Audio Engineering Society

Convention Paper

Presented at the 123rd Convention
2007 October 5–8 New York, NY, USA

Simulating the Directivity Behavior of Loudspeakers with Crossover Filters

Stefan Feistel[1], Wolfgang Ahnert[2], and Charles Hughes[3], and Bruce Olson[4]

[1] Ahnert Feistel Media Group, Arkonastr.45-49, 13189 Berlin, Germany
sfeistel@afmg.eu

[2] Ahnert Feistel Media Group, Arkonastr.45-49, 13189 Berlin, Germany
wahnert@ada-acousticdesign.de

[3] Excelsior Audio Design & Services, LLC, Gastonia, NC, USA
charlie@excelsior-audio.com

[4] Olson Sound Design, Brooklyn Park, MN, USA
bcolson@olsonsound.com

ABSTRACT

In previous publications the description of loudspeakers was introduced based on high-resolution data, comprising most importantly of complex directivity data for individual drivers as well as of crossover filters. In this work it is presented how this concept can be exploited to predict the directivity balloon of multi-way loudspeakers depending on the chosen crossover filters. Simple filter settings such as gain and delay and more complex IIR filters are utilized for loudspeaker measurements and simulations, results are compared and discussed. In addition advice is given how measurements should be made particularly regarding active and passive loudspeaker systems.

1. INTRODUCTION

Over the last decade digitally controlled loudspeaker systems became an industry-wide standard for professional installations and live sound applications. Due to the abundance and availability of fast and reliable computer components, digital signal processors (DSP) are used nowadays as a means to complement or replace analog networks for tapering and filtering. Today this approach is not only widely accepted but it also offers more flexibility and better control in various aspects. In particular the directivity behavior of multi-way loudspeaker systems, column loudspeakers and similar devices can be changed quite significantly with specific filters applied to the individual transducers. Modern DSP controllers make this technology much more accessible to the end user both from a costeffective point of view and also from a usability point of view.

However, until recently acoustic prediction software packages were principally not able to image this movement towards dynamic and adjustable loudspeaker radiation properties. The authors addressed this very essential problem in detail in previous publications ([1], [2]) and solutions in form of a generalized loudspeaker description format were proposed. In particular the introduced new Generic Loudspeaker Library (GLL) data format can handle active and passive loudspeaker systems including explicitly defined filter settings.

Based on the GLL format, this work presents the new concept of predicting the directivity behavior of multi-way loudspeakers depending on the chosen crossover filters. Until today, loudspeakers were often designed mainly looking at their on-axis frequency response and their opening angle. Seldom were 3D verification measurements made until the design was finished. Later on, problems in the coverage pattern of the device may have been found but they could not be solved easily. In contrast, the GLL format along with the newly developed EASE SpeakerLab software ([3]) allows the user to enter the location and orientation of individual transducers in the box and then specify corresponding acoustic data, such as directivity balloons, for each of them. In a second step, filter settings can be applied and the resulting overall directivity balloon can be calculated almost instantaneously. For the loudspeaker development process this function can save hours of time previously spent on repeatedly tuning filters and re-measuring the system.

In this respect it is crucial to note that while a variety of commercial and free loudspeaker design software packages already exist ([4]), the nature of EASE SpeakerLab is different. Especially the use of broadband, full-sphere balloon data for each of the transducers, defined either as impulse response or complex frequency response data is new and allows investigations at a much higher level of accuracy and information detail. To be able to assemble a GLL model of a loudspeaker system based on a nearly unlimited amount of individual sources, each with its own filter set, directivity balloon and sensitivity data, makes the software a very general modeling tool. The ability to combine boxes into clusters and line arrays in a natural way extends these modeling and prediction capabilities. Finally, facilitating the direct use of the created GLL files for the purpose of electro- and room-acoustic simulation in a software like EASE ([5]) completes the scope of the software.

In the following sections we utilize the simulation software EASE SpeakerLab as well as the measuring software EASERA ([6]) to compare and discuss measurement and prediction results for several multi-way loudspeakers. We explain the principle concepts and present some simple case studies as examples in part 2. In part 3 we investigate cases with more complex filter settings such as IIR high-pass, low-pass and parametric filters and predict and optimize the resulting radiation patterns. How loudspeaker measurements should be made to obtain reliable prediction results is discussed in part 4. Finally, we draw conclusions in part 5.

2. CONCEPTS

In this section we will give an overview about the GLL concept of modeling loudspeaker systems. Based on this we will also present some examples for the calculation of loudspeaker radiation patterns utilizing directivity data for the individual transducers as well as the applied filter settings.

2.1. Generic Loudspeaker Library

2.1.1. Data Format

The Generic Loudspeaker Library was developed to overcome decade-old limitations of conventional, tabular data formats used to describe loudspeakers. To characterize any loudspeaker by a single point source is not an adequate solution, in particular for systems with DSP-controlled radiation behavior. The GLL description language takes a more fundamental approach to create a loudspeaker model for prediction purposes. First of all it allows including mechanical and electronic properties in addition to the acoustic characteristics of the device. Accordingly, a multi-way loudspeaker is principally described by a set of sources, each with its own directivity balloon data and sensitivity.

Like in the real world, the format specifies how the electric inputs of the loudspeaker are linked to the acoustic outputs. This filter network, whether it is an analogue network or a set of filters realized by a DSP controller, can be entered as a group of IIR or FIR filter curves in matrix form, meaning for each link between an input and an output. The format also defines threedimensionally where the acoustic sources are located relative to each other and relative to the enclosure as well as how they are oriented (Fig. 1). Naturally this information must be aligned to the circumstances under which the sources have been measured.

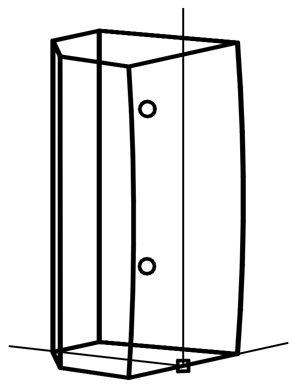

Fig. 1. 3D-Model for a two-way loudspeaker in EASE SpeakerLab, circles indicate the locations of the acoustic sources.

It is important to note that the GLL data files are built in a two-step process. First a text-based configuration file is created by the designer or manufacturer of the loudspeaker. It contains all of the essential information outlined above but also defines which parameters can be viewed or changed by the end user. After compilation of the GLL file the user can view the data and manipulate those mechanical and electronic settings that have been made available to him. As an example, for an active two-way loudspeaker the user would be able to configure the filter settings for each way individually, while for a passive two-way system he would only be able to adjust the filters for the single input.

An important feature of great practical value is the possibility to directly exchange filter settings between prediction software and DSP control software. Thus the designer of a sound reinforcement system can prepare crossover and equalizer settings before the installation. Later on, the installer of the system can start from this data and fine-tune the setup on-site. Vice versa, filter data can also be transferred from a DSP controlled loudspeaker system into the simulation software if trouble-shooting is needed and the accessibility of the venue is limited.

2.1.2. Prediction

As described earlier, the loudspeaker model is based on a set of point sources, each with its own radiation characteristics. To calculate the acoustic output of the whole ensemble we perform a complex summation that includes the filter curves. The complex pressure of a point source can be expressed by ([7]):

$$\tilde{p}(\vec{r}) = \frac{\tilde{A}}{|\vec{r}|} \exp\left[-j\left(\vec{r}\vec{k}\right)\right] \qquad (1)$$

with $\vec{r}$ being the location of interest, $\vec{k}$ being the wave vector and $\tilde{A}$ representing the angle- and frequency-dependent complex correction (magnitude and phase) for the particular source. To calculate the response at a defined location the pressure contributions of all sources i are summed in a complex manner:

$$\tilde{p}_{Sum}(\vec{r}) = \sum_i \tilde{p}_i\left(\vec{r}\right) \qquad (2)$$

Filters can be included in a straight-forward way by adding their complex transfer function $\tilde{h}_i$ to the contribution of each source:

$$\tilde{p}_{Sum}(\vec{r}) = \sum_i \tilde{h}_i \tilde{p}_i\left(\vec{r}\right) \qquad (3)$$

We state that the prediction model used here does not take into account diffraction effects or shadowing.

2.1.3. Comparison Measurements

The GLL model provides the means to investigate the performance of a complex loudspeaker system directly and quickly in the software domain. For that, the model has to be created with a sufficient degree of detail and complexity to be satisfyingly accurate. To ensure the validity of the GLL model test measurements should be made and compared to the prediction results. We will make use of this method in most of the next sections.

In particular, we will use on-axis frequency response measurements and balloon measurements to compare measurement and simulation. All of the loudspeaker measurements have been made in the approximate far field of the device, propagation effects and other influences have been compensated for. The point of rotation (POR) was chosen to be either at the location of the individual sources or at a point central to the device, such as the port. More details of this procedure are given in part 4.

Our comparisons in this work are based mainly on frequency response data, mostly smoothed to 1/24th octave bandwidth, as well as on vertical and horizontal polar data, derived from full-sphere balloon measurements and usually smoothed to 1/3rd octave bandwidth. All polar plots are scaled to a radius of 40 dB and follow common layout conventions. The on-axis direction of the loudspeaker is denoted by 0°, a value of 90° represents the upward direction for the vertical polar plots and the left-hand direction for the horizontal polar plots, when looking out of the device. Other typical plots include frequency-dependent directivity index and beamwidth as well as mappings of the vertical radiation as a function of frequency and angle.

2.2. Simple Crossover Modeling

2.2.1. On-Axis Response

To illustrate the relationship given by Eq. 3 we now discuss the on-axis response of a simple two-way loudspeaker as an example, namely a Peavey SP-1G. For this case, Eq. 3 contains 4 unknown entities, the pressure contribution of horn and woofer as well as the corresponding low-pass and high-pass filter. Measurements for all four curves are given in Figures 2a-b. The results expressed by Eq. 3 can now be compared with a full-range measurement of the two sources with the filters in place. Figure 2c shows a good agreement of the predicted and the measured on-axis full-range frequency response.

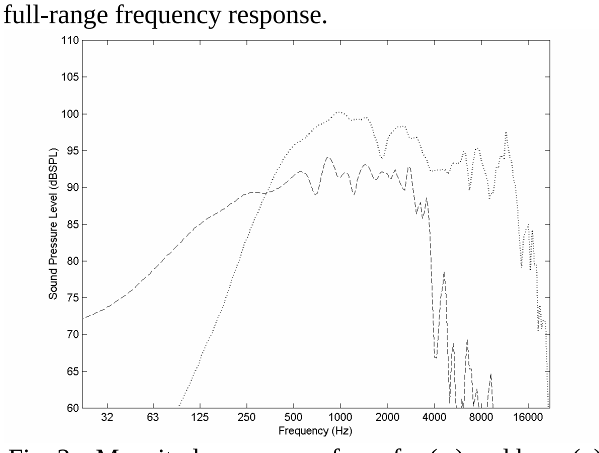

Fig. 2a. Magnitude response of woofer (--) and horn (..) measured without filtering, at 1/24th octave bandwidth.

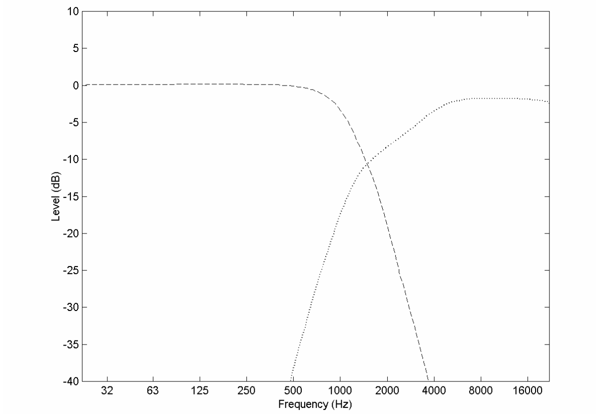

Fig. 2b. Measured magnitude response of low-pass filter applied to woofer (--) and high-pass filter applied to horn (..).

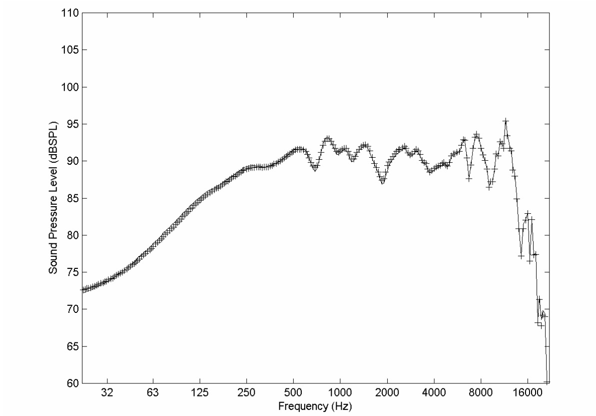

Fig. 2c. Magnitude response for full-range measurement (+) and response reconstructed from individual measurements (-), at 1/24th octave bandwidth.

2.2.2. Directivity

As a second step, we want to look at a more detailed example, namely the change of the vertical directivity pattern of a two-way loudspeaker by adjusting filter gain and delay. For this study we use a PNX121 from Renkus-Heinz, consisting of a horn and a woofer which are normally separated at a frequency of about 1.6 kHz. We show that by attenuating the horn versus the woofer the crossover frequency is increased, while by delaying the woofer against the horn the main lobe is tilted in the vertical domain.

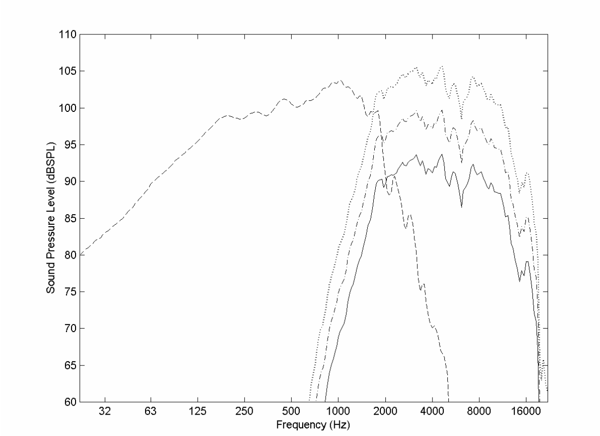

Fig. 3. Magnitude response of the woofer (--), horn (..), horn attenuated by 6 dB (-.-) and horn attenuated by 12 dB (-).

Figure 3 shows the effect of attenuating the horn by 6 dB or by 12 dB, namely that the crossover region shifts toward higher frequencies, for these particular attenuation values to 2 kHz. Figures 4a-d depict both predicted and measured results for the vertical polar data. It can be seen that interference effects due to the two transducers interacting at similar pressure amplitudes shift from 1.6 kHz to 2 kHz. Except for some small differences at angles where horn and woofer almost cancel each other out (in this case especially at an angle of +15°), the calculated and measured polar data match very well. An example for a direct comparison between measurement and calculation is given in Figure 4e.

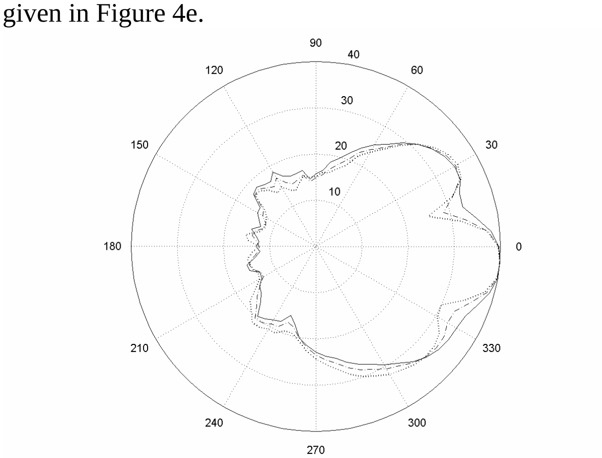

Fig. 4a. Predictions for the undamped horn (..), the horn attenuated by 6 dB (-.-) and the horn attenuated by 12 dB (-), 1600 Hz at 1/3rd octave bandwidth.

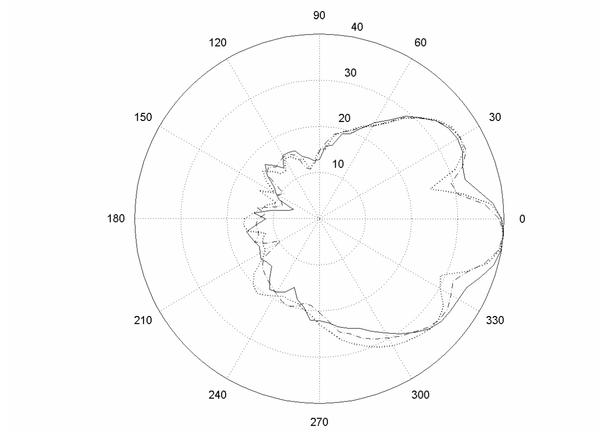

Fig. 4b. Measurements for the undamped horn (..), the horn attenuated by 6 dB (-.-) and the horn attenuated by 12 dB (-), 1600 Hz at 1/3rd octave bandwidth.

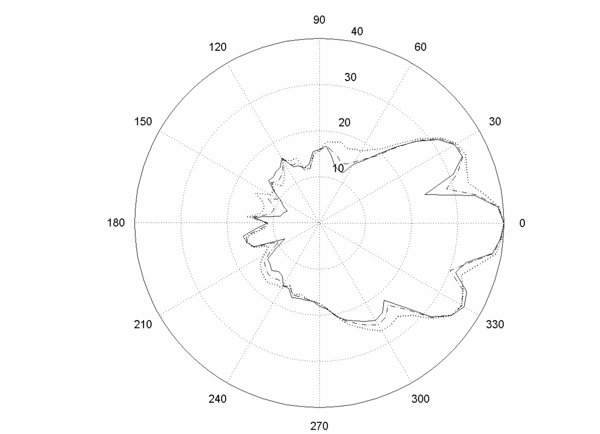

Fig. 4c. Predictions for the undamped horn (..), the horn attenuated by 6 dB (-.-) and the horn attenuated by 12 dB (-), 2000 Hz at 1/3rd octave bandwidth.

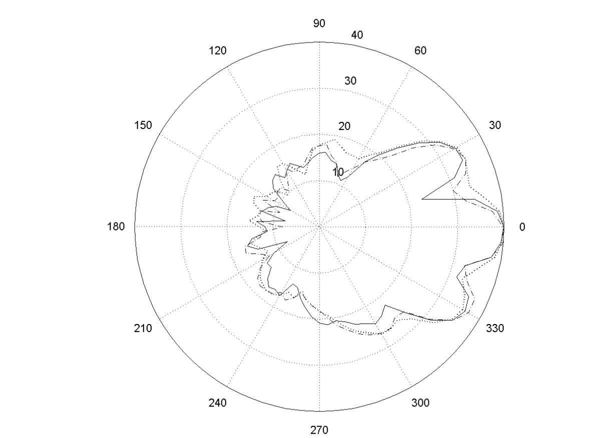

Fig. 4d. Measurements for the undamped horn (..), the horn attenuated by 6 dB (-.-) and the horn attenuated by 12 dB (-), 2000 Hz at 1/3rd octave bandwidth.

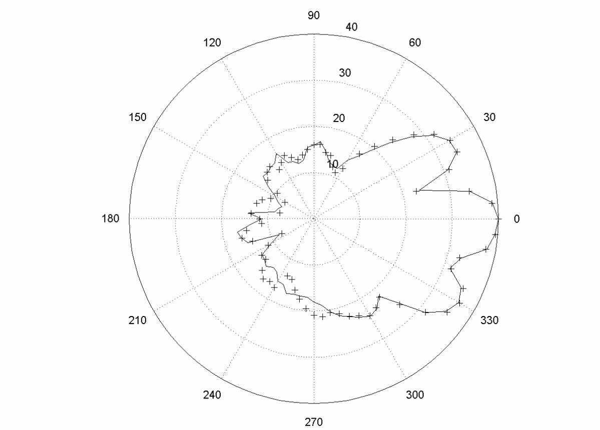

Fig. 4e. Prediction (-) and measurement (+) for the horn attenuated by 12 dB, 2000 Hz at 1/3rd octave bandwidth.

Figures 5a-b display the vertical polars for prediction and measurement of the same two-way system, but instead of attenuating the HF unit, different delays were applied to the LF unit, namely 0.167 ms and 0.334 ms. At the crossover frequency, the system consists of two sources of equivalent strength and can be considered as a very small line array. In this picture the delays correspond to down-steering angles of approximately 10° and 20°, which can also be identified in the graph. Measured and calculated polars agree very well for this case, Figure 5c shows the direct comparison for the woofer delayed by 0.334 ms.

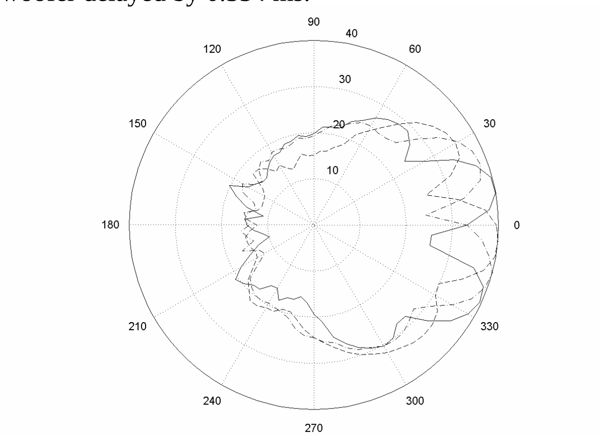

Fig. 5a Predictions for the undelayed woofer (--), the woofer delayed by 0.167 ms (-.-) and the woofer delayed by 0.334 ms (-), 1600Hz at 1/3rd octave bandwidth.

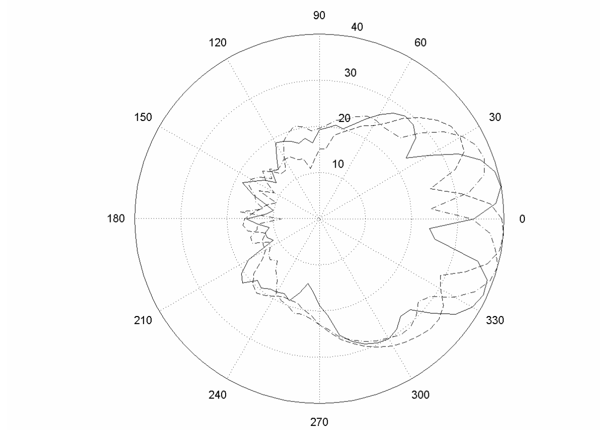

Fig. 5b. Measurements for the undelayed woofer (--), the woofer delayed by 0.167 ms (-.-) and the woofer delayed by 0.334 ms (-), 1600 Hz at 1/3rd octave bandwidth.

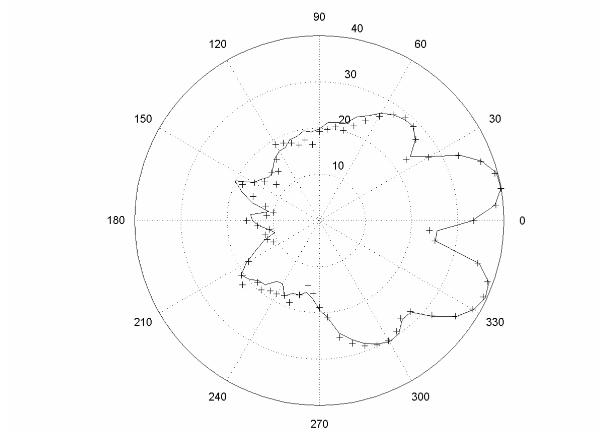

Fig. 5c. Prediction (-) and measurement (+) for the woofer delayed by 0.334 ms, 1600 Hz at 1/3rd octave bandwidth.

2.3. Conclusions

After introducing the GLL concept and the mathematical model for calculations in this respect, we have investigated the simulation results for two different two-way loudspeaker models based on the measurements of the individual transducers and of the filter curves. We summarize that the predicted directivity characteristics for the whole loudspeaker match very well with the full-range measurements. Accordingly, Eq. 3 can be seen as a viable means in practice to calculate the effects of applying different filter settings to a multi-way loudspeaker.

3. CROSSOVER DESIGN

In this part we present the central idea of this work. Essentially, we demonstrate how software based on GLL data can be used to predict the entire sound pressure field radiated by a multi-way loudspeaker depending on the chosen filter settings. Starting with an initial configuration and the corresponding measurement data, the simulation software can then facilitate the optimization of the loudspeaker system in an entirely virtual way. Different filter settings can be applied in the software domain and their effect on the loudspeaker performance can be calculated and viewed immediately. Consequently, we show that an optimal configuration can be determined directly without re-measuring the device many times ([12]).

3.1. Simple Multi-Way System

As a first step we now consider a two-way loudspeaker and extend the introduced concept to more complex filter settings. For this purpose, the SG/SGX 151 from Renkus-Heinz was used as an example (Model in Fig. 1). The same model was used in two different configurations, namely as a passive loudspeaker and as an active loudspeaker. The passive setup utilizes an analog filter network that is built into the loudspeaker box. Its crossover frequency is located at about 2 kHz. The active setup was realized using an external DSP controller with the crossover frequency approximately at 1600 Hz. For both setups the transfer functions of the filters have been measured. Their magnitude is shown in Figure 6a-b.

After that, full-range balloon measurements have been made and compared with the prediction that utilizes the individual sources and the crossover filters. The results are shown in Figures 7a-b. Clearly, the agreement is very good. The simulation of the full-range loudspeaker is capable of imaging the real full-range system for different filter settings, both using analog filter networks and DSP-implemented filters. In addition to that, figures 8a-b show the directivity index as derived from the full-sphere balloon data for both measurement and calculation results. This underlines that the introduced approach can be utilized to predict characteristic figures of loudspeakers as well.

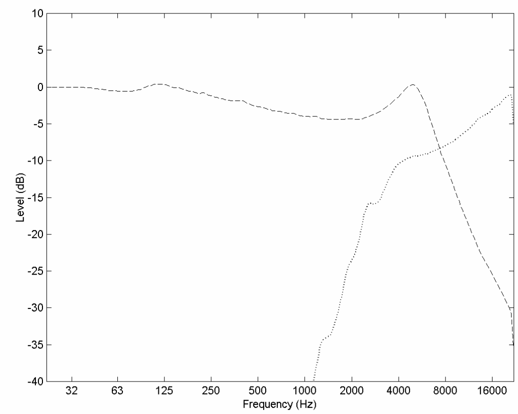

Fig. 6a. Magnitude response of crossover filters for horn and woofer of the passive loudspeaker.

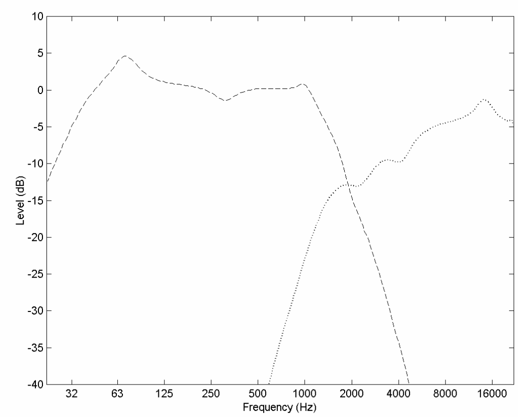

Fig. 6b. Magnitude response of crossover filters for horn and woofer of the active loudspeaker.

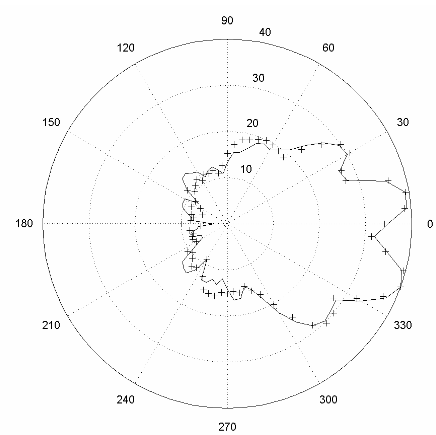

Fig. 7a. Prediction (-) and measurement (+) for the passive loudspeaker, vertical polar for 2000 Hz at 1/3rd octave bandwidth.

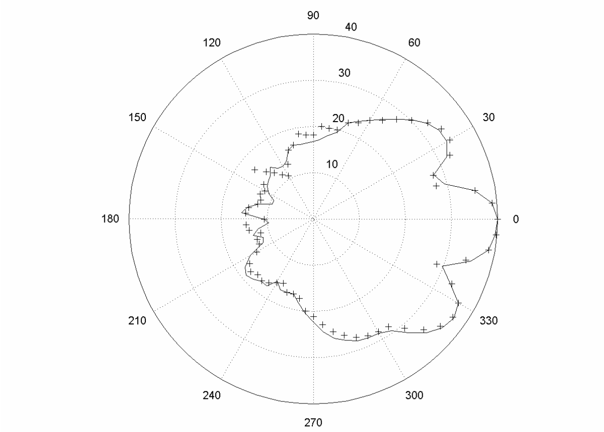

Fig. 7b. Prediction (-) and measurement (+) for the active loudspeaker, vertical polar for 1600 Hz at 1/3rd octave bandwidth.

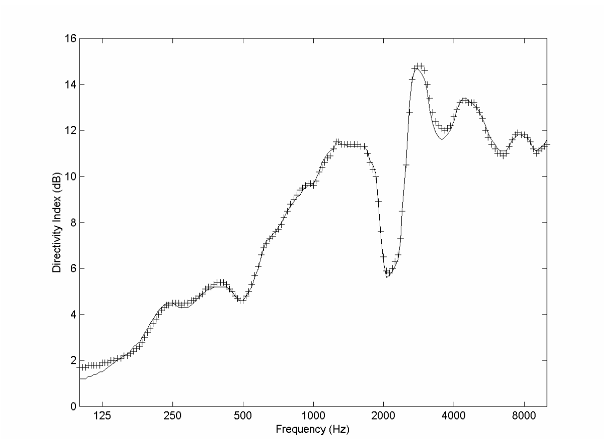

Fig. 8a. Prediction (-) and measurement (+) for the passive loudspeaker, directivity index at 1/3rd octave bandwidth.

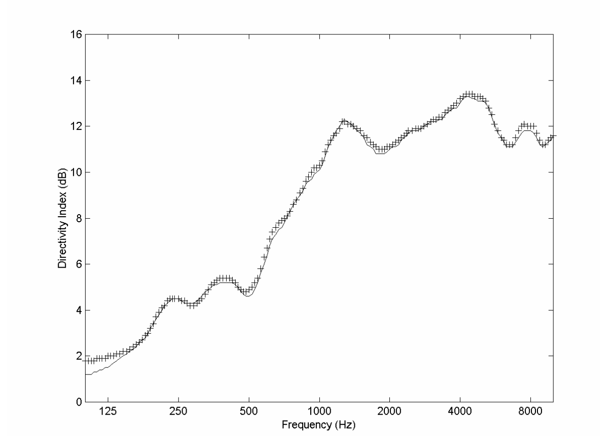

Fig. 8b. Prediction (-) and measurement (+) for the active loudspeaker, directivity index at 1/3rd octave bandwidth.

It should be noted that the small deviations towards the lower frequencies may be caused by the increased influence of the time windowing that was slightly different for the full-range and for the woofer measurement. We also note that both filter settings are of preliminary nature as the manufacturer indicated.

3.2. Directivity Optimization

3.2.1. Overview

The process of directivity optimization of a loudspeaker system using GLL prediction is not all that different from conventional methods currently being used in the industry. One of the prime differentiating factors is that once the individual source (transducer) directivity balloons have been measured the full range system directivity is calculated. Any arbitrary filter, gain or delay may be applied to each source individually and the system directivity recalculated. Therefore, the time consuming process of measuring and re-measuring the system directivity can be eliminated during the design phase.

3.2.2. Optimization Steps

A 2-way loudspeaker system from TCS Audio (TM112) comprised of a 12 inch woofer and a similar sized horn is used to demonstrate a typical optimization process. The layout of this loudspeaker system places the horn above the woofer, yielding horizontal symmetry and vertical asymmetry. A GLL model was created with the measured directivity balloons of each individual source. This GLL was then used to apply crossover and equalization filters to these sources. As an example for the software realization, Figures 17a and 17b in the appendix show screenshots from EASE SpeakerLab and its implementation of the TM112 GLL model.

A cursory look at the vertical and horizontal beamwidth plots of the LF and HF sections shows that an acoustical crossover in the 1 – 2 kHz region should work well. The appropriate signal delay was applied to the LF section and constant directivity horn EQ to the HF section. Fourth order Linkwitz-Riley LP and HP filters at 1.6 kHz were applied to the LF and HF sections, respectively. The effects of this filtering are shown in Figures 9a-c.

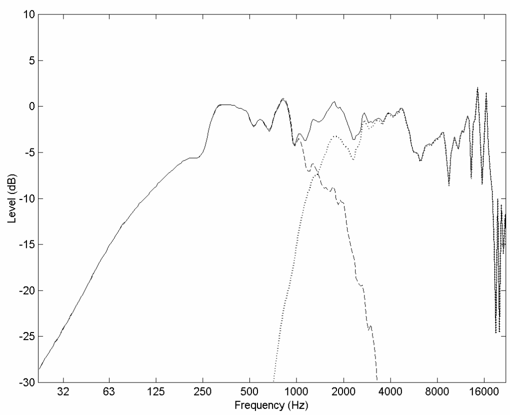

Fig. 9a. Normalized magnitude response of woofer (--), horn (..) and combined system (-) using initial Linkwitz-Riley filtering.

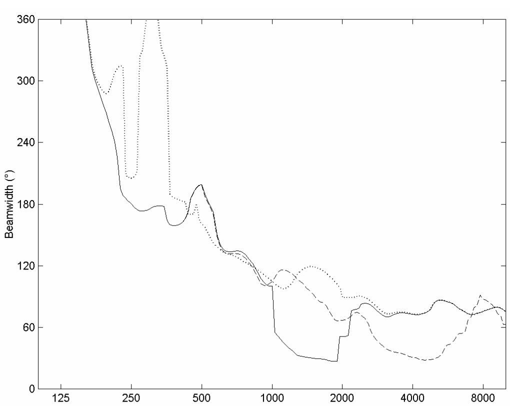

Fig. 9b. Vertical beamwidth of woofer (--), horn (..) and combined system (-) using initial Linkwitz-Riley filtering, at 1/3rd octave bandwidth.

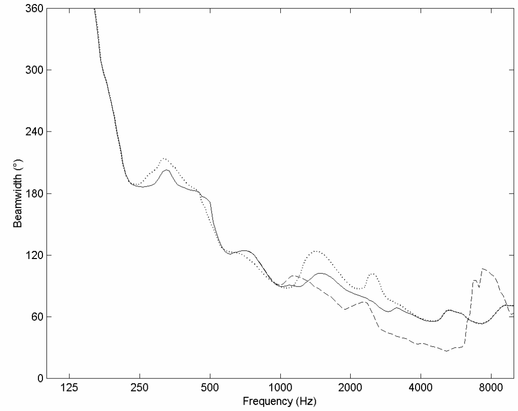

Fig. 9c. Horizontal beamwidth of woofer (--), horn (..) and combined system (-) using initial Linkwitz-Riley filtering, at 1/3rd octave bandwidth.

While the on-axis magnitude response and the horizontal beamwidth are as expected, it is easily seen that the vertical beamwidth is far from what would be desired. Note the significant narrowing from 1kHz to 2kHz shown in Figure 9b. This is partially a consequence of the vertical displacement of the LF and HF sources.

By taking into account the acoustical response of the LF and HF sources (both magnitude and phase), more appropriate LP and HP filter functions were selected. These new filter functions better complement the acoustic response of the transducers to yield a better overall system response as shown in Figures 10a-c. The new filters were a fourth order Butterworth LP at 1.6 kHz and a fifth order Butterworth HP at 2.0 kHz. Some additional minor equalization was also employed.

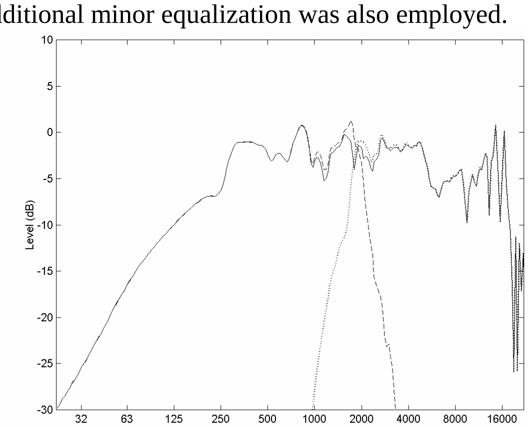

Fig. 10a. Normalized magnitude response of woofer (--), horn (..) and combined system (-) using asymmetrical Butterworth filtering

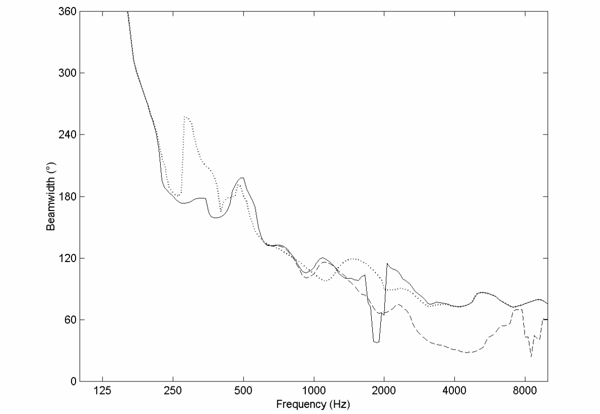

Fig. 10b. Vertical beamwidth of woofer (--), horn (..) and combined system (-) using asymmetrical Butterworth, at 1/3rd octave bandwidth.

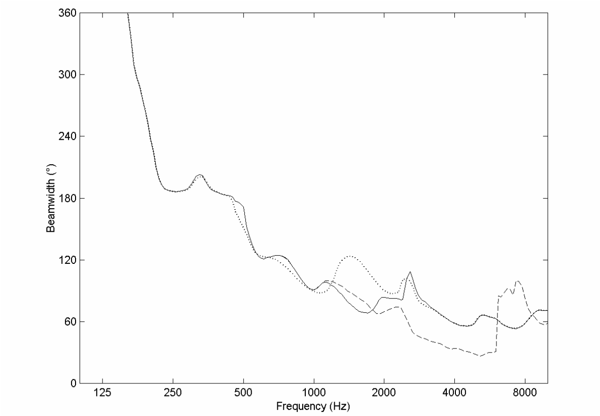

Fig. 10c. Horizontal beamwidth of woofer (--), horn (..) and combined system (-) using asymmetrical Butterworth, at 1/3rd octave bandwidth.

The vertical beamwidth through the crossover region is now much more consistent. The bandwidth over which the coverage angle decreases has been greatly narrowed; from more than one octave to approximately 0.2 octave. The horizontal beamwidth and on-axis magnitude response are still very uniform and desirable.

Note that the beamwidth of the horn is not well defined for the lower frequencies. Similarly the beamwidth for the woofer is not defined in the high frequency range. As a consequence the beamwidth plots show some artifacts for these regions.

While beamwidth plots only yield a snapshot of the coverage, they are useful for rough comparison. The real detail of the directivity in the vertical plane is seen in the vertical coverage map. A comparison of each set of filters is shown in Figures 11a-b.

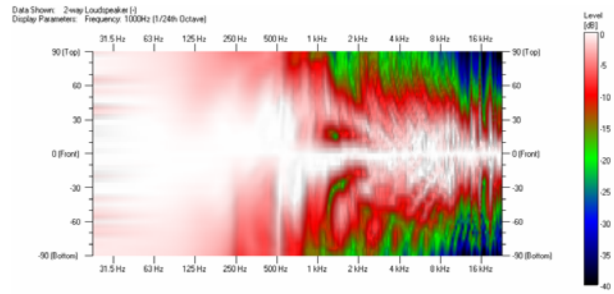

Fig. 11a. Vertical map of system response using initial Linkwitz-Riley filtering, at 1/24th octave bandwidth.

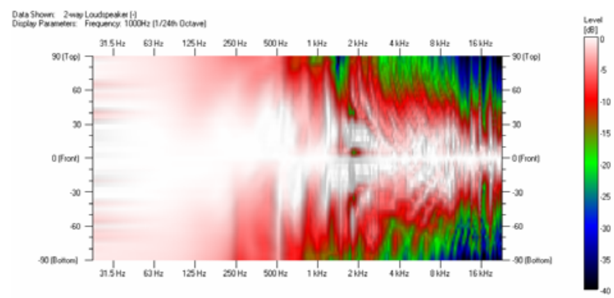

Fig. 11b. Vertical map of system response using asymmetrical Butterworth filtering, at 1/24th octave bandwidth.

3.2.3. Measured Comparison

To verify the prediction, the optimized directivity filters (asymmetrical Butterworth) were implemented on a readily available DSP unit (Biamp Audia). This DSP was used to drive two identical amplifier channels that powered the LF and HF sections of the loudspeaker system. The polar plots of the GLL and the measured system are shown in Figures 12a-b. These are at the crossover frequency of 1.9 kHz. This shows that there is good agreement between the prediction and the measurement.

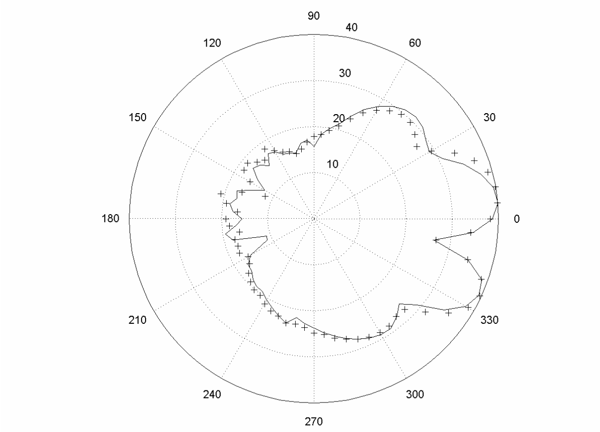

Fig. 12a. Prediction (-) and measurement (+) for system response using asymmetrical Butterworth filtering, vertical polar for 1900 Hz at 1/3rd octave bandwidth.

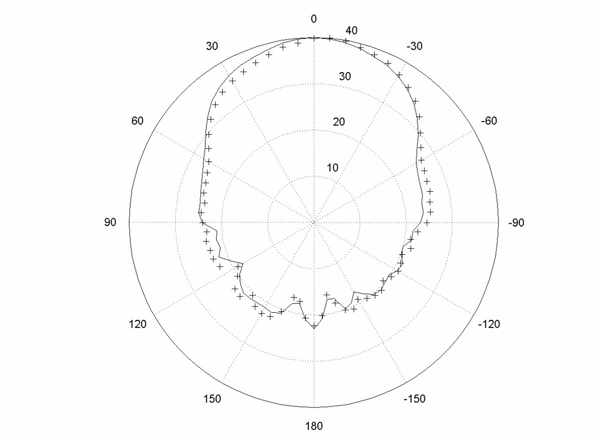

Fig. 12b. Prediction (-) and measurement (+) for system response using asymmetrical Butterworth filtering, horizontal polar for 1900 Hz at 1/3rd octave bandwidth.

3.2.4. Passive Filters

It is also quite possible to use the GLL concept to develop optimized passive crossover filters for a loudspeaker system. For this endeavor one must be careful not to apply items within the GLL that cannot be implemented passively. One primary example of this is signal delay to align LF and HF sources. Similarly, equalization filters with gain (boost) should not be used. However, attenuation equalization (cut only) can be utilized.

The best philosophy for developing a passive crossover using the GLL (or with any other method) is to keep the design simple. A passive crossover will not have buffering circuitry to isolate different filter sections as active circuitry or a DSP does. The impedance seen and caused by passive components will load adjacent components and circuit subsections.

With these limitations in mind, the same process should be followed as for the optimization with active filters. Once an acceptable system response is obtained the individual LP and HP filter transfer functions should be exported from the GLL. These transfer functions can then be imported into passive crossover modeling software for use as target functions in the development of passive filters (Figures 13a-b). They can also be opened in measurement software for comparison to the measured response of the passive circuit.

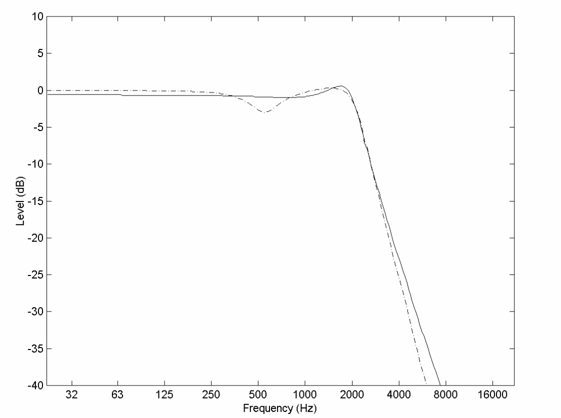

Fig. 13a. Low pass filter transfer function; GLL export (-.-) and modeled/measured passive circuit (-).

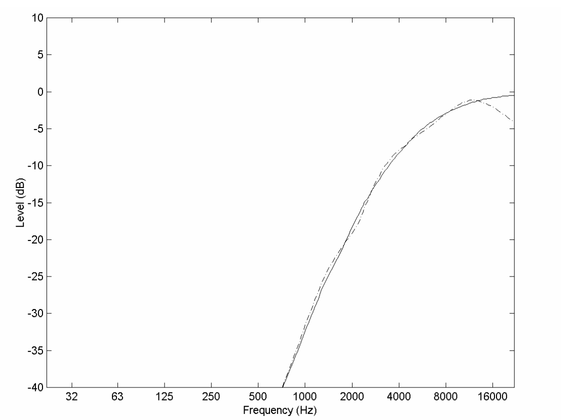

Fig. 13b. High pass filter transfer function; GLL export (-.-) and modeled/measured passive circuit (-).

After the passive circuits have been refined to sufficiently match the targets, their measured response can be imported into the GLL. This will allow the GLL to show a calculated system response using the actual passive crossover filters. A comparison of the vertical directivity using the initial GLL filters and the final passive filters is shown in Figures 14a-b. These graphs show very minor differences through the crossover region as a result of the well matched passive circuit response to the target transfer functions.

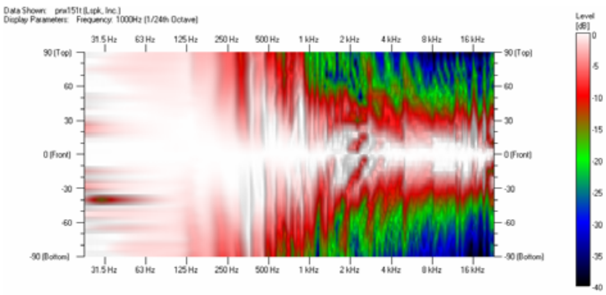

Fig. 14a. Vertical map of system response using GLL filters to develop a passive crossover.

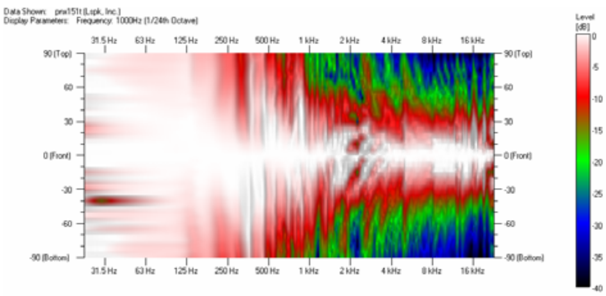

Fig. 14b. Vertical map of system response using measured passive filters.

3.3. Conclusions

We have shown that optimizing the response of a loudspeaker system using the GLL prediction method is similar to manipulating the filter functions of a conventional DSP unit. Whatever can be done in DSP can be done within the GLL. The system response can then be calculated with fairly accurate results compared to measurements.

This predictive modeling technique is equally valid for active and passive analog filter implementations.

4. LOUDSPEAKER MEASUREMENTS

To complete our investigation we want to discuss the measuring conditions and accuracy needed to successfully apply GLL-based predictions of multi-way loudspeakers.

4.1. Acquisition of Balloon Data

4.1.1. Data Resolution and Validation

When performing balloon measurements for an acoustic source, it is important to take the purpose of the data into account. For use in a prediction that combines several sources to calculate complex pressure sums, impulse response or complex frequency response data must be acquired. Magnitude data without phase data can lead to erroneous results in many cases (see e.g. [1], [8], [9]).

Equally important is the choice of the appropriate frequency and angular resolution as well as the location of the point of rotation (POR) during the measurement. Conditions for these three parameters have been derived in recent publications ([1], [8]). As shown there, all of the conditions required for a satisfyingly accurate software simulation of a loudspeaker arrangement are already met by modern measuring laboratories and available PC performance. For GLL data, which is composed mostly of several directivity balloons for the individual transducers, an angular resolution of 5° and frequency resolution of 1/24th octave is generally sufficient. Also, the choice of the POR is principally not critical when complex data is acquired and the measured system is not too large in size. Usually, the individual transducers of small and medium size devices can be measured about the same point of rotation, like the center of geometry, and thus remounting of the loudspeaker is not needed. As a rule of thumb ([8]), for an acceptable phase error at frequencies up to 8 kHz the acoustic source should be located no more than 0.25 m away from the POR when the measuring distance is about 6 m. In any case measurements should be made in the approximate far field of the device or transducer. This corresponds to a measuring distance that is approximately ten times greater than the characteristic dimension of the source.

Nevertheless the question arises to what extent the balloon measurements are reproducible. In practice, one cannot measure the full balloon data of a loudspeaker several times, because usually it already takes several hours for the acquisition of a full data set. However, there are various measures one can take to verify and ensure data validity. At first, in most measuring setups multiple on-axis measurements are made. These data can be examined and the mean deviation for the on-axis measurement can be derived. Thus severe errors due to mechanical or environmental changes during the rotation of the loudspeaker around its axis can be excluded very easily. Furthermore, on-axis measurements can be utilized by the simulation software, such as EASE SpeakerLab ([3]), to individually renormalize each set of data taken from the front to the back of the loudspeaker. By this means, small amounts of temporal drift during the series of measurements can be compensated.

It is also clear that full-range measurements of the whole ensemble can be made to verify the prediction of the loudspeaker arrangement. Mostly, on-axis measurements are already sufficient to detect significant measuring errors, as demonstrated in section 2.2.

Although models for the low-frequency behavior of loudspeakers and their enclosures already exist (for example [10]), they are still limited in their capabilities. The point source approach presented here does not account for diffraction, mutual coupling, boundary loading and other wave-based effects. Also shadowing is not taken into account. Therefore it is strongly recommended that any measurements of a loudspeaker are made close to its real-world application. That means that a transducer should be measured inside the box and a line array cabinet should be measured with its upper and lower neighbor cabinet in place if possible.

4.1.2. Measuring Environment

In noisy or slightly unstable environments noise suppression procedures can be applied to the measuring engine. Multiple time averages during the measurement can reduce the random noise floor significantly. Time windows and filters can be applied to remove systematic errors, such as side wall reflections, from the measurement of the loudspeaker's frequency response. Modern measuring platforms, such as EASERA ([6]), also allow including compensation files for the frequency response of the AD/DA hardware and the microphone as well as reference measurements to normalize the measuring process.

Air Temperature

Temperature conditions in the measuring environment represent another significant influence. A small change of the air temperature between measurements can lead to increased or decreased signal propagation times and thus to different contributions of the propagation delay to the phase data. It is of great importance for any multi-way loudspeaker measurement that one considers compensation for more than just the measuring distance at a fixed speed of sound. For example, for a two-way loudspeaker it must be ensured that the measurement of the horn and of the woofer either take place at the same air temperature or that temperature effects in the signal arrival time are removed properly. In fact, this requires recording air temperature on a regular basis. Fortunately, the GLL concept simplifies the removal of such effects as the balloon is always stored relative to the on-axis response of the measured source. As a result, only the on-axis response needs to be corrected. The effect of the air temperature ϑ (in °C) on the speed of sound c_{air} is given by ([7]):

$$c_{air} = 331.3\sqrt{1+\frac{\vartheta}{273.15}}\,\frac{m}{s} \qquad (4)$$

It is similarly important to monitor air temperature during the measurement series. Since full balloon measurements can take several hours to complete, it is imperative that the temperature remains approximately constant during the series or that changes are detected and compensated for in the balloon data. In this respect the import functions of EASE SpeakerLab allow to renormalize each measuring series from front-to-back relative to the corresponding on-axis measurement, so that temporal drift due to temperature changes during the measuring series is normally already accounted for.

It should be noted that temperature effects are more significant for greater measuring distances. While a large distance between microphone and loudspeaker is a typical requirement for far field measurements, it increases the vulnerability for environmental influences at the same time.

This large measurement distance necessitates a physically large space. The enclosed volume of these spaces can be sufficiently large that the environment should not be assumed homogenous with respect to temperature. Greater elevations within the space are often accompanied by greater temperature. This is typically not the case for changes of location within a given horizontal plane unless heating/cooling sources are present and their thermal effects not taken into account during the design of the space.

In any event, it is recommended that the temperature at or very near the source (loudspeaker) and receiver (measurement microphone) locations be monitored. If there is a significant difference, indicating the presence of a thermal gradient, its effects on the measured data should be quantified.

Thermal Gradients

Sometimes the effects of inhomogeneous media on sound propagation can be compensated ([11]). For a measurement facility employing multiple microphones, the problem posed by a thermal gradient could contaminate the measured complex data. The primary issue is the difference in arrival times at the microphones as a result of the difference in the speed of sound along the transmission paths due to the thermal gradient.

This issue can be overcome by measuring and setting the distance from the POR of the loudspeaker to each microphone acoustically. A reference loudspeaker is aimed directly at each microphone and the impulse response (IR) measured. One microphone is designated as the reference. All other microphones are positioned so that the peak of the IR at that microphone is synchronous with the IR of the reference microphone.

Another issue, usually of secondary importance, is refraction. This can become primary if the gradient is sufficiently large. This has the potential to affect not only the arrival time of the wave front at a microphone (phase data) but also the magnitude of the pressure due to the bending of the wave front as it propagates.

From ([7]), the angular refraction of the propagating wave front can be calculated using Snell's Law.

$$\frac{\sin\phi}{c(x)} = \frac{\sin\phi_0}{c_0} \qquad (5)$$

If the error in propagation due to refraction is to be confined to less than 0.5° the thermal gradient should be less than 18° F (10° C).

4.2. Filter Transfer Functions

In addition to measuring the directivity characteristics of an acoustic source or transducer as balloon data the transfer functions of the filters used also have to be determined. A closer look reveals that it must be distinguished among a number of cases.

4.2.1. Passive Filters

With respect to passive multi-way loudspeakers without configuration possibilities for their electronic properties the balloon data can be acquired with the filters in place. However, this has the disadvantage that later changes to the filter network of the box will require the full balloon to be re-measured.

Therefore, it might be more feasible to measure the individual source to obtain the unfiltered, so-called "raw" balloon data and then measure the filters independently. For passive loudspeakers with user-switchable filter settings this is required in any case. With respect to analogue filter networks care must be taken that the setup is chosen so that the filter measurements include the full impedance of the system, that is to say, also of the transducer.

Typically passive crossover filters are not terminated with a pure resistance. They are instead most often terminated with the impedance of the transducer to which they supply a signal. When measuring the transfer function of a passive network, it is imperative that the filter is measured with the proper terminating load, i.e. its transducer. For this reason it is recommended that the measurement be made with the test leads connected to the input terminals of the transducer while in its enclosure as shown in Figure 15a.

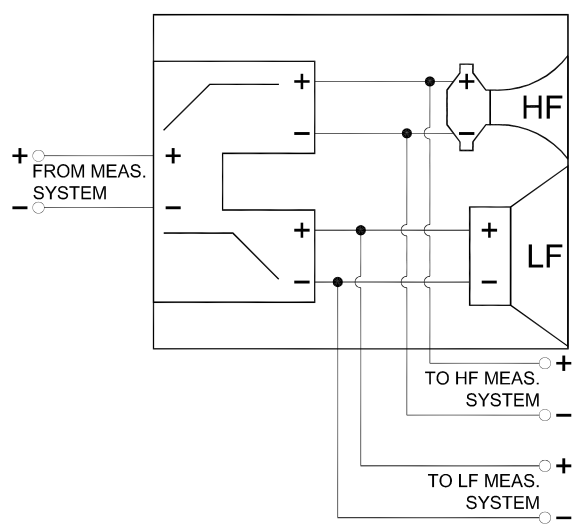

Fig. 15a. Setup for the measurement of the crossover in a two-way loudspeaker.

Of equal importance when measuring passive filter transfer functions is the conservation of polarity (within the measurement) of the signal supplied to the transducer. To accomplish this, the positive (+) test lead must be connected to the positive (+) input of the transducer and the negative (-) test lead must be connected to the negative (-) input of the transducer. If the polarity of the passive filter output is inverted (as is typical for the HF filter of some systems using second order filters as shown in Figure 15b) this connection scheme will cause a problem for single ended (unbalanced) measurement hardware since the positive (+) output of the driving amplifier will be shorted to ground of the measurement system.

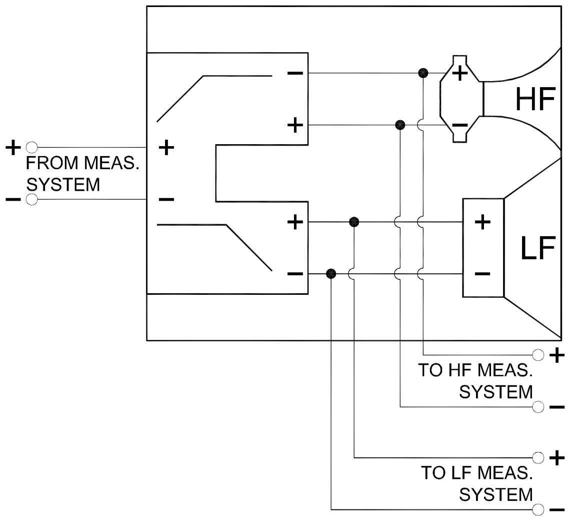

Fig. 15b. Setup for the measurement of the crossover in a two-way loudspeaker with inverted HF filter.

It is therefore recommended that only measurement systems with balanced inputs be used for this type of measurement. The non-inverting input (+) should be connected to the transducers positive (+) input, while the inverting input (-) should be connected to the transducers negative (-) input. The ground for the input to the measurement hardware should be left floating (not connected). This connection scheme will accurately capture the signal polarity presented to a transducer by a passive filter.

4.2.2. DSP Based Filters

Active loudspeaker systems, particularly those that are DSP-controlled, can be measured in a relatively straight-forward way. As an example, the full-range measurement setup of a loudspeaker may be given by Figure 16a. Its components, HF unit, LF unit, high-pass und low-pass filter, are then measured individually, see Figures 16b-e.

It is noteworthy that, while DSP-based loudspeakers often offer IIR filter selection via a software interface, implementations of these filters are not standardized. Depending on the model and on the manufacturer the transfer function of IIR filters with the same "front panel" parameters may look different from platform to platform ([13]). As a result it is recommended that the end user measure the transfer function of the DSP and apply it directly to the GLL model. Alternatively the user can make verification measurements to ensure that the DSP realization of the specified IIR filter setting is sufficiently close to the filter used in the prediction software.

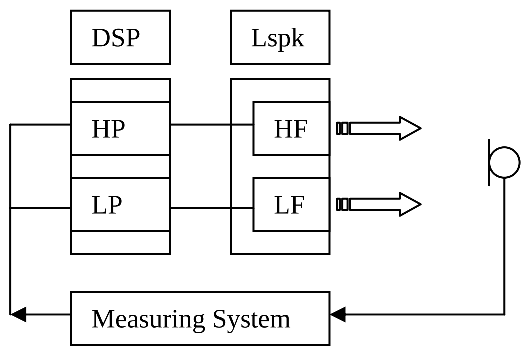

Fig. 16a. Setup for the full-range measurement of a two-way loudspeaker.

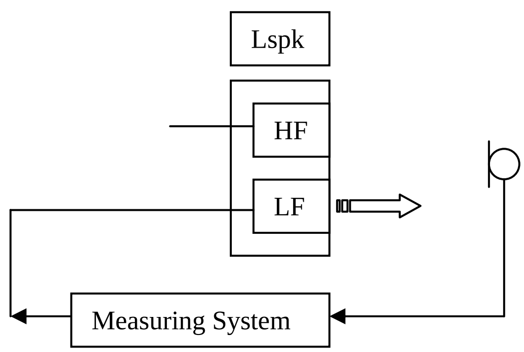

Fig. 16b. Setup for the measurement of the LF unit.

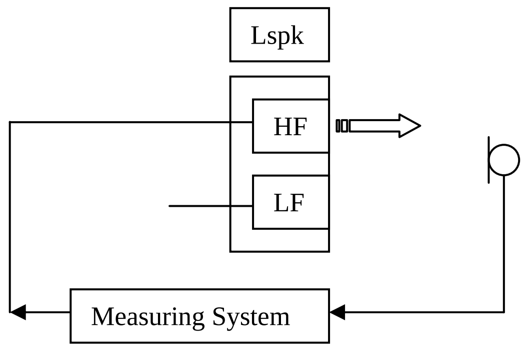

Fig. 16c. Setup for the measurement of the HF unit.

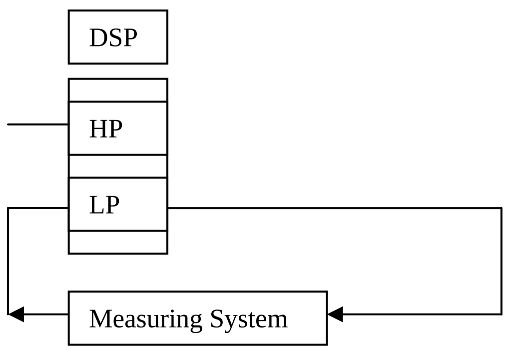

Fig. 16d. Setup for the measurement of the low-pass filter.

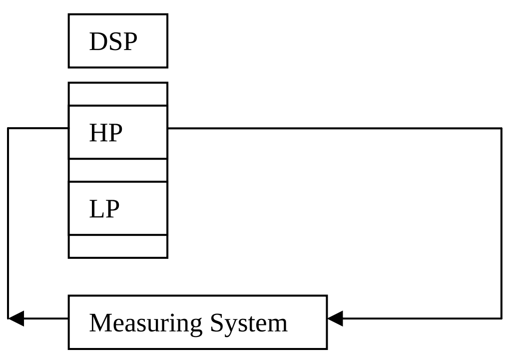

Fig. 16e. Setup for the measurement of the high-pass filter.

5. CONCLUSIONS

With this work we presented several practical applications of the recently introduced concept of the Generic Loudspeaker Library. Our research was particularly focused on the prediction of multi-way loudspeakers and the optimization of their directivity patterns in the software domain, as demonstrated here with EASE SpeakerLab. Utilizing fully-descriptive acoustic data for the individual transducers of the loudspeaker as well as being given filter settings for the individual pass bands, the software facilitates the calculation of the resulting performance in 3D. It was shown that simulation results are in good agreement with measurements.

This new and universally applicable approach simplifies the design of loudspeakers and of crossovers in specific. It can save designers and manufacturers of loudspeakers from spending many hours on manually optimizing the radiation behavior based on sequences of repeatedly measuring and tuning the device. It also allows the loudspeaker designer to detect and eliminate design mistakes without building or modifying the loudspeaker in the real world. In this regard EASE SpeakerLab can be compared to a room modeling software like EASE because it allows the simulation of what-if scenarios in an entirely virtual way. It is noteworthy that the concepts presented here can be applied similarly to other types of loudspeakers with multiple transducers, such as steered columns.

To conclude, the authors feel that with the flexible GLL description format, a prediction software like EASE SpeakerLab and adequate measuring software readily available, the newly introduced software-based crossover design methods represent a great step forward in the time-efficient development and improvement of loudspeaker models.

6. ACKNOWLEDGEMENTS

The authors wish to thank Ron Sauro for performing numerous loudspeaker measurements at the facility of NWAA Labs and the people at Renkus-Heinz for measuring and providing in-depth data for several of their loudspeaker models.

7. REFERENCES

[1] S. Feistel and W. Ahnert, “Modeling of Loudspeaker Systems Using High-Resolution Data“, *J. Audio Eng. Soc.*, vol. 55, pp. 571-597 (2007 July/August).

[2] S. Feistel, W. Ahnert, and S. Bock, “New Data Format to Describe Complex Sound Sources,” presented at the 119th Convention of the Audio Engineering Society, *J. Audio Eng. Soc.*

(Abstracts), vol. 53, pp. 1239, 1240 (2005 Dec.), convention paper 6631: GLL format specification, http://www.ada-acousticdesign.de, http://www.sda.de.

[3] EASE SpeakerLab software, http://www.ada-acousticdesign.de.

[4] LEAP software, http://www.linearx.com : LspCAD software, http://www.ijdata.com : AudioCad software, http://www.audiocad.de : SoundEasy software, http://www.interdomain.net.au/~bodzio/ : CALSOD software, http://www.sonicdesign.se.

[5] EASE software, http://www.ada-acousticdesign.de.

[6] EASERA software, http://www.sda.de.

[7] L. Kinsler, A. Frey, A. Coppens, and J. Sanders, *Fundamentals of Acoustics*, 4th ed. (Wiley, New York, 2000).

[8] S. Feistel and W. Ahnert, "The Significance of Phase Data for the Acoustic Prediction of Combinations of Sound Sources," presented at the 119th Convention of the Audio Engineering Society, *J. Audio Eng. Soc. (Abstracts)*, vol. 53, p. 1240 (2005 Dec.), convention paper 6632.

[9] W. Ahnert, S. Feistel, J. Baird, and P. Meyer, "Accurate Electroacoustic Prediction Utilizing the Complex Frequency Response of Far-Field Polar Measurements," presented at the 108th Convention of the Audio Engineering Society, *J. Audio Eng. Soc. (Abstracts)*, vol. 48, p. 357 (2000 Apr.), preprint 5129.

[10] M. Urban, C. Heil, C. Pignon, C. Combet and P. Bauman, "The Distributed Edge Dipole (DED) Model for Cabinet Diffraction Effects", *J. Audio Eng. Soc.*, vol. 52, pp. 1043-1059 (2004 October).

[11] C. Hughes, "How Accurate is Your Directivity Data?", technical white paper published October 2005; www.excelsior-audio.com.

[12] C. Hughes, "Alternative Ways of Viewing Polar Data", Synergetic Audio Concepts Tech Topics Vol. 29, No. 3 (Summer 2001); www.excelsior-audio.com.

[13] P. Brown, "DSP Comparison, Part 2", Synergetic Audio Concepts Newsletter Vol. 33, No. 4, pp. 20-23 (Fall 2005).

8. APPENDIX

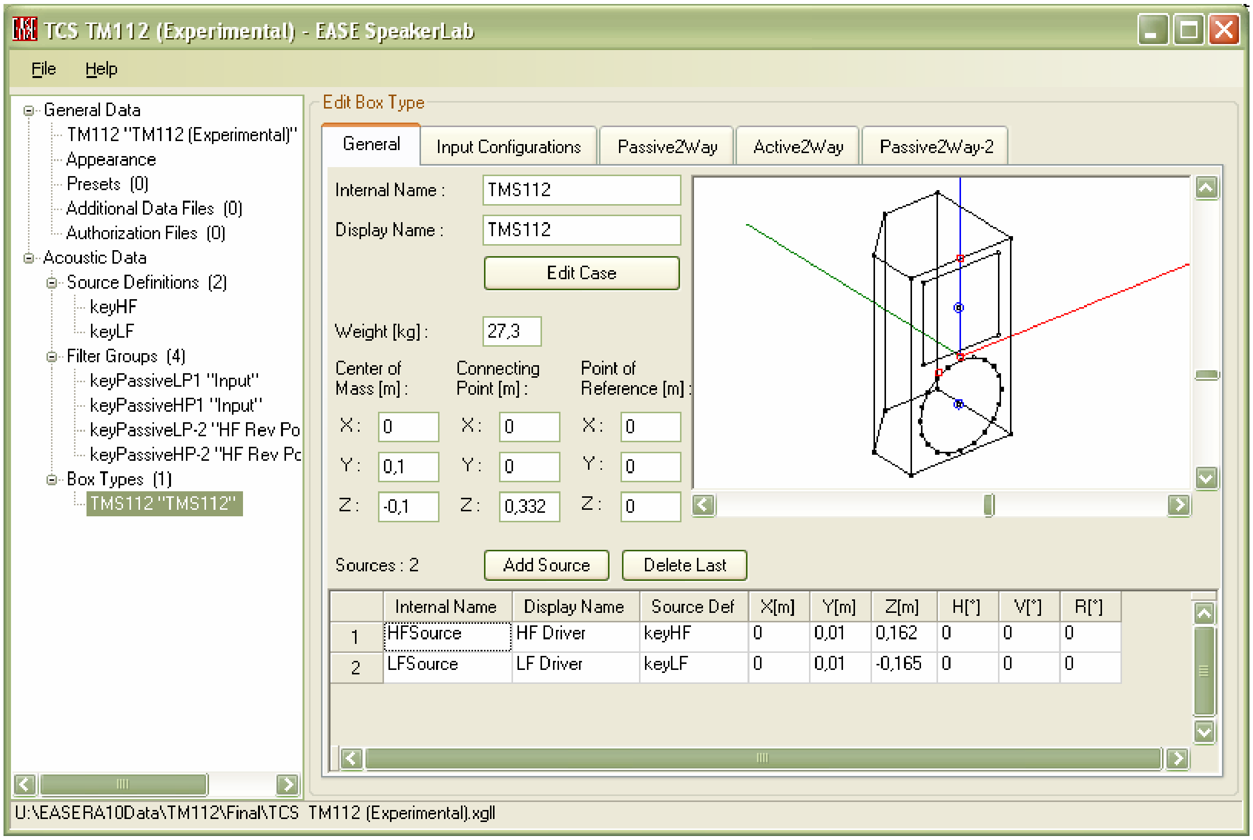

Fig. 17a. GLL model of the two-way loudspeaker as displayed in EASE SpeakerLab.

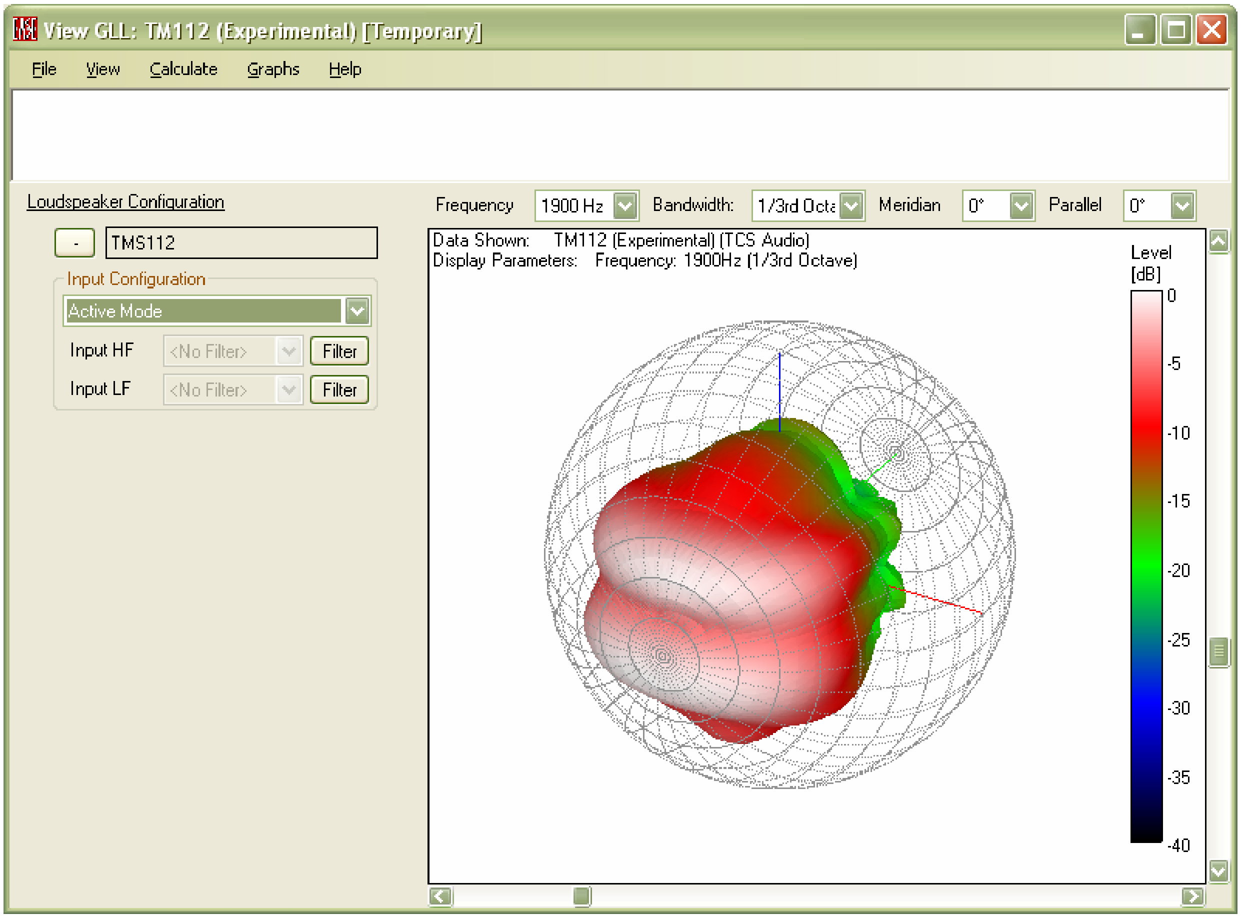

Fig. 17b. Exemplary balloon data calculated for the two-way loudspeaker as displayed in EASE SpeakerLab.