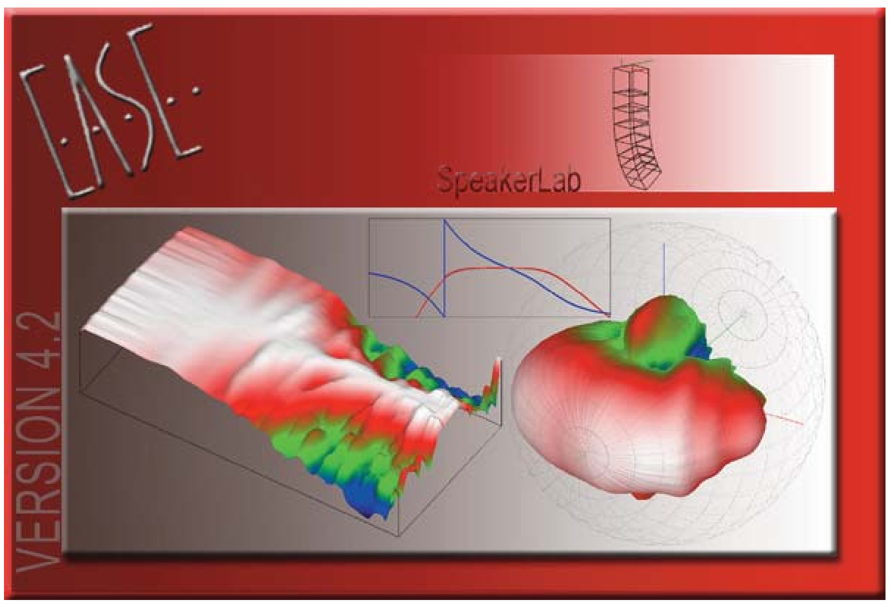
GLL (Generic Loudspeaker Library)

A New Standard for Measuring and Storing Loudspeaker Performance Data



An
AFMG (Ahnert Feistel Media Group)
Engineering White Paper

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GLL (Generic Loudspeaker Library): A New Standard for Measuring and Storing Loudspeaker Performance Data

But first, a little computer history…

EASE, the first major project of Ahnert Feistel Media Group (AFMG), was introduced in 1990. That year, IBM's PS/2 Model 65 desktop computer included a 16 MHz 80386SX processor, 2 MB of RAM, and a 60 MB SCSI hard drive. Today's IBM ThinkPad notebook computers have surpassed the predictions of Moore's Law, which states that the number of transistors on an integrated circuit for minimum component cost doubles every 24 months. They include Intel Core 2 Duo processors operating at up to 2.33 GHz, up to 4 GB of RAM and hard drives up to 160 GB.

The Generic Loudspeaker Library: A More Complete, Higher Resolution Data Format

The EASE software suite, now in version 4.2, has kept pace with the exponential expansion and acceleration of commonly available computing resources. Four years ago, it became apparent that the EASE SPK Database, the tabular data format for characterizing loudspeaker systems to be modeled in EASE, was limiting the accuracy and realism of EASE performance predictions. Since even laptops now have the processor speed and storage capacity to handle expanded data sets, the restrictions and compromises of the original data format were no longer necessary or useful.

The result of AFMG's three-year development project is known as the Generic Loudspeaker Library (GLL). This new data format is designed for modern software practices such as object-oriented programming. At the same time, it allows loudspeaker systems to be described with greater completeness, and performance measurements to be stored with higher resolution. Both factors are important in building more accurate models of sound system performance.

More Complete Descriptions of Today's More Complex Loudspeakers

Loudspeaker design has evolved rapidly since the introduction of EASE in 1990. Integrated multi-way systems, a rarity in the late 1980's and early 1990's, have become the norm. Now firmly established in the loudspeaker marketplace, the system concept has continued to expand and evolve. The first loudspeaker systems included multiple transducers, waveguides and passive filter networks in a single enclosure. Today's designs can also include power amplifiers and analog or digital signal processing – even separate amplifiers and DSP channels for each transducer.

The system concept has expanded past the single enclosure to encompass multi-enclosure arrays. Both horizontal clusters and vertical line arrays are now designed as systems. The hardware that connects the enclosures has become an integral element of these systems, since it determines the spatial relationships between separate acoustic sources. The directivity balloon of such a system is no longer fixed, but can be altered drastically by reconfiguring the array hardware, the signal processing, or both.

Clearly, both designers and installers of these systems need computer models that can accurately describe and simulate the dynamic and adjustable loudspeakers used in today's sound reinforcement systems. It is no longer enough to model every loudspeaker system or array as a single acoustic source with a fixed directivity balloon. A new data format is required to support more complex and complete modeling algorithms.

The GLL: Benefits

Better Predictions, Better Products

Loudspeakers have typically been designed by looking at the on-axis frequency response and the opening angle. 3D verification measurements have not normally been made until the design is finished. If they revealed problems in the coverage pattern of the device, those problems could not be easily solved. In contrast, the GLL format along with the newly developed EASE SpeakerLab software allows the developer to enter the location and orientation of each transducer in the box and then specify its directivity balloons and other acoustic data. In a second step, filter settings can be applied and the resulting overall directivity balloon can be calculated almost instantaneously. This can save hours of repetitive filter tuning and re-measuring.

EASE SpeakerLab uses broadband, full-sphere balloons for each transducer, defined either as impulse response or complex frequency response data. This unique and new capability allows a much higher level of accuracy and information detail. The program can assemble virtual loudspeaker models with a nearly unlimited amount of individual sources, each with its own filter set, directivity balloon and sensitivity data: this 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. When the development cycle is finished and the product is released, the GLL files can be used directly for the purpose of electro- and room-acoustic simulation in a software suite like EASE.

Better Models, Better Designs

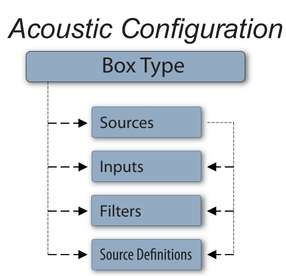
Because the GLL data object describes the modern loudspeaker system more completely and with greater resolution, it can be used to create better models of sound system performance. The working premise of EASE is that more information leads to better decisions. This has been true throughout the 17 years since the program's introduction. The GLL provides a higher quality and quantity of information to sound system designers, and their decisions will be better informed and more effective as a result of using it.

Better Systems, Better Installations

The GLL includes much information that in the past was excluded from the basic model of acoustical sources. In particular, configurable elements of the modern loudspeaker system such as rigging hardware and signal processing are now part of the data object. Measurement software such as EASERA and EASERA SysTune, and prediction software such as EASE, EASE SpeakerLab, EASE AURA, EASE Focus, etc. can now use the same data objects. This creates an opportunity to close the loop between modeling, measurement and manufacturer-specific applications that are used to configure hardware and/or DSP. System tuning will then become part of the modeling and design process, with verification and final adjustments performed during the commissioning stage of an installation.
Using GLL data objects, filter settings can be directly exchanged between prediction software such as EASE and measurement software such as EASERA or SysTune and DSP control software like RHAON, the Renkus-Heinz Audio Operations Network. The sound system designer can use EASE to prepare delay and equalizer settings before the installation. The installer of the system will start with these settings and fine-tune them on site. If trouble-shooting is needed later on, DSP data can be transferred from a DSP controlled loudspeaker system into the simulation or measurement software. The system's performance can then be analyzed and corrected without restricting the normal availability of the venue itself.

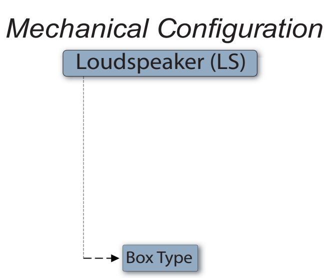
The GLL Object Hierarchy

Object-oriented programming provided a useful template to follow in designing the new format. Software objects have multiple properties and methods. The Generic Loudspeaker Library uses this template to describe both simple and complex loudspeaker systems.



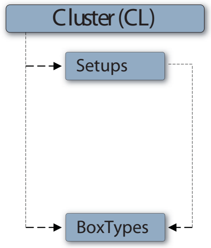
The most basic GLL object is known as a Box Type: this object defines the mechanical, acoustical and geometrical properties of a loudspeaker cabinet. Each box type in turn consists of a set of acoustic sources and input configurations. Each source is described by a position and orientation relative to the box.

The source description may include a directivity balloon along with other acoustically relevant data, such as rated bandwidth and maximum voltage. Input configurations allow the manufacturer to map filter groups to specific inputs for each box type. Filter groups specify internal crossovers, external adjustment possibilities by the user, as well as dedicated complex filters for signal processing.



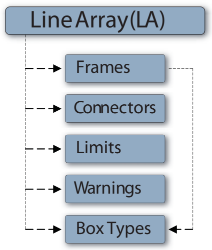
Box Types can be part of larger objects in the GLL format. The simplest of these is the familiar Loudspeaker, which is described as a Box Type. Of course the Box Type includes all the properties described above.

Mechanical Configuration



Secondly there is the loudspeaker Cluster, which is described as one or more fixed configurations or Setups of specific Box Types.

Mechanical Configuration



The most complex object in the GLL format is the line array, which is described as a configurable set of box types suspended in one or more rigging frames – the links between boxes are termed connectors. Limits and Warnings allow manufacturers to put constraints on the software object that reflect the real-world limitations of the actual line array.

Building a GLL data object is a two-step process. First the designer or manufacturer of the loudspeaker, cluster or line array creates a text-based configuration file using EASE SpeakerLab for example. It contains all of the information outlined above, and also defines which parameters can be viewed or changed by the end user. Once this GLL file has been compiled in EASE SpeakerLab, the user can view the data and manipulate any mechanical and/or electronic settings that the object makes available (or "exposes," to use a bit of computer-speak). For instance, the user might be able to configure the filter settings of an active two-way loudspeaker separately for LF and HF; the user of a passive two-way system would only be able to adjust the filters for the single input.

GLL object definitions have been developed from DLL (Dynamic Link Library) formats that have been part of EASE since v4.0. They are much more fully descriptive of the variety of modern loudspeaker systems than the EASE SPK Database format that was defined in 1990. 17 years ago, EASE SPK reduced everything to a single acoustic source in order to work within the limitations of the available processor speed and data storage. All loudspeaker systems, no matter how complex, were described as simple point sources.

This level of description (a simple point source) is now the lowest in the GLL object hierarchy; more complex objects can be described and modeled much more accurately and realistically. For instance, line arrays can be configured within EASE rather than imported as DLLs defined in manufacturer-specific software or in the manufacturer-independent EASE Focus line array configuration and aiming program. One advantage of the GLL is that both manufacturers and users of EASE can examine the specific characteristics of a data object using the free GLL Viewer.

Another advantage of the object model is that objects are extensible. For example, the Line Array object in EASE Focus software includes aiming lines for individual boxes. Future extensions could include performance at frequencies below 20 Hz and diffraction effects of line array enclosures, among many possibilities.

Higher Resolution Acoustical Sources

The GLL object model provides the means to calculate the sound pressure field and thus the directivity balloon of a complex loudspeaker system directly and quickly in software. Of course, the data object has to be created with a sufficient degree of detail and complexity: more accurate underlying data means more reliable performance predictions and sound system design decisions.

The facilities required in order to measure a loudspeaker system for the purpose of creating a GLL data object include:

— A dedicated space large enough for measurements to be taken in the approximate far field of the device under investigation

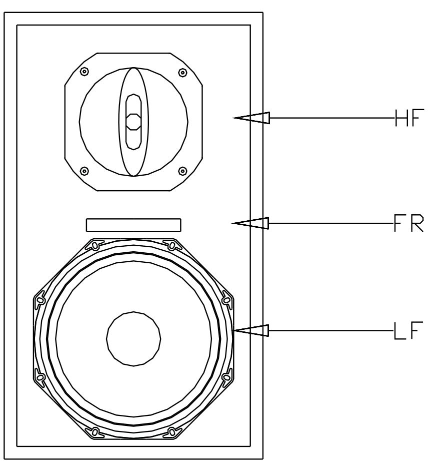
— A platform on which the loudspeaker can be mounted and rotated by known increments in two dimensions (can also be 1 dimension and a mic arc).
— At least one measurement microphone.
— A PC-based measurement system.

AFMG's research naturally uses EASERA PRO and EASE SpeakerLab. Both of these programs can process raw measurements, and provide much more detailed information about the loudspeaker assuming that the original measurements are true impulse response or complex frequency response data that include both magnitude and phase information.

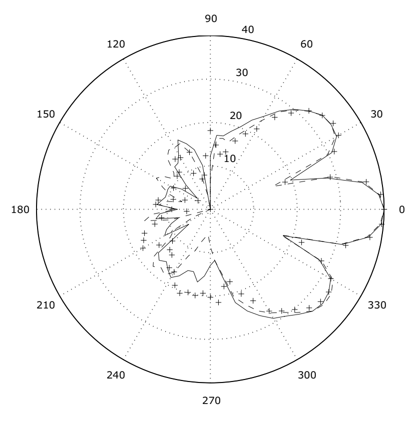
Modern measuring laboratories and currently available PCs are fully capable of acquiring the high-resolution data needed by the GLL. For directivity balloons of the individual transducers, an angular resolution of 5° and frequency resolution of about 1/24th octave is generally sufficient. Almost any practical point of rotation can be chosen 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 using the same point of rotation, like the center of geometry, so the loudspeaker does not have to be remounted. As a rule of thumb, 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.

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. The GLL data model does not presently include wave effects such as diffraction, or shadowing: therefore the loudspeaker should be as close to its real-world configuration as possible. Individual transducers should be measured inside the box; if possible, a line array cabinet should be measured with its upper and lower neighbor cabinets in place.

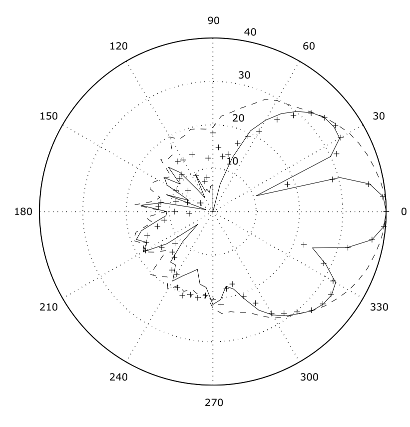
Other measures one can take to verify and ensure data validity include: deriving the mean deviation for on-axis measurements, in order to exclude severe errors introduced by mechanical or environmental changes during the rotation of the loudspeaker; using simulation software such as EASE SpeakerLab to individually renormalize each set of data taken from the front to the back of the loudspeaker (this compensates for temporal drift); multiple time averages during measurement to lower the random noise floor in noisy or slightly unstable environment; applying time windows and filters to remove systematic errors, such as side wall reflections, from frequency response measurements; including compensation files for the frequency response of the AD/DA hardware and the microphone; taking reference measurements to normalize the process.



A simple Loudspeaker such as the Renkus-Heinz PNX121T can be measured using a single point of rotation, if the measurements gather complex magnitude-and-phase data at each point.

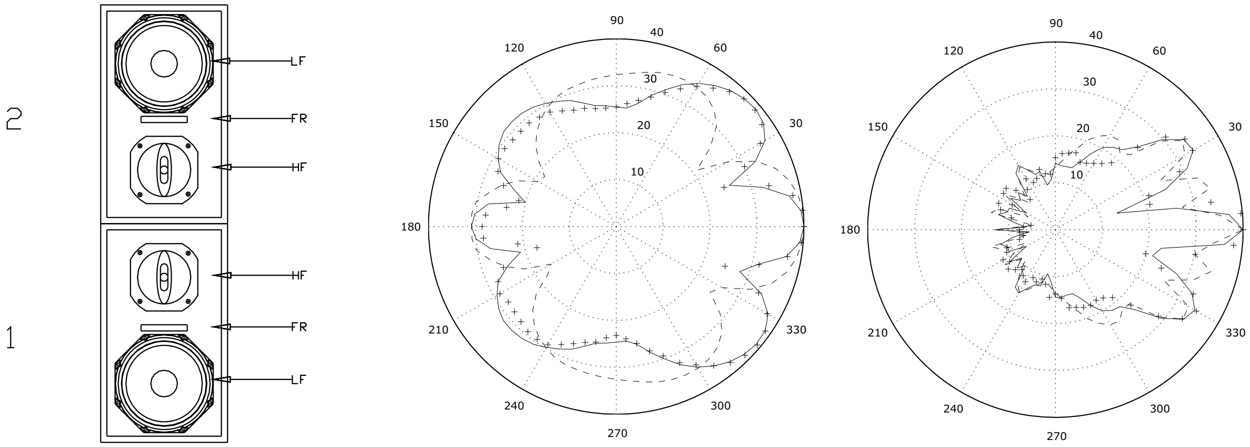


Complex data (———), magnitude-only data (-----) and measurements (++++) based on individual HF & LF points of rotation: 1/24 octave @ 1600 Hz.

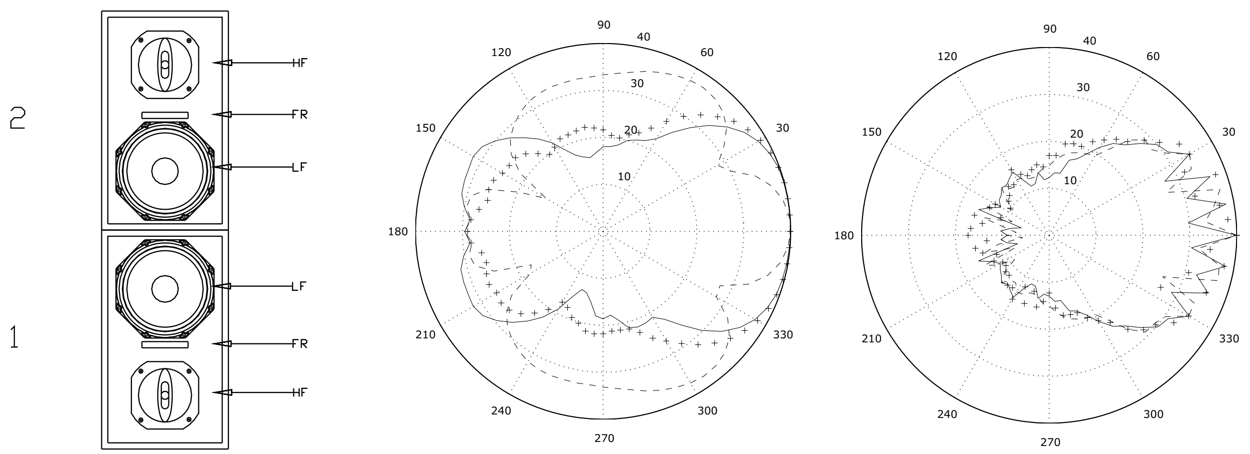


Complex data (———), magnitude-only data (-----) and measurements (++++) based on a common point of rotation at the port: 1/24 octave @ 1600 Hz. Note the large discrepancy between the magnitude-only data and the actual measurement.

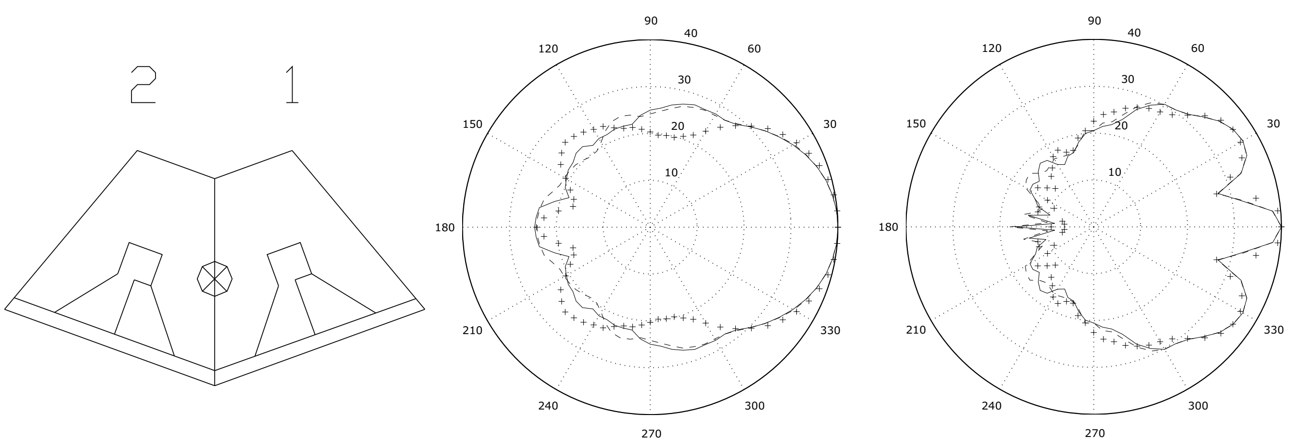
In the GLL, a Loudspeaker such as the PNX121T can be part of Clusters such as the three shown below.



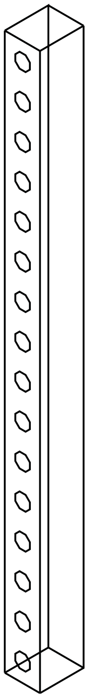
Comparison of predictions based on complex data (———) and magnitude-only data (------) with measurements (+++++) of a horn-to-horn stacked array of two PNX121T's at 500 (l) and 2000 (r) Hz: 1/3 octave resolution



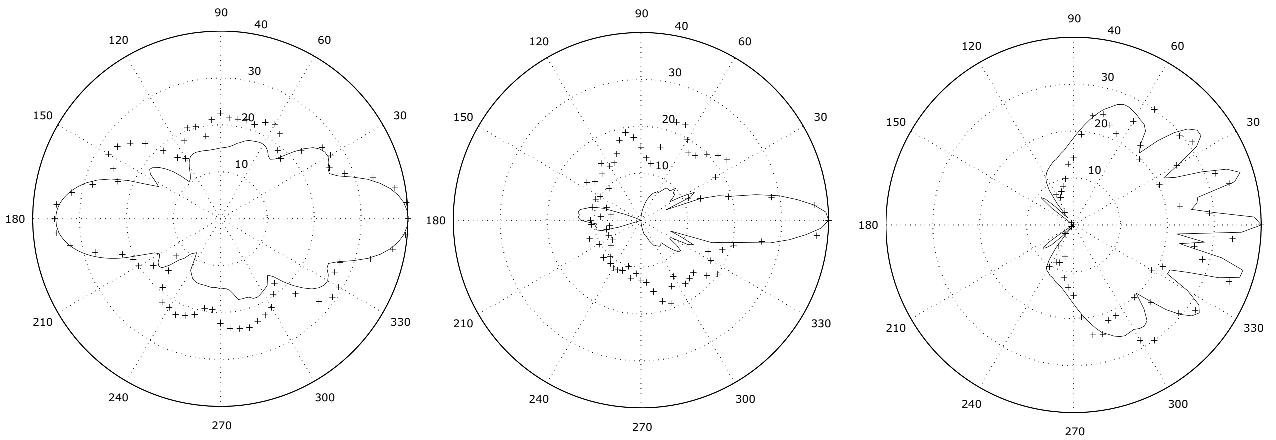
Comparison of predictions based on complex data (———) and magnitude-only data (------) with measurements (+++++) of a woofer-to-woofer stacked array of two PNX121T's at 500 (l) and 2000 (r) Hz: 1/3 octave resolution



Comparison of predictions based on complex data using two full range sources (———) and using four sources (two HF and two LF) (------) with measurements (+++++) of a side-by-side array of two PNX121T's: horizontal polars at 500 (l) and 2000 (r) Hz: 1/3 octave resolution



For more complex systems in which separate filters and/or delays can be applied to individual sources, it is obviously necessary to measure each source separately using its own point of rotation; if all sources are identical, only one needs to be measured. A very complex system such as the Renkus-Heinz Iconyx IC16 Digitally Steerable Array must be modeled as 16 separate sources. This is an example of a system whose directivity can be dynamically altered by manipulating DSP filters.



Comparison of predictions based on complex data (———) with measurements (+++++) of the IC16 at 500 Hz (l), 2000 Hz (c), and 10 kHz (r), at 1/3 octave resolution.

Virtual Filter Design Using the GLL

Greater Efficiency Means More Time to Optimize Performance

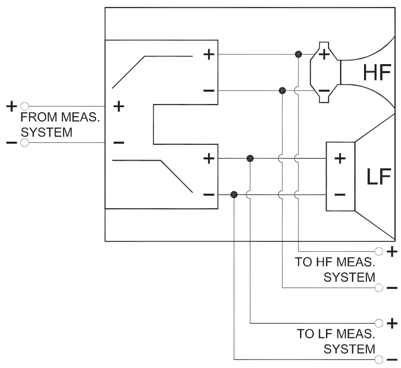
The GLL and simulation software such as EASE SpeakerLab can “virtually” eliminate one of the most time-consuming and tedious design chores – measuring and re-measuring a system’s directivity to evaluate changes in signal processing parameters such as filters, gain and delay. Once the directivity balloon of each transducer has been measured, the system’s directivity can be calculated. Provided that the transfer function of the signal processing is properly modeled, its effects on system directivity can also be calculated. Modern computers can do this in far less time than it takes to re-measure the system. The designer can spend more time optimizing, and can see the results of design changes very rapidly.

As with directivity balloons for individual transducers, proper procedures must be followed when measuring active or passive filters. Passive crossover filters must be measured with the test leads connected to the input terminals of the transducer while in its enclosure.

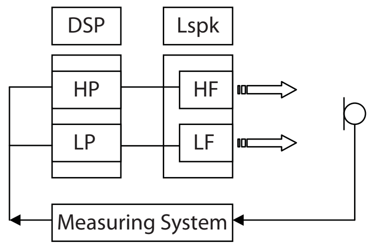
The measurement system must have a balanced input, and its input ground should be left floating (not connected). Otherwise, an HF filter with reverse polarity (quite common in many passive systems with second order filters) will short the amplifier when the positive (+) test lead is connected to the positive input of the driver and the negative (-) test lead is connected to the negative input of the same driver.

Active filtering today means DSP, but not all DSPs are identical in performance. In particular, the implementation of IIR (Infinite Impulse Response) filters is not standardized. The transfer function of the DSP should be measured directly as part of the GLL build process: if that is not possible, verification measurements should be made to determine the degree of correspondence between the actual DSP filters and the software model.

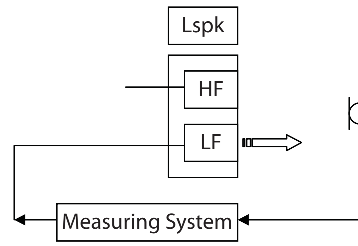
FILTER MEASUREMENT SETUPS



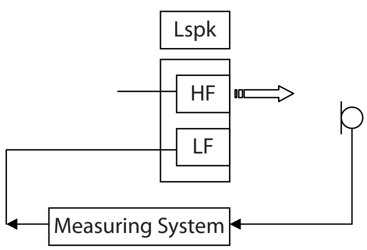
2-way passive system with non-inverted polarity HF filter



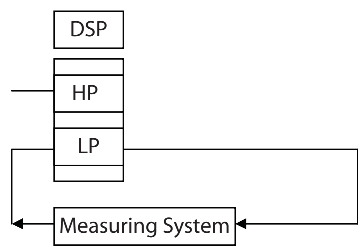
2-way active: fullrange measurement



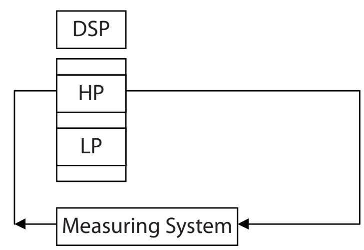
2-way active: LF measurement



2-way active: HF measurement



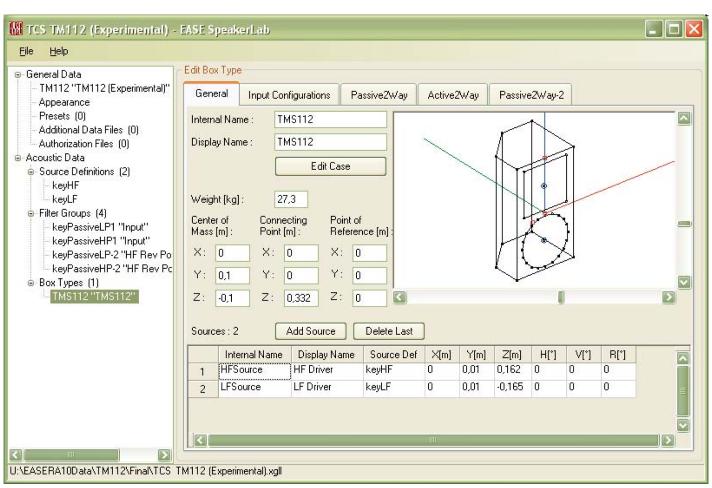
2-way active: LF filter measurement

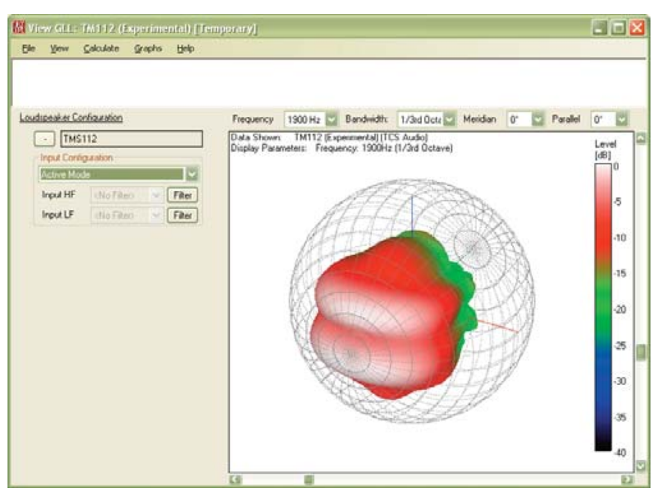


2-way active: HF filter measurement

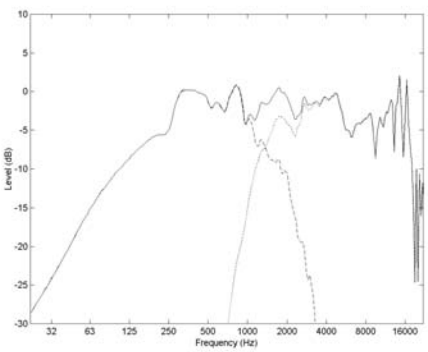
Crossover Design and Directivity Optimization with the GLL & EASE SpeakerLab

DSP filters for TCS Audio's TM112, a 12-inch 2-way system with a horn-loaded compression driver, can illustrate the design process in a virtual software environment. The first step was to create a GLL data object with the measured directivity balloons of each individual source, as shown below. Note that the TM112's conventional horn-above-woofer layout provides horizontal symmetry and vertical asymmetry.

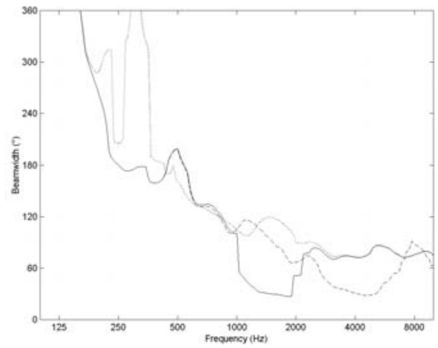




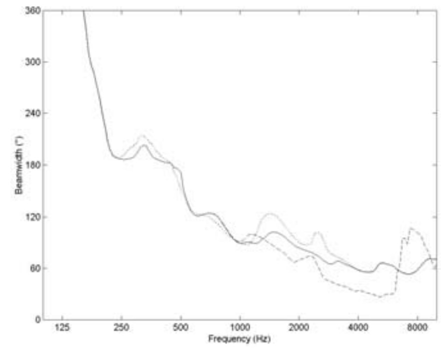
Vertical and horizontal beamwidth plots of the LF and HF indicated that a crossover in the 1 – 2 kHz octave should work. The first approximation included delay on the LF to align it with the HF, and constant-directivity horn EQ for the HF. Crossover filters are symmetrical 4th-order Linkwitz-Riley lowpass and highpass at 1.6 kHz. The plots below show (l-r) normalized magnitude response, vertical beamwidth and horizontal beamwidth.



Normalized Magnitude Response



Vertical Beamwidth

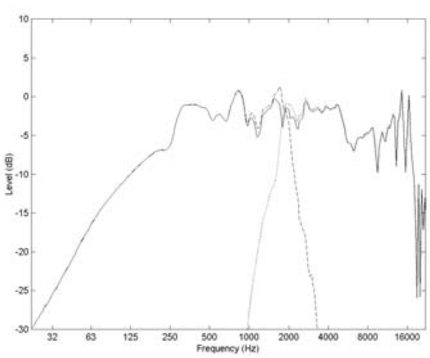


Horizontal Beamwidth

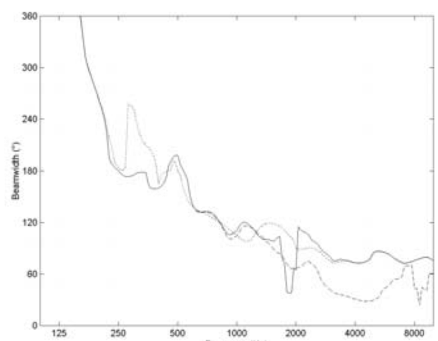
Woofer - - - - Horn ... Fullrange —

Both on-axis magnitude response and horizontal beamwidth look good. However, there is significant vertical narrowing in the 1 - 2 kHz octave, partly due to the horn-above-woofer layout.

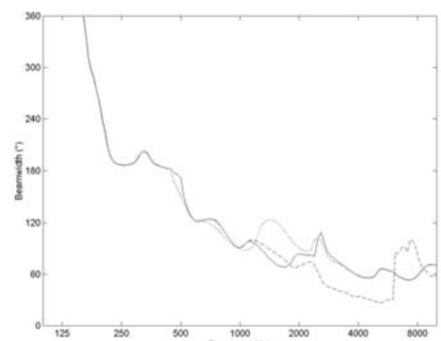
Looking more closely at the complex frequency response (magnitude and phase) of the LF and HF sources helps identify a better set of filter functions. A little minor EQ along with offset filters (4th-order Butterworth lowpass at 1.6 kHz and 5th-order Butterworth highpass at 2 kHz) produces the response shown below.



Normalized Magnitude Response



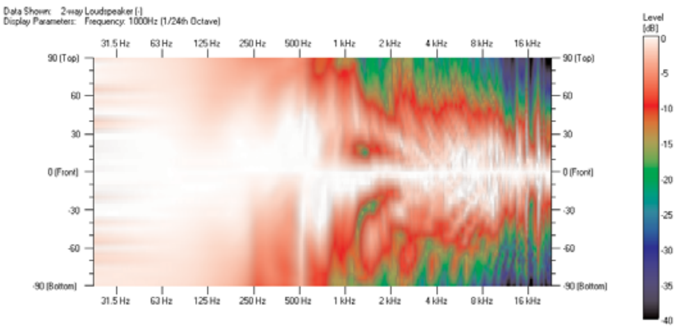
Vertical Beamwidth



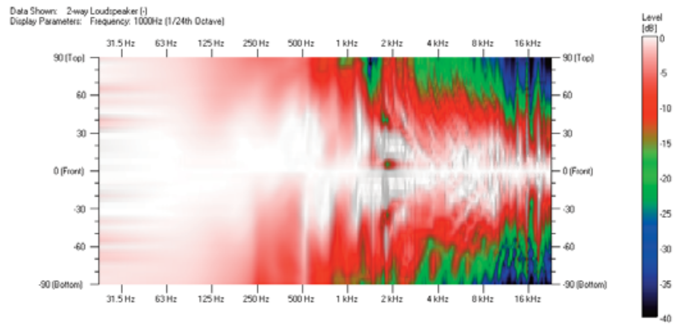
Horizontal Beamwidth

Woofer - - - Horn ... Fullrange —

Without introducing any problems in the on-axis response or the horizontal directivity of this system, the new filters have reduced the bandwidth over which vertical coverage decreases from more than an octave to about 0.2 octave. For more detail, compare the vertical maps of TM112 response using the symmetrical Linkwitz-Riley and asymmetrical Butterworth filters shown on the next page.

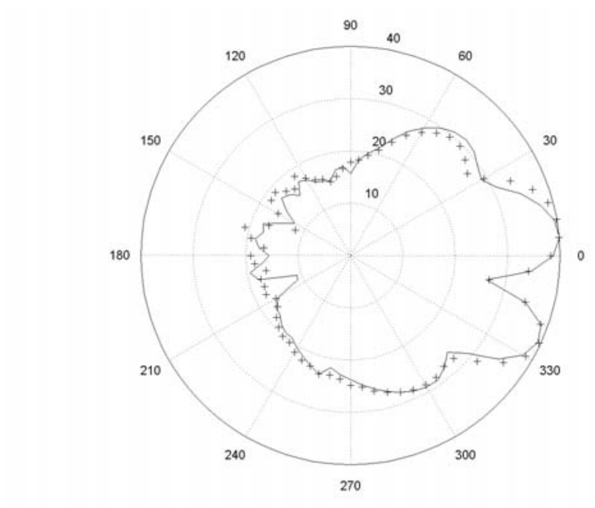


Symmetrical Linkwitz-Riley

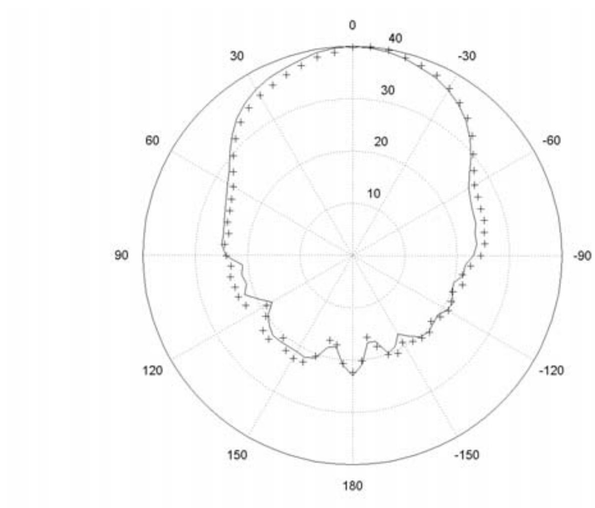


Asymmetrical Butterworth

To verify the predictions of the GLL and SpeakerLab, the delay, EQ and filter settings were implemented in a Biamp Audia DSP, driving two identical amplifier channels that were used to power the TM112. Comparison of measured and predicted polar plots at 1.9 kHz shows good agreement between the predicted and actual performance of the system.



Predicted (-) and measured (+) vertical polars



Predicted (-) and measured (+) horizontal polars

The design process for passive filters is similar, although of course driver-alignment delay and boost EQ cannot be used in a passive design. As with the process described above, accurate predictions are dependent on measuring directivity balloons for each source separately.

Future Possibilities

The current GLL model diffraction does not take shadowing into account. It also ignores wave-based effects such as mutual coupling, boundary loading etc. Very low frequencies below 20 Hz could be included in the model, since they are of interest in many applications. As the rate of increase in the capabilities of commercially available computers and data storage devices shows no signs of slowing down, we can expect to continue extending the GLL object definition model and increasing the resolution of measured data.

Further Information

For more information on the GLL specification, visit the AFMG web sites http://www.sda.de and http://www.ada-acousticdesign.de.

The following sources may also be of interest to those seeking more detailed, in-depth information on the relationship between acoustical measurements and modeling software:

- S. Feistel, W. Ahnert, C. Hughes, B. Olson, "Simulating the Directivity Behavior of Loudspeakers with Crossover Filters," presented at the 123rd Convention of the Audio Engineering Society, New York City, NY, 2007 October 5-8, convention paper 7254.

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- C.Hughes, Excelsior Audio White Papers, "Using Crossovers in the Real World", http://www.excelsior-audio.com/Publications/Crossover/Crossover1.html.

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Ahnert Feistel Media Group:
ADA (Acoustic Design Ahnert) & SDA (Software Design Ahnert GmbH)
Bringing the Science of Acoustics to the Professional Audio Workplace

Ahnert Feistel Media Group (AFMG) originated in the late 1980's. Through his work with AKG on the Delta Stereophonic System, Prof. Dr.-Ing. habil. Wolfgang Ahnert realized the importance of accurate acoustical design and analysis software. After scientific studies at Lomonossov University in Moscow, he teamed up with Dr. rer. nat. habil. Rainer Feistel, Professor of Theoretical Physics at Rostock University, to develop a computer program capable of scientifically simulating the Delta Stereophony concept. Dr. Ahnert applied his thorough theoretical training and practical experience in acoustics to design the software concepts. Dr. Feistel contributed both advanced calculation algorithms and user interface programming.

Soon after the Berlin Wall fell in 1989, ADA Acoustic Design Ahnert was founded. The previously-developed software prototype was given a functional interface and a now-familiar name: EASE (Electro-Acoustic Simulator for Engineers). EASE was first introduced to the professional audio industry in 1990 at the 88th AES Convention in Montreux. Shortly thereafter, Renkus-Heinz became the worldwide distributor of the English-language version.

The rapid acceptance of EASE by professional acousticians and sound system designers encouraged the development team to expand the program's functionality and refine its accuracy, leading in 1994 to the last release for MS-DOS – EASE 2.1. Dipl.-Phys. Stefan Feistel joined the development team in 1995 to port EASE to the Windows operating system. EASE 3.0 was released in 1999 as the first version of EASE to run in the more user-friendly Windows environment. The growing availability of both desktop and laptop computing power, along with increased demand for modular extensions to the EASE software suite, led to the decision to form a new company in order to put the software development on a professional basis. SDA Software Design Ahnert GmbH was founded in 1999. Its first goal was the development of a new software-based measurement platform that would unify all existing approaches in a modern format. The result, EASERA, was released in late 2004. Shortly after that the manufacturer-independent line array aiming and modeling program EASE Focus was released, along with other software packages engineered for professionals in the audio industry.

Today's Ahnert Feistel Media Group is composed of both ADA, which has expanded with satellite offices in Cairo, Egypt and Doha, Qatar, and SDA as well as the non-profit ADA - Foundation gGmbH. The mission of the ADA - Foundation is to support research into the science of acoustics by collaborating with leading universities in countries around the world, and by making academic discounts on AFMG-developed software available to students and researchers.

EASE and its related programs are an ongoing joint development project of ADA and SDA. EASE has become the de facto industry standard for acoustic design and analysis. This is due to a number of factors: open databases, a foundation in rigorous academic research, and continual improvements in functionality and user interface that take advantage of exponential growth in computer processing power and speed.

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