Engineering Conference, 2014, (to be published)
- [5] ALBERS, Albert ; LANDES, David ; BEHRENDT, Matthias ; WEBER, Christian ; SIEGEL, Antje ; HUSUNG, Stephan: Determination of the Near-Field-Acoustics of Primary Vehicle Sound Sources in Relation to Indoor Pass-by Noise Testing for the Verification of a Virtual Acoustic Vehicle Model. In: SCHARFF, Peter (Hrsg.): Ilmenau Scientific Colloquium – Shaping the Future by Engineering : 8 - 12 September 2014 ; programme. Ilmenau : Univ.-Verl, 2014 (to be published)
- [6] JANSSENS, K., AARNOUTSE, P., GAJDATSY, P., BRITTE, L., DEBLAUWE, F., and van der AUWERAER, H., 2011. “Time-domain source contribution analysis method for in-room pass-by noise”. In SAE 2011 Noise & Vibration Conference and Exhibition.

CONTACTS

Univ.-Prof. Dr.-Ing. C. Weber
Dipl.-Ing. A.Siegel
Dr.-Ing. S. Husung
Prof. Dr.-Ing. Dr. h.c. A. Albers
Dipl.-Ing. D. Landes
Dr.-Ing. M. Behrendt

christian.weber@tu-ilmenau.de
antje.siegel@tu-ilmenau.de
stephan.husung@tu-ilmenau.de
albert.albers@kit.edu
david.landes@kit.edu
matthias.behrendt@kit.edu