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# Audio Engineering Society

# Convention e-Brief

Presented at the 142nd Convention  
2017 May 20–23 Berlin, Germany

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## Fully Digital Development of Automotive Audio Systems

Alfred J. Svobodnik<sup>1</sup>, Marc Levasseur<sup>1</sup>, and Christof Fallner<sup>2</sup>

<sup>1</sup> MVOID Group, Karlsruhe, Germany

<sup>2</sup> Illusonic GmbH, Uster, Switzerland

Correspondence should be addressed to Alfred J. Svobodnik (alfred.svobodnik@mvoid-group.com)

### ABSTRACT

This paper describes the building blocks of a fully digital development environment for automotive audio systems. The whole development process, including all major engineering disciplines, has been virtualized - up to the realistic audibility of the sound systems by means of auralizations. All building blocks are based on simulations, and thus fully digital prototypes can be used already in the early concept phase. Hence, product quality, i.e. reproduced sound performance, can be assessed, and improved, long before any hardware exists.

### 1 Introduction

The term virtualization in product development is often associated with the finite element method, being a numerical analysis scheme to solve physical phenomena to predict a product's performance in the virtual domain. Since its invention in the middle of the last century [1] there has been much progress on applicable physics and its industrial applications. The finite element method was probably a key technology to drive the development of numerical simulations in general, also based on other numerical schemes, resulting in a significant change of the whole development paradigm, which is nowadays called simulation driven engineering.

While this new paradigm was first introduced in the aerospace and automotive industry, it is today a general trend to follow for almost all industries [2]. More and more, advanced engineering analysis methods are used already in the early development phases to set up digital prototypes for assessing the performance of products already at concept level.

This trend can also be seen in the audio industry. For more than a decade now, advanced engineering analysis methods are used on a regular basis for the

multiphysical simulation of loudspeakers long before any hardware exists [3]. Recently, the complex listening environment of an automobile has been included in the simulation models as well [4].

There is a significant difference in the assessment of product performance between the audio industry and many other industries. While for most technical products its performance can be described by reference metrics or parameters, an audio system's performance, i.e. the reproduced and perceived sound quality, cannot be described by a closed form definition with a distinct metric that could be measured or simulated. Typically, the assessment of sound quality is based on subjective methods by means of listening tests [5].

This implies that the simulation model must be capable of making the acoustic results audible, so that listening tests of sound systems can be performed, and hence its sound quality assessed and subsequently improved already at early development stages. Therefore, auralization techniques need to be coupled to the multiphysical simulation model.

Especially for the automotive industry, this is of crucial importance as there is a strong trend towards

a fully digital development environment, where advanced engineering analysis methods are moved upfront in the development process, with the goal of increasing the engineering efficiency and using less physical prototypes.

In the following, a fully digital development environment for automotive audio systems is described, that allows to listen to virtual prototypes, and thus opens the capability of assessing the sound quality already at concept level where no real prototypes in hardware exist.

## 2 Extending Multiphysical Simulations to Multidisciplinary Simulations for a Fully Digital Development Environment

Adding the capability of auralizations to our multiphysical simulation model actually means to extend it to a multidisciplinary model. Additional engineering disciplines need to be added to multiphysical engineering analysis:

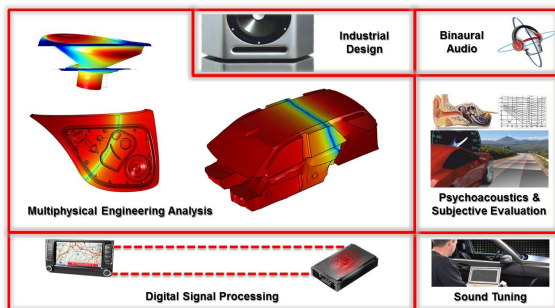


Figure 1. Engineering disciplines needed for a fully digital development environment for automotive audio systems

The building blocks needed for a total virtualization of the development process, based on the above-mentioned engineering disciplines, are:

- Multiphysical simulation
- Virtual tuning
- Auralization

These building blocks are described in the following sections.

## 3 Multiphysical Simulation

On loudspeaker level, the simulation model needs to capture at least the most important physical domains:

- Electromagnetics
- Mechanics (structural dynamics)
- Acoustics

The following figure shows the governing equation for each domain and the resulting coupled system:

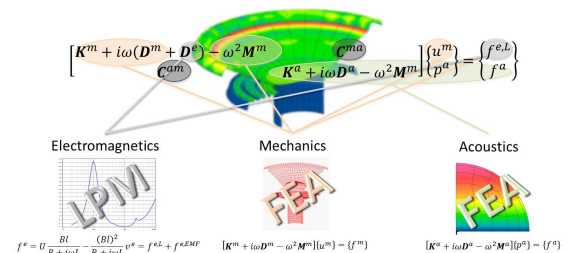


Figure 2. Governing equations that define the coupled multiphysical model

A detailed description of this model can be found in [6]. In the context of auralizations we are dealing with system level simulations. I.e., the whole system including the interaction of the loudspeaker with an enclosure and a listening space needs to be modelled. Because of the scale differences between the loudspeaker’s motor system (the air gap between voice coil and magnet is only several 10<sup>th</sup> of millimeters) and the passenger compartment (size is several meters), we use a mixture of 1D and 3D models. A 1D lumped parameter model is used for the motor system, while finite elements are used for the mechanical and acoustical domain.

The usage of finite elements opens a modeling approach to directly include the mechanical and acoustical effects of enclosures and listening spaces. That can be done by extending the geometrical domain to these additional spaces [7]. However, there are numerical limitations for the finite element method in simulating the whole audible frequency range for the passenger compartment. Hence, we use a hybrid model based on finite elements for the low and mid frequency region, and a raytracing model for high frequencies.

This approach gives us finally a room impulse response (RIR) at arbitrary locations in the 3D space that we will later use as a starting point for sound tuning and auralization. This RIR covers the vibro-electroacoustic sound path from the speaker terminals, where voltage is applied, to specific measurement points in the passenger compartment, including also the location of the listener's ears. As the finite element simulations are done in the frequency domain, and the raytracing simulation in the time domain, we need to apply transformation operations to the results from time to frequency domain, and vice versa, and to merge the results from the two simulation methods, to get a full bandwidth impulse or frequency response.

#### 4 Virtual Tuning

The major factors of perceived sound quality can be described as:

- Frequency response smoothness on and off-axis
- Perceived bass extension
- Naturalness

Where naturalness can be defined as “the feeling of space”, i.e., the correct sense of ambience, or the appropriate levels of direct and reflected information.

A passenger compartment is a highly challenging listening environment that leads to many degradations of the above-mentioned major factors of perceived sound quality. In the lower frequency region, especially below 500 Hz, modal effects due to the small room result in huge variation of sound pressure level, sometimes up to 30 dB. This variation can be observed both in terms of a frequency response for a given listening location, and a seat-to-seat variation. Misalignment of loudspeakers, because of different locations of speakers that are being crossed over, and severe packaging and integration constraints, are additional sources of sound performance degradation.

By using methods of digital signal processing, these degradations in sound quality are typically improved during the process of tuning, where for each loudspeaker channel gain, delay and filters (equalization) are applied. In a first tuning phase the

frequency response, as a graphical information, is used as a reference to improve the perceived sound quality, while additional refinement steps are based on listening tests.

A similar process is executed in the virtual domain, where the simulation of the RIR is used as a virtual measurement instead of a real measurement. However, in both the real and virtual domain, a careful definition of measurement points defining the response at a specific seating location needs to be considered.

As the perception of sound in such small spaces as an automobile's passenger compartment is related to sound power, our measurement process is based on the localized sound power method [8]. This method uses an array of six microphones to measure sound pressure, and perform a spatial averaging as an efficient approximation to local sound power. The term “local” here means the proximity of a listener's ears. As per definition, the location of the microphone covers the “99% ear ellipsoid”. That is the zone where 99% of the driver's ears are located.



Figure 3. Typical measurement set-up for the localized sound power method

The output of the multiphysical simulation model is the 3D sound field in the passenger compartment, and hence we can directly derive the RIRs at the microphone locations as defined by the localized sound power method.

These RIRs are used to perform a real-time tuning for each channel in the virtual domain. Gain, delay and filters are applied, and the resulting spatially averaged response, i.e. the localized sound power, is

calculated as a first visual metric for the sound quality. This is done individually for each channel, but also for the summation of all channels to represent the full system response at a specific seating location. Real-time here means, that when changing any tuning parameter the resulting change in the frequency response curve of the localized sound power is immediately visualized.

The result of such a typical tuning can be seen in the following figure:



Figure 4. Raw (left) and tuned (right) virtual frequency response

The figure shows the localized sound power response from 20 Hz – 20 kHz with 6<sup>th</sup> octave smoothing for the summation of all channels (thick orange line) of a system containing subwoofer (blue), woofers (green), midranges (white) and tweeters (red). Looking at the tuned results, it is clear that the first two major factors of perceived sound quality, “frequency response smoothness on and off-axis” and “perceived bass extension” are significantly improved. These two factors are classified as spectral attributes, and thus can be detected in the frequency domain.

As there is almost no indication of spatial attributes in a frequency response, at least not for automotive listening spaces, the third major factor of perceived sound quality, “naturalness”, cannot be assessed by a frequency response. There is no alternative to a listening test to evaluate spatial attributes. As in the concept phase no hardware exists, there is only a possibility for a listening test in the virtual domain by means of auralizations based on the simulated RIRs.

## 5 Auralization

Major psychoacoustic effects, namely spatial effects and localization of sound, need to be added to the room impulse response, so that a realistic listening

experience in the virtual domain is created. At our two ears, sound events arrive with differences in time and level, caused by reflection and diffraction due to our head and torso. These binaural cues need to be added to the ear sound pressure, which is done by applying a head-related-transfer-function (HRTF) to the simulated RIRs at left and right ear location.

There are several ways to derive HRTFs. It can be based on measurements from individual persons or artificial heads (dummy heads). To circumvent the disadvantage that such individualized HRTFs might not fit well for listeners with different physiology, we use an analytical model with a simplified geometry [9].

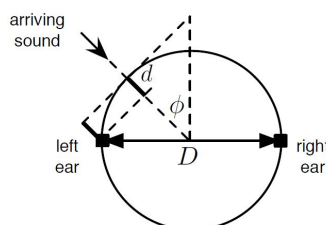


Figure 5. HRTF model with simplified geometry

As a rigid sphere is used as a head model, it is possible to derive a closed form solution for an efficient approximation of left ear’s and right ear’s HRTFs due to time and level differences of incident sound waves. It must be noted, that the model is a full 3D model and thus HRTFs vary with azimuth (i.e. horizontally) as well as elevation (vertically) for arriving sound waves.

These HRTFs are then applied to the simulated RIRs, so that finally the binaural room impulse response (BRIR) is calculated. Despite typical binaural audio applications, where usually only one microphone located at the center of the head is used to derive a single RIR, we use two RIRs from the simulation model corresponding to left and right ear. Thus, the time difference is directly based on the multiphysical simulation model, resulting in an improved accuracy and a more natural listening experience.

In our multiphysical model, the RIR is simulated at the location of the ear canal entrance, and in the

HRTF, the sound path through the ear canal is not included. This aligns well with the reproduction system based on headphones, which uses the listener's real ear canal during playback.

The HRTF filters are implemented in the discrete time domain, using first order IIR filters and fourth order Lagrange fractional delays, to enable real-time processing. Real-time processing is of crucial importance to realize a virtual reality (VR) like environment for tuning an audio system.

In a last step, the left and right BRIRs are convolved with acoustic test files, and the resulting signal is sent to headphones. The following figure shows the principal signal flow of our auralization environment:



Figure 6. Signal flow for a VR-like tuning environment

The test signal passes a routing matrix, which routs the two-channel test signal to the individual channels of the sound system. Then each channel's tuning parameters are applied, and the resulting signal is sent through the simulation data, the RIRs. Finally, the BRIRs are calculated by applying the HRTFs, and the resulting signal is sent to headphones for playback.

## 6 Conclusions

The building blocks of a VR-like fully digital development environment for automotive audio systems has been presented. Starting with a multiphysical simulation model of the loudspeaker, enclosure and listening space, it has been shown how to add additional engineering disciplines, namely digital signal processing, sound tuning, psychoacoustics and binaural audio, to enable tuning and listening in the virtual domain. By adding the blocks virtual tuning and auralization to the multiphysical simulation model, a fully digital VR-like development environment is realized, which can be used for assessing an audio system's sound quality - and developing improvements - already in

the concept phase, where no real, hardware based prototypes exist. This environment has been validated by means of listening tests based on A/B comparisons between real and virtual cars, and thus its maturity could be demonstrated.

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