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Computer Aided Engineering of Public Address Loudspeakers

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Abstract
Sound reinforcement in public venues requires high sound pressure levels in case of an evacuation and a precisely determined directivity of the loudspeaker systems to ensure a sufficient STI distribution in the auditorium.

Nowadays, numerical methods, CAD-Design and advanced post-processing methods can be combined to form a powerful development tool for designing the loudspeaker cabinet including components like horns and wave guides. The presentation illustrates how the different methods and software tools can be combined to predict loudspeaker performance as precisely as possible using computer aided engineering.

Keywords: loudspeaker, horn, numerical methods, directivity, CAD

1 Introduction

Loudspeaker systems can be found in nearly all public buildings and spaces, like train and underground stations, airports, tunnels, shopping malls, football stadiums, arenas, churches etc.. The scope of applications of the public address system varies from pure speech reproduction or background music at low levels in shopping malls to full range high level sound systems for football stadiums. In many cases the public address system is also used as voice alarm system to give instructions, for instance, in case of an emergency situation. Accordingly, the sound system is an essential component to ensure the safety of public spaces.
Primary goal in the design of the sound system is to achieve a sufficient speech intelligibility. In modern buildings, concrete, glass and other materials with poor absorption coefficients are commonly used which, unfortunately, comes along with long reverberation times (RT). The requirement for high speech intelligibility is contrary to long reverberation times, but as

architectural and budgetary considerations often have a higher priority than room-acoustical needs, this has to be considered as fixed boundary condition in the design of the sound system.
To meet the requirements for the variety of applications in different venues a number of basic loudspeaker concepts have been developed over the last decades and for each concept there exists a variety of cabinet sizes, power handling capabilities, frequency ranges, radiation characteristics, etc.. To achieve a high speech transmission index (STI) in public venues, it can be stated that the radiation characteristic is one of the most important features of a public address system. The rule of thumb of sound system design is simply to direct the sound to where it is needed and avoid radiation onto unoccupied areas. Furthermore, the sound pressure level at the listener positions has to be high enough to ensure a sufficient signal-to-noise ratio. Especially in stadiums, this can become a crucial factor for choosing a suitable loudspeaker concept.
The significance of the loudspeaker's radiation characteristic is also reflected by the increasing accuracy of loudspeaker databases used in electro-acoustic simulation tools. This paper gives an overview of today's methods used in loudspeaker development focussing on the tools to simulate and optimize the radiation characteristic during development.

2 Directivity Engineering

Loudspeaker development starts with a definition of properties and components of the system to be developed. Key feature of the design is the desired radiation characteristic. First, the here used numerical method is described briefly and then a simple two-way loudspeaker will serve as example to demonstrate how numerical methods and post-processing tools can be used to predict and optimize directivity of loudspeaker systems during the development process.

2.1 Numerical simulation of radiation using the Boundary Element Method (BEM)

The Boundary Element Method (BEM) is a well known tool in acoustics for the calculation of radiation from vibrating surfaces. It was described in a couple of papers dealing with more or less specific horn related topics, for example in [1]. Generally speaking, the method enables one to simulate the exterior (or interior) sound-field of a closed surface with a surface admittance and vibrating regions on the surface. Accordingly, using this methodology in a more general way for loudspeaker development seems to be straight-forward. Unfortunately, numerical methods like BEM or FEM need a lot of computational power and, in the case of the BEM, computation time and needed memory size increase unproportionally with the size of the numerical problem. But with increasing performance of desktop PCs and affordable huge memory, this drawback was compensated for over the last years making the BEM an effective tool in loudspeaker development and optimisation of radiation properties.
The idea of the BEM and how it is used in practice is shown in Figure 1. Basis is the Helmholtz-integral equation which describes the relation between pressure and velocity distribution on a closed surface and the exterior and interior sound-field of the particular configuration. To solve the equation numerically, the surface has to be discretised and the radiation problem has to be described in a suitable way. As an example, Figure 2 shows a quarter section of a small horn. The horn surface and cabinet are described by many small triangles, so called elements. Each element has its own velocity and admittance. In this example, a homogeneous velocity distribution is defined (red area) at the interface between

horn driver and horn (Figure 2, right side). Hence, the driver is replaced by a velocity distribution, which is assumed to be equivalent to a real distribution generated by the sound source. More sophisticated source descriptions can be used and combined with numerical methods [3], but that would go beyond the scope of this paper.

$$\iint_S \left[p(y)\frac{\partial g(x,y)}{\partial n(y)} - \frac{\partial p(y)}{\partial n(y)} g(x,y) \right] ds = \begin{cases} p(x), & x \in B_e \\ C(x)p(x), & x \in S \\ 0, & x \in B_i \end{cases}$$

outside
surface
inside

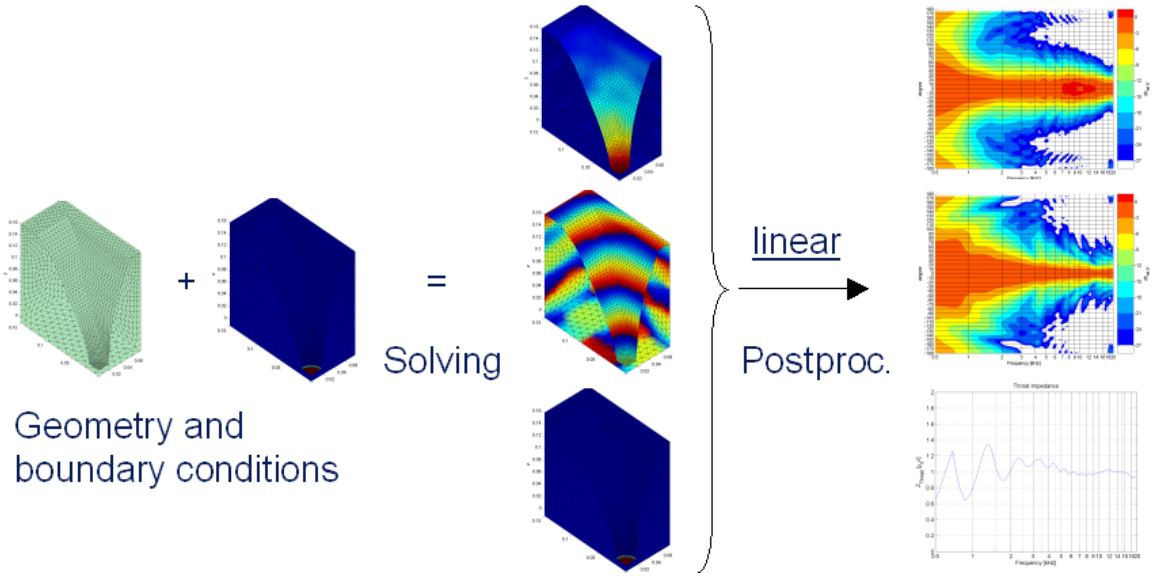


Figure 1 Basic steps of the Boundary Element Method

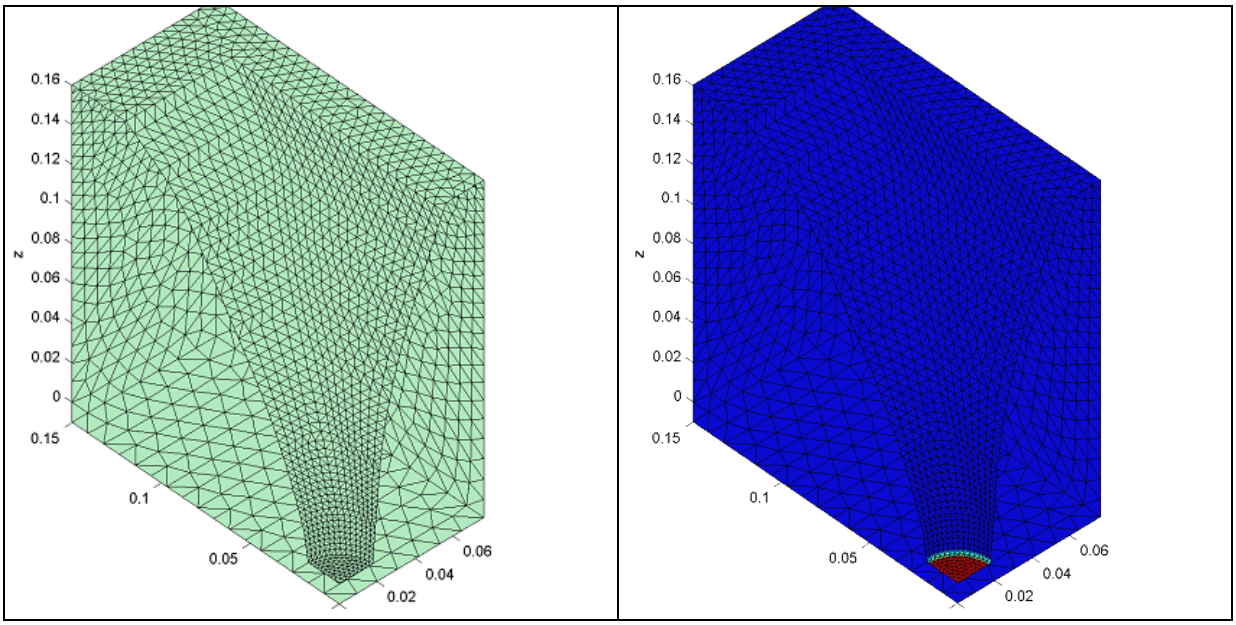


Figure 2 Quarter section of a small horn loudspeaker: geometry (left) and boundary conditions (right)

The three parameters, the geometry of the problem, the distribution of velocity and the distribution of admittance are the boundary conditions of the radiation problem. Together with

the boundary conditions, the equation (middle row) can be solved for the unknown pressure distribution p(x) on the surface. Now, pressure magnitude and phase are known, so that integrating the complex pressure over the surface yields the pressure outside the surface at any point in the sound-field. This last step is called post-processing. The most important results are the directivity of the loudspeaker and the loading impedance on the vibrating panels of the surface. Using an advanced post-processing tool, a good estimation of non-linear distortion generated by the horn itself can be calculated, too [3].

2.2 Basic considerations for a simple two-way 60°x40° loudspeaker

The use of numerical methods in directivity engineering shall be demonstrated with the help of a simple two-way public address loudspeaker.
The desired directivity of the example loudspeaker is assumed to be 60° -6dB-beamwidth in the horizontal direction and 40° -6dB-beamwidth in the vertical direction (60°x40°). The loudspeaker is to be realized using a simple 1"-horn driver combined with a horn and an 8"-mid range woofer. Furthermore, the horn should be rotatable, so that it is also possible to obtain a 40°x60° pattern by 90° rotation of the horn. The exact cross-over frequency between 8"-woofer and 1"-high frequency horn is not known at this stage of development. Typical cross-over-frequencies of 1"-horn drivers are in the range between 1 kHz and 2 kHz, depending on the type of driver and steepness of the used cross-over. From these few boundary conditions, it is possible to sketch the speaker concept and make first simulations. Figure 3 shows a simple set-up of the front baffle, indicating the maximum dimensions of the horn and the estimated frequency ranges of the woofer and tweeter. Now, the optimization of the complete system can be started. In a first step, the horn itself has to be optimized under consideration of the approximate front baffle dimensions. From the estimated cross-over frequency, one can deduce, that the 1"-horn will cover a range of four octaves, which means that its directivity should be as close as possible to the desired pattern within these four octaves. In a second step, the interaction between woofer and horn depending on various cross-over functions is studied. To do this, the complete system, consisting of woofer and horn, will be simulated and superimposed with suitable cross-over functions. The type of cross-over and the steepness of its slope will influence a range of about 1 or 2 octaves. The frequency range below this is mainly determined by the diameter of the woofer's membrane and diffraction at the cabinet and, thus, is a function of baffle dimensions and position of the woofer on the baffle.

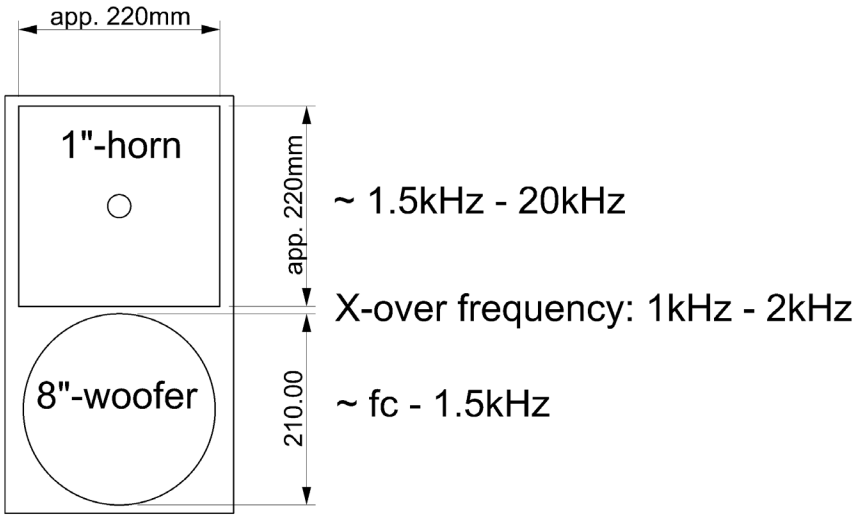


Figure 3 Approximate definition of maximum dimensions for the front baffle layout and estimated cross-over frequency.

2.3 Horn optimisation

In the first step the horn should be designed to achieve the desired directivity (60°x40°). To be as close as possible to the final configuration, the dimension of the baffle and position of the horn should be considered. In this particular case of a rotatable horn, it is sufficient to use only the width of the baffle during optimisation. Accordingly, the task is to find a horn geometry which can be mounted using a quadratic flange not exceeding 220 mm. The assumed cabinet width for the simulation model is 250 mm. The horn's length is not defined and will result from the simulation. To save computational time, only a quarter section of the model is processed, using the fact that the geometry has two planes of symmetry.
Figure 4 shows the procedure for optimising the horn with respect to a desired directivity or other properties like throat impedance or efficiency. The optimisation begins with an initial set of parameters to define the geometry to start with, boundary conditions and settings for the mesh generation and post-processing. From the geometry parameters, the edges of the geometry are generated and from the set of edges the simulation mesh is computed. After solving and post-processing, a set of results is calculated, like horizontal and vertical directivity pattern, throat impedance or estimates for distortion caused by the horn. Based on the results, the parameters of the geometry are modified and the next iteration is started. With some experience in horn design it is possible to reach the desired result after 10 - 50 iterations. Computation time using a PC with quad-core 2.6GHz CPU is about 1-2 s for each frequency for a mesh with 1000 nodes. During the first iterations, the basic behaviour of the horn can be optimised by processing only few frequencies. A cycle of iteration needs about 1 minute for a middle-sized horn with two planes of symmetry. With ongoing optimisation more detailed results require more frequencies to be processed and probably a refined mesh with more elements to get smoother results at high frequencies. Computation time can be 30 minutes or more, then, for one iteration cycle, but this can easily be reduced by increasing the number of CPUs involved.

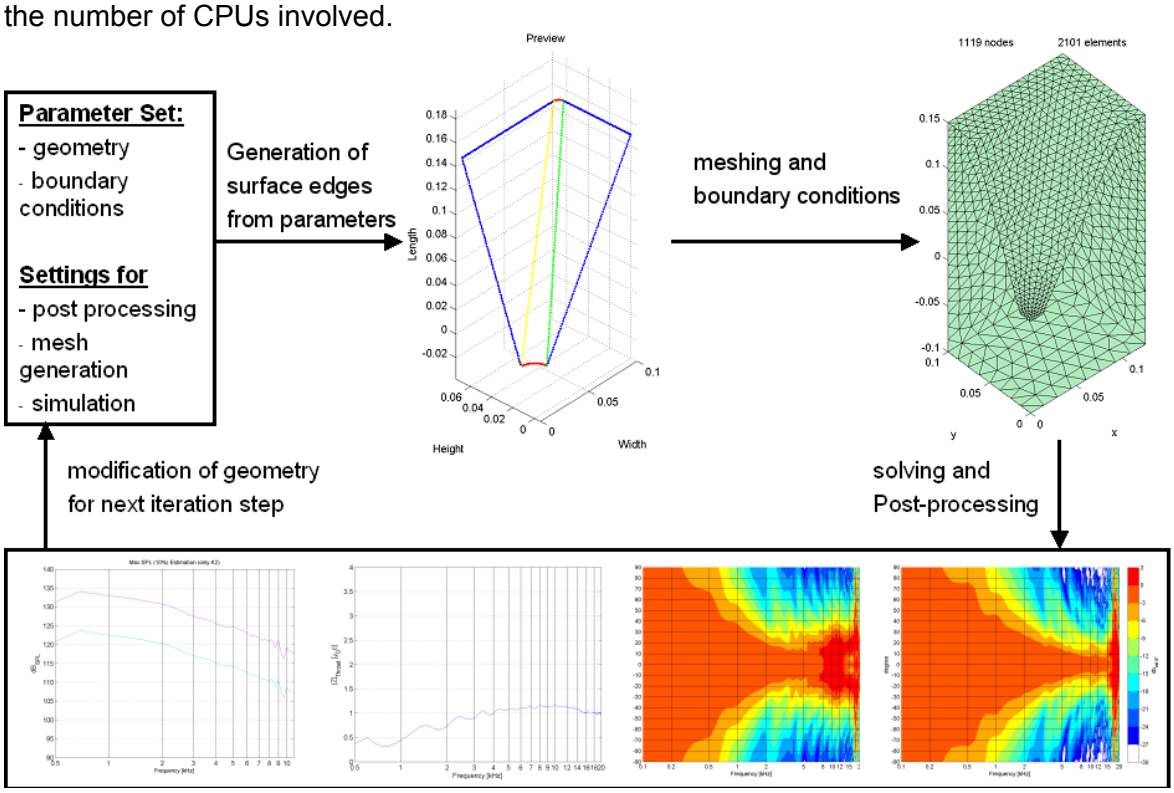


Figure 4 Horn optimisation using BEM and post-processing.

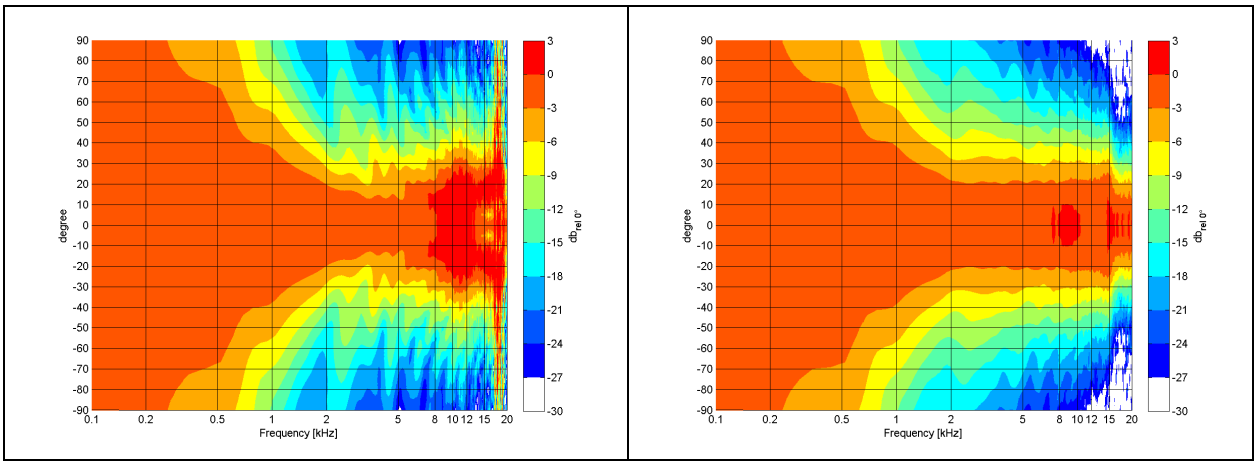


Figure 5 Horizontal directivity after the first iteration step (left) and after 14 iterations (right)

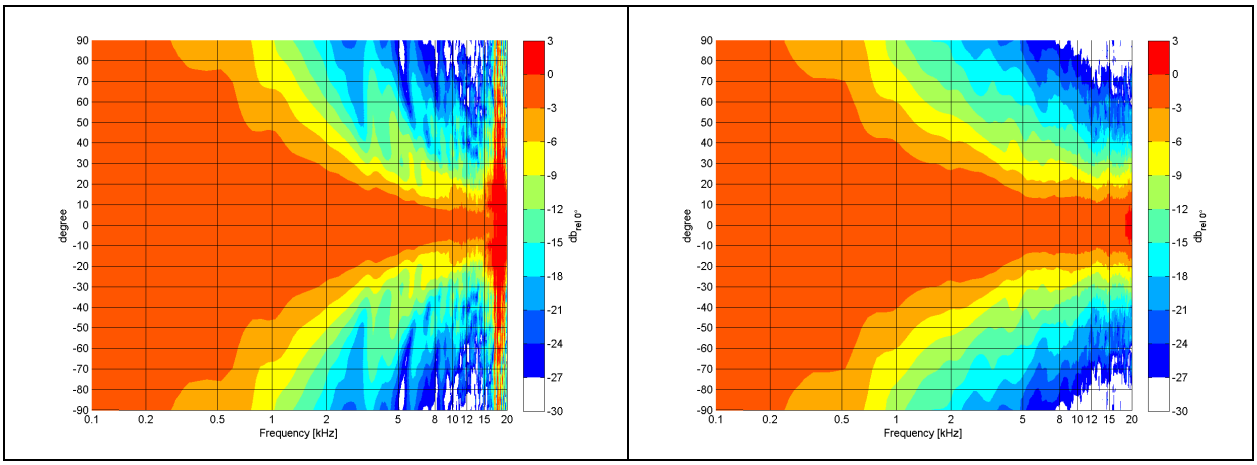


Figure 6 Vertical directivity after the first iteration step (left) and after 14 iterations (right)

Figure 5 shows the horizontal directivity of the 60°x40° horn at the beginning of horn optimization and after 14 iterations. Starting from 2 kHz, the horizontal dispersion shows a nearly perfect behaviour of 60° -6dB-beamwidth up to 15 kHz. Off-axis sound pressure decreases very smoothly and no peaks or dips are visible. Figure 6 shows the results for the vertical direction. One can clearly see that the 40°-beamwidth can not be reached over the complete frequency range using a conventional horn of this size. Anyhow, optimisation is stopped at this point and the next step, studying the interaction of woofer, horn and x-over, is to be done.

2.4 Loudspeaker system design

To study the complete system, the 8"-woofer and optimised 1"-horn have to be combined in a CAD-model. Figure 7 shows the CAD of the test-setup corresponding to Figure 3. It has to be noted that some space between horn and woofer is left which will be occupied by the horn flange and the chassis of the woofer in the "real life model". Now, a surface mesh is created and the radiation is calculated separately for all membranes. Accordingly, for this two-way cabinet, one set of results for the woofer membrane and one set of results for the 1"-horn is

obtained. Computation time for this mesh with about 3500 nodes is 50 s for each frequency using a quad-core 2.6GHz CPU. To get a general idea of the speaker behaviour, the simulation can be started with 1/3 octave frequency stepping which needs about 15 minutes in this case.

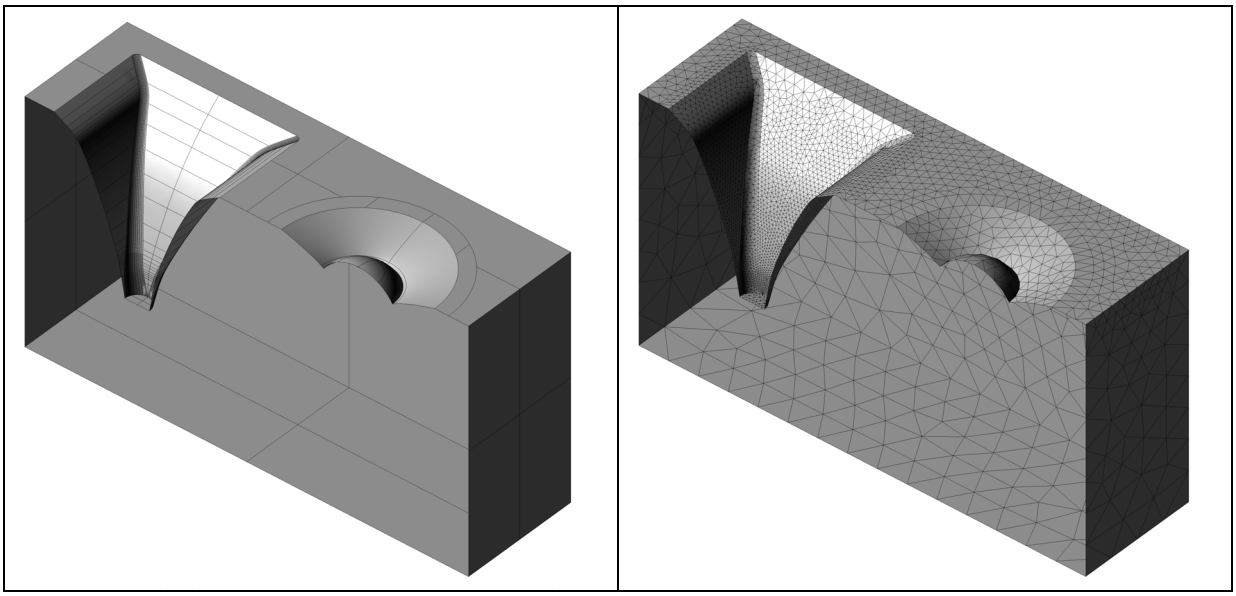
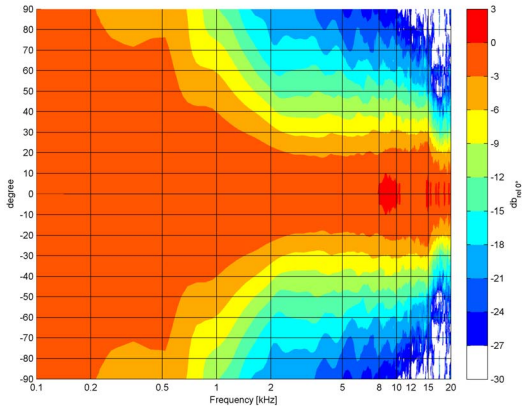
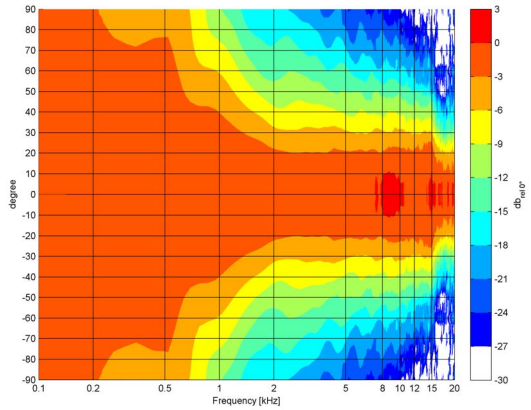


Figure 7 CAD-Model of the 8" woofer cabinet with optimised 60°x40° 1"-horn (left) and surface mesh (right). The mesh has about 3500 nodes.

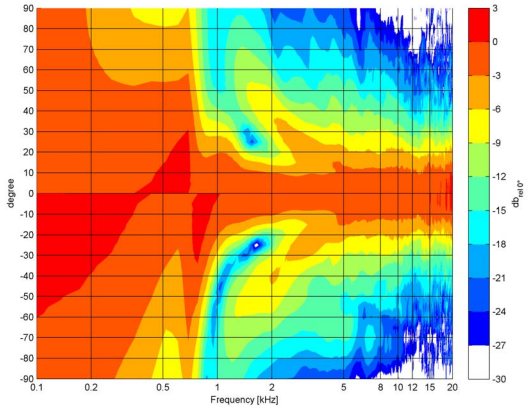
Directivity, loading impedances and also coupling impedances between the various membranes can be obtained by linear post-processing of the BEM. At this point of the development it is also possible to couple lumped element models or two-port models to the loading impedance and, thus, the sensitivity and the power limited maximum SPL for each way could be calculated very accurately [5]. The directivity of the system is obtained by multiplying the inverted frequency responses of each channel with x-over functions and superimposing the results. This procedure corresponds to using inverting FIR-filters [4] to create the EQing for the single ways of the system.
Figure 8 shows directivity and directivity index when using two different x-over functions, a Linkwitz-Riley-filter of 2. order with cut-off at 1500 Hz and a Linkwitz-Riley-filter of 8. order with cut-off at 1000 Hz. The second order set-up is a typical configuration when using passive networks or in order to increase the directivity index in the x-over region between 1 kHz and 2 kHz. In contrast, the 8. order setup would be used to get a wider coverage in the vertical direction. Now, other settings can be tested and studied to achieve a good compromise between vertical coverage and directivity index.
At this stage of loudspeaker development, one can clearly evaluate if a desired directivity can be achieved or not and which kind of x-over and cut-off frequency will be necessary. Accordingly, if a concept does not reach the directivity requirements, this would be the right stage to discard everything or to make conceptual changes. The crucial point is, that no prototype is necessary to test the basic concept and also no time consuming measurements have to be made. This is even more significant if more complex and larger systems are to be developed, as costs for prototyping can be quite high in this case.



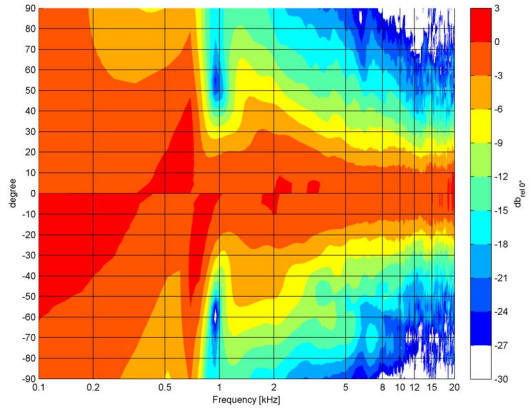
horizontal directivity with LR O2 1500 Hz



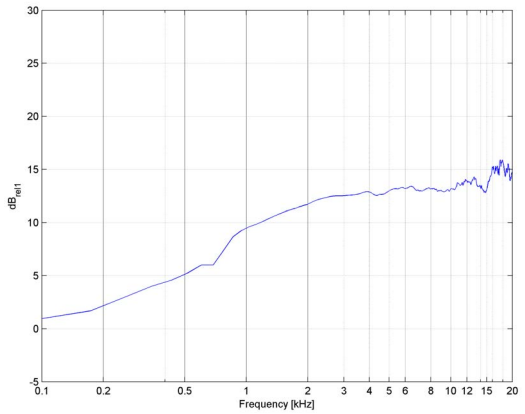
horizontal directivity with LR O8 1000 Hz



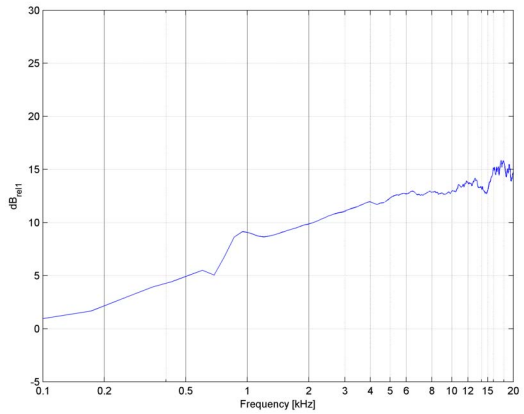
vertical directivity with LR O2 1500 Hz



vertical directivity with LR O8 1000 Hz



directivity index with LR O2 1500 Hz



directivity index with LR O8 1000 Hz

Figure 8 Directivity results for two filter settings: left column is calculated for a Linkwitz-Riley-filter of order 2 and x-over at 1500 Hz, left column is the result for a Linkwitz-Riley-filter of order 8 with cut-off at 1000 Hz.

3 Conclusions and Outlook

Numerical simulations are an indispensable tool for directivity engineering of public address loudspeakers. Optimisation of components, as waveguides, horns, diffusers and the simulation of different loudspeaker concepts including cabinet and x-over design can be done with high accuracy without building a prototype for each iteration during loudspeaker development. With increasing computational power of desktop PCs and affordable PC memory, the use of numerical methods for loudspeaker design will become more and more attractive in future.
Another very powerful application is to use post-processing results of a simulation model, like shown in Figure 7, together with electro-acoustic simulations for designing complete sound systems in large venues. This would allow to test loudspeaker models even if they only exist as rough concept in realistic venues under consideration of their interaction with room-acoustics. The investigation and verification of such extensive simulations will be an interesting task in future.

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