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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.

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

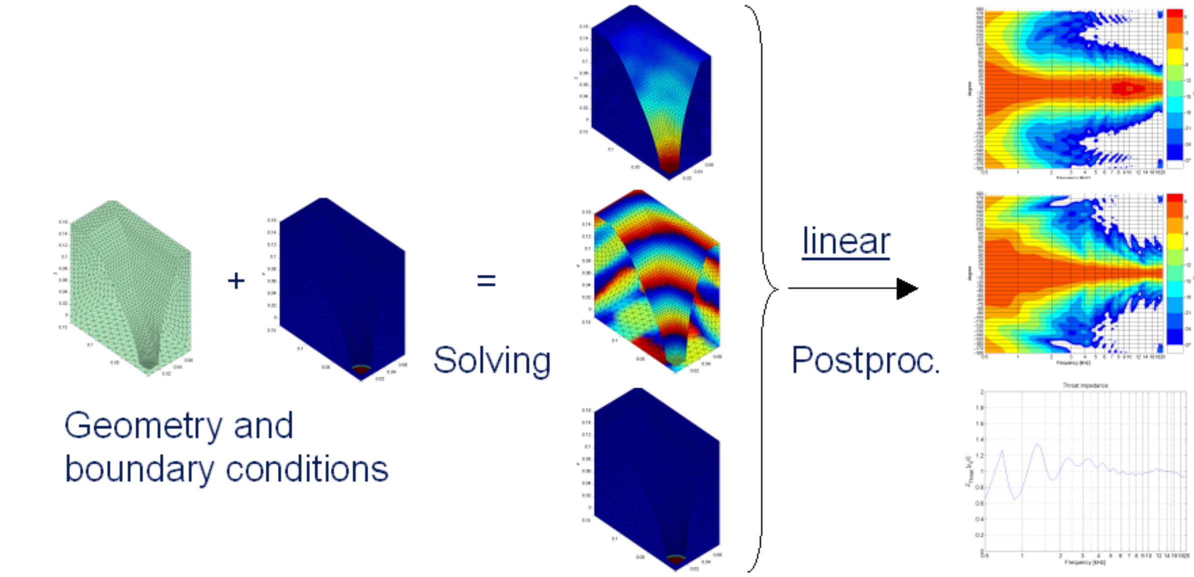

Figure 1 Basic steps of the Boundary Element Method

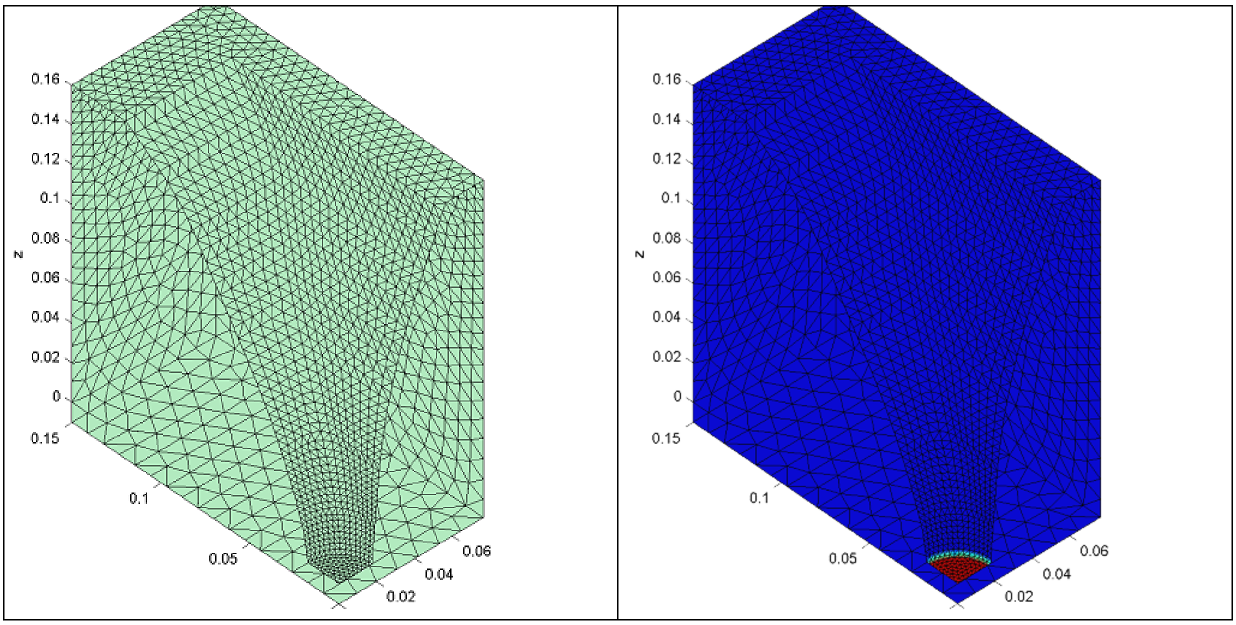

Figure 2 Quarter section of a small horn loudspeaker: geometry (left) and boundary conditions (right)

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.

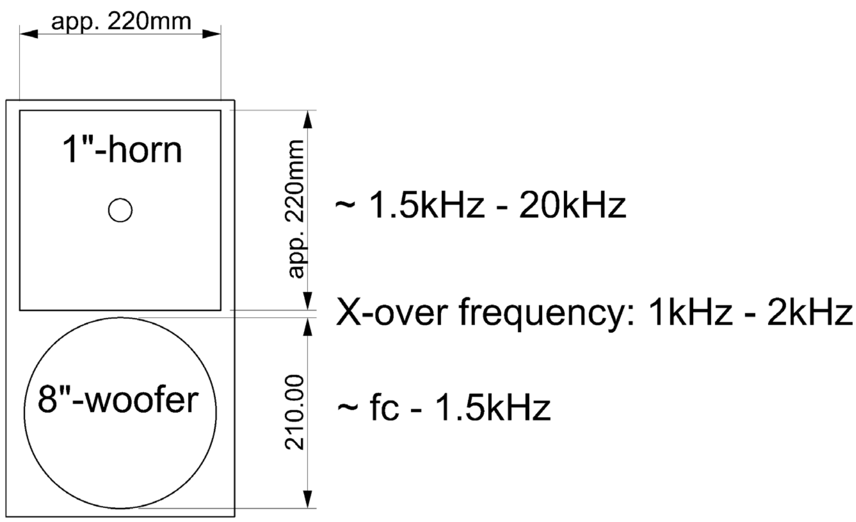

Figure 3 Approximate definition of maximum dimensions for the front baffle layout and estimated cross-over frequency.